Implementing Cisco IP Telephony and Video, Part 2 (CIPTV2)

Number: 300-075
Passing Score: 800
Time Limit: 120 min
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This exam tests candidates seeking CCNP Collaboration on their ability for implementing a Cisco Unified Collaboration solution in a multisite environment. It covers Uniform Resource Identifier (URI) dialing, globalized call routing, Intercluster Lookup Service and Global Dial Plan Replication, Cisco Service Advertisement Framework and Call Control Discovery, tail-end hop-off, Cisco Unified Survivable Remote Site Telephony, Enhanced Location Call Admission Control (CAC) and Automated Alternate Routing (AAR), and mobility features such as Device Mobility, Cisco Extension Mobility, and Cisco Unified Mobility. The exam also describes the role of Cisco Video Communication Server (VCS) Control and the Cisco Expressway Series and how they interact with Cisco Unified Communications Manager.

Exam A

QUESTION 1

A local gateway is registered to Cisco TelePresence Video Communication Server with a prefix of 7. The administrator wants to stop calls from outside the organization being routed through it. Which CPL configuration accomplishes this goal?

Exhibit A (exhibit):

```
<?xml version="1.0" encoding="UTF-8" ?>
<cpl xmlns="urn:ietf:params:xml:ns:cpl"
  xmlns:taa="http://www.tandberg.net/cpl-extensions"
  xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
  xsi:schemaLocation="urn:ietf:params:xml:ns:cpl cpl.xsd">
  <taa:routed>
    <taa:rule-switch>
      <taa:rule originating-zone="TraversalZone" destination="7(.*)">
        <!—External calls are not allowed to use this gateway -->
        <!-- Reject call with a status code of 403 (Forbidden) -->
        <reject status="403" reason="Denied by policy"/>
      </taa:rule>
      <taa:rule origin="(.*)" destination="(.*)">
        <!-- All other calls allowed -->
        <proxy/>
      </taa:rule>
    </taa:rule-switch>
  </taa:routed>
</cpl>
```


Exhibit B (exhibit):

```
<?xml version="1.0" encoding="UTF-8" ?>
<cpl xmlns="urn:ietf:params:xml:ns:cpl"
  xmlns:taa="http://www.tandberg.net/cpl-extensions"
  xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
  xsi:schemaLocation="urn:ietf:params:xml:ns:cpl cpl.xsd">
  <taa:routed>
    <taa:rule-switch>
      <taa:rule originating-zone="NeighborZone" destination="7(.*)">
        <!-External calls are not allowed to use this gateway -->
        <!-- Reject call with a status code of 403 (Forbidden) -->
        <reject status="403" reason="Denied by policy"/>
      </taa:rule>
      <taa:rule origin="(.*)" destination="(.*)">
        <!-- All other calls allowed -->
        <proxy/>
      </taa:rule>
    </taa:rule-switch>
  </taa:routed>
</cpl>
```

Exhibit C (exhibit):

```xml
<?xml version="1.0" encoding="UTF-8" ?>
<cpl xmlns="urn:ietf:params:xml:ns:cpl"
  xmlns:taa="http://www.tandberg.net/cpl-extensions"
  xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
  xsi:schemaLocation="urn:ietf:params:xml:ns:cpl cpl.xsd">
  <taa:routed>
    <taa:rule-switch>
      <taa:rule originating-zone="TraversalSubZone" destination="7(.*)">
        <!—External calls are not allowed to use this gateway -->
        <!-- Reject call with a status code of 403 (Forbidden) -->
        <reject status="403" reason="Denied by policy"/>
      </taa:rule>
      <taa:rule origin="(.*)" destination="(.*)">
        <!-- All other calls allowed -->
        <proxy/>
      </taa:rule>
    </taa:rule-switch>
  </taa:routed>
</cpl>
```

Exhibit D (exhibit):

```xml
<?xml version="1.0" encoding="UTF-8" ?>
<cpl xmlns="urn:ietf:params:xml:ns:cpl"
  xmlns:taa="http://www.tandberg.net/cpl-extensions"
  xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
  xsi:schemaLocation="urn:ietf:params:xml:ns:cpl cpl.xsd">
  <taa:routed>
    <taa:rule-switch>
      <taa:rule originating-zone="TraversalZone" destination="7(.\d)">
        <!—External calls are not allowed to use this gateway -->
        <!-- Reject call with a status code of 403 (Forbidden) -->
        <reject status="403" reason="Denied by policy"/>
      </taa:rule>
      <taa:rule origin="(.*)" destination="(.*)">
        <!-- All other calls allowed -->
        <proxy/>
      </taa:rule>
    </taa:rule-switch>
  </taa:routed>
</cpl>
```

Exhibit E (exhibit):

```xml
<?xml version="1.0" encoding="UTF-8" ?>
<cpl xmlns="urn:ietf:params:xml:ns:cpl"
  xmlns:taa="http://www.tandberg.net/cpl-extensions"
  xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"
  xsi:schemaLocation="urn:ietf:params:xml:ns:cpl cpl.xsd">
  <taa:routed>
    <taa:rule-switch>
      <taa:rule originating-zone="TraversalZone" destination="(.*)">
        <!—External calls are not allowed to use this gateway -->
        <!-- Reject call with a status code of 403 (Forbidden) -->
        <reject status="403" reason="Denied by policy"/>
      </taa:rule>
      <taa:rule origin="(.*)" destination="(.*)">
        <!-- All other calls allowed -->
        <proxy/>
      </taa:rule>
    </taa:rule-switch>
  </taa:routed>
</cpl>
```

A. Exhibit A
B. Exhibit B
C. Exhibit C
D. Exhibit D
E. Exhibit E

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 2
Which type of search message appears in the Cisco TelePresence Video Communication Server search history page when it receives a H.323 call from a RAS-enabled endpoint that originates from an external zone?

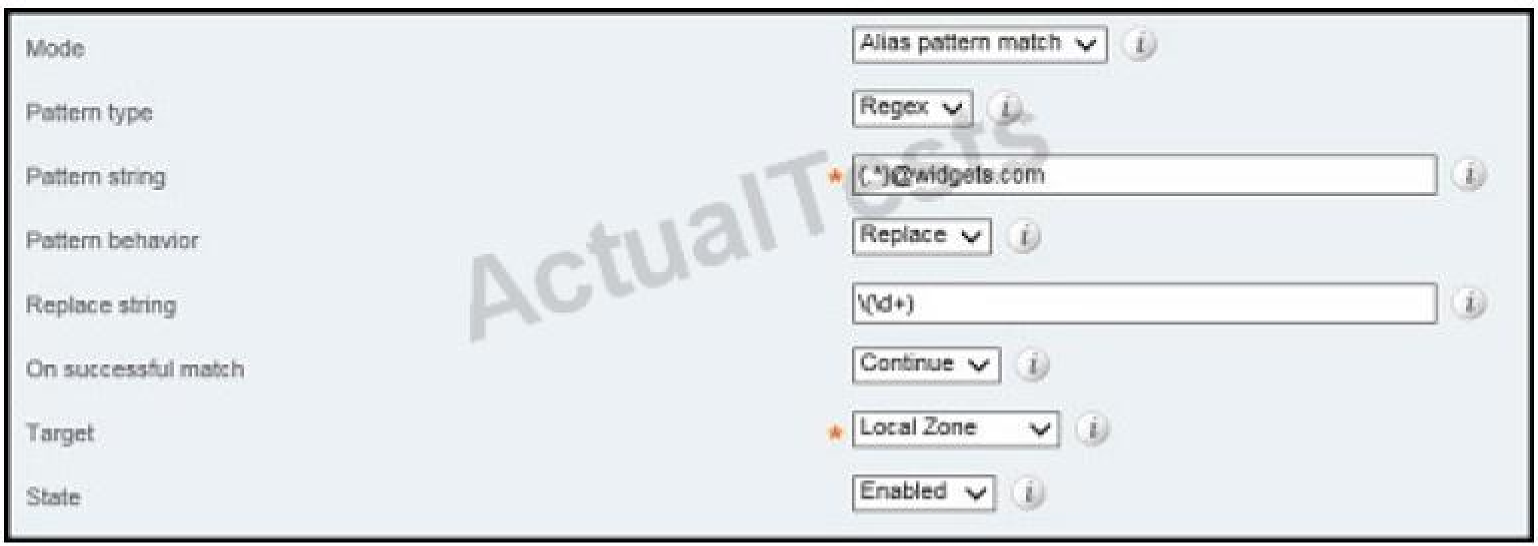
A. ARQ
B. SETUP
C. LRQ
D. INVITE
E. OPTIONS

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 3
Widgets.com's Cisco TelePresence Video Communication Server allows SIP and H.323 registrations. Which local zone search rule configuration allows SIP registered endpoints to connect to H.323 endpoints that register with an H.323 E.164 number only?

A.



B.


Mode
Alias pattern match
Pattern type
Regex
Pattern string
(.+)@widgets.com
Pattern behavior
Replace
Replace string
*
On successful match
Continue
Target
Local Zone
State
Enabled

C.


Mode
Alias pattern match
Pattern type
Regex
Pattern string
(.+)@widgets.com
Pattern behavior
Replace
Replace string
\$
On successful match
Continue
Target
Local Zone
State
Enabled

D.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 4
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 5
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 6
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 7
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 8
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 9
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 10
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 11
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 12
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 13
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 14
What is the standard Layer 3 DSCP media packet value that should be set for Cisco TelePresence endpoints?

A. CS3 (24)
B. EF (46)
C. AF41 (34)
D. CS4 (32)

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 15
When you configure QoS on VCS, which settings do you apply if traffic through the VCS should be tagged with DSCP AF41?

A. Set QoS mode to DiffServ and tag value 32.
B. Set QoS mode to IntServ and tag value to 34.
C. Set QoS mode to DiffServ and tag value 34.
D. Set QoS mode to IntServ and tag value to 32.
E. Set QoS mode to ToS and tag value to 32.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 16
What is the default DSCP/PHB for video conferencing packets in Cisco Unified Communications Manager?

