

Cisco.300-075.v2018-03-13.q93

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NEW QUESTION: 1

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. ccm-manager secondary dialtone 0
- D. dial-peer voice 1 pots
secondary dialtone 0

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 2

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 3

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):

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

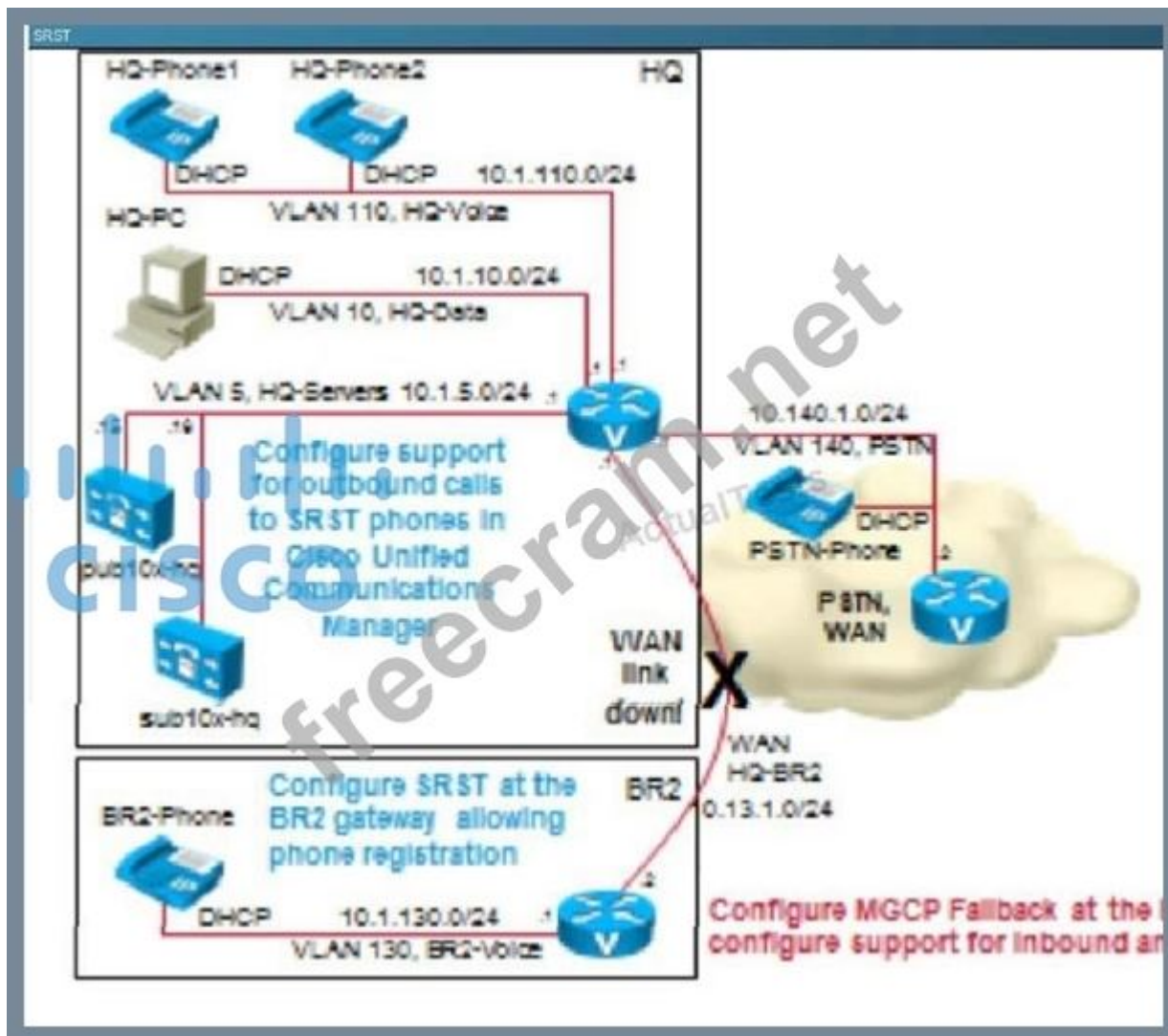
Locations (exhibit):

Location	Default	Region	Date/Time
Hub_None	Default	Default	CMLocal
Phantom	Default	Default	CMLocal
Shadow	Default	Default	CMLocal

CSS (exhibit):

CSS Name	Default	Region	Date/Time
All-Devices	Default	Default	CMLocal
All-Devices	Default	Default	CMLocal
All-Devices	Default	Default	CMLocal

SRST (exhibit):



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. 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

B. 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

C. dial-peer voice 1 pots

incoming called-number 228822...

direct-inward-dial

port 0/0/0:13

D. The router does not need to be configured.

E. dial-peer voice 1 pots

incoming called-number 228822...

direct-inward-dial

port 0/0/0:15

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 4

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

A. 6

B. 10

C. 1

D. 4

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 5

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

A. A VCS Expressway located in the public network or DMZ acts as the firewall traversal client.

B. VCS supports either the Assent or the H.460.18/19 protocol for H.323 traversal calls.

C. If the Assent protocol is configured, a TCP/TLS connection is established from the traversal client to the traversal server for SIP signaling.

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

E. VCS supports either the Assent or the H.460.18/19 protocol for SIP traversal calls.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 6

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 7

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. Redirect CCD advertising and requesting services to another Cisco Unified Communications Manager.
- C. Place the Cisco Unified Communications Manager node in standby mode.
- D. Delete the trunk from the CCD requesting service node.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 8

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:

The screenshot shows the configuration for a Local Zone Search Rule. The interface includes a left sidebar with labels: Mode, Pattern type, Pattern string, Pattern behavior, Replace string, On successful match, Target, and State. The main configuration area on the right contains the following settings:

- Alias pattern match: (dropdown menu)
- Pattern type: Regex (dropdown menu)
- Pattern string: (*)@widgets.com (text input)
- Pattern behavior: Replace (dropdown menu)
- Replace string: \d+ (text input)
- On successful match: Continue (dropdown menu)
- Target: Local Zone (dropdown menu)
- State: Enabled (dropdown menu)

A large, semi-transparent watermark reading "ActualTuts.com" and "freecramp" is overlaid on the center of the interface. A "CISCO" logo is visible at the bottom center.

B:

Mode: Alias pattern match ⓘ

Pattern type: Regex ⓘ

Pattern string: (+)@widgets.com ⓘ

Pattern behavior: Replace ⓘ

Replace string: \1

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: \5

On successful match: Continue ⓘ

Target: Local Zone ⓘ

State: Enabled ⓘ

D:

Mode: Alias pattern match ⓘ

Pattern type: Regex ⓘ

Pattern string: (+)@widgets.com ⓘ

Pattern behavior: Replace ⓘ

Replace string: \1

On successful match: Continue ⓘ

Target: Local Zone ⓘ

State: Enabled ⓘ

A. Option C

B. Option B

C. Option D

D. Option A

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 9

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

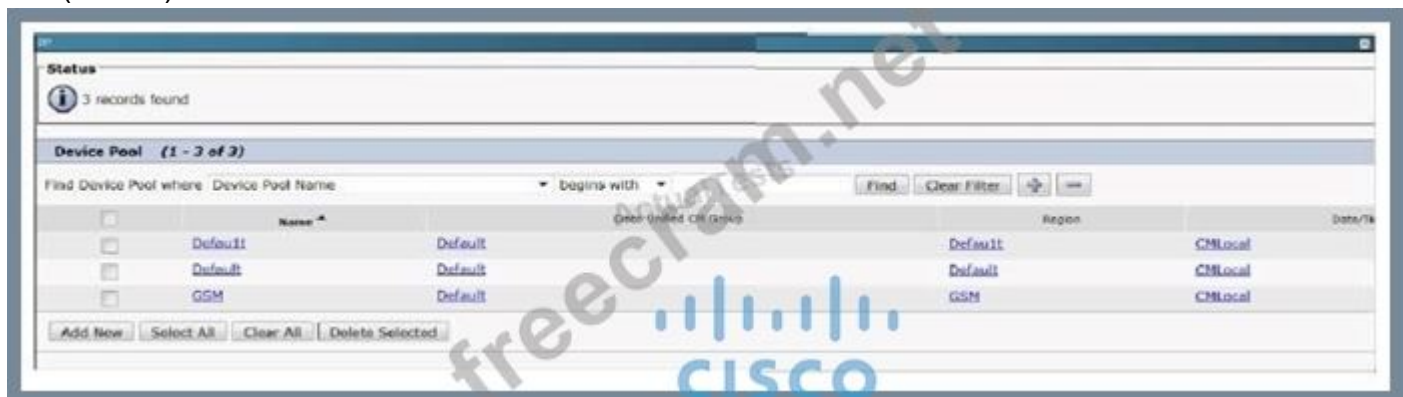
DX650 Configuration (exhibit):

DX650 Configuration	
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 >

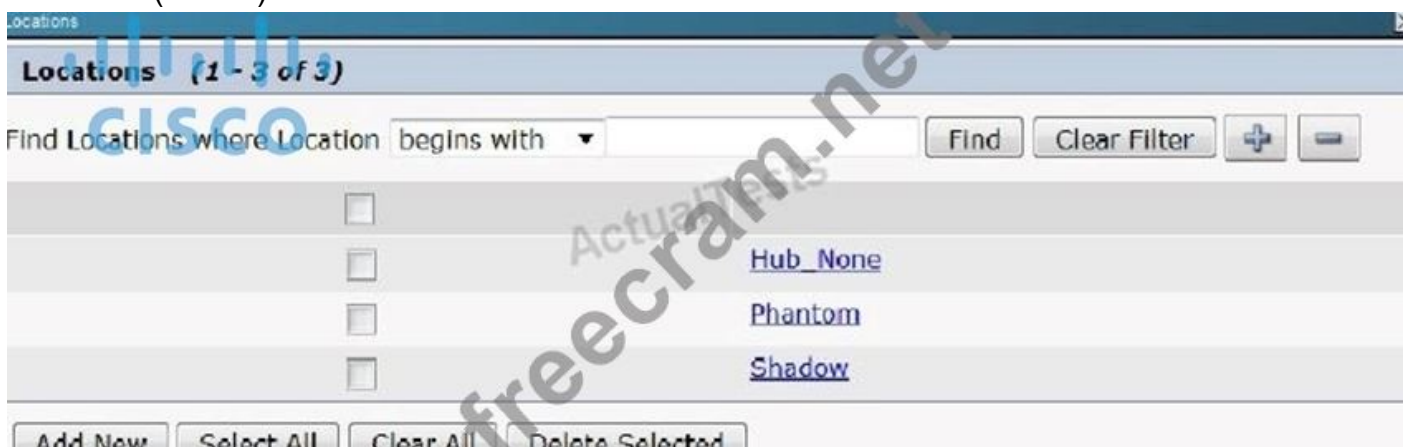
MRGL (exhibit):



DP (exhibit):



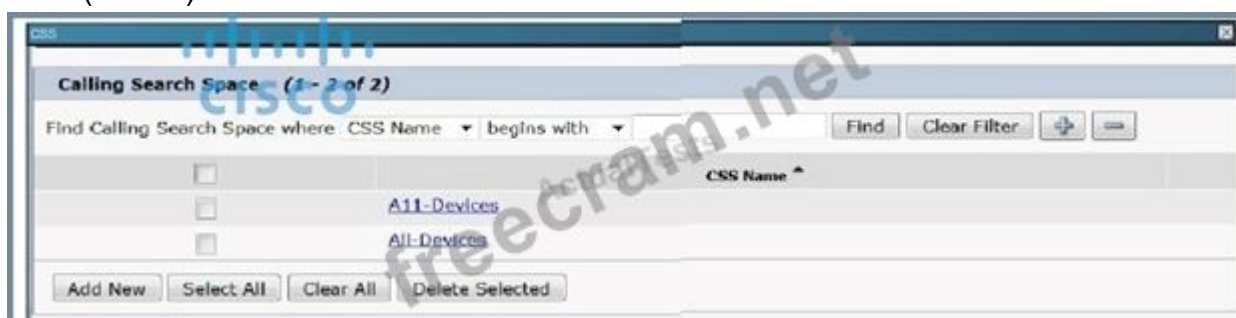
Locations (exhibit):



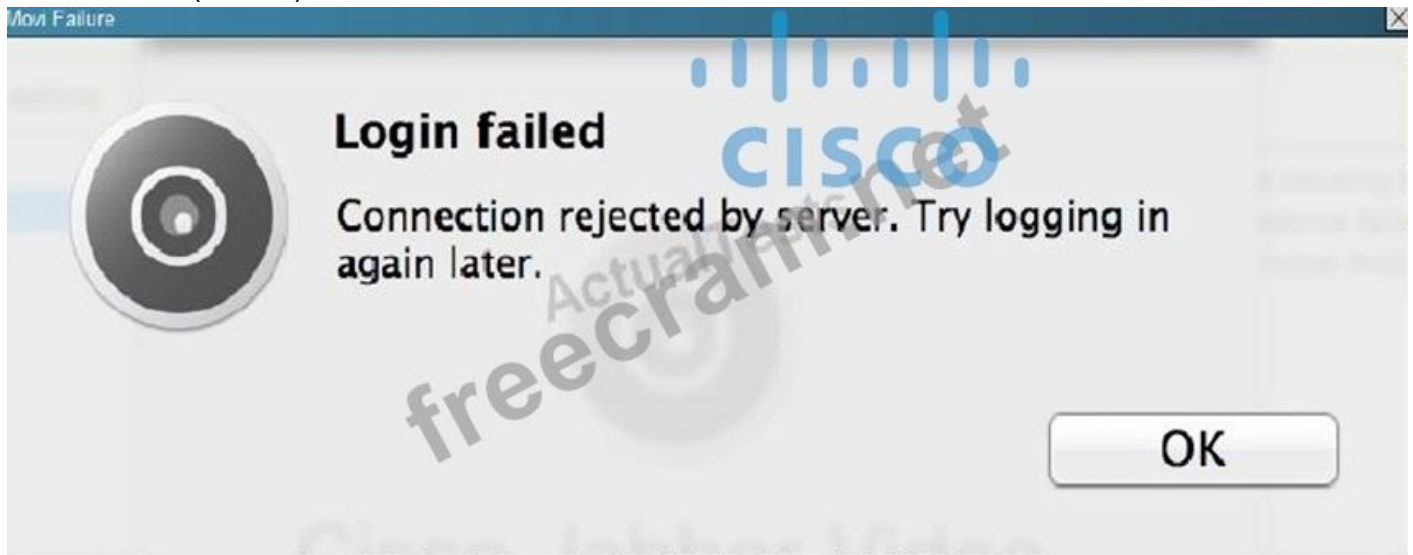
AARG (exhibit):