A. EF/46
B. CS6/48
C. AF41/34
D. CS3/24

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 17
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 18
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 19
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 20
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 21
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 22
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 23
Which configuration change is needed to enable NANP international dialing during MGCP fallback?

```
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 24
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 25
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 26
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 27
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. show running-config gateway
F. show fallback-mgcp ccm-manager

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 28
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 29
Which command displays the detailed configuration of all the Cisco Unified IP phones, voice ports, 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 30
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 31
Which gateway does the Cisco Unified Communications Manager control all call activity?

A. SIP
B. MGCP
C. H.323
D. Media

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 32
Which two options should be selected in the SIP trunk security profile that affect the SIP trunk pointing to the VCS? (Choose two.)

A. Accept Unsolicited Notification
B. Enable Application Level Authorization
C. Accept Out-of-Dialog REFER
D. Accept Replaces Header
E. Accept Presence Subscription

Correct Answer: AD
Section: (none)
Explanation

Explanation/Reference:

QUESTION 33
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 34
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 35
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 36
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 37
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 38
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 39
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 40
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 41
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 42
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 43
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 44
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 45
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 46
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 47
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. (Choose two).

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

Correct Answer: BC
Section: (none)
Explanation

Explanation/Reference:

QUESTION 48
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.

Correct Answer: B
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.

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?

SX20 System information (exhibit):

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

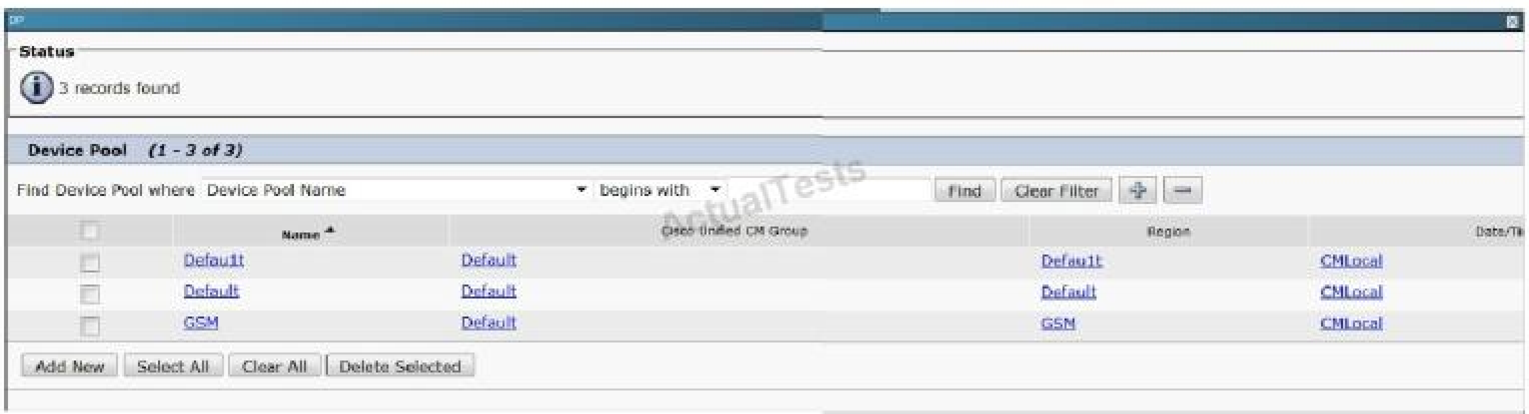
SIP Proxy 1

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

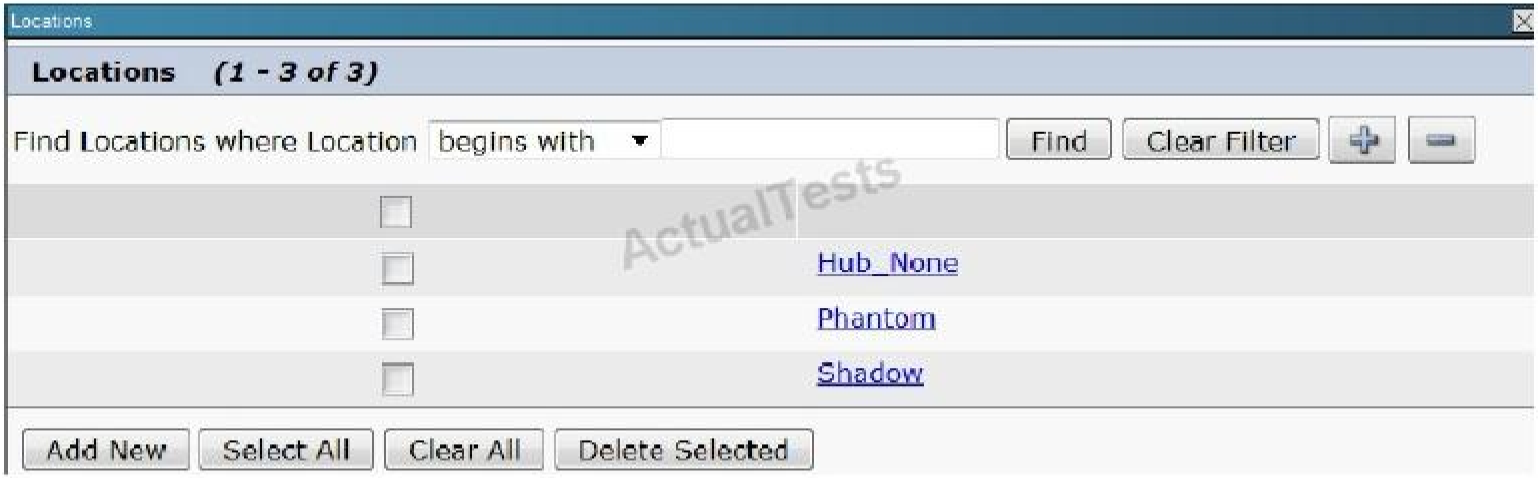
DX650 Configuration (exhibit):

MRGL (exhibit):

DP (exhibit):



Locations (exhibit):

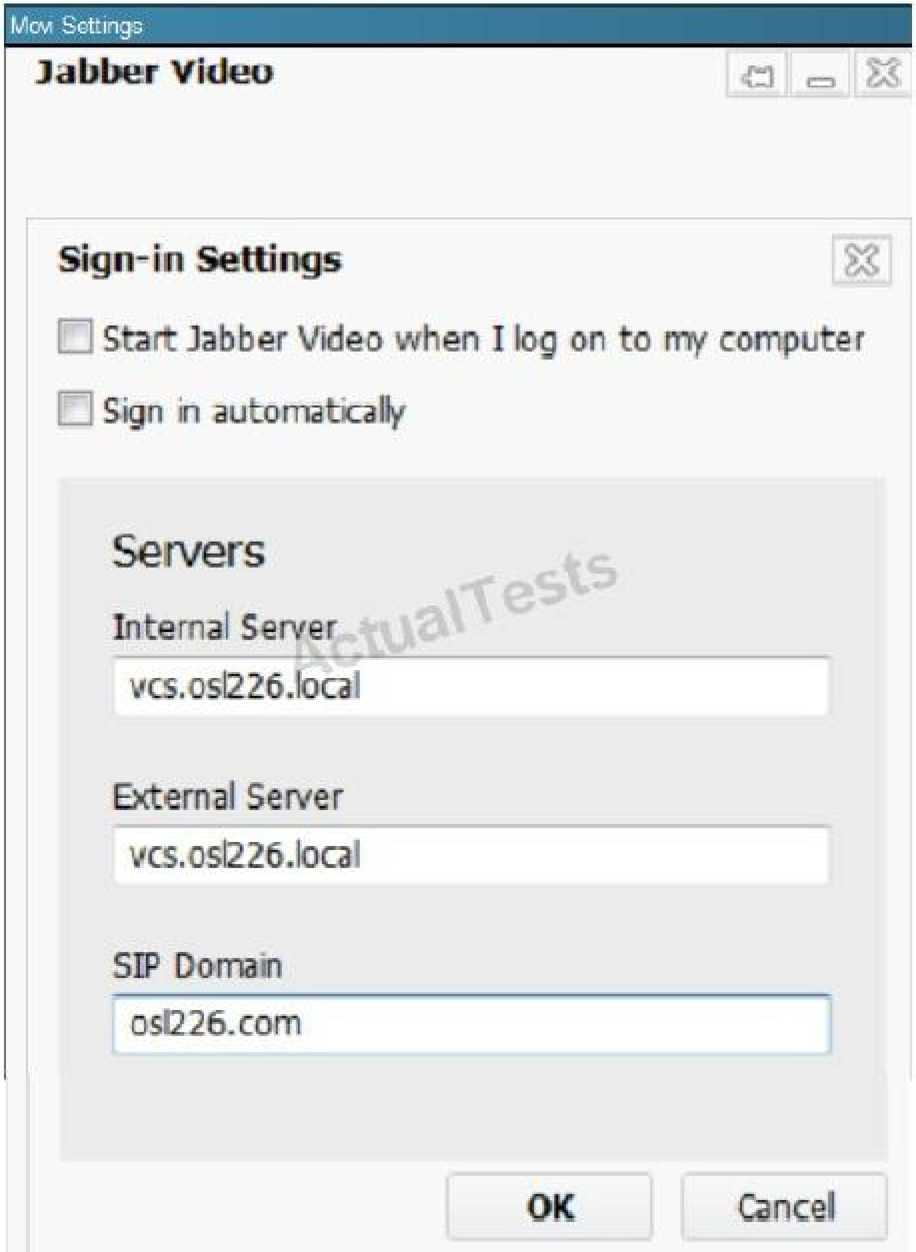


AARG (exhibit):

CSS (exhibit):

Movi Failure (exhibit):

Movi Settings (exhibit):


Mov 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: E
Section: (none)
Explanation

Explanation/Reference:

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 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)

SX20 System information (exhibit):

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

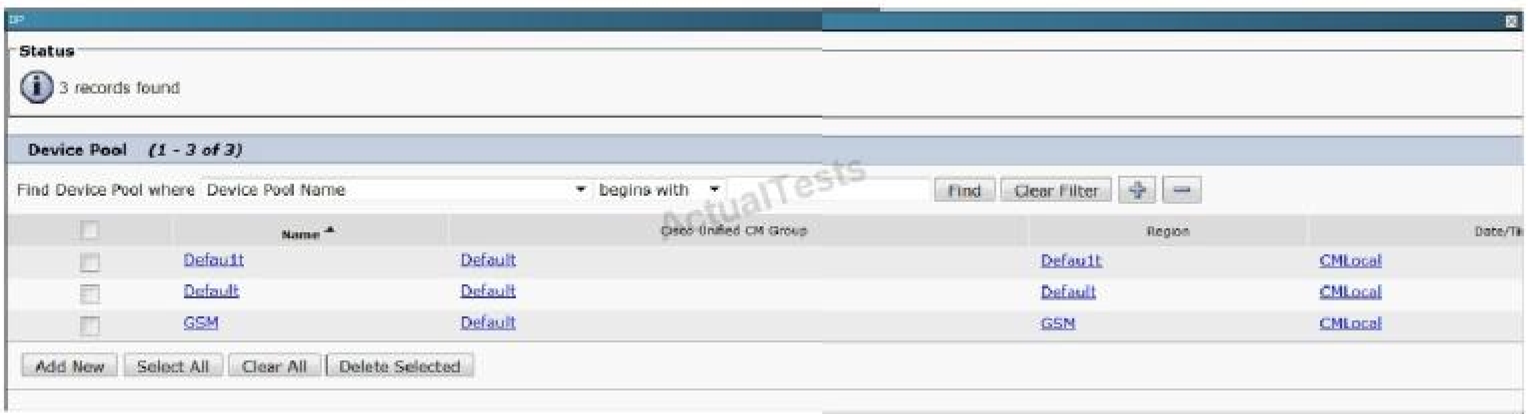
SIP Proxy 1

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

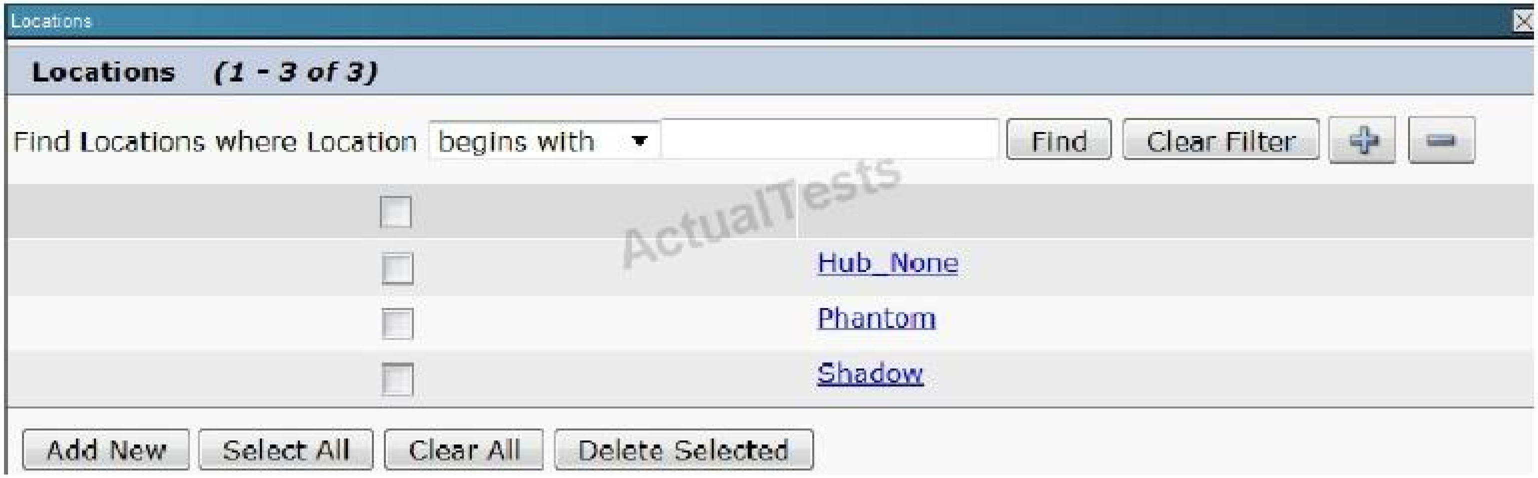
DX650 Configuration (exhibit):

MRGL (exhibit):

DP (exhibit):



Locations (exhibit):

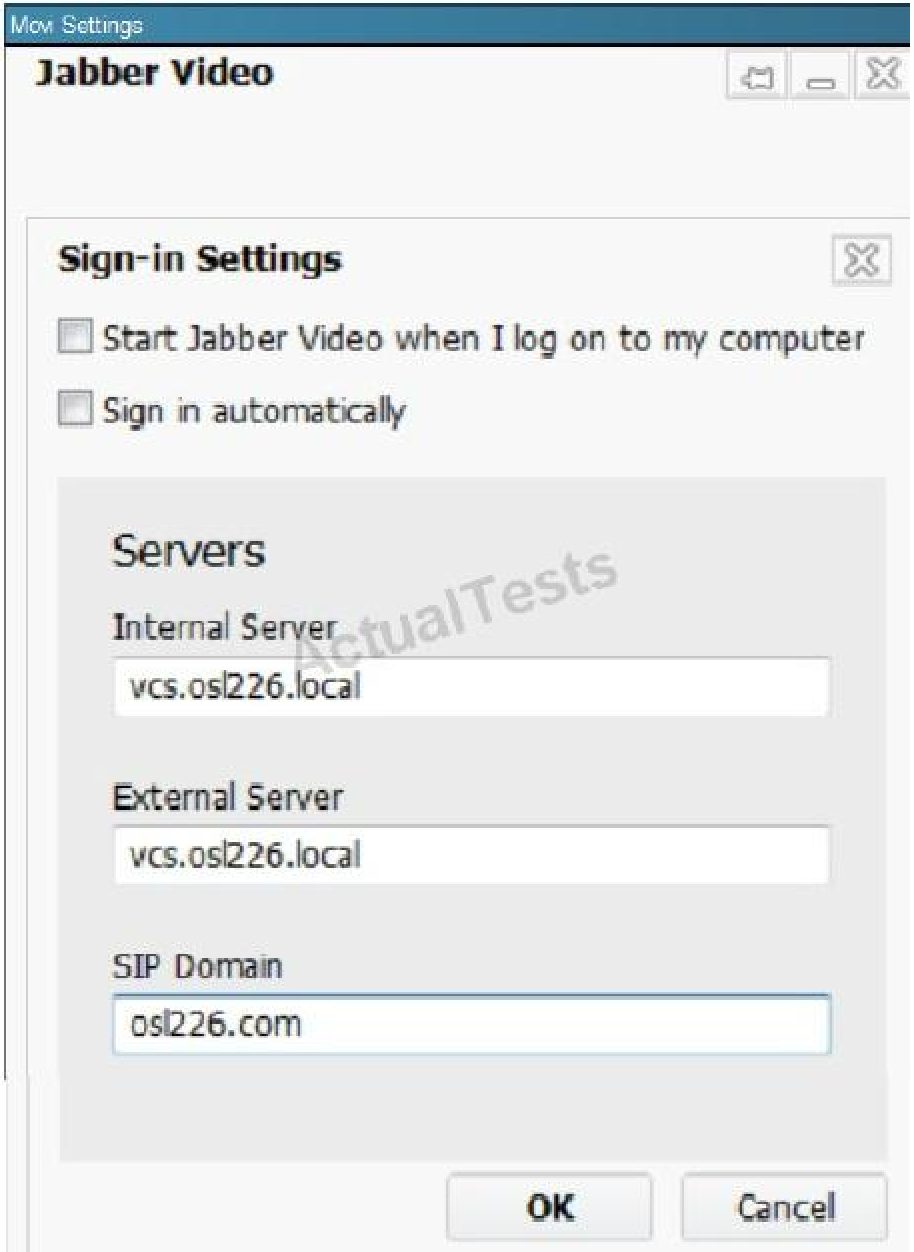


AARG (exhibit):

CSS (exhibit):

Movi Failure (exhibit):

Movi Settings (exhibit):


Movi 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
ActualTests

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 51
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.

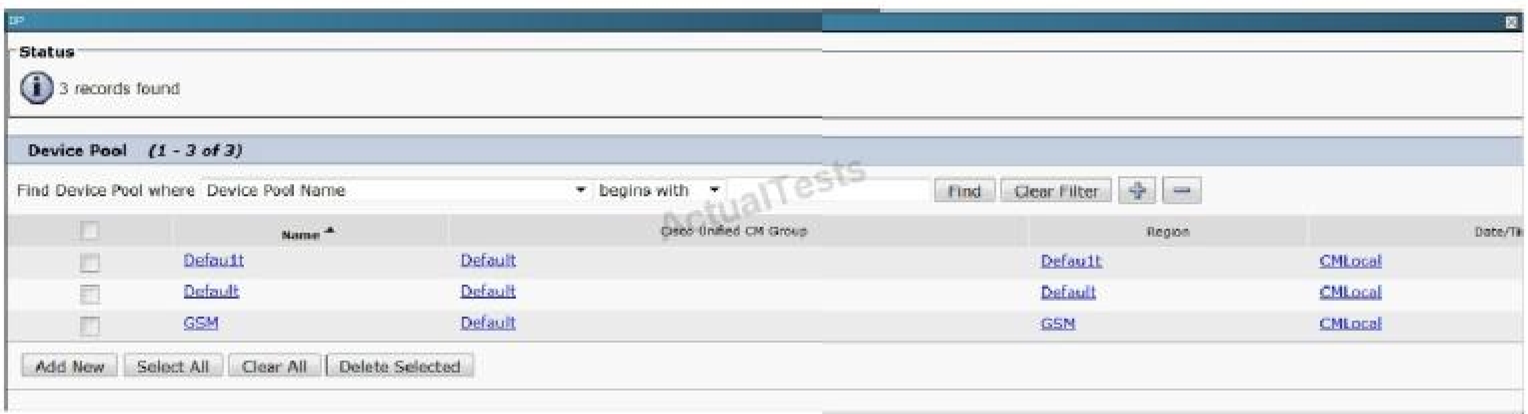
Which device configuration option will allow an administrator to assign a device to specific rights for making calls to specific DNs?

SX20 System information (exhibit):

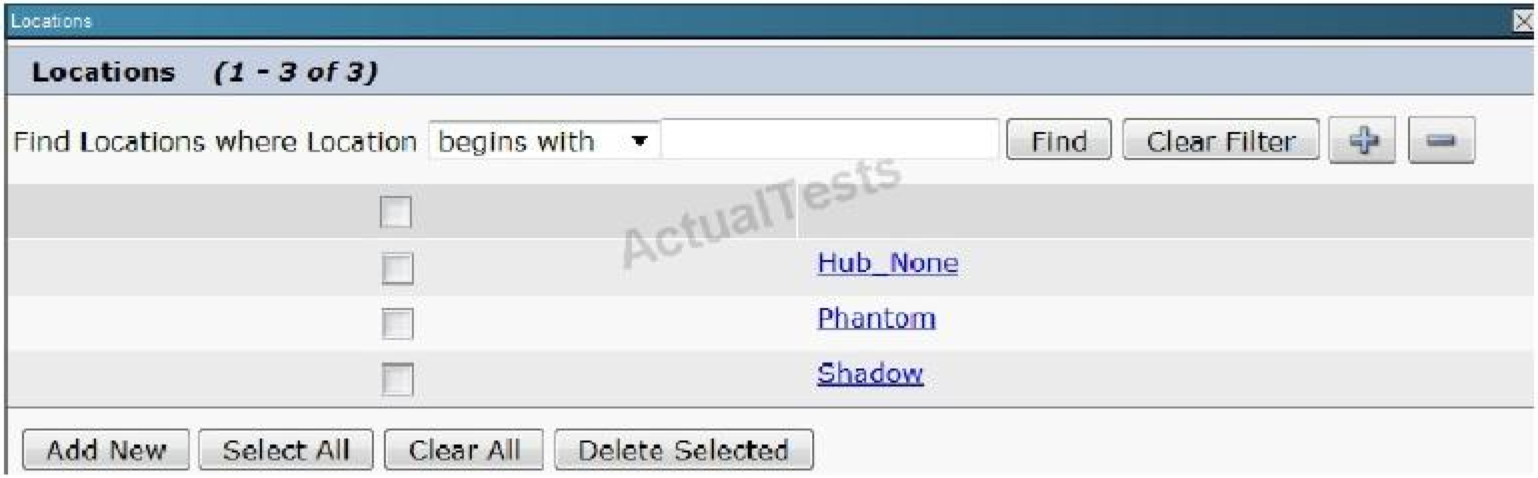
DX650 Configuration (exhibit):

MRGL (exhibit):

DP (exhibit):



Locations (exhibit):



AARG (exhibit):

CSS (exhibit):

Movi Failure (exhibit):

Movi Settings (exhibit):

Movi 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

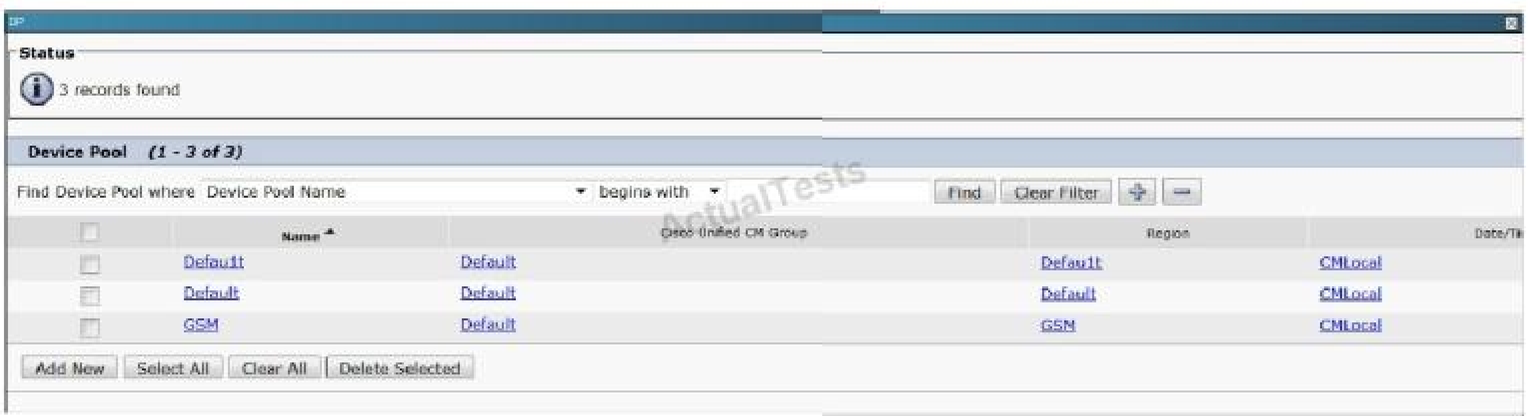
Correct Answer: E
Section: (none)
Explanation

Explanation/Reference:

QUESTION 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.

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?

DP (exhibit):


Status

3 records found

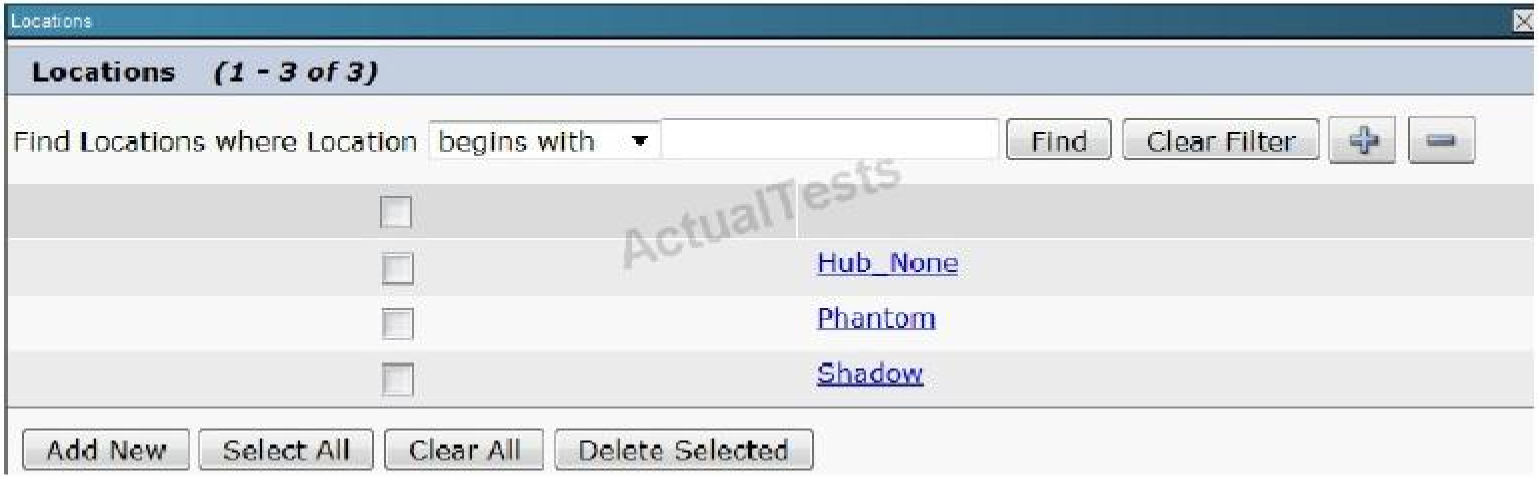
Device Pool (1 - 3 of 3)

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

	Name	Cisco Unified CM Group	Region	Date/Ti
	Default	Default	Default	CMLocal
	Default	Default	Default	CMLocal
	GSM	Default	GSM	CMLocal

Add New | Select All | Clear All | Delete Selected

Locations (exhibit):



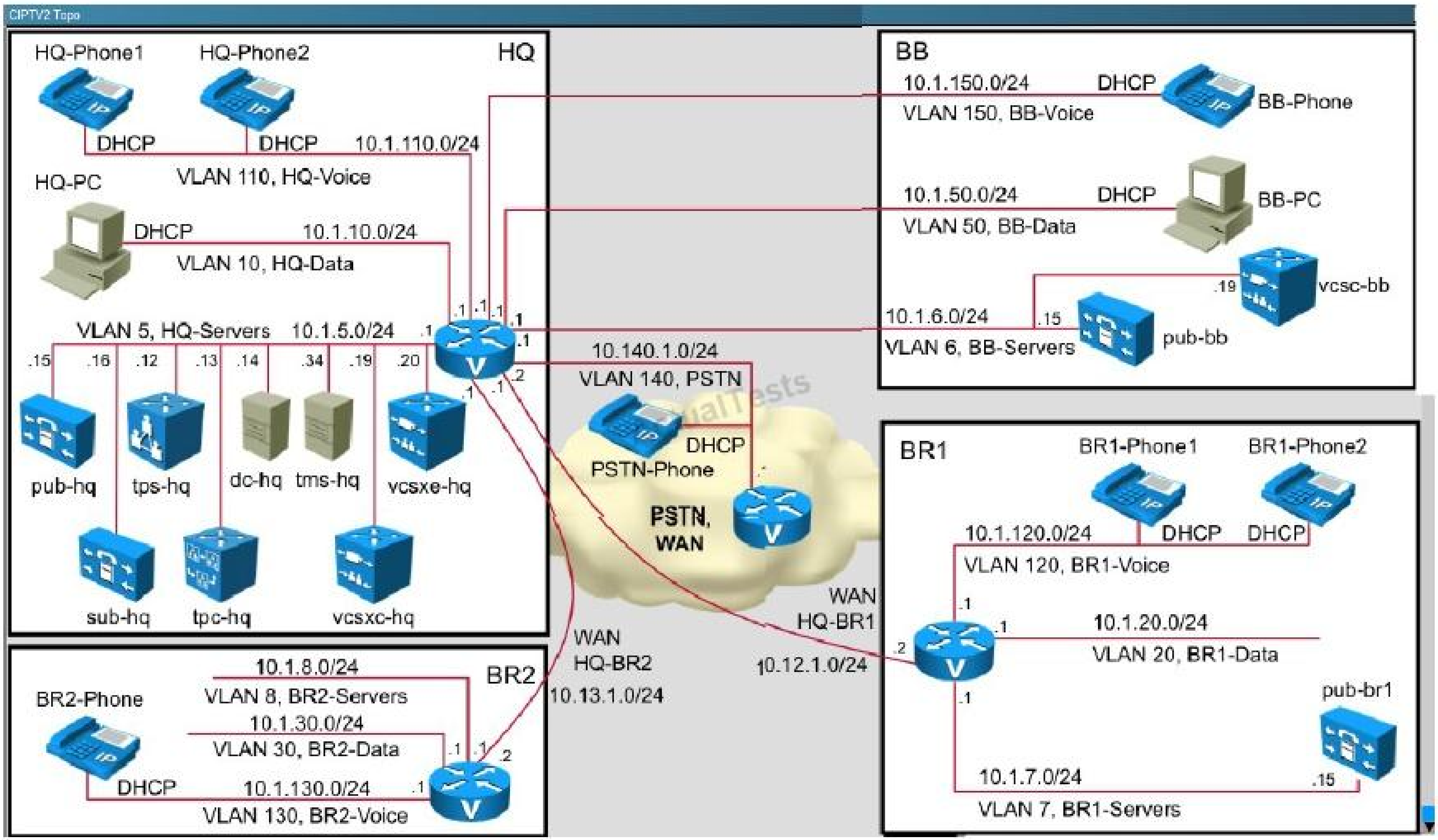
CSS (exhibit):

Movi Failure (exhibit):

Movi Settings (exhibit):


Movi 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

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

Links (exhibit):

Links

You are here: Configuration ▸ Bandwidth ▸ Links

	Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth used	Actions
☐	DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 kbps	View/Edit
☐	DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 kbps	View/Edit
☐	DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 kbps	View/Edit
☐	SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 kbps	View/Edit
☐	SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 kbps	View/Edit
☐	TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 kbps	View/Edit
☐	VCS_HQ - toHQ	HQ	to_HQ	to_HQ_pipe		1	128 kbps	View/Edit

Pipe (exhibit):

Pipe

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.

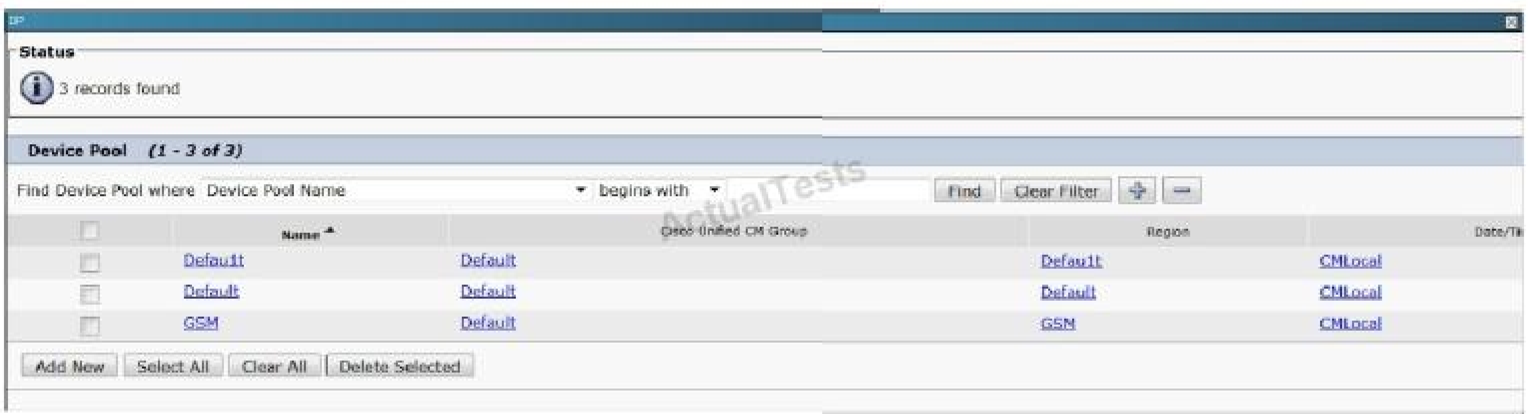
Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

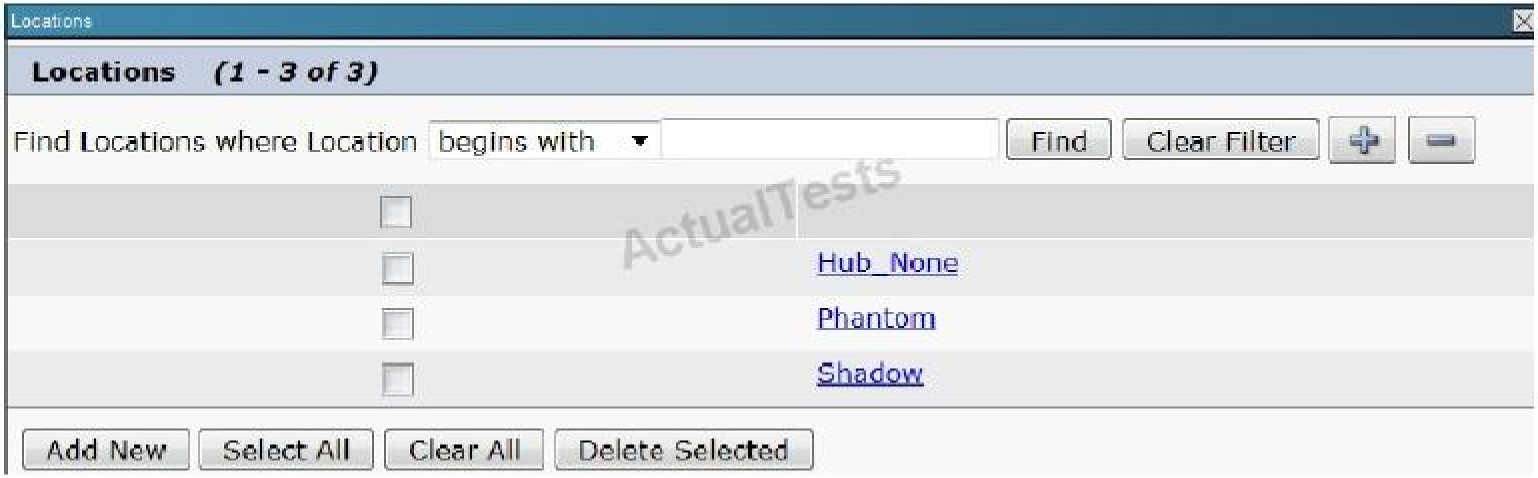
QUESTION 53
Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phone. The Cisco VCS is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows

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

DP (exhibit):



Locations (exhibit):



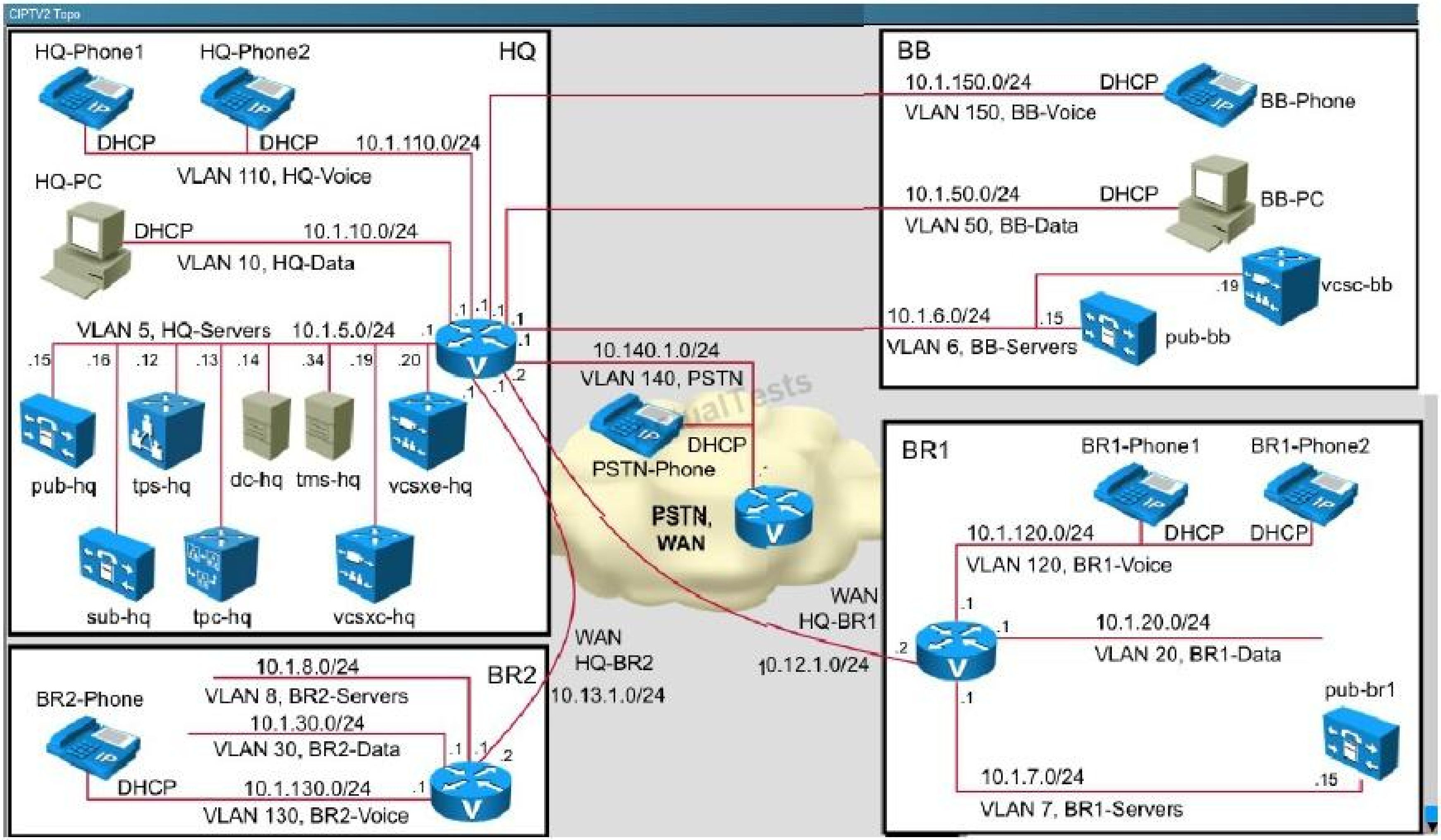
CSS (exhibit):

Movi Failure (exhibit):

Movi Settings (exhibit):


Movi 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
ActualTests

CIPTV2 Topology (exhibit):



Subzone (exhibit):

Links

You are here: Configuration ▸ Bandwidth ▸ Links

Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth used	Actions
DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 kbps	View/Edit
DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 kbps	View/Edit
DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 kbps	View/Edit
SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 kbps	View/Edit
SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 kbps	View/Edit
TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 kbps	View/Edit
VCS_HQ - toHQ	HQ	to_HQ	to_HQ_pipe		1	128 kbps	View/Edit