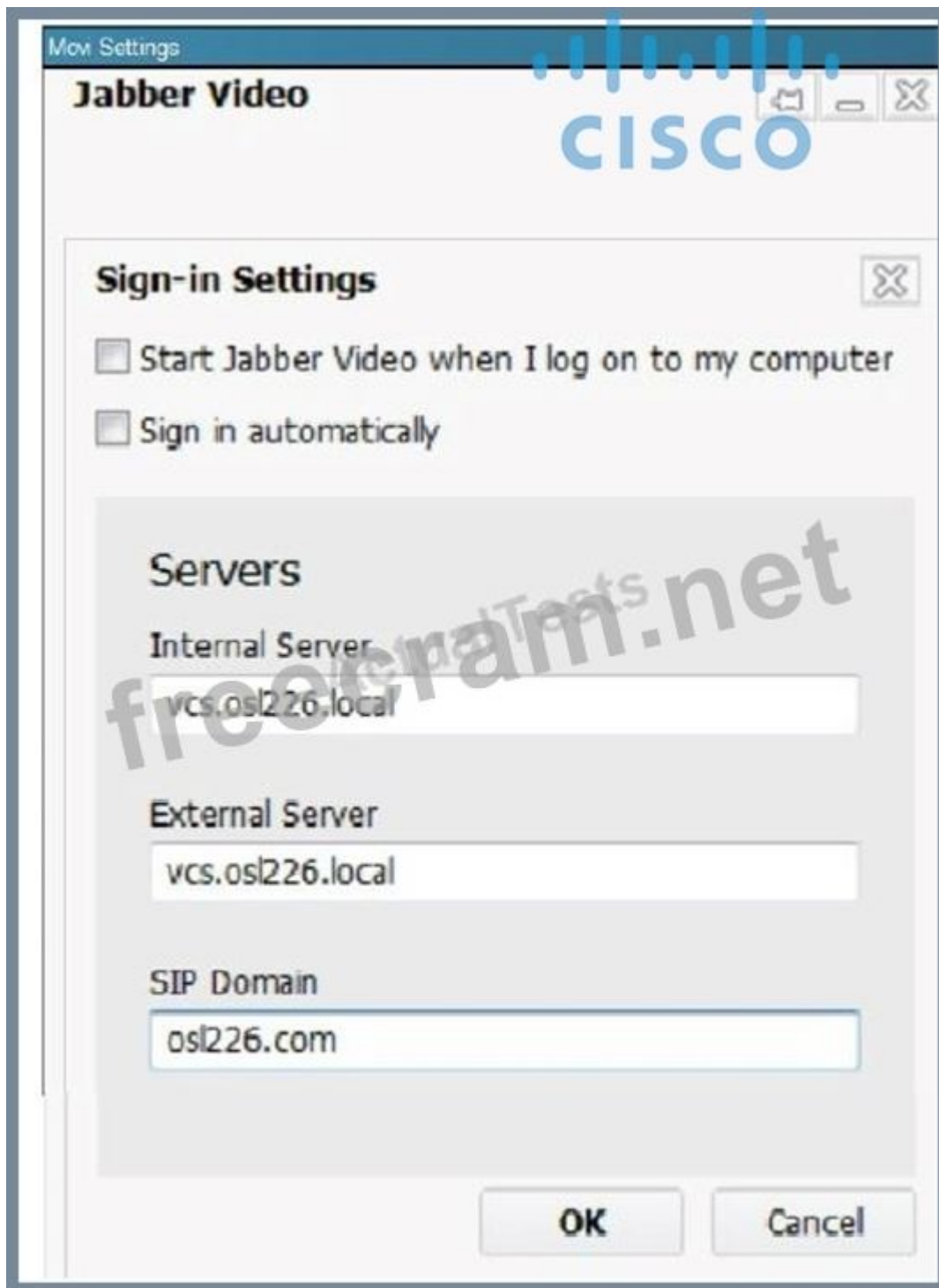
CSS (exhibit):



Movi Failure (exhibit):



Movi Settings (exhibit):



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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 10

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 11

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 12

Which code snippet is required for SAF to be initialized?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 13

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 waits for the primary Cisco Unified Communications Manager server to come alive.

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 14

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 15

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 16

When a SIP trunk is added for Call Control Discovery, which statement is true?

- A. The SIP trunk is added by selecting Call Control Discovery Trunk and then selecting SIP as the protocol to be used.
- B. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Enable SAF check box should be selected.
- C. 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.
- D. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Trunk Service Type should be Call Control Discovery.

Answer: ([SHOW ANSWER](#))

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NEW QUESTION: 17

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

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

Answer: ([**SHOW ANSWER**](#)**)**

NEW QUESTION: 18

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

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

Answer: ([**SHOW ANSWER**](#)**)**

NEW QUESTION: 19

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

- A. Directory URI replication does not need to be enabled individually for each cluster.
- 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. 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.
- D. Cisco Unified Communications Manager uses the ILS to support intercluster URI dialing.
- 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.

Answer: ([**SHOW ANSWER**](#)**)**

NEW QUESTION: 20

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

- A. keepalive
- B. limit-dn
- C. max-ephones
- D. call-manager-fallback
- E. ip source-address

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 21

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 22

Which option is a benefit of implementing CFUR?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 23

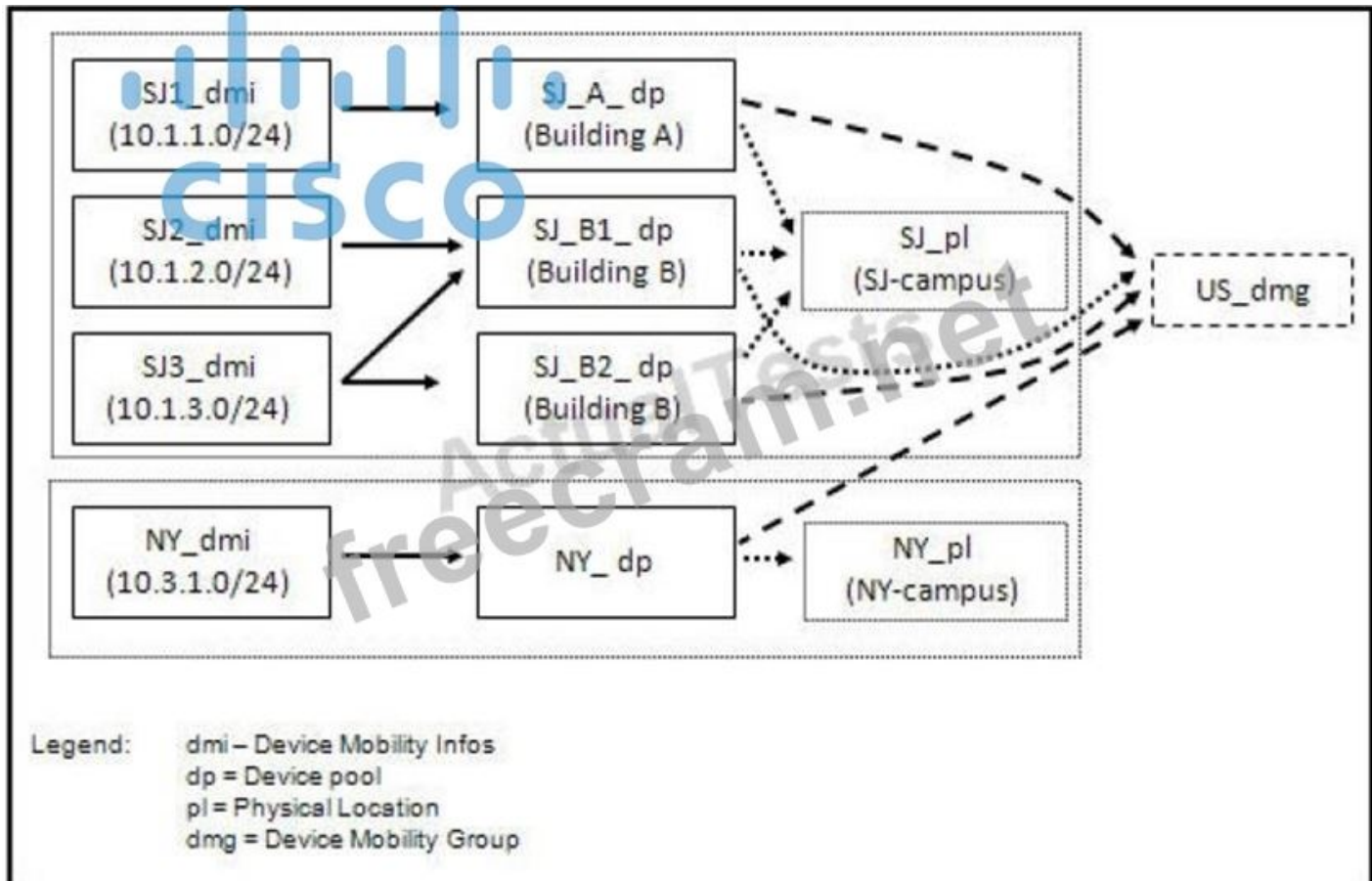
When implementing a dial plan for multisite deployments, what must be present for SRST to work successfully?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 24

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 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.
- B. 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.
- C. The physical locations are different, so the roaming-sensitive parameters of the roaming device pool are applied.
- D. The device mobility groups are the same, so the Device Mobility-related settings are applied in addition to the roaming-sensitive parameters.
- E. The physical locations are not different, so the configuration of the phone is not modified.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 25

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

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

D. _collab-edge._tls.example.com

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 26

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 27

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 TFTP service on the primary call manager.

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 Unified CM service on the primary call manager. Devices should reregister with the backup Unified CM in the CallManager Group.

- D. 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.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 28

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

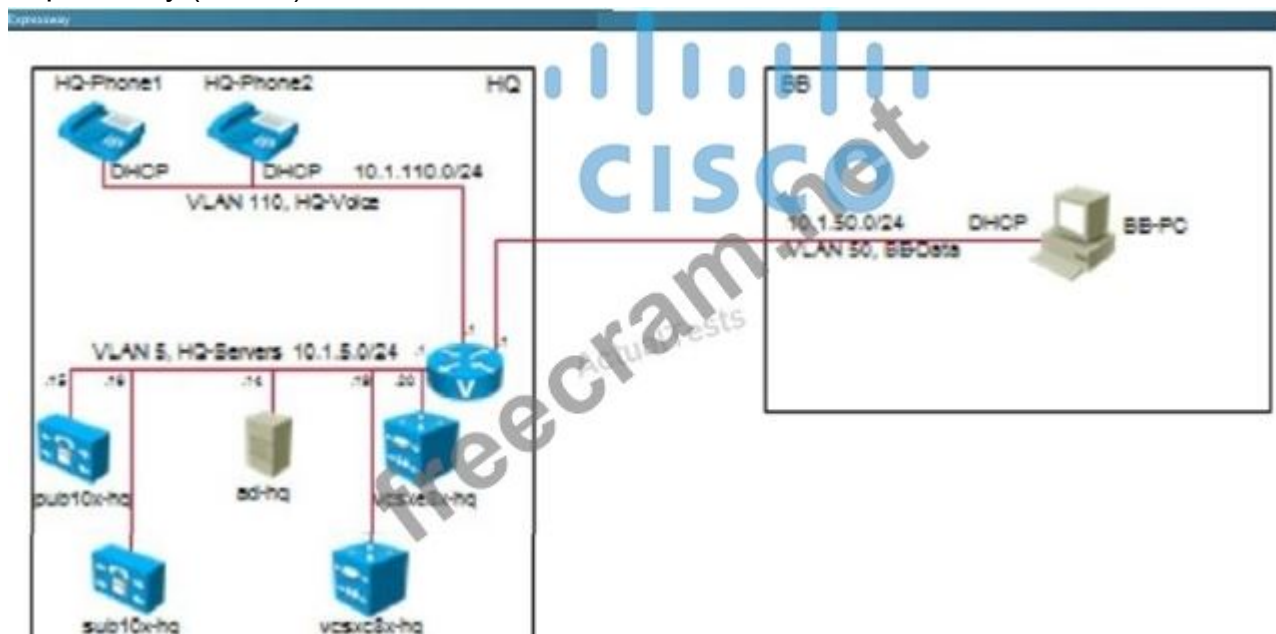
Device Pool (1 - 3 of 3)

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

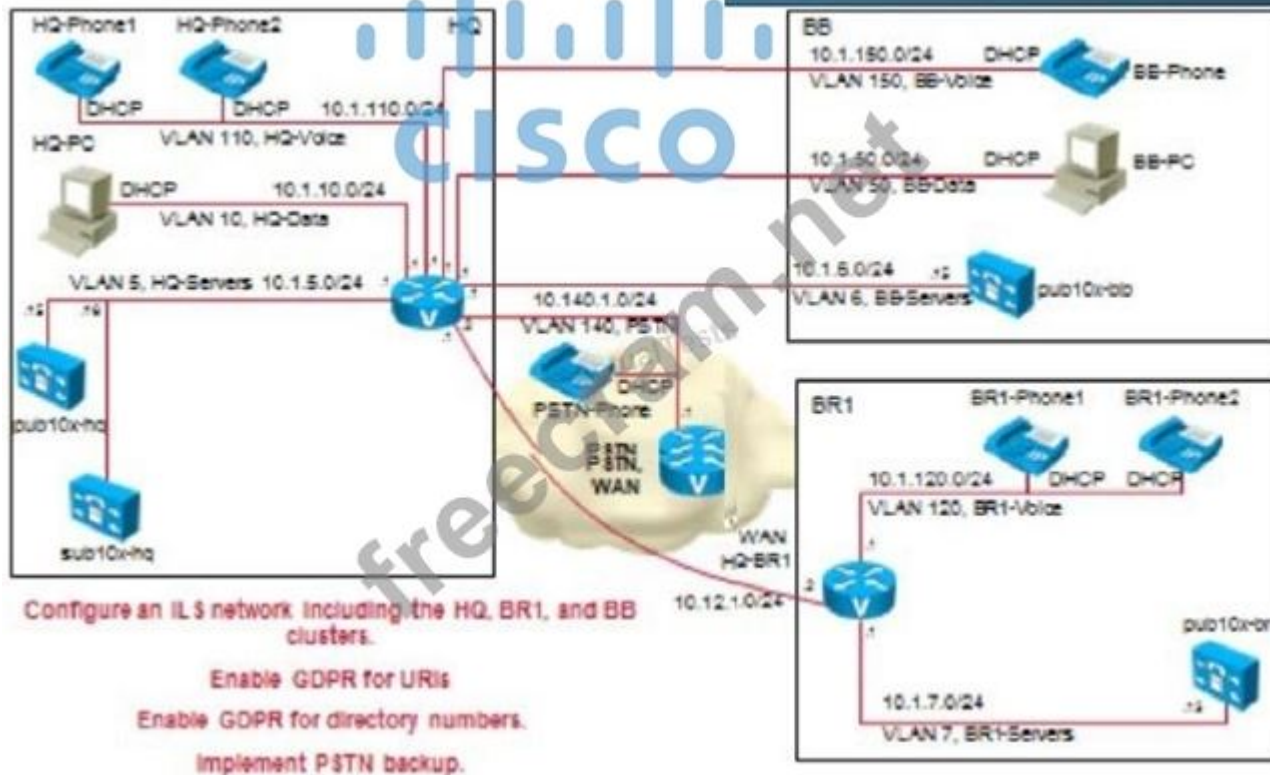
	Name	Device Pool Group	Region	CM Local
☐	Default	Default	Default	CM Local
☐	Default	Default	Default	CM Local
☐	GSM	Default	GSM	CM Local