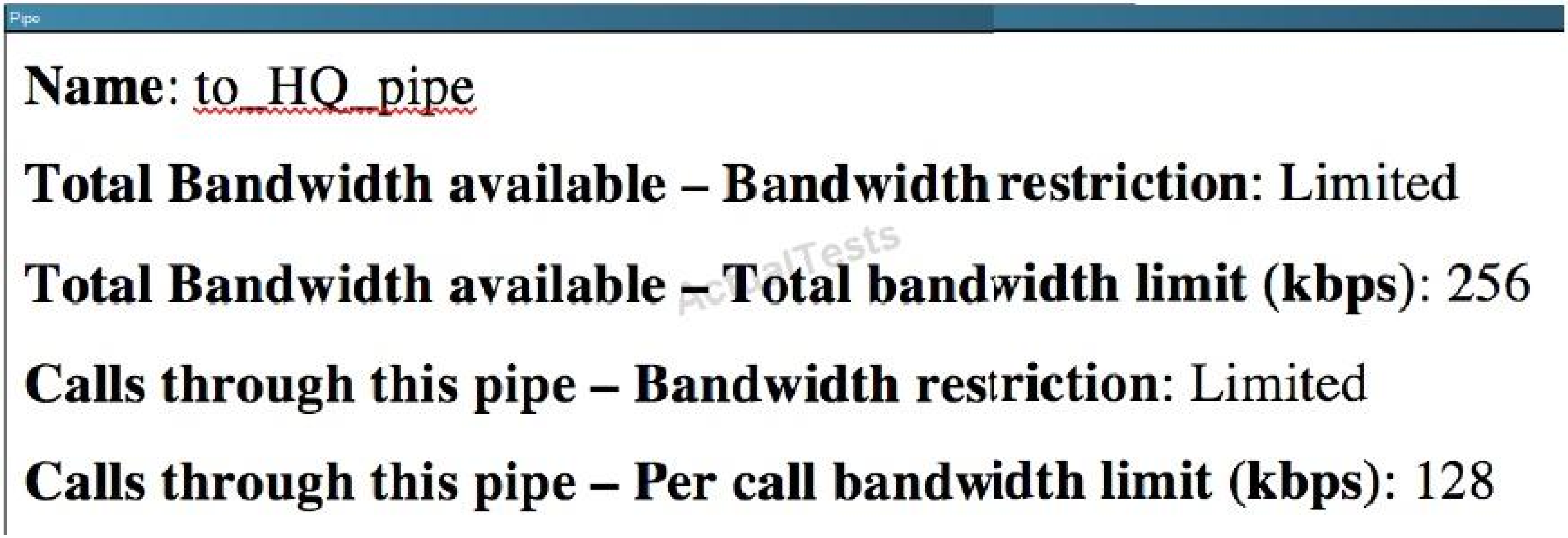
Links (exhibit):

Links

You are here: Configuration ▸ Bandwidth ▸ Links

Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth used	Actions
DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 kbps	View/Edit
DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 kbps	View/Edit
DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 kbps	View/Edit
SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 kbps	View/Edit
SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 kbps	View/Edit
TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 kbps	View/Edit
VCS_HQ - toHQ	HQ	to_HQ	to_HQ_pipe		1	128 kbps	View/Edit

Pipe (exhibit):



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.

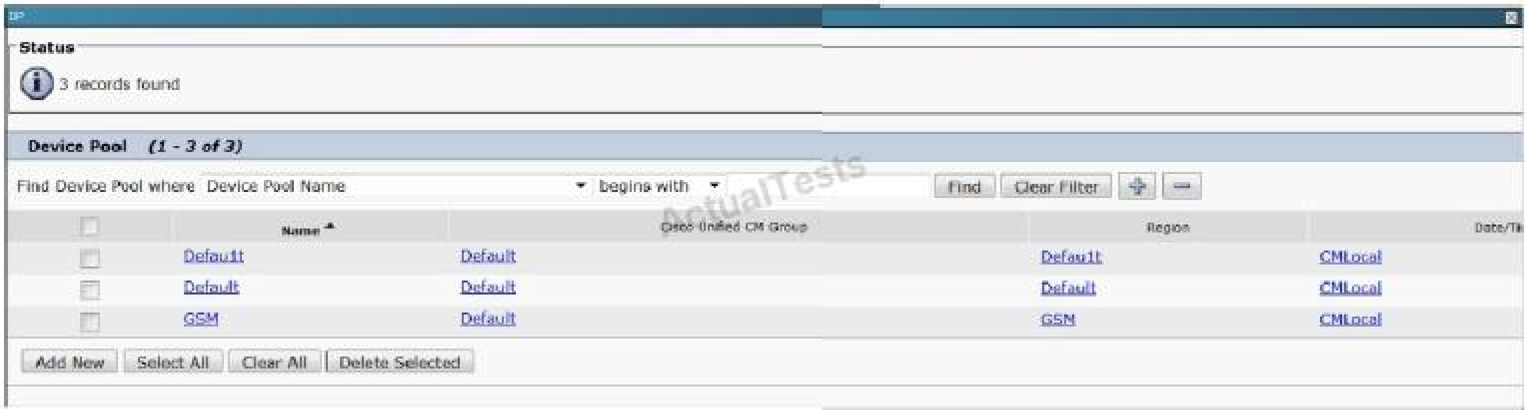
Correct Answer: BD
Section: (none)
Explanation

Explanation/Reference:

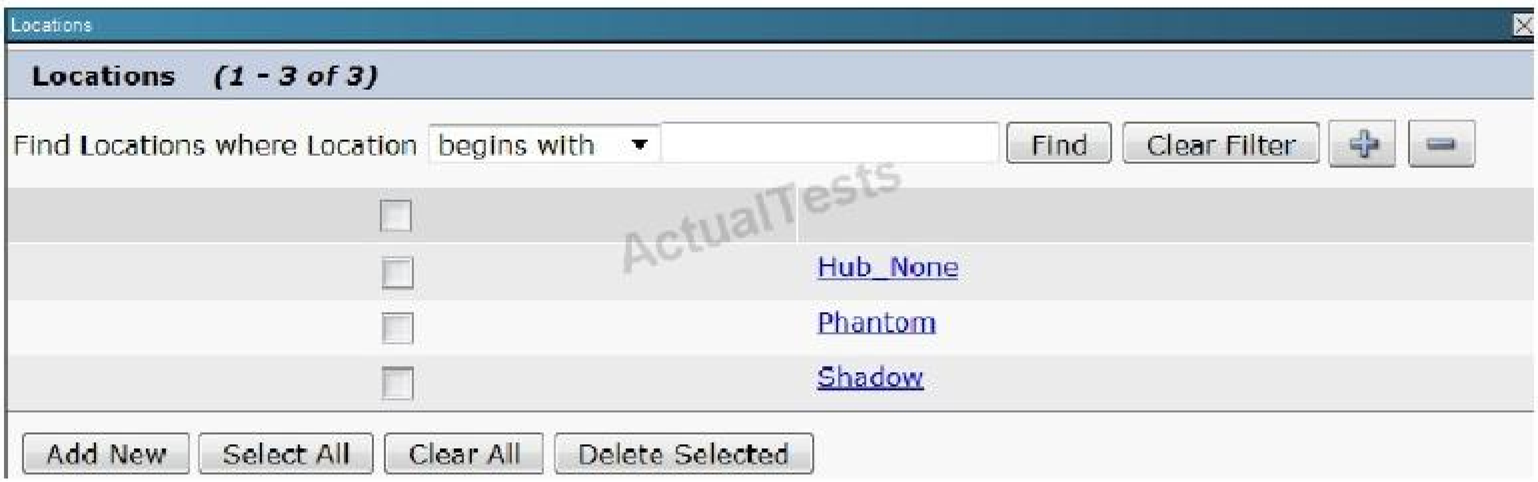
QUESTION 54
Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phone. The Cisco VCS is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

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?

DP (exhibit):



Locations (exhibit):



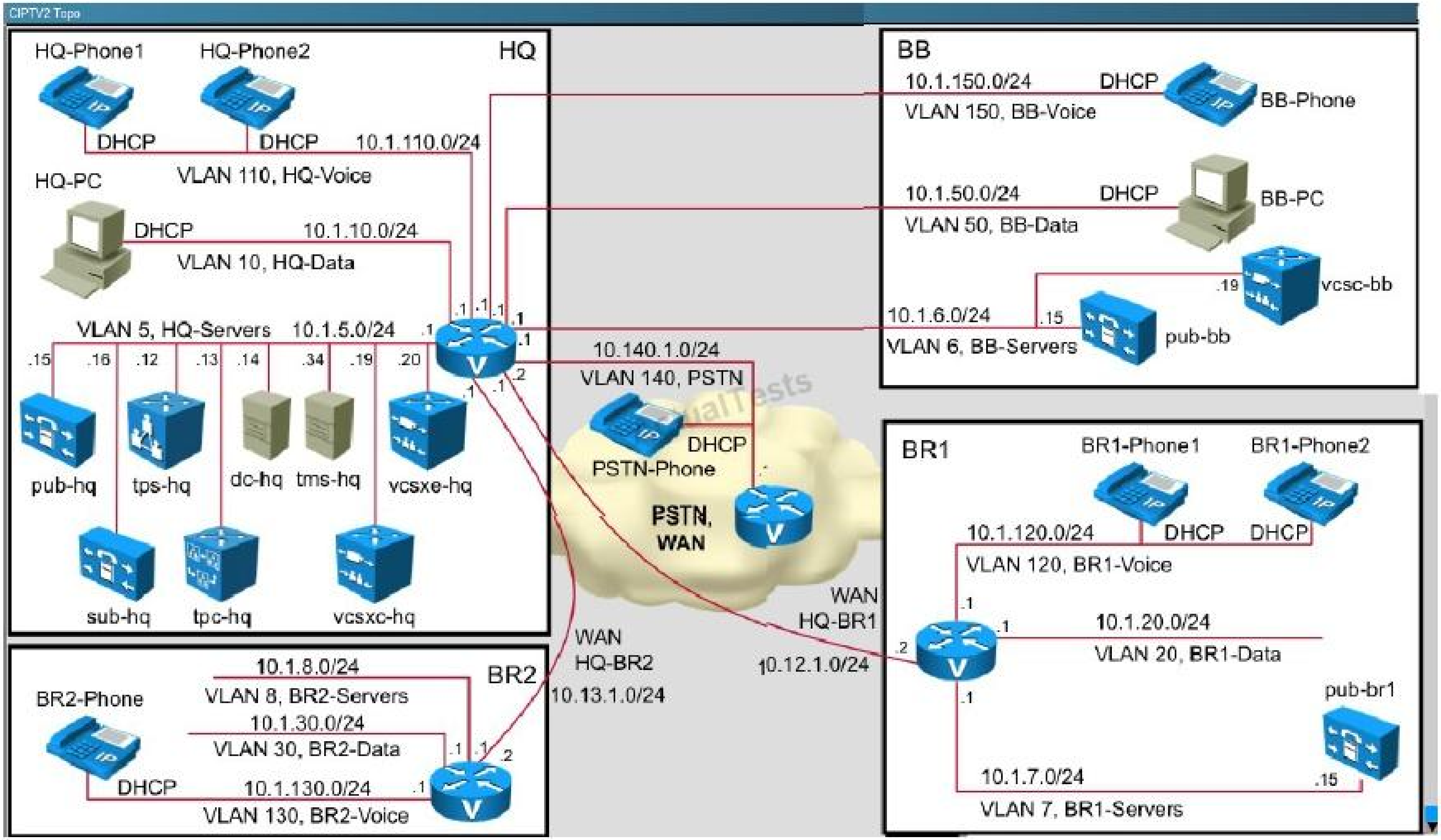
CSS (exhibit):

Movi Failure (exhibit):

Movi Settings (exhibit):


Movi 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

CIPTV2 Topology (exhibit):



Subzone (exhibit):

Links

You are here: Configuration ▸ Bandwidth ▸ Links

Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth used	Actions
DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 kbps	View/Edit
DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 kbps	View/Edit
DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 kbps	View/Edit
SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 kbps	View/Edit
SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 kbps	View/Edit
TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 kbps	View/Edit
VCS_HQ - toHQ	HQ	to_HQ	to_HQ_pipe		1	128 kbps	View/Edit

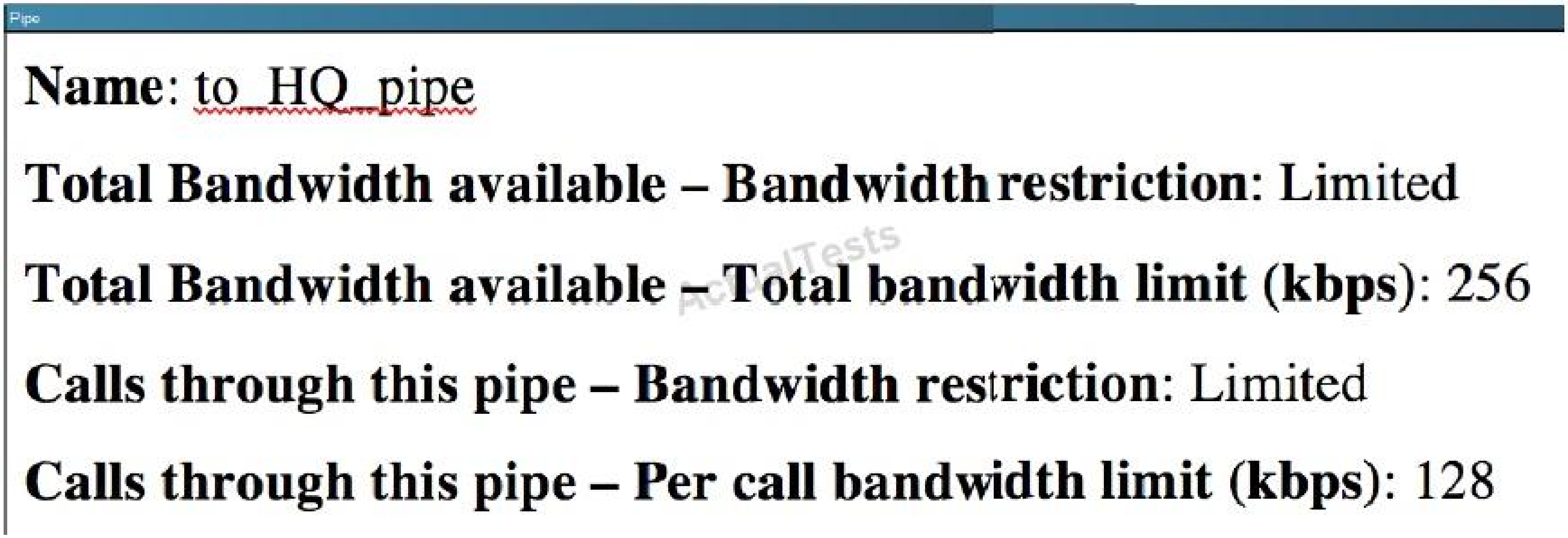
Links (exhibit):

Links

You are here: Configuration ▸ Bandwidth ▸ Links

Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth used	Actions
DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 kbps	View/Edit
DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 kbps	View/Edit
DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 kbps	View/Edit
SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 kbps	View/Edit
SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 kbps	View/Edit
TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 kbps	View/Edit
VCS_HQ - toHQ	HQ	to_HQ	to_HQ_pipe		1	128 kbps	View/Edit

Pipe (exhibit):



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:

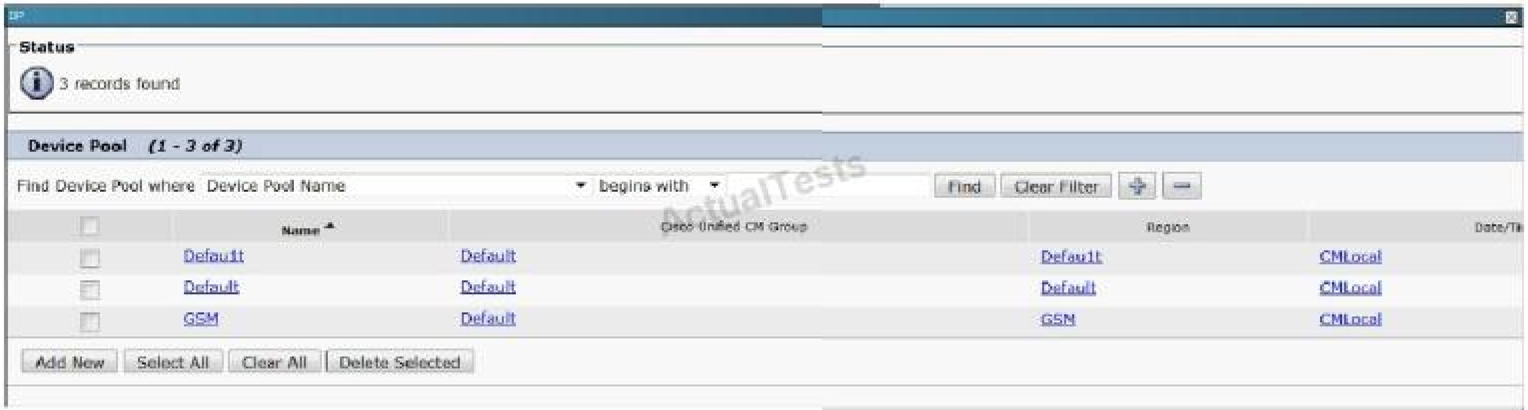
QUESTION 55

Scenario:

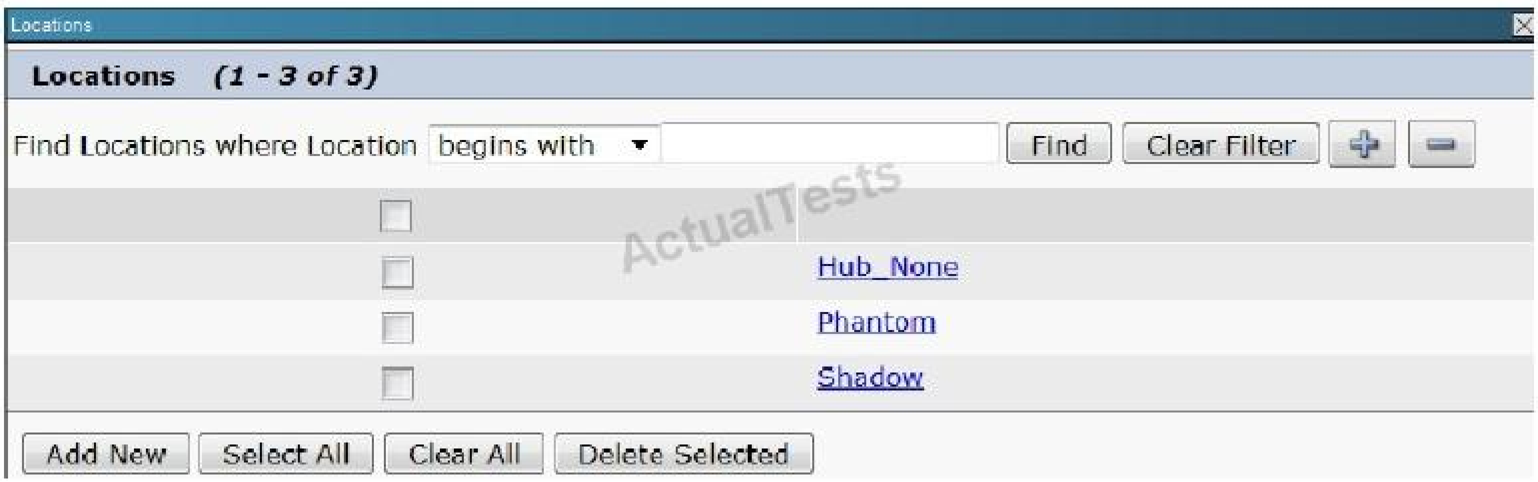
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 9971 Video IP Phone. The Cisco VCS and TMS control the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

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

DP (exhibit):

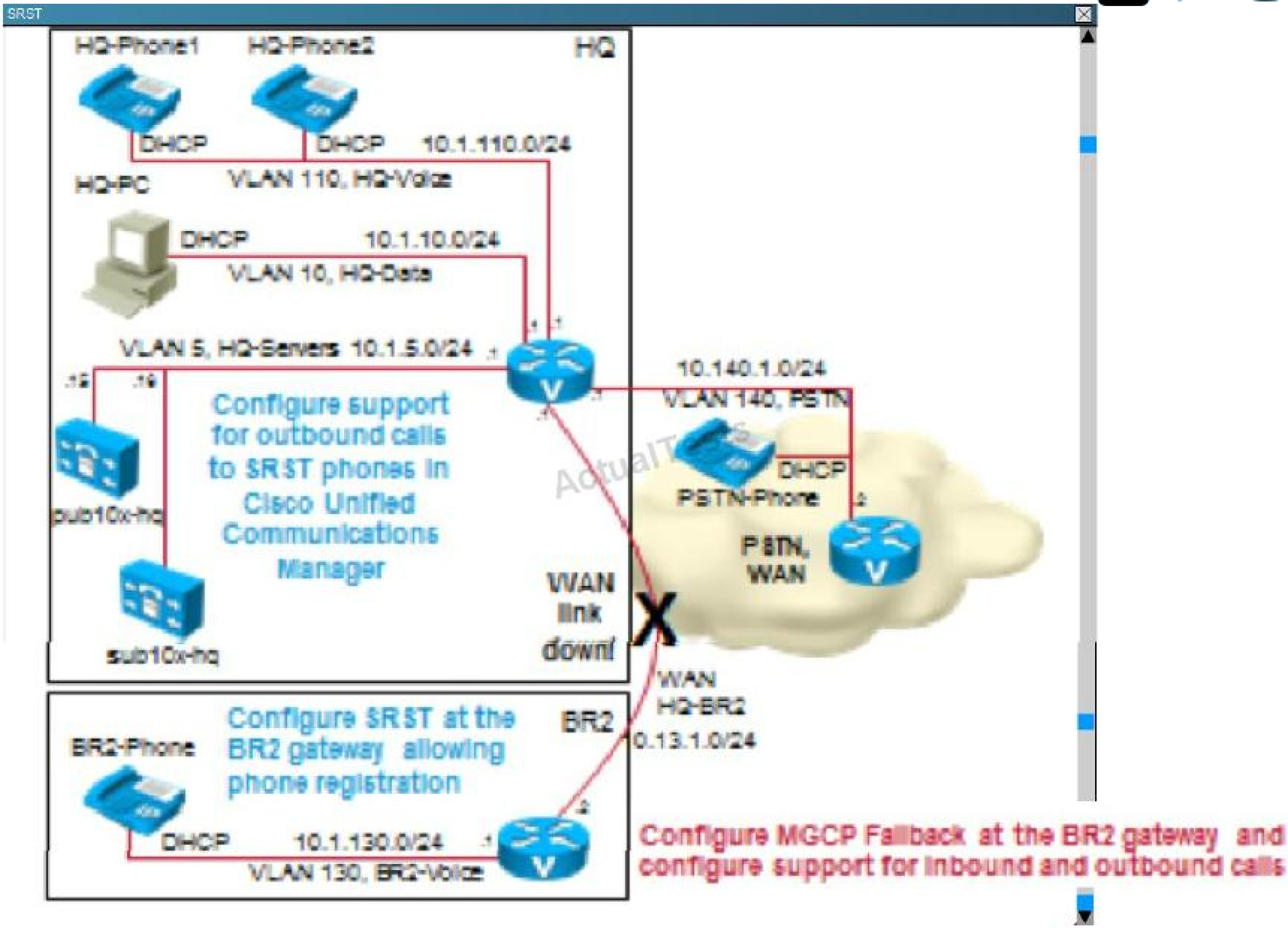


Locations (exhibit):

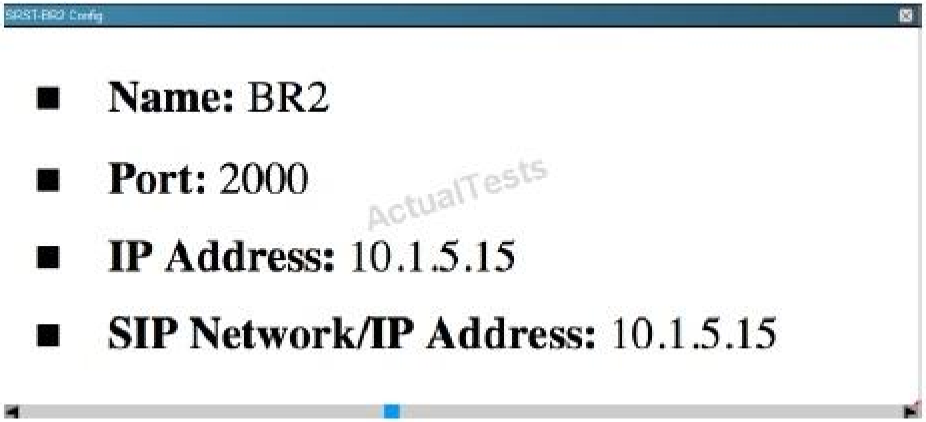


CSS (exhibit):

SRST (exhibit):


SRST
HQ-Phone1
HQ-Phone2
HQ
DHCP
DHCP
10.1.110.0/24
HQ-PC
VLAN 110, HQ-Voice
DHCP
10.1.10.0/24
VLAN 10, HQ-Data
VLAN 5, HQ-Servers 10.1.5.0/24
10.140.1.0/24
VLAN 140, PSTN
Configure support for outbound calls to SRST phones in Cisco Unified Communications Manager
pub10x-hq
DHCP
PSTN-Phone
PSTN, WAN
WAN link down!
sub10x-hq
WAN
HQ-BR2
0.13.1.0/24
Configure SRST at the BR2 gateway allowing phone registration
BR2
BR2-Phone
DHCP
10.1.130.0/24
VLAN 130, BR2-Voice
Configure MGCP Fallback at the BR2 gateway and configure support for inbound and outbound calls

SRST-BR2-Config (exhibit):



BR2 Config (exhibit):