Add New Select All Clear All Delete Selected

Expressway (exhibit):



ILS (exhibit):



Locations (exhibit):

Locations (1 - 3 of 3)

Find Locations where Location begins with Find Clear Filter

☐	Hub
☐	None
☐	Phantom
☐	Shadow

Add New Select All Clear All Delete Selected

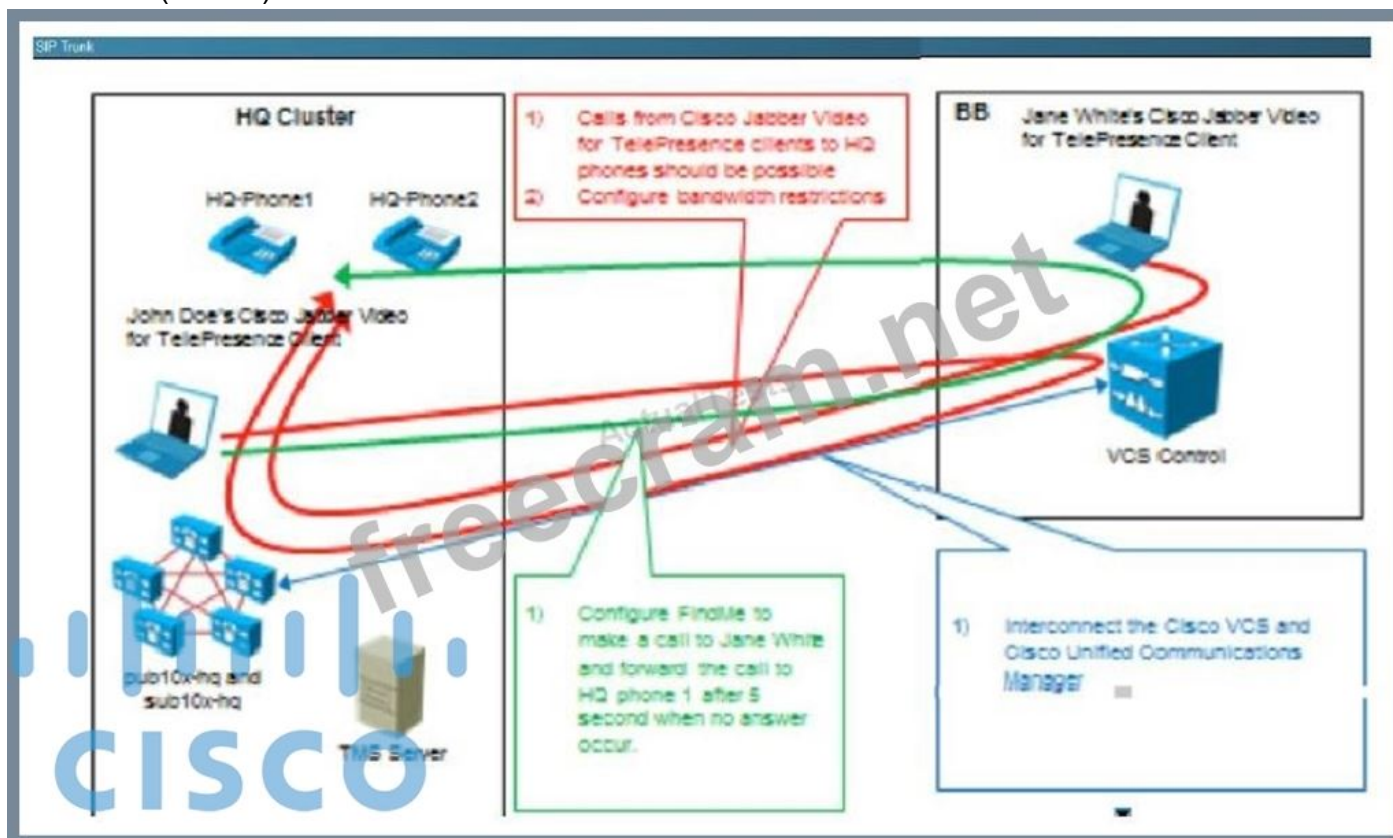
MRA (exhibit):

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

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 internal DNS Service (SRV) records need to be updated on the DNS Server.

B. Add the appropriate DNS SRV for the Internet entries on the DNS Server.

- C. The DNS AOR records are wrong.
- D. The DNS server has the wrong IP address.
- E. Flush the DNS Cache on the client.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 29

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 30

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

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

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 31

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

Answer: ([SHOW ANSWER](#))

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NEW QUESTION: 32

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 33

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 34

What is the purpose of the local route group?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 35

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 36

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 37

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 38

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 39

Which statement about TEHO is true?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 40

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):

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

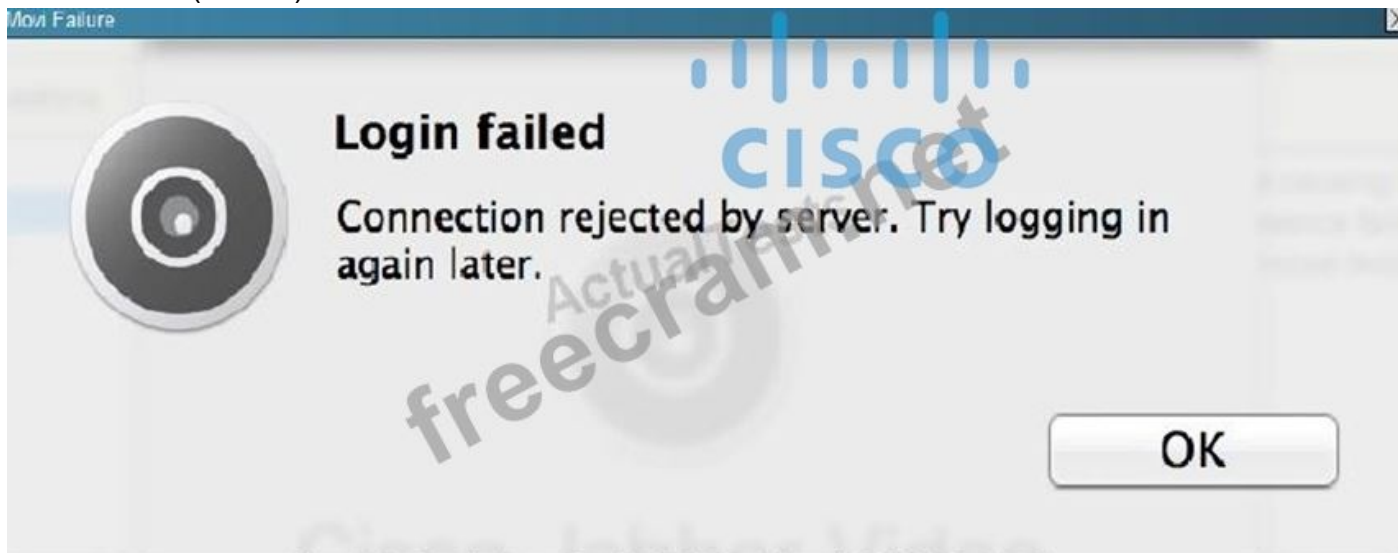
Locations (exhibit):

Name	Hub	Location
Hub_None		
Phantom		
Shadow		

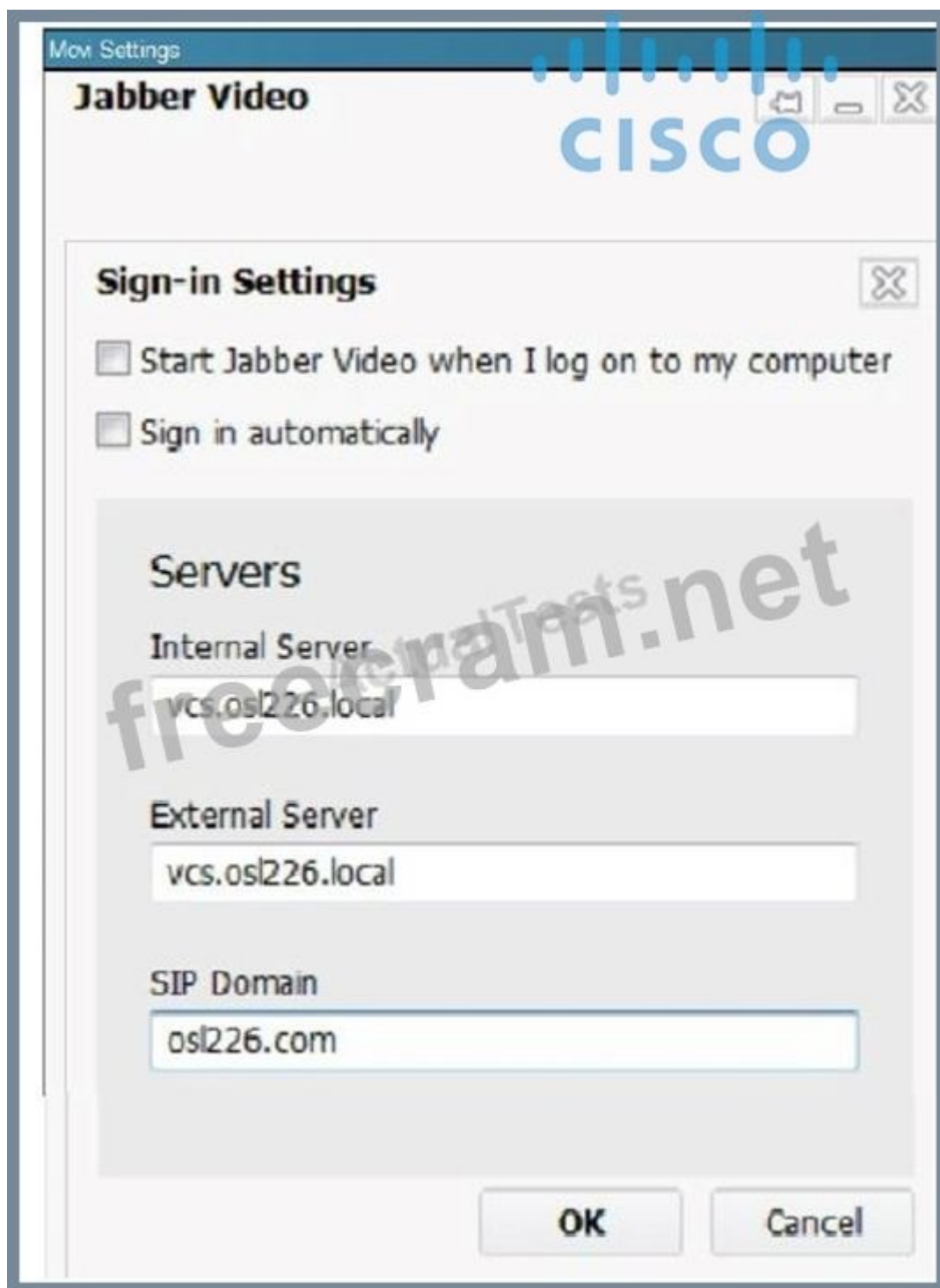
CSS (exhibit):



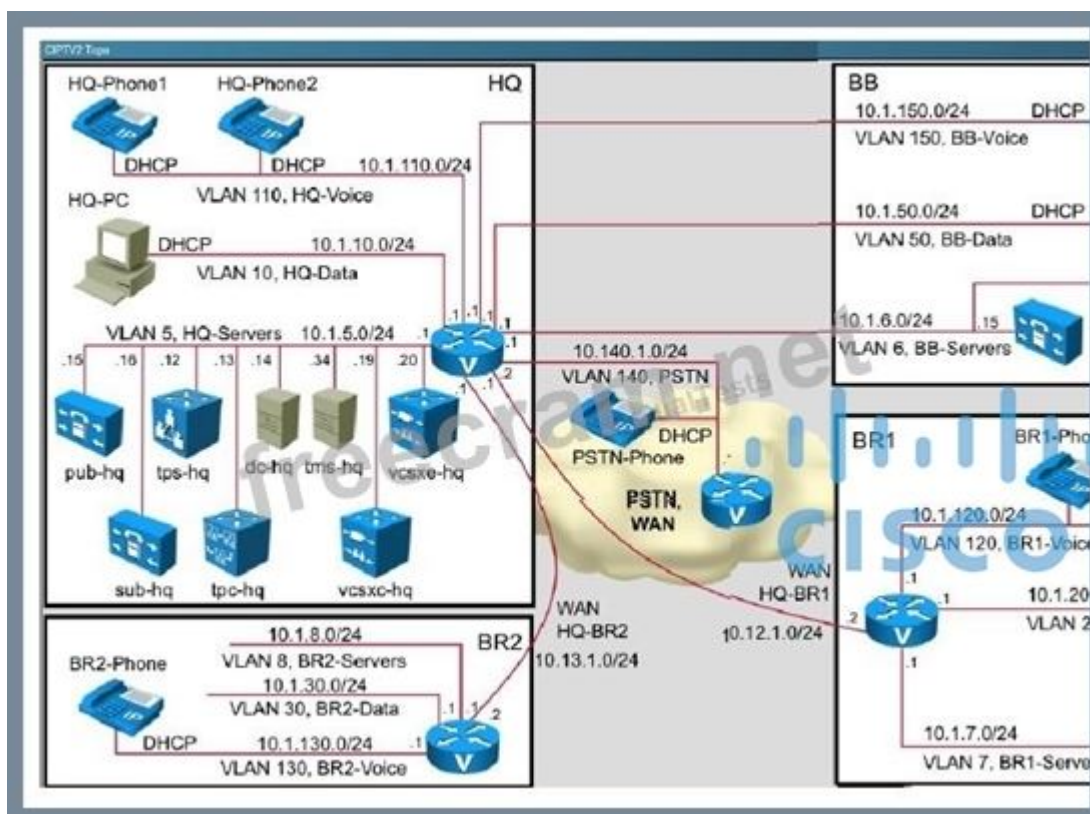
Movi Failure (exhibit):



Movi Settings (exhibit):



CIPTV2 Topology (exhibit):



Subzone (exhibit):

Links		You are here: Configure			
Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls
☐ DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0
☐ DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0
☐ DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0
☐ SubZone001ToDefaultSZ	HQ	DefaultSubZone			0
☐ SubZone001ToTraversalSZ	HQ	TraversalSubZone			0
☐ TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0
☐ VCS HQ - toHQ	HQ	to HQ	to HQ pipe		1

Links (exhibit):

Links

</

Pipe (exhibit):

Pipe
Name: <u>to HQ pipe</u>
Total Bandwidth available – Bandwidth restriction:
Total Bandwidth available – Total bandwidth limit
Calls through this pipe – Bandwidth restriction: Lin
Calls through this pipe – Per call bandwidth limit (I

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 41

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. the QoS service parameter in Cisco Unified Communications Manager

- B. on the actual Cisco phone itself because the DSCP setting is not part of its configuration file downloaded at registration
- C. the service parameters in the VCS Control
- D. 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.
- E. on the MGCP router at the edge of both networks

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 42

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 43

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 44

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 45

What is the standard Layer 3 DSCP media packet value that should be set for Cisco TelePresence endpoints?