```
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
```

SRSTPSTNCall (exhibit):

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. 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.

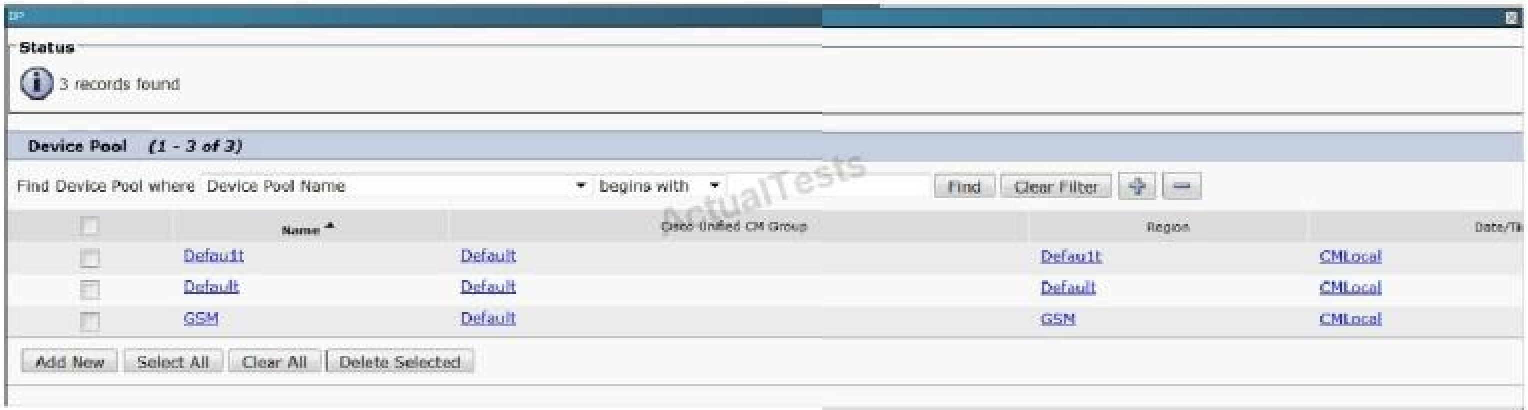
Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

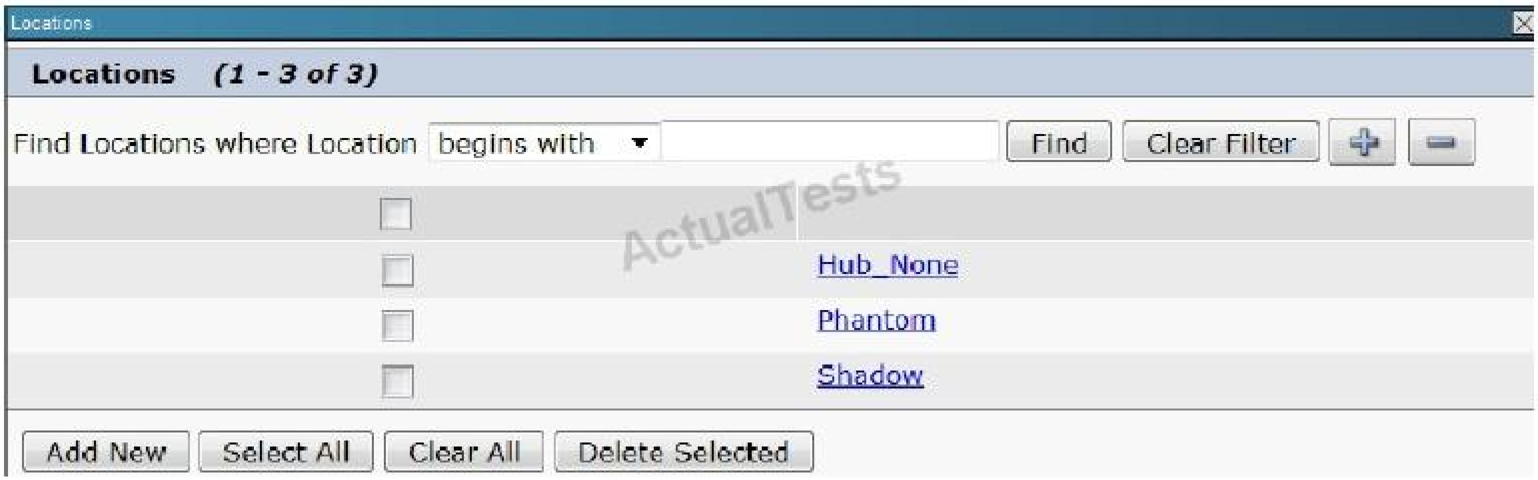
QUESTION 56
Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 9971 Video IP Phone. The Cisco VCS and TMS control the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

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.)

DP (exhibit):

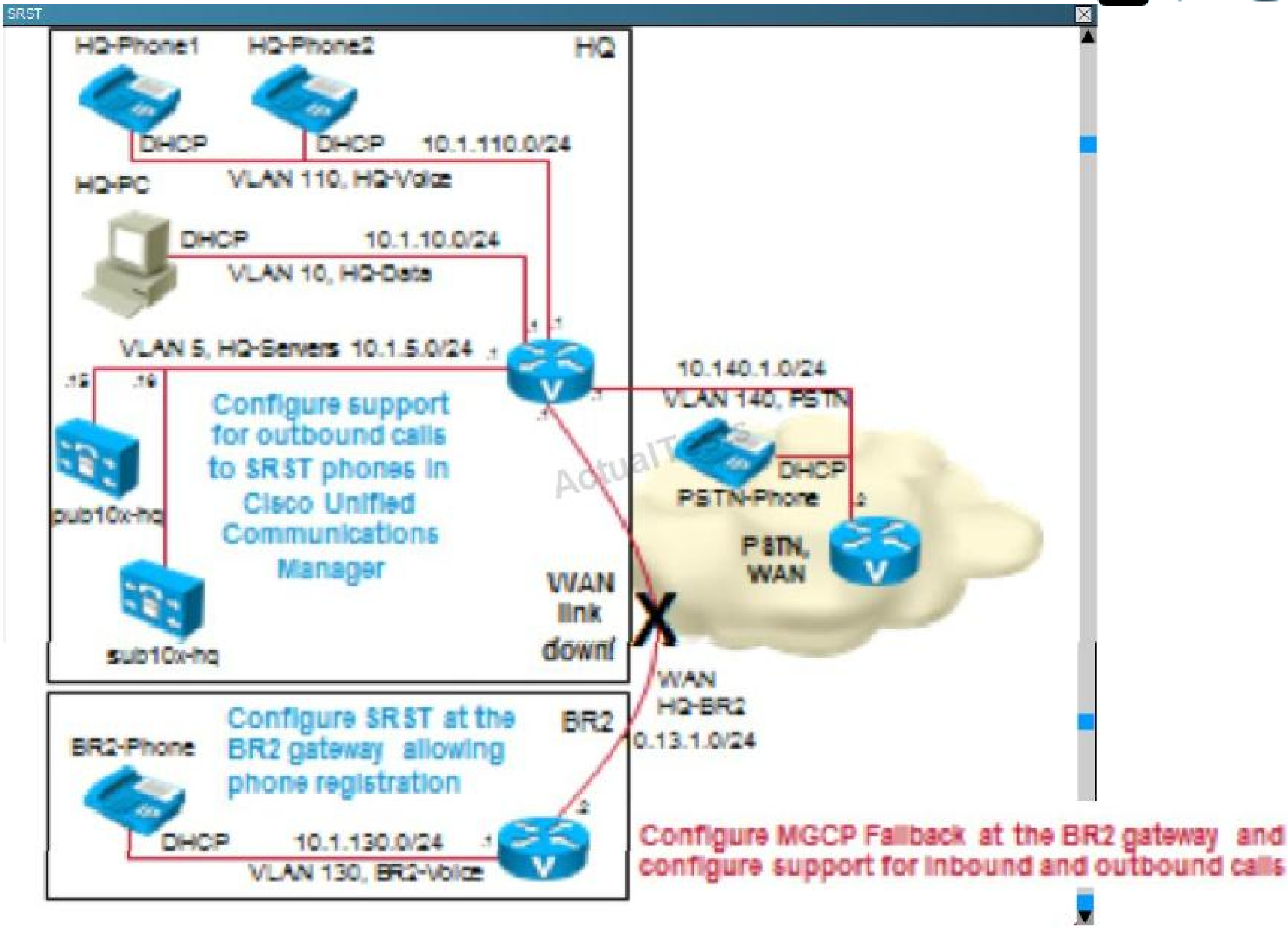


Locations (exhibit):

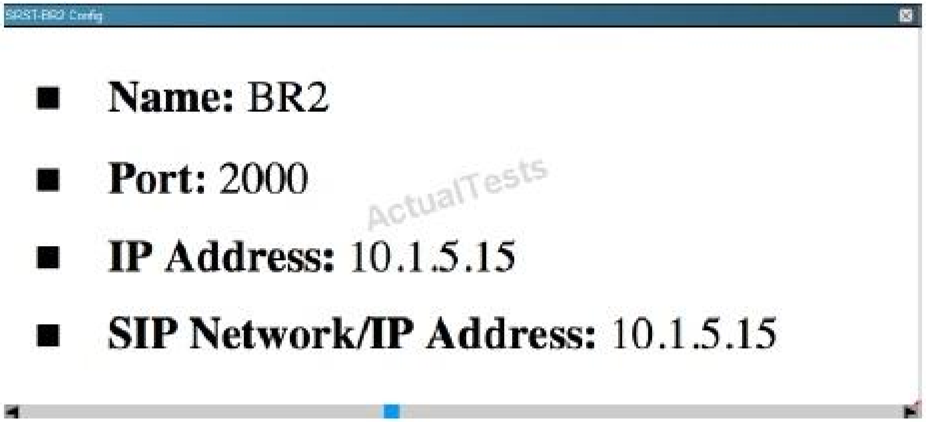


CSS (exhibit):

SRST (exhibit):


SRST
HQ-Phone1
HQ-Phone2
HQ
DHCP
DHCP
10.1.110.0/24
HQ-PC
VLAN 110, HQ-Voice
DHCP
10.1.10.0/24
VLAN 10, HQ-Data
VLAN 5, HQ-Servers 10.1.5.0/24
10.140.1.0/24
VLAN 140, PSTN
Configure support for outbound calls to SRST phones in Cisco Unified Communications Manager
DHCP
PSTN-Phone
PSTN, WAN
pub10x-hq
sub10x-hq
WAN link down!
X
WAN HQ-BR2
10.13.1.0/24
Configure SRST at the BR2 gateway allowing phone registration
BR2
BR2-Phone
DHCP
10.1.130.0/24
VLAN 130, BR2-Voice
Configure MGCP Fallback at the BR2 gateway and configure support for inbound and outbound calls

SRST-BR2-Config (exhibit):



BR2 Config (exhibit):