- A. CS4 (32)
- B. AF41 (34)

C. CS3 (24)

D. EF (46)

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 46

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 Cisco Unified Communications Manager locations.

B. Configure Cisco Unified Communications Manager AAR.

C. Configure Cisco Unified Communications Manager RSVP-enabled locations.

D. Configure an SIP trunk to the ISR.

Answer: ([SHOW ANSWER](#))

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<https://www.examdiscuss.com/Cisco/exam/300-075/premium/> (220 Q&As Dumps, **35%OFF**

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NEW QUESTION: 47

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

A. 3

B. 2

C. 1

D. 5

E. 4

F. 6

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 48

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 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 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 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="TraverseallZone" 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 D
- B. Exhibit A
- C. Exhibit E
- D. Exhibit C
- E. Exhibit B

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 49

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 50

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

Answer: ([SHOW ANSWER](#))

Explanation/Reference:

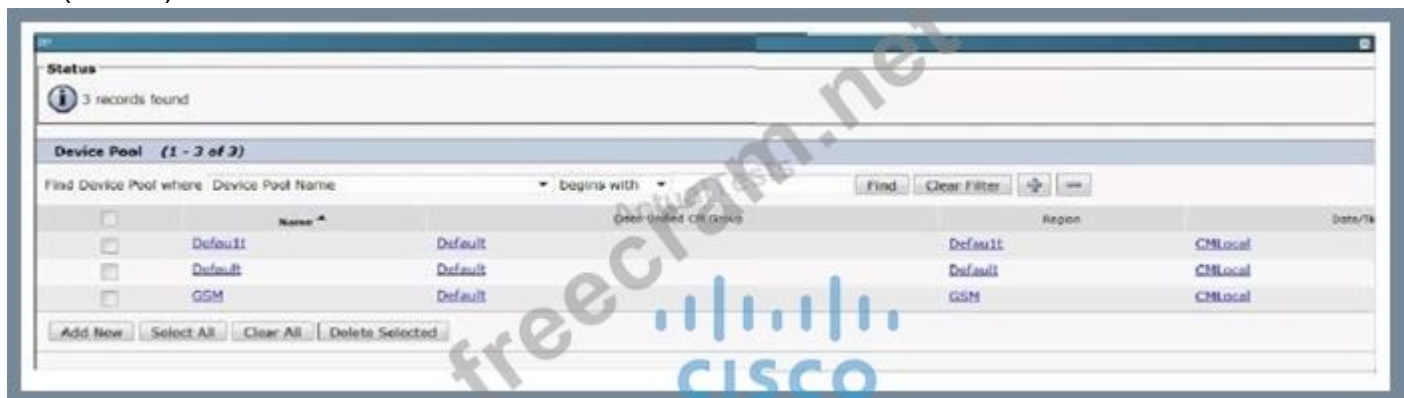
NEW QUESTION: 51

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.

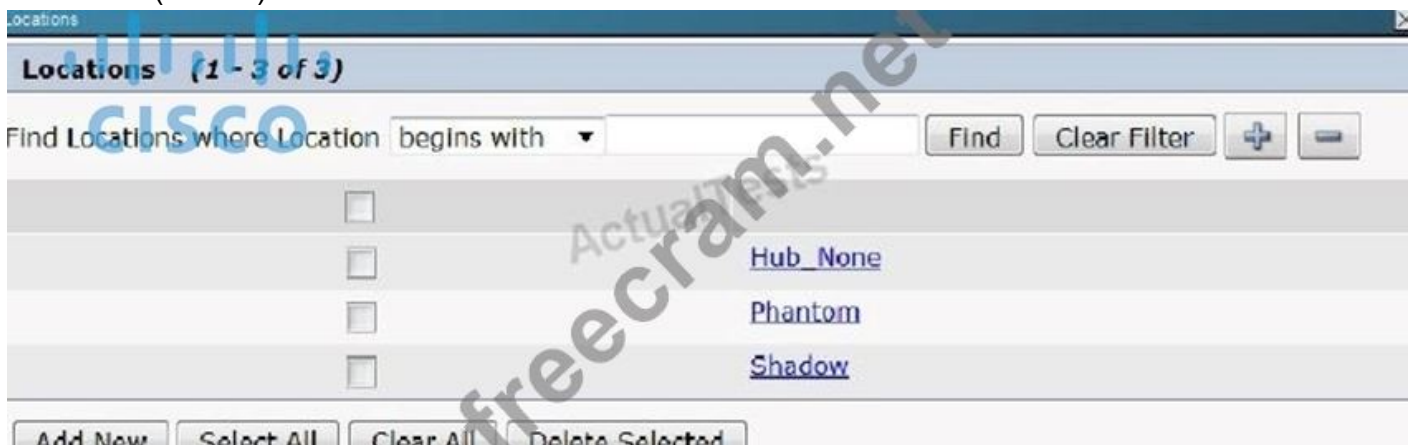
A third collaboration call fails between the backbone site and the HQ site. After reviewing the exhibits, which of the following reasons could be causing this failure?

DP (exhibit):



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

Locations (exhibit):



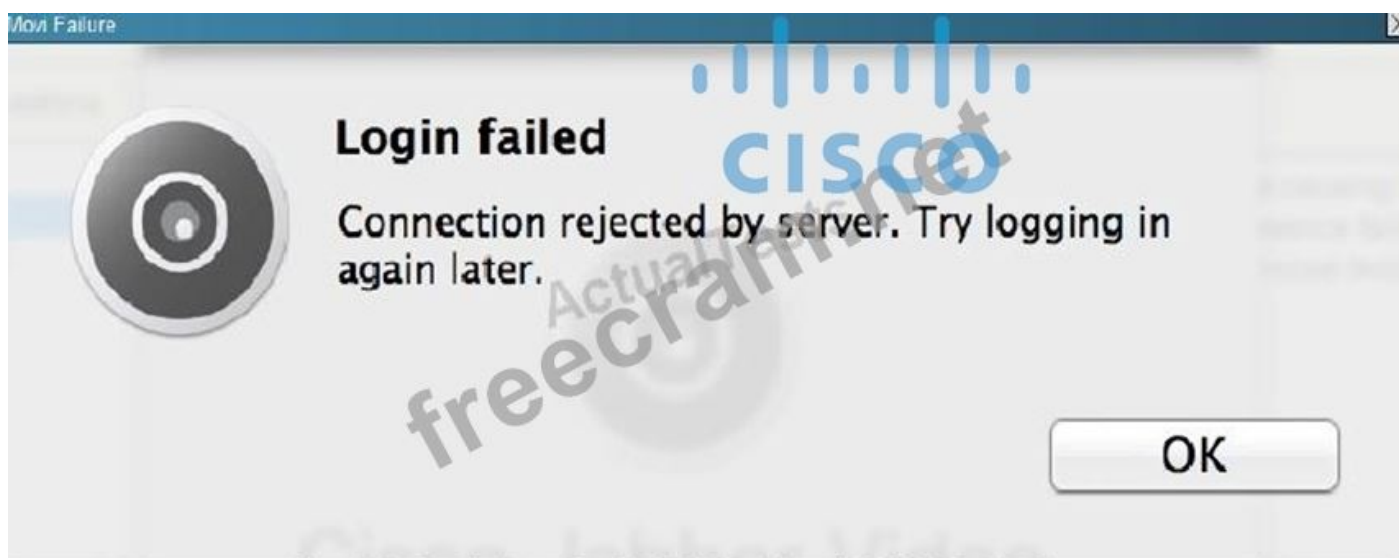
Name	Hub	Location
Hub_None	Hub_None	Phantom
Phantom	Hub_None	Shadow
Shadow	Hub_None	

CSS (exhibit):

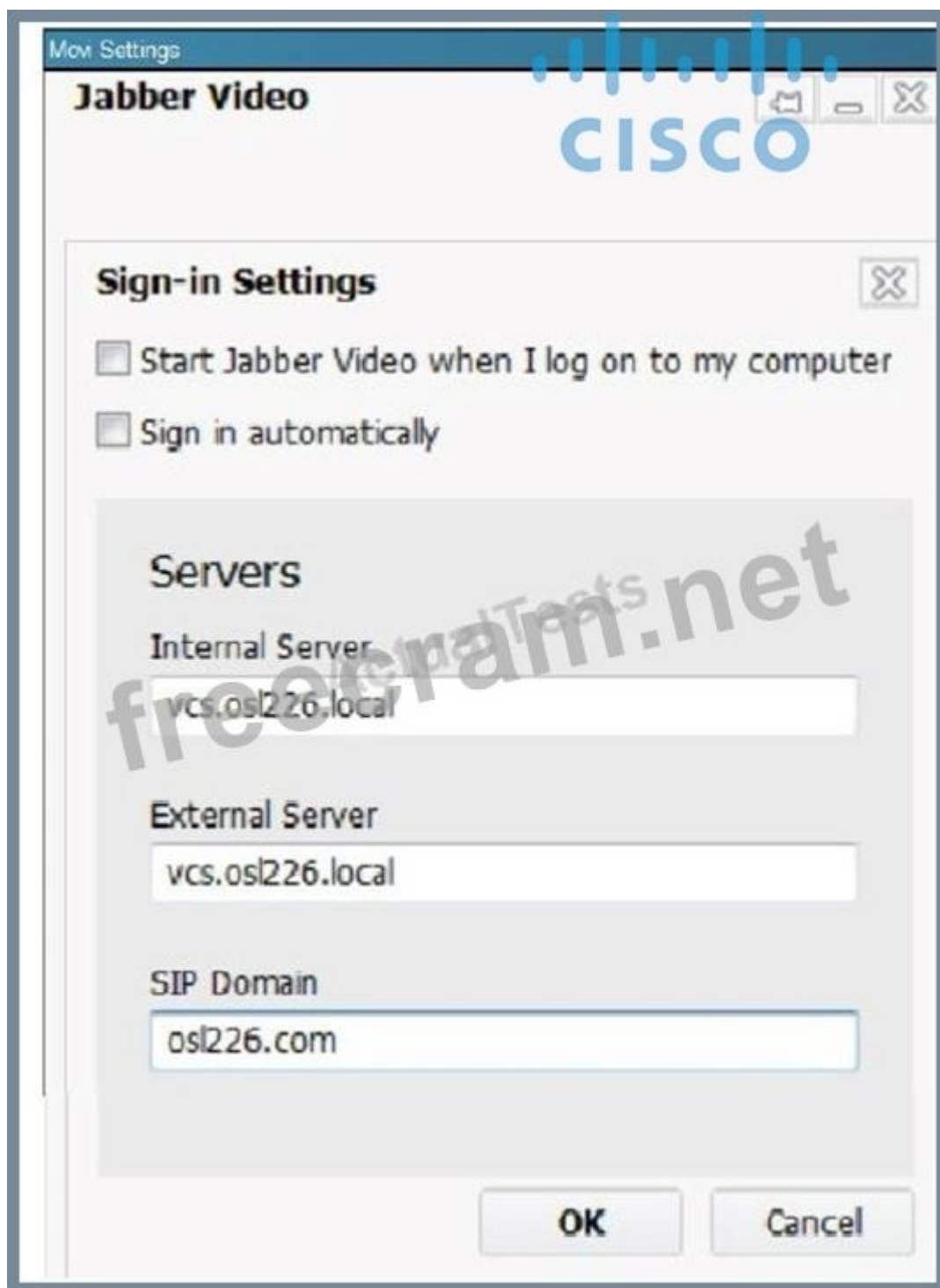


Name	CSS Name	CSS
All-Devices	All-Devices	All-Devices
All-Devices	All-Devices	All-Devices
All-Devices	All-Devices	All-Devices

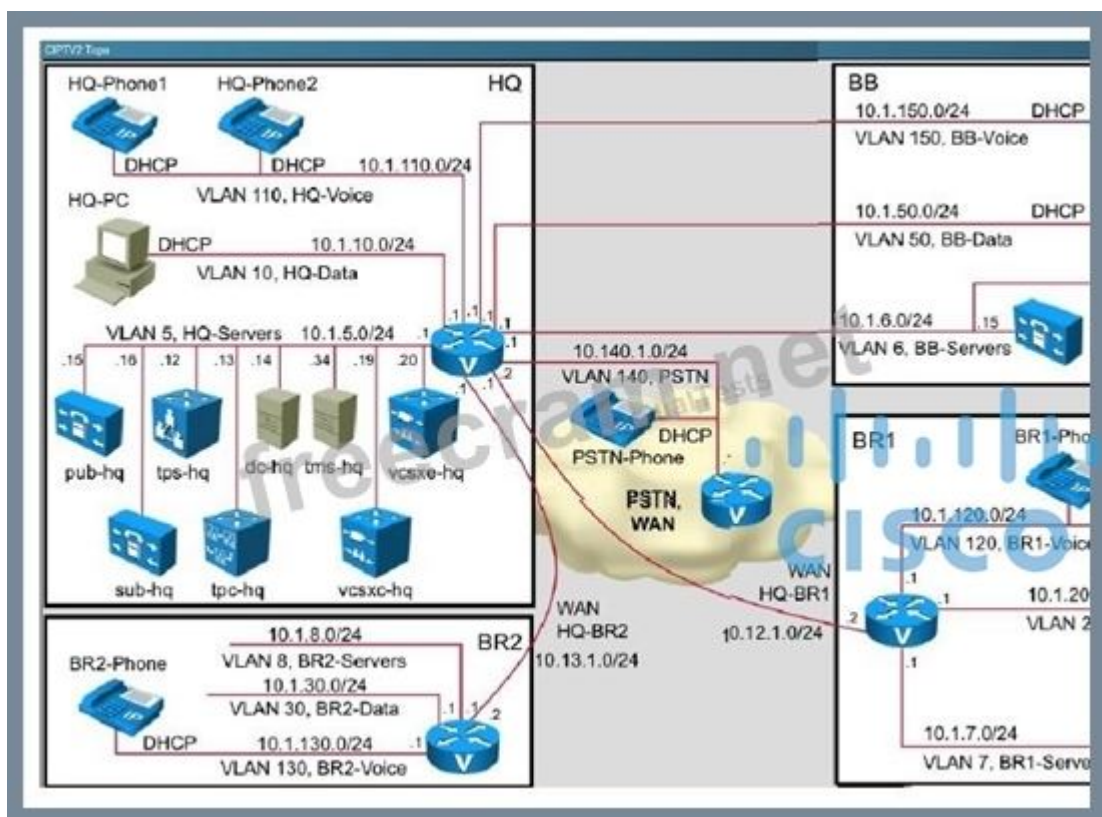
Movi Failure (exhibit):



Movi Settings (exhibit):



CIPTV2 Topology (exhibit):



Subzone (exhibit):

Links		You are here: Configure			
Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls
☐ DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0
☐ DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0
☐ DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0
☐ SubZone001ToDefaultSZ	HQ	DefaultSubZone			0
☐ SubZone001ToTraversalSZ	HQ	TraversalSubZone			0
☐ TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0
☐ VCS HQ - toHQ	HQ	to HQ	to HQ pipe		1

Links (exhibit):

Links		You are here: Configure				
Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth
☐ DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 k
☐ DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 k
☐ DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 k
☐ SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 k
☐ SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 k
☐ TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 k
☐ VCS HQ - toHQ	HQ	to HQ	to HQ pipe		1	12

Pipe (exhibit):

Pipe
Name: <u>to HQ pipe</u>
Total Bandwidth available – Bandwidth restriction:
Total Bandwidth available – Total bandwidth limit
Calls through this pipe – Bandwidth restriction: Lin
Calls through this pipe – Per call bandwidth limit (I

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 52

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

- A. The traversal client zone and the traversal server zone Media encryption mode must be set to Auto.
- B. The traversal server zone on Expressway-C must have a TLS verify subject name configured.
- C. The traversal client zone and the traversal server zone must be set to SIP TLS with TLS verify mode set to On.
- D. The traversal client zone and the traversal server zone Media encryption mode must be set to Force encrypted.

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 53

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 54

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 55

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

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

Answer: ([SHOW ANSWER](#))

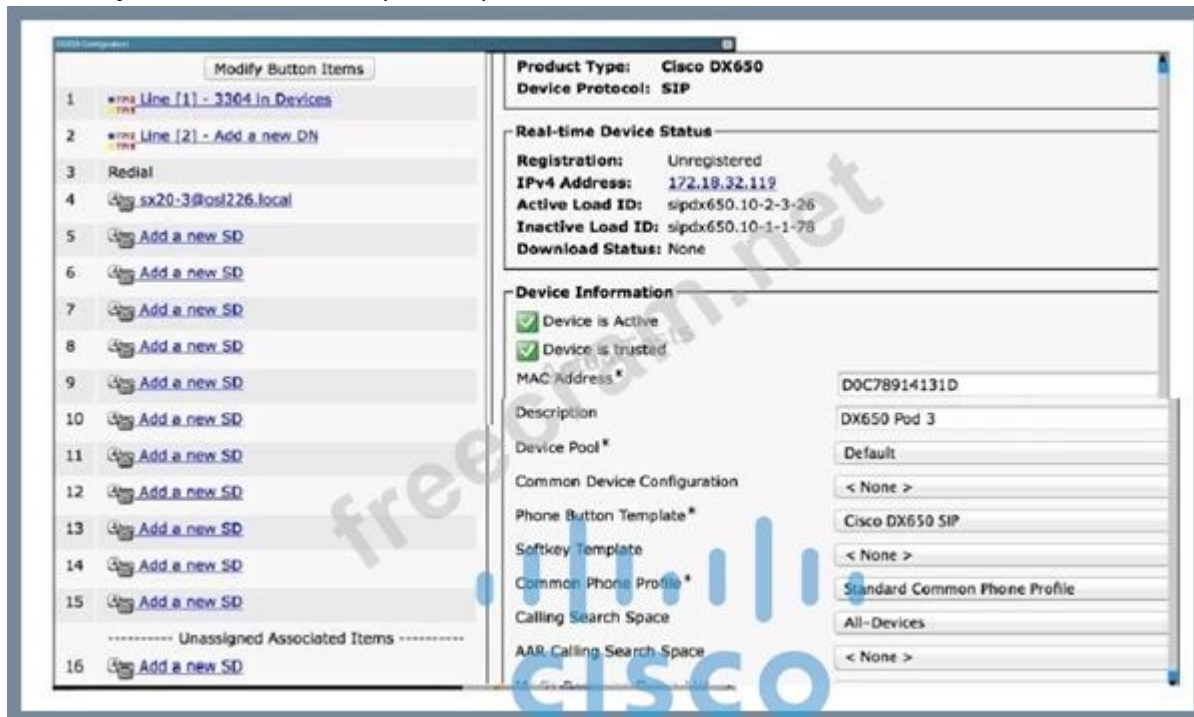
NEW QUESTION: 56

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 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 DN's?

SX20 System information (exhibit):



Modify Button Items

1 Line [11] - 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: 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

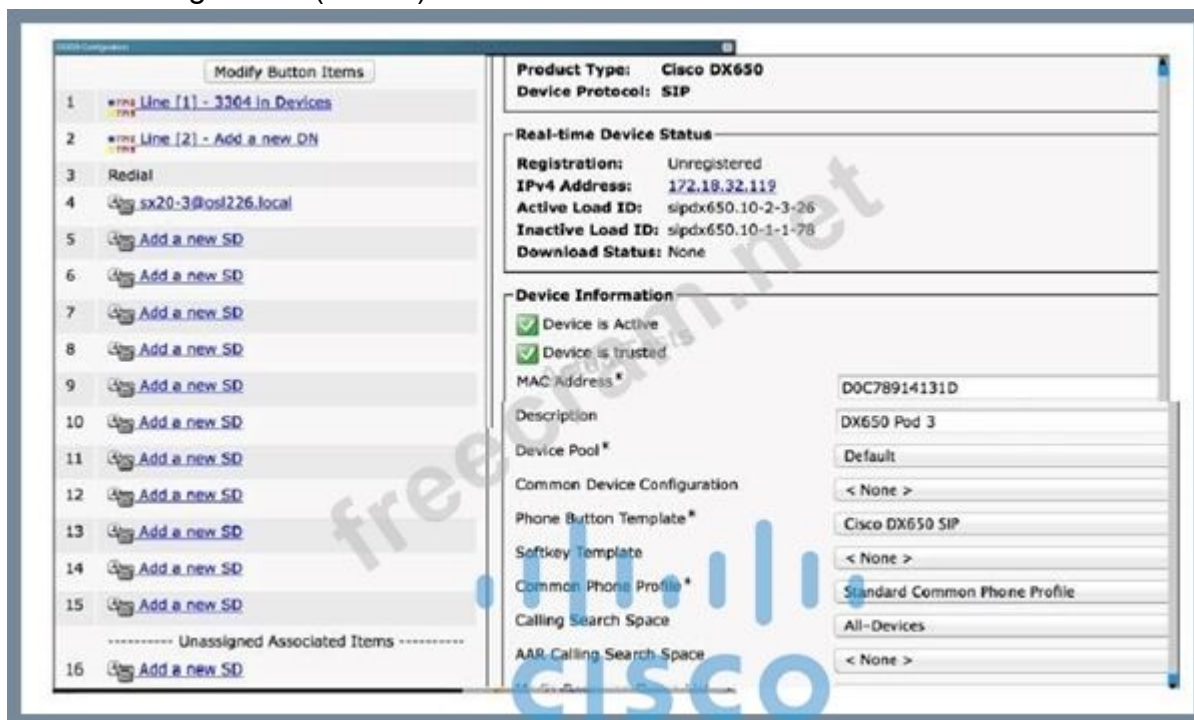
Softkey Template <None>

Common Phone Profile * Standard Common Phone Profile

Calling Search Space All-Devices

AAR Calling Search Space <None>

DX650 Configuration (exhibit):



Modify Button Items

1 Line [11] - 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: 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

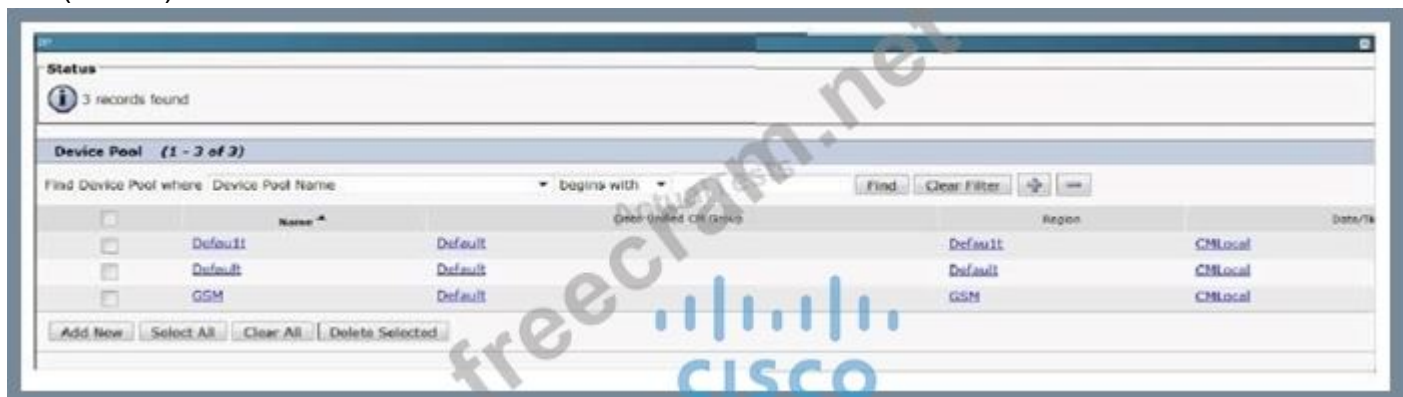
Calling Search Space All-Devices

AAR Calling Search Space <None>

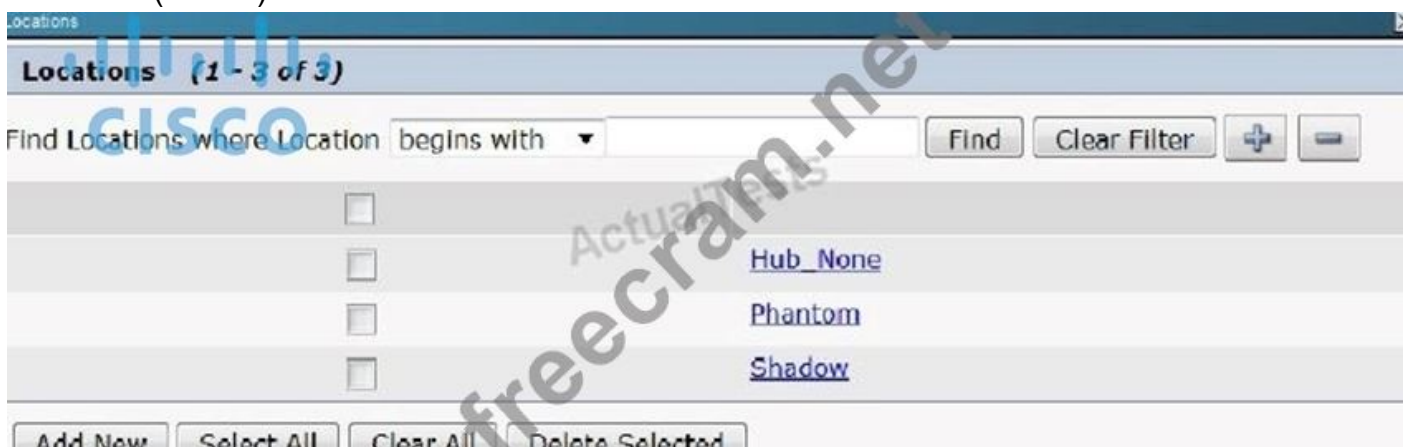
MRGL (exhibit):



DP (exhibit):



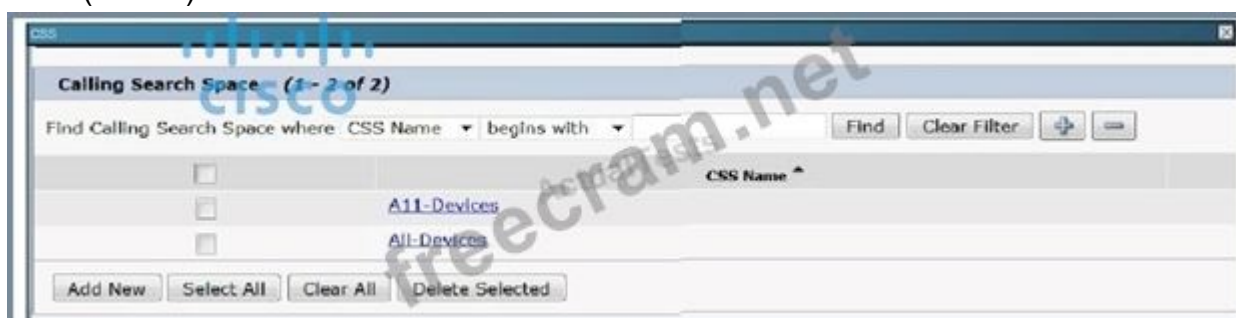
Locations (exhibit):



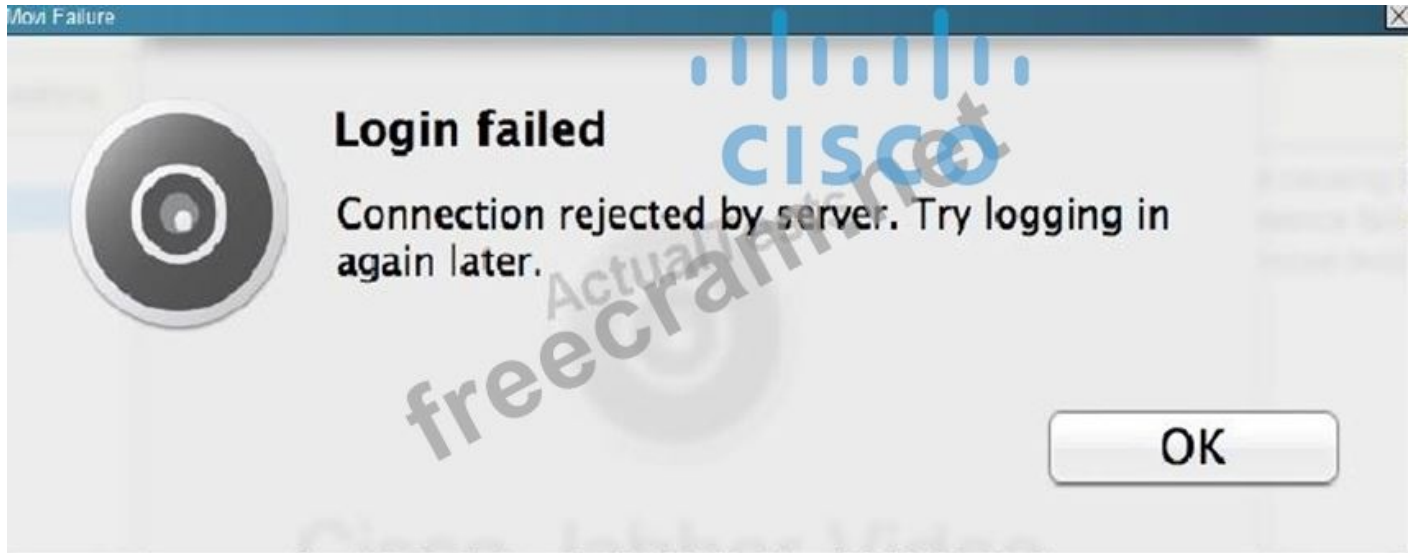
AARG (exhibit):