```
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
```

SRSTPSTNCall (exhibit):

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.

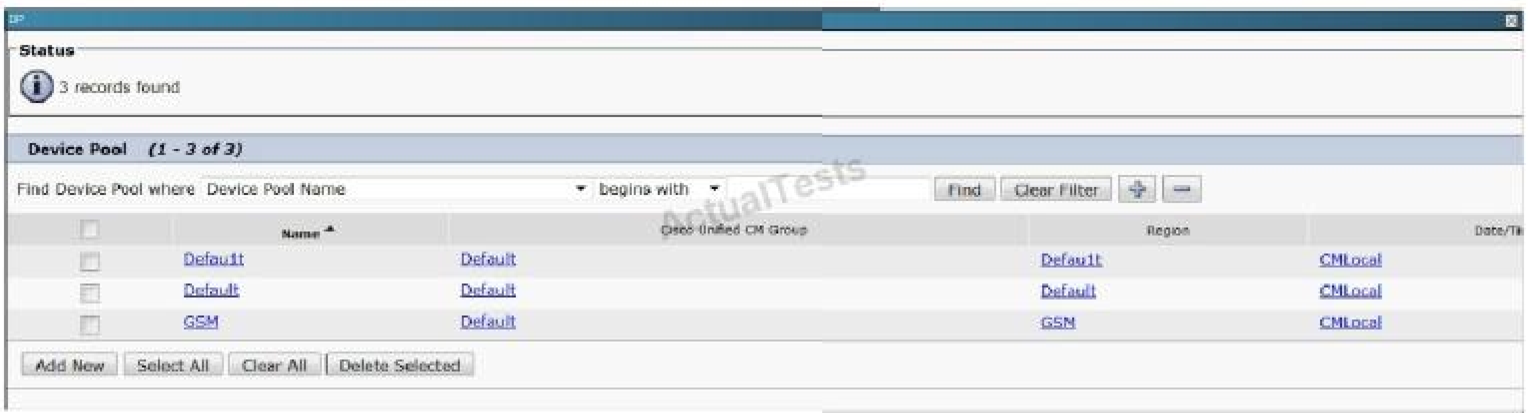
Correct Answer: AD
Section: (none)
Explanation

Explanation/Reference:

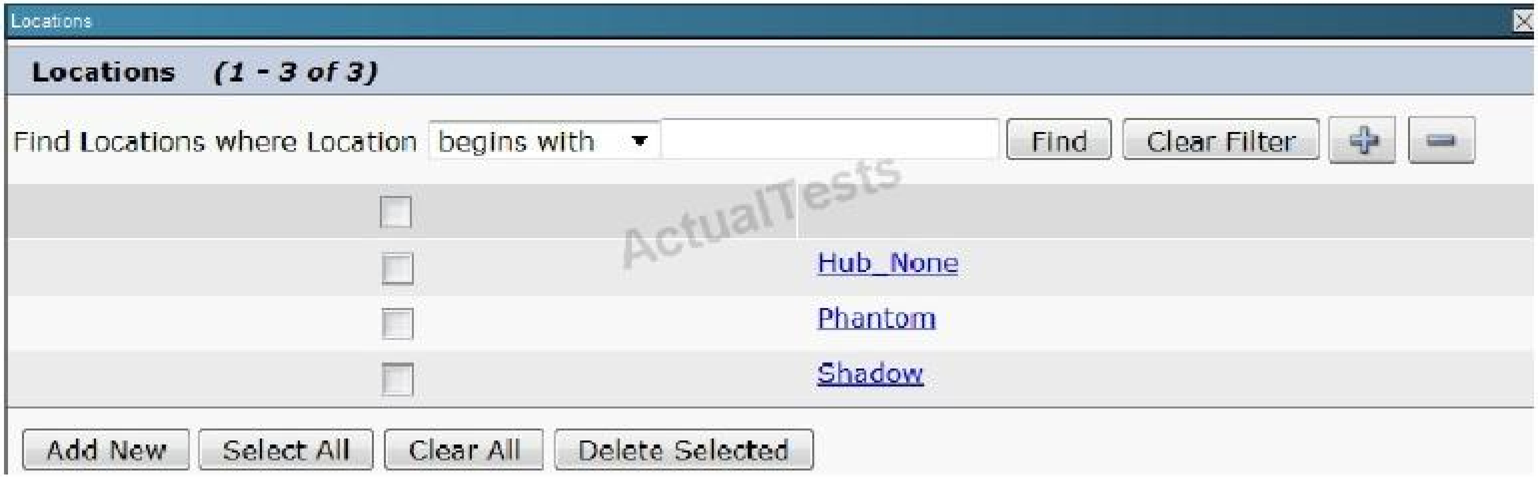
QUESTION 57
Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 9971 Video IP Phone. The Cisco VCS and TMS control the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows

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

DP (exhibit):

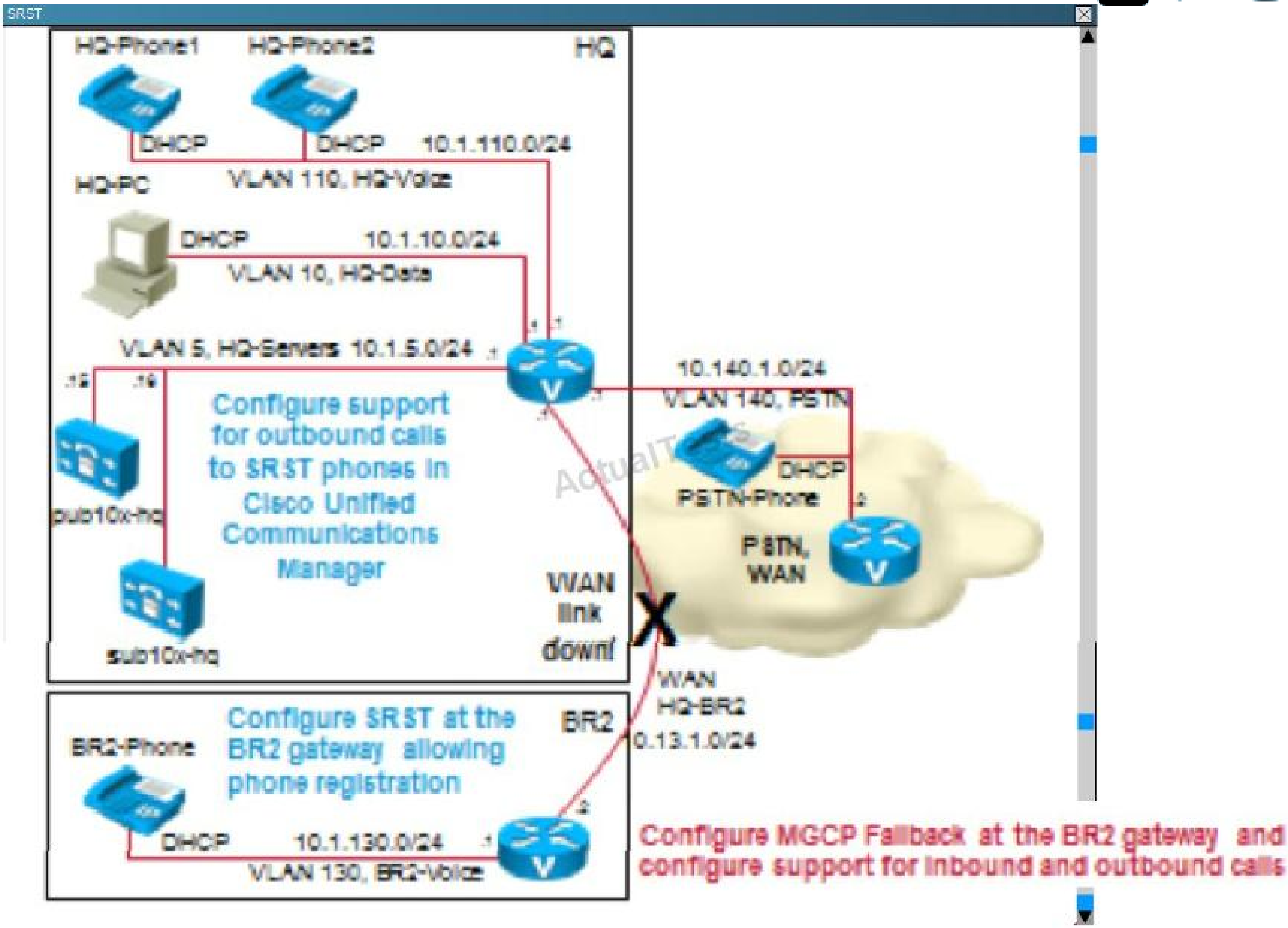


Locations (exhibit):



CSS (exhibit):

SRST (exhibit):


SRST
HQ-Phone1
HQ-Phone2
HQ
DHCP
DHCP
10.1.110.0/24
HQ-PC
VLAN 110, HQ-Voice
DHCP
10.1.10.0/24
VLAN 10, HQ-Data
VLAN 5, HQ-Servers 10.1.5.0/24
10.140.1.0/24
VLAN 140, PSTN
Configure support for outbound calls to SRST phones in Cisco Unified Communications Manager
DHCP
PSTN-Phone
pub10x-hq
PSTN, WAN
WAN link down!
sub10x-hq
WAN
HQ-BR2
10.13.1.0/24
Configure SRST at the BR2 gateway allowing phone registration
BR2
BR2-Phone
DHCP
10.1.130.0/24
VLAN 130, BR2-Voice
Configure MGCP Fallback at the BR2 gateway and configure support for inbound and outbound calls

SRST-BR2-Config (exhibit):

SRST-BR2 Config

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

BR2 Config (exhibit):