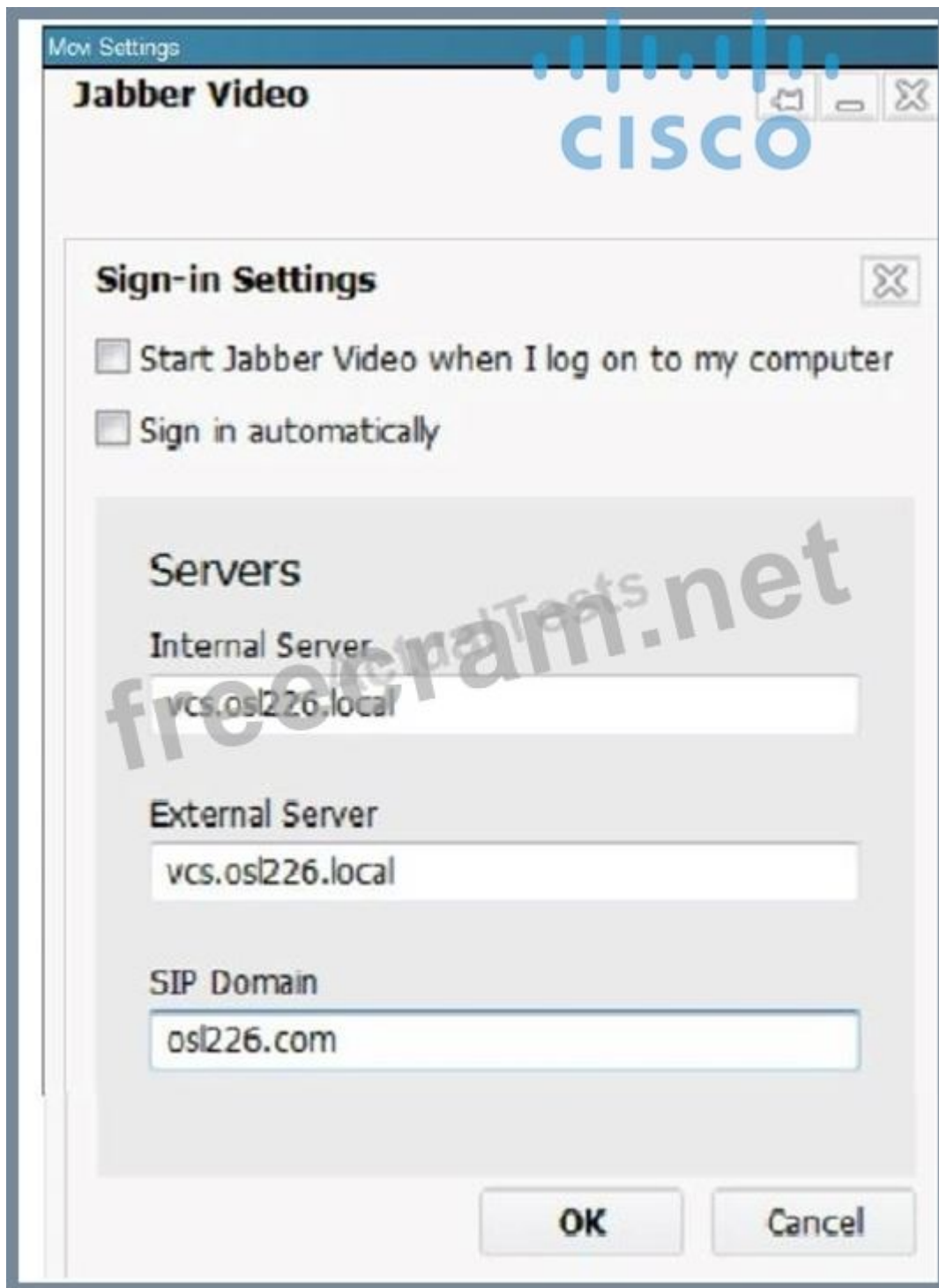
CSS (exhibit):



Movi Failure (exhibit):



Movi Settings (exhibit):



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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 57

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

Answer: ([SHOW ANSWER](#))

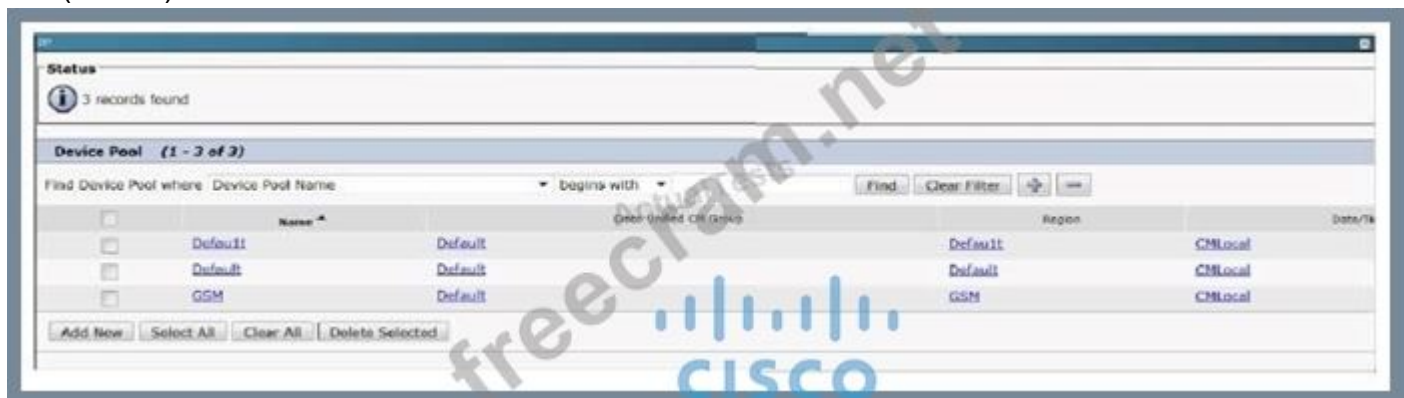
NEW QUESTION: 58

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.

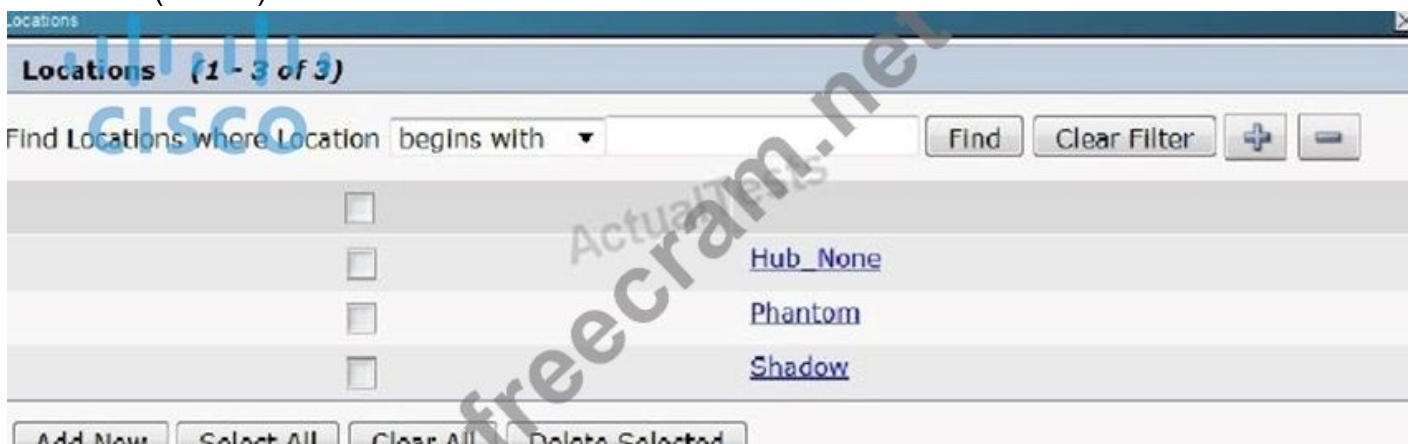
Both of the Cisco TelePresence Video for Windows clients are able to log into the server but can't make any calls. After reviewing the exhibits, which of the following reasons could be causing this failure?

DP (exhibit):



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

Locations (exhibit):

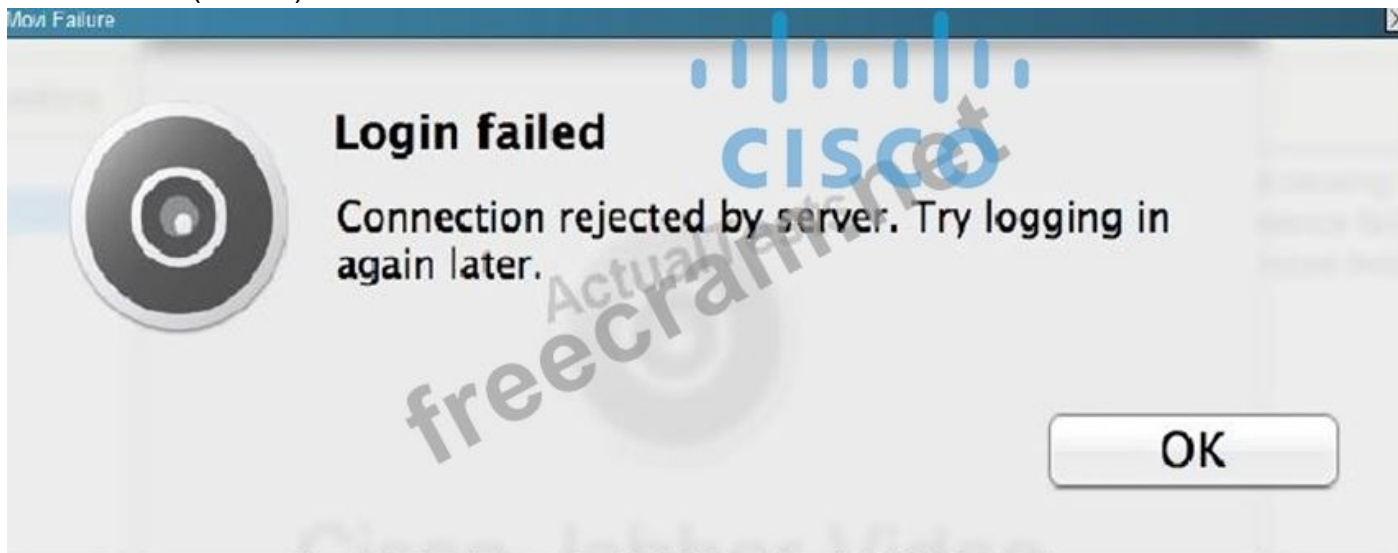


Name	Hub	Location
Hub_None	None	
Phantom	None	
Shadow	None	

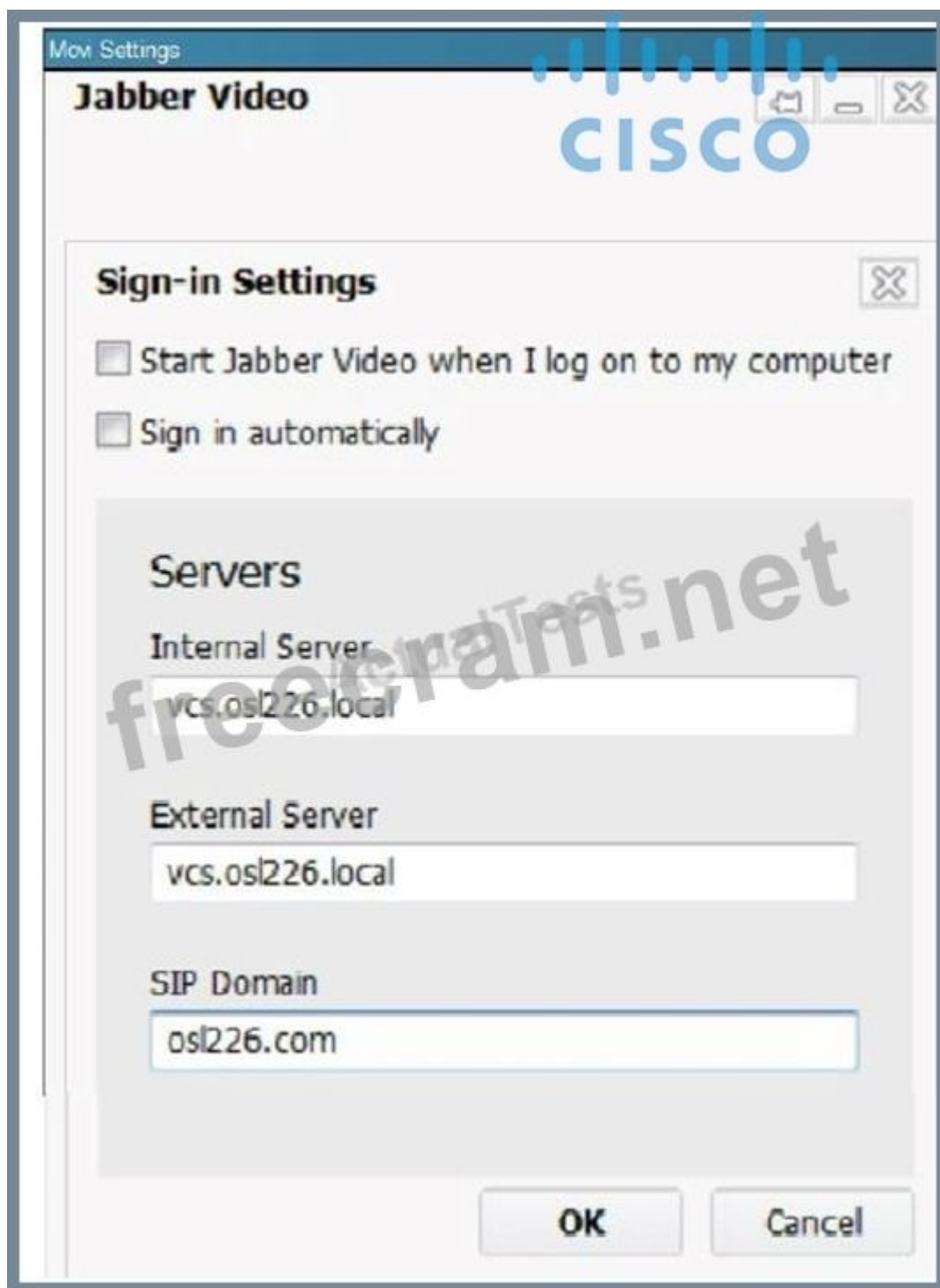
CSS (exhibit):



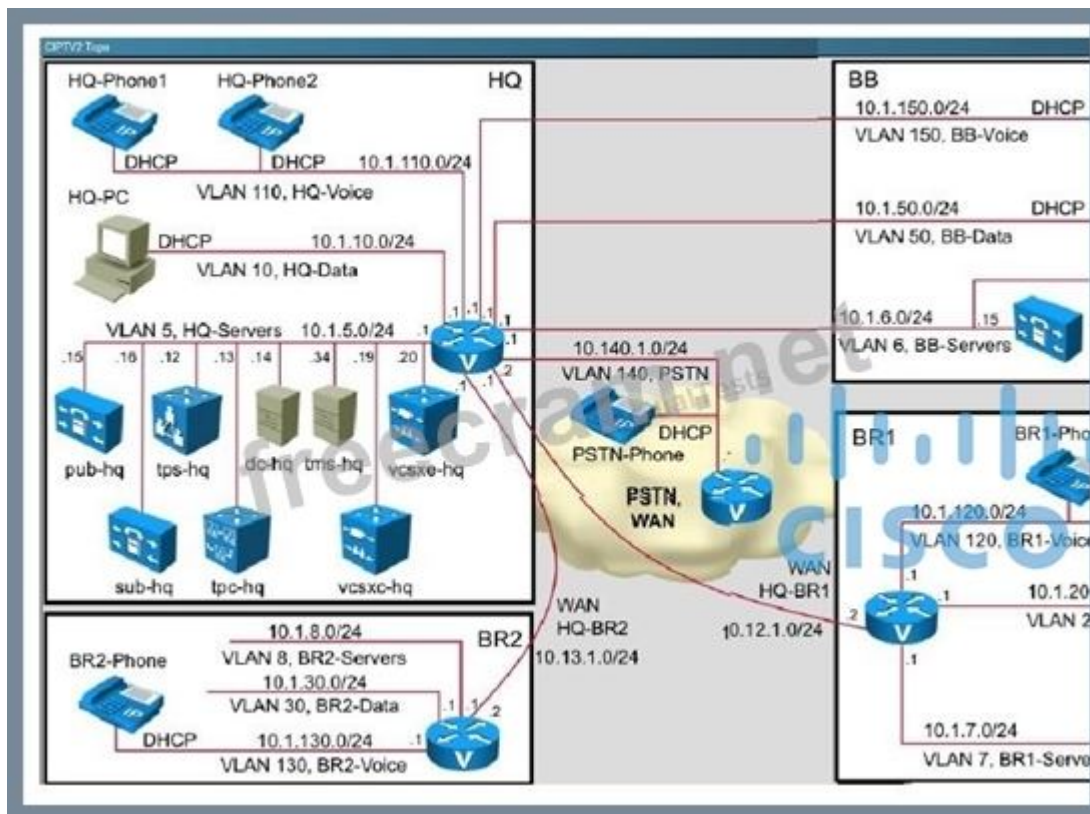
Movi Failure (exhibit):



Movi Settings (exhibit):



CIPTV2 Topology (exhibit):



Subzone (exhibit):

- **Name:** HQ
- **Authentication policy:** Treat as authenticated
- **Total bandwidth available - Bandwidth restriction:** Lin
- **Total bandwidth available - Total bandwidth limit (kb**
- **Calls into or out of this subzone - Bandwidth restrictic**
- **Calls into or out of this subone - Total bandwidth lin**
- **Calls entirely within this subzone Bandwidth restric**
- **Calls entirely within this subzone Total bandwidth li**

Links (exhibit):

Links		You are here: Configure				
Name	Node 1	Node 2	Pipe 1	Pipe 2	Calls	Bandwidth
☐ DefaultSZtoClusterSZ	DefaultSubZone	ClusterSubZone			0	0 kbps
☐ DefaultSZtoDefaultZ	DefaultSubZone	DefaultZone			0	0 kbps
☐ DefaultSZtoTraversalSZ	DefaultSubZone	TraversalSubZone			0	0 kbps
☐ SubZone001ToDefaultSZ	HQ	DefaultSubZone			0	0 kbps
☐ SubZone001ToTraversalSZ	HQ	TraversalSubZone			0	0 kbps
☐ TraversalSZtoDefaultZ	TraversalSubZone	DefaultZone			0	0 kbps
☐ VCS HQ - toHQ	HQ	to HQ	to HQ pipe		1	12 kbps

Pipe (exhibit):

Name: <u>to HQ pipe</u> Total Bandwidth available – Bandwidth restriction: Total Bandwidth available – Total bandwidth limit: Calls through this pipe – Bandwidth restriction: Limit: Calls through this pipe – Per call bandwidth limit (kbps):
--

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 59

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 60

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 61

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

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

Answer: ([SHOW ANSWER](#))

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NEW QUESTION: 62

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 since the last restart of the Cisco IP Voice Streaming Application that a Cisco Unified Communications Manager connection was lost.
- B. 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.
- 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 that a call through locations failed due to the lack of bandwidth.

Answer: ([SHOW ANSWER](#))

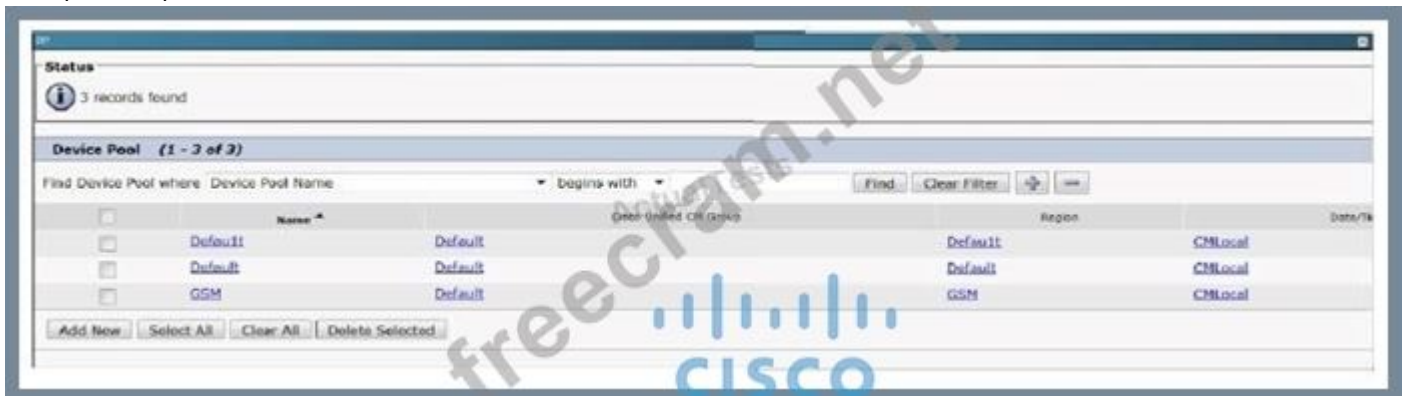
NEW QUESTION: 63

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

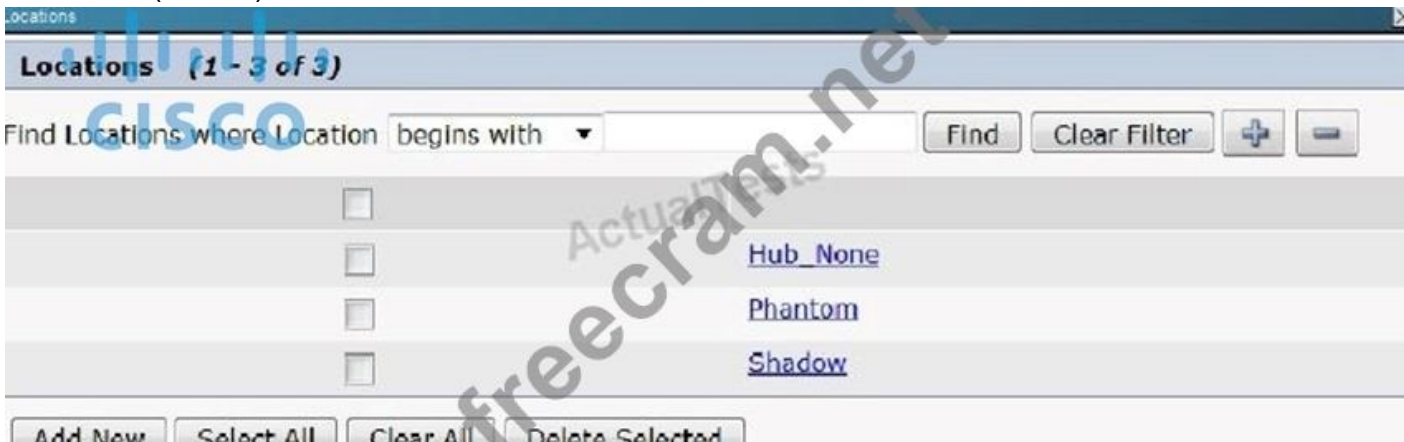
Which device configuration option will allow an administrator to control bandwidth between calls placed between branches?

DP (exhibit):



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

Locations (exhibit):



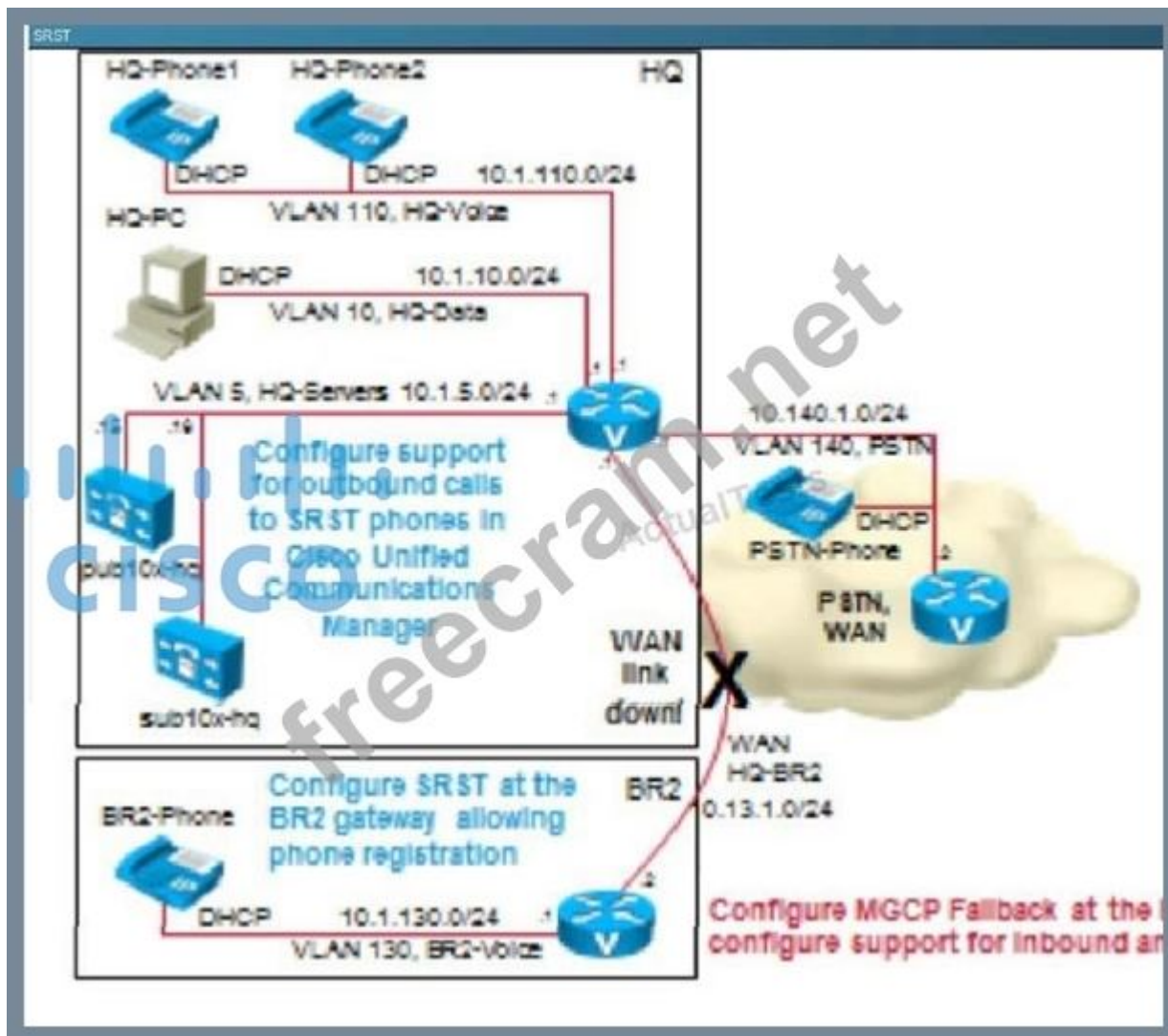
Name	Hub	Date/Time
Hub_None	Hub_None	Hub_None
Phantom	Phantom	Phantom
Shadow	Shadow	Shadow

CSS (exhibit):



Name	Date/Time	CSS Name
All-Devices	All-Devices	All-Devices
All-Devices	All-Devices	All-Devices

SRST (exhibit):



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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 64

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

- A.** VCS rewrites the Layer 3 source address of outbound SIP and H.323 packets to 209.165.200.230
- B.** The Advanced Networking or Dual Network Interfaces option key has been installed.
- 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.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 65

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

- A.** The H.323 trunk is added by selecting Call Control Discovery Trunk and then selecting H.323 as the protocol to be used.
- 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 Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Enable SAF check box should be selected in the trunk configuration.
- 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.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 66

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 67

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 68

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

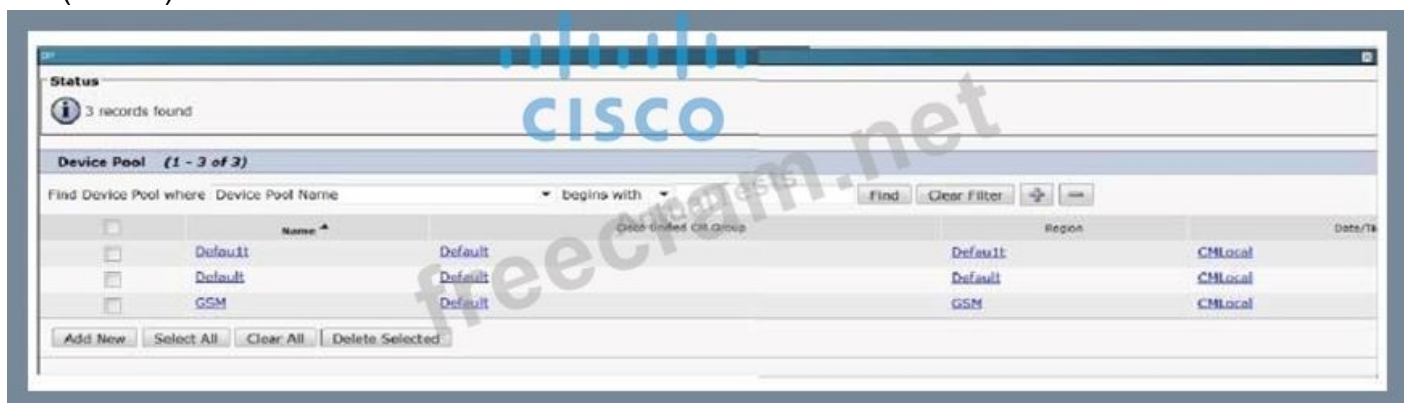
DX650 Configuration (exhibit):