```
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
```

SRSTPSTNCall (exhibit):

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

Correct Answer: C
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.

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

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
```

Device Pool (exhibit):

Device Pool

Status

3 records found

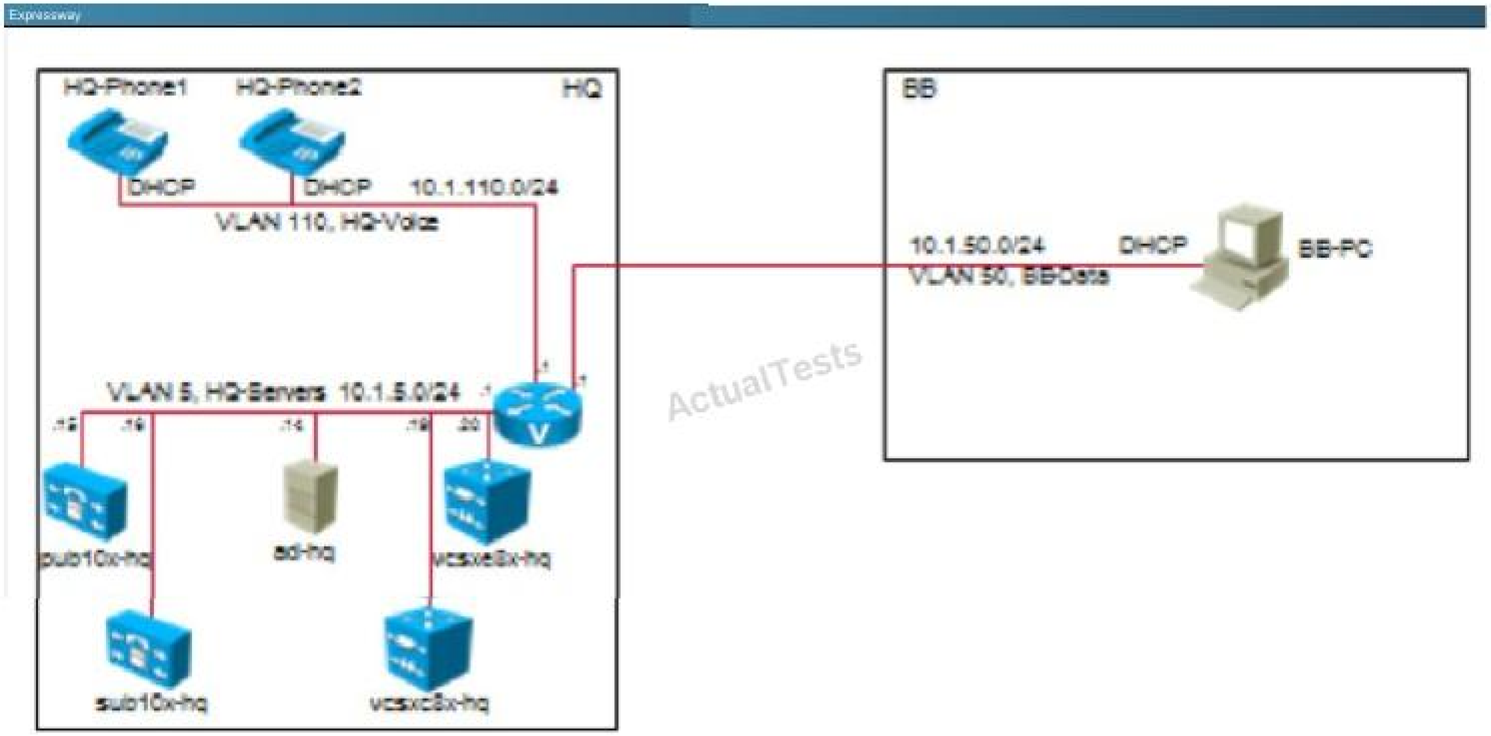
Device Pool ***(1 - 3 of 3)***

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

	Name	Cisco Unified CM Group	Region	Date/Ti
	Default	Default	Default	CMLocal
	Default	Default	Default	CMLocal
	GSM	Default	GSM	CMLocal

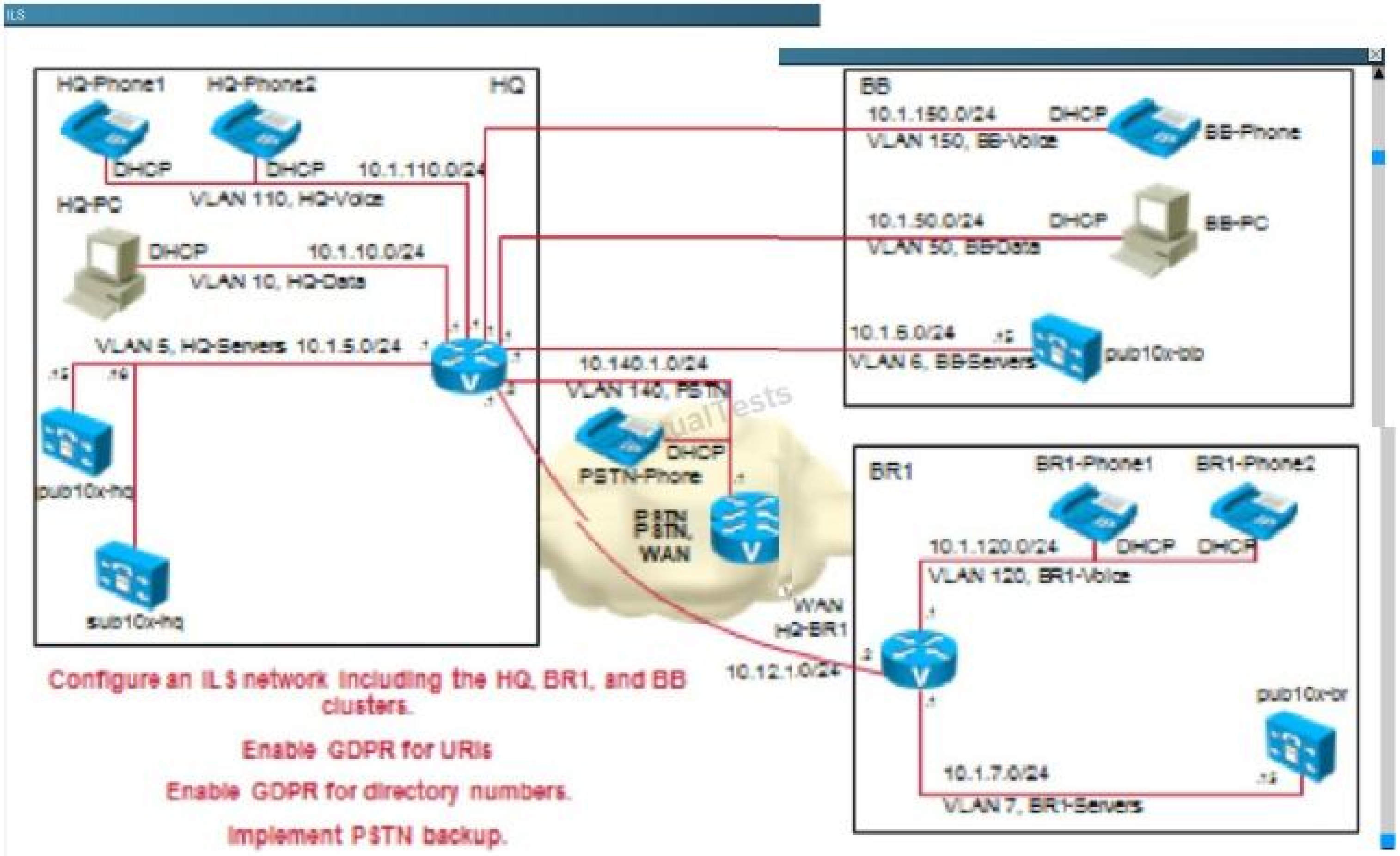
Add New | Select All | Clear All | Delete Selected

Expressway (exhibit):