DX650 Configuration	
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 >	

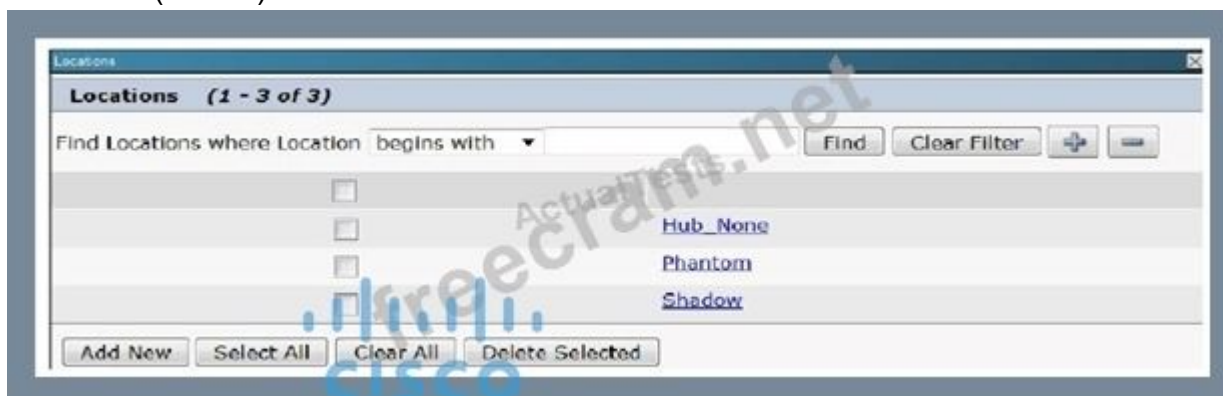
MRGL (exhibit):



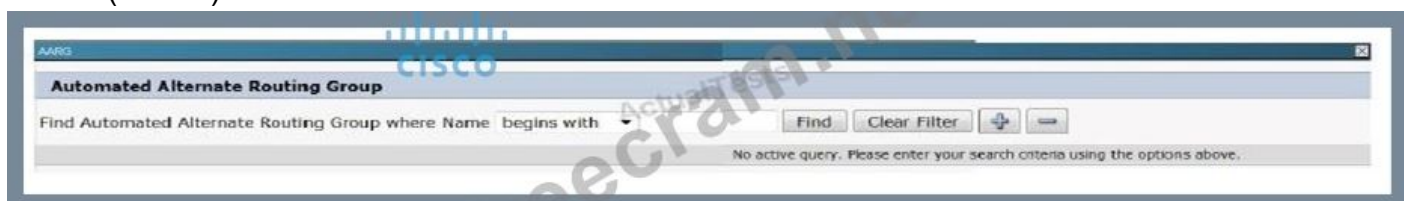
DP (exhibit):



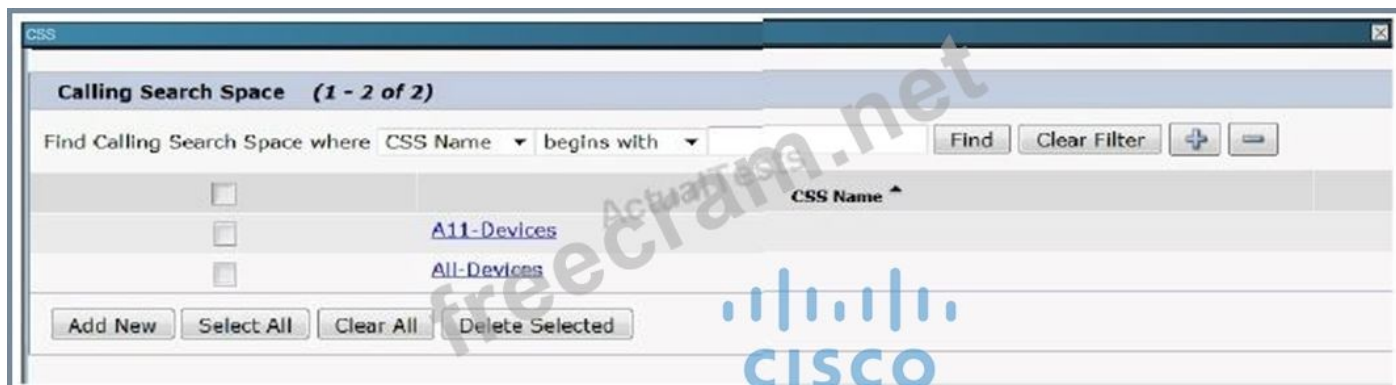
Locations (exhibit):



AARG (exhibit):



CSS (exhibit):



Movi Failure (exhibit):



Movi Settings (exhibit):



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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 69

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 70

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 71

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 72

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

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

F. 5

Answer: ([SHOW ANSWER](#))

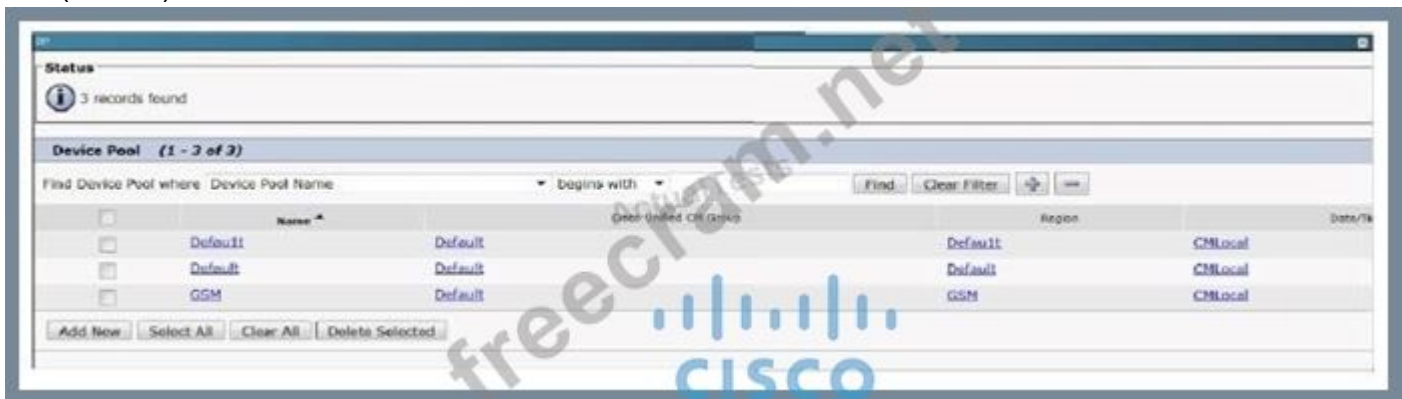
NEW QUESTION: 73

Scenario:

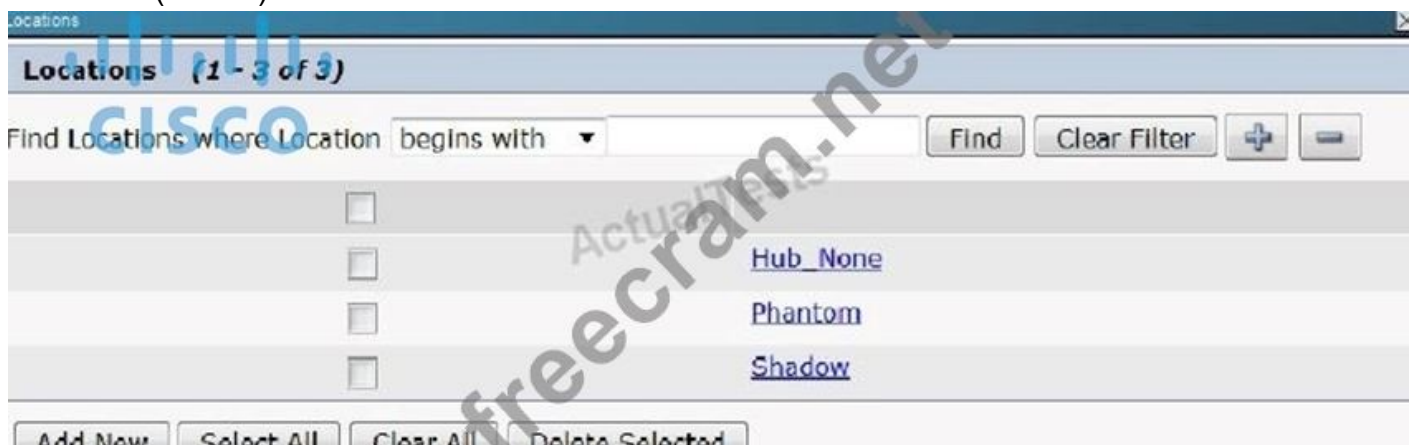
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 9971 Video IP Phone. The Cisco VCS and TMS control the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

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):



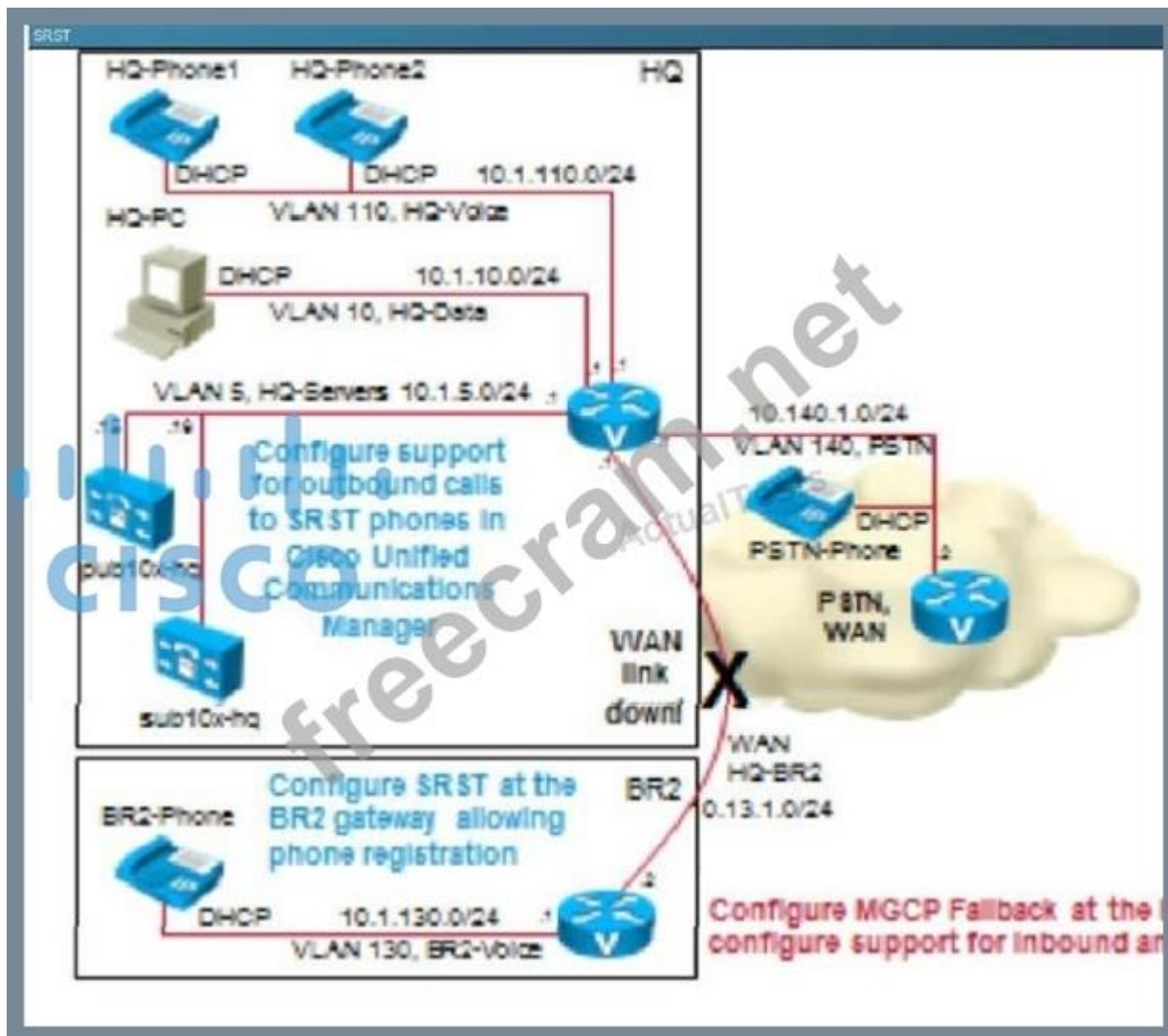
Locations (exhibit):



CSS (exhibit):



SRST (exhibit):



SRST-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. The router does not support SRST.
- B. The Cisco UCM is pointing to the wrong IPv4 address of the BR router.
- C. The SRST enabled router is not configured correctly.
- D. Device Pool cannot be default.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 74

Which statement about Service Advertisement Framework is true?

- A. SAF has no dependency on the underlying routing protocol, as long as it is a dynamic routing protocol.

- B. SAF operates on any dynamic or static IP routing configuration. SAF is totally independent of the underlying routing protocol.
- C. SAF requires that the EIGRP be configured on all routers, including non-SAF routers.
- D. SAF requires that the EIGRP be configured only on SAF routers. Non-SAF routers act as an IP cloud.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 75

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

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

Answer: A ([LEAVE A REPLY](#))

NEW QUESTION: 76

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

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

Answer: ([SHOW ANSWER](#))

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NEW QUESTION: 77

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 string
- C. route pattern
- D. URI

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 78

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 79

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 80

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 81

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

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

Answer: ([SHOW ANSWER](#))

NEW 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. Route Pattern under Route/Hunt
- C. Device Information under Phone Configuration
- D. +E.164 alternate number under Directory Number Settings

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 83

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 84

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

Answer: B ([LEAVE A REPLY](#))

NEW QUESTION: 85

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 requires a rule to allow all traffic from the DMZ to pass to the same network that the VCS Control is on.
- C.** A TMS server is needed to allow the firewall traversal to occur between the VCS Expressway and the VCS Control servers.
- D.** The firewall at Company X must have all SIP ALG functions disabled.

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 86

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 87

Which task must you perform before deleting a transcoder?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 88

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 89

Which component is needed to set up SAF CCD?

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 90

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 91

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

Answer: ([SHOW ANSWER](#))

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<https://www.examdisscuss.com/Cisco/exam/300-075/premium/> (220 Q&As Dumps, **35%OFF**
Special Discount Code: freeexam)

NEW QUESTION: 92

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

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

Answer: ([SHOW ANSWER](#))

NEW QUESTION: 93

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

Answer: ([SHOW ANSWER](#))

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