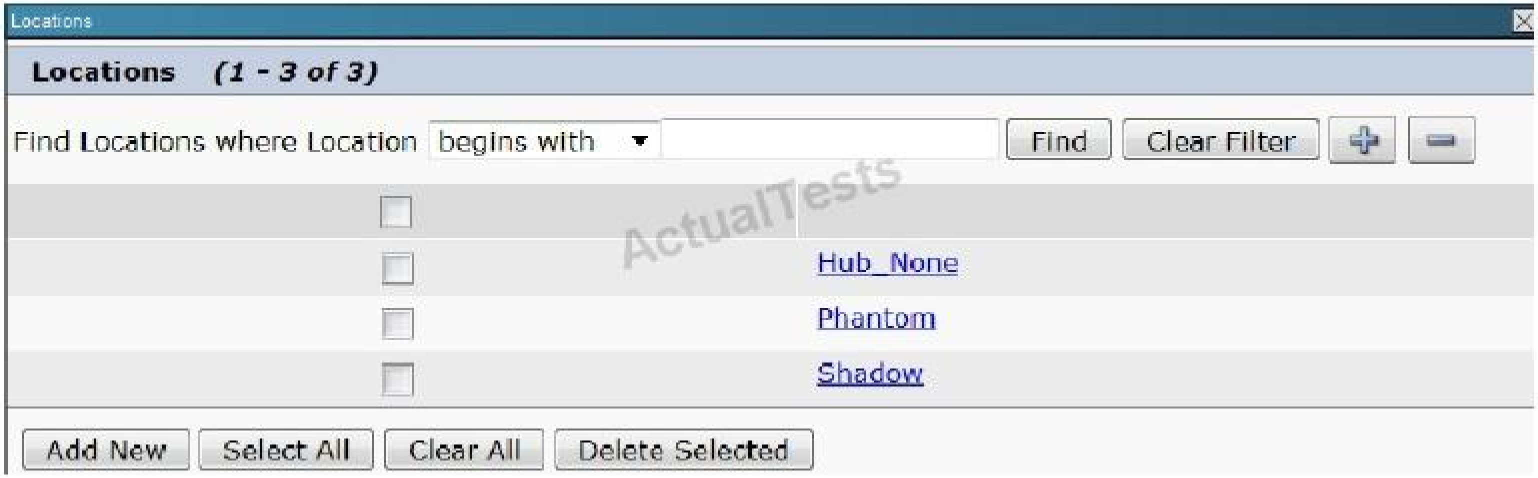


Expressway
HQ-Phone1
HQ-Phone2
HQ
DHCP
DHCP
10.1.110.0/24
VLAN 110, HQ-Voice
VLAN 5, HQ-Servers 10.1.5.0/24
.1
.1
.1
pub10x-hq
ad-hq
vcsxe8x-hq
sub10x-hq
vcsxc8x-hq
BB
10.1.50.0/24
DHCP
VLAN 50, BBData
BB-PC
ActualTests

ILS (exhibit):



Locations (exhibit):

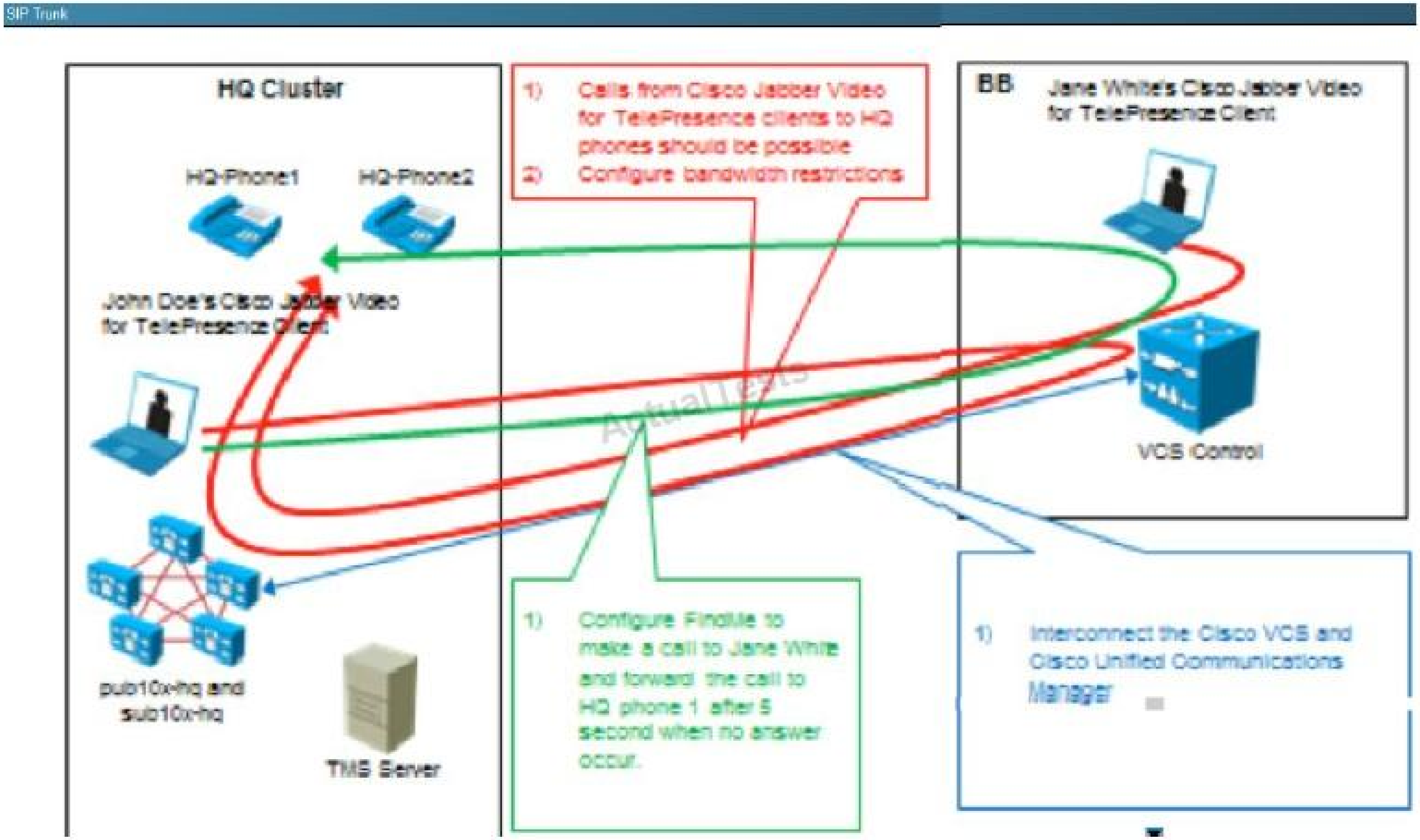


MRA (exhibit):

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

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 59
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 60
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 61
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 62
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 63
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 64
Refer to exhibit.

```
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)

    Class-map: class-default (match-any)
      10 packets, 560 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match: any
      Queueing
      queue limit 64 packets
      (queue depth/total drops/no-buffer drops/flowdrops) 0/0/0/0
      (pkts output/bytes output) 10/560
      Fair-queue: per-flow queue limit 16
```

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

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

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 65
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 66
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 67
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 68
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 69

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 70
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 71
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 reregister with the backup

Unified CM in the CallManager Group.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 72
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 73
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 74
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 75
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 76
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 77
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 78
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 79
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 80
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 81
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 82
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:

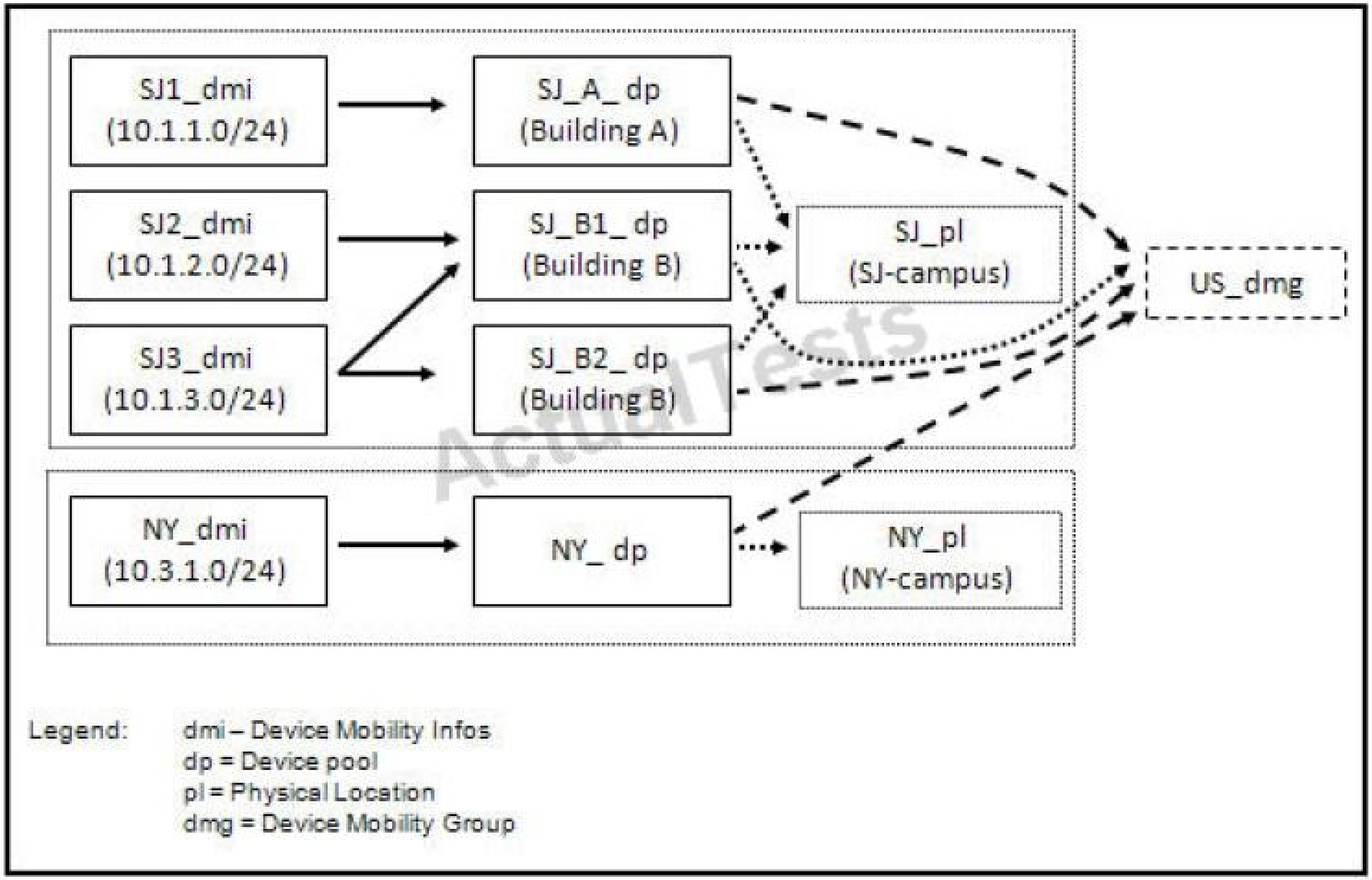
QUESTION 83
Which option is known as the location attribute that the global dialplan replication uses to advertise its dial plan information?

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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 84
Refer to the exhibit.



If an IP phone in San Jose roams to New York, which two IP phone settings will be modified by Device Mobility so that the phone can place and receive calls in New York? (Choose two.)

A. The physical locations are not different, so the configuration of the phone is not modified.

B. The physical locations are different, so the roaming-sensitive parameters of the roaming device pool are applied.

C. The device mobility groups are the same, so the Device Mobility-related settings are applied in addition to the roaming-sensitive parameters.

D. The Device Mobility information is associated with one or more device pools other than the home device pool of the phone, so one of the associated device pools is chosen based on a round-robin load-sharing algorithm.

E. The Device Mobility information is associated with the home device pool of the phone, so the phone is considered to be in its home location. Device Mobility will reconfigure the roaming-sensitive settings of the phone.

Correct Answer: BC
Section: (none)
Explanation

Explanation/Reference:

QUESTION 85
What happens when a user logs in using the Cisco Extension Mobility Service on a device for which the user has no user device profile?

A. The Extension Mobility log in fails.
B. The device takes on the default device profile for its type.
C. The user can log in but does not have access to any features, soft key templates, or button templates.
D. The device uses the first device profile assigned to the user in Cisco Unified Communications Manager.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 86
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 87
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 88
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 89
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 90
When a SIP trunk is added for Call Control Discovery, which statement is true?

A. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Enable SAF check box should be selected.
B. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Trunk Service Type should be Call Control Discovery.
C. The SIP trunk is added by selecting Call Control Discovery Trunk and then selecting SIP as the protocol to be used.
D. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The destination IP address field is configured as ‘SAF’ to indicate that this trunk is used for SAF.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 91
When an H.323 trunk is added for Call Control Discovery, which statement is true?

A. The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Enable SAF check box should be selected in the trunk configuration.
B. The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Trunk Service Type should be Call Control Discovery.
C. The H.323 trunk is added by selecting Call Control Discovery Trunk and then selecting H.323 as the protocol to be used.
D. The H.323 trunk is added by selecting H.323 Trunk, and selecting Inter-Cluster Trunk as the Device Protocol. The destination IP address field is configured as ‘SAF’ to indicate that this trunk is used for SAF.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 92
Which Cisco IOS command is used to verify that a SAF Forwarder that is registered with Cisco Unified Communications Manager has established neighbor relations with an adjacent SAF Forwarder?

A. show eigrp service-family ipv4 neighbors
B. show eigrp address-family ipv4 neighbors
C. show voice saf dndball
D. show saf neighbors
E. show ip saf neighbors

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 93
Which Cisco IOS command is used to verify that the Cisco Unified Communications Manager Express has registered with the SAF Forwarder?

A. show eigrp service-family ipv4 clients
B. show eigrp address-family ipv4 clients
C. show voice saf dndb all
D. show saf registration
E. show ip saf registration

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 94
Which statement about Service Advertisement Framework is true?

A. SAF requires that the EIGRP be configured on all routers, including non-SAF routers.
B. SAF requires that the EIGRP be configured only on SAF routers. Non-SAF routers act as an IP cloud.
C. SAF has no dependency on the underlying routing protocol, as long as it is a dynamic routing protocol.
D. SAF operates on any dynamic or static IP routing configuration. SAF is totally independent of the underlying routing protocol.

Correct Answer: D
Section: (none)
Explanation

Explanation/Reference:

QUESTION 95
What is the purpose of the local route group?

A. minimize PSTN costs
B. help in the selection of the PSTN egress gateway
C. eliminate the need for a route list
D. allow manipulation of digits at the cost point to egress

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 96
Which action configures PSTN backup for calls that are rejected by the gatekeeper CAC?

A. Configure AAR in Cisco Unified Communications Manager.
B. Configure CFUR in Cisco Unified Communications Manager.
C. Configure a route pattern, a route list, and route groups to a trunk and a gateway in Cisco Unified Communications Manager.
D. Configure a route pattern to a gateway in Cisco Unified Communications Manager.

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 97
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.

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 98
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

Correct Answer: C
Section: (none)
Explanation

Explanation/Reference:

QUESTION 99
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

Correct Answer: B
Section: (none)
Explanation

Explanation/Reference:

QUESTION 100
With Media Gateway Control Protocol configuration on the voice gateway, which three types of messages are involved in the call flow between the call agent and the voice gateway (Choose three.)

A. audit endpoint
B. modify endpoint
C. create connection
D. delete notification
E. restart in progress
F. end connections

Correct Answer: ACE
Section: (none)
Explanation

Explanation/Reference:

QUESTION 101
Which statement about TEHO is true?

A. The dial plan is simplified with local route groups.
B. Local route groups add complexity to the dial plan.
C. Toll charges can be reduced when TEHO is implemented with CAC.
D. Toll charges can be reduced when TEHO is implemented with MGCP fallback.

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference:

QUESTION 102
How are Cisco IP Phones directly configured to utilize local route groups?

A. with Cisco Unified Communications Manager device pools
B. with Cisco Unified Communications Manager CSS and partitions
C. with Cisco Unified Communications Manager regions
D. with Cisco Unified Communications Manager locations
E. with Cisco Unified Communications Manager AAR

Correct Answer: A
Section: (none)
Explanation

Explanation/Reference: