

300-075 Test-king

Number: 300-075
Passing Score: 800
Time Limit: 120 min
File Version: 14.1



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300-075

Implementing Cisco IP Telephony & Video, Part 2 v1.0

Version 14.1

Exam A

QUESTION 1

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 2

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 3

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 4

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



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- 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 5

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 6

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 7

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 8

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 9

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 10

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 11

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 12

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 13

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 14

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 15

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 16

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 17

Refer to the exhibit.

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

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

- 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 18

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 19

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 20

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 21

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 22

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 23

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 24

How many active gatekeepers can you can 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 25

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 26

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 27

Company X has a Cisco Unified Communications Manager cluster and a VCS Control server with video endpoints registered on both systems. Users find that video endpoints registered on Call manager can call each other and likewise for the endpoints registered on the VCS server. The administrator for Company X realizes he needs a SIP trunk between the two systems for any video endpoint to call any other video endpoint. Which two steps must the administrator take to add the SIP trunk? (Choose two.)

- A. Set up a SIP trunk on Cisco UCM with the option Device-Trunk with destination address of the VCS server.
- B. Set up a subzone on Cisco UCM with the peer address to the VCS cluster.
- C. Set up a neighbor zone on the VCS server with the location of Cisco UCM using the menu option VCS Configuration > Zones > zone.

- D. Set up a SIP trunk on the VCS server with the destination address of the Cisco UCM and Transport set to TCP.
- E. Set up a traversal subzone on the VCS server to allow endpoints that are registered on Cisco UCM to communicate.

Correct Answer: AC

Section: (none)

Explanation

Explanation/Reference:

QUESTION 28

Which sign is prefixed to the number in global call routing?

- A. -
- B. +
- C. #
- D. @
- E. &
- F. *

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 29

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

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

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 30

What is an advantage of TEHO?

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

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 31

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

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

Correct Answer: B

Section: (none)

Explanation

Explanation/Reference:

QUESTION 32

Which statement is true regarding the configuration of SAF Forwarder?

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

Correct Answer: A

Section: (none)

Explanation

Explanation/Reference:

QUESTION 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 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 35

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

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

Correct Answer: BE

Section: (none)

Explanation

Explanation/Reference:

QUESTION 36

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

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

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 37

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

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

Correct Answer: BD

Section: (none)

Explanation

Explanation/Reference:

QUESTION 38

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

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

Correct Answer: D

Section: (none)

Explanation

Explanation/Reference:

QUESTION 39

Which 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 40

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 41

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

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

Correct Answer: BC

Section: (none)

Explanation

Explanation/Reference:

QUESTION 42

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.
SX20 System information:


System Information


General

Product:	Cisco TelePresence SX20
Last boot:	Last Wednesday at 21:43
Serial number:	ABCD12345678
Software version:	TC7.3.0
Installed options:	PremiumResolution
System name:	MySystem
IPv4:	192.168.1.120
IPv6:	2001:DB8:1001:2002:3001:1000:1000:1000
MAC address:	01:23:45:67:89:AB
Temperature:	50.5°C / 137.3°F

DX650 Configuration:

Modify Button Items


1  [Line \[1\] - 3304 in Devices](#)


2  [Line \[2\] - Add a new DN](#)

3 Redial

4  [sx20-3@os1226.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](#)

MRGL:

MRGL

Status

 1 records found

Media Resource Group List (1 - 1 of 1)

Find Media Resource Group List where Name

<input type="checkbox"/>
<input type="checkbox"/>

DP:

Status

3 records found

Device Pool (1 - 3 of 3)

Find Device Pool where

<input type="checkbox"/>	Name ▲	
<input type="checkbox"/>	Default	Default
<input type="checkbox"/>	Default	Default
<input type="checkbox"/>	GSM	Default

Locations:

Locations

Locations

(1 - 3 of 3)

Find Locations where Location

begins with

Add New

Select All

Clear All

Delete Selected

AARG:

AARG

Automated Alternate Routing Group

Find Automated Alternate Routing Group where Name

CSS:

CSS

Calling Search Space (1 - 2 of 2)

Find Calling Search Space where

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

Movi Failure:



Login failed

Connection rejected by server.
again later.

Movie Settings:

Jabber Video



Sign-in Settings



- ☐ Start Jabber Video when I log on to my computer
- ☐ Sign in automatically

Servers

Internal Server

vcs.osl226.local

External Server

vcs.osl226.local

SIP Domain

osl226.com

OK

Cancel

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

- 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 43

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

DP:

Status

3 records found

Device Pool (1 - 3 of 3)

Find Device Pool where

<input type="checkbox"/>	Name	
<input type="checkbox"/>	Default	Default
<input type="checkbox"/>	Default	Default
<input type="checkbox"/>	GSM	Default

Locations:

Locations

Locations (1 - 3 of 3)

Find Locations where Location begins with ▼

<input type="checkbox"/>	
<input type="checkbox"/>	H
<input type="checkbox"/>	P
<input type="checkbox"/>	S

Add New

Select All

Clear All

Delete Selected

CSS:

CSS

Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name ▼ begins w

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

Add New

Select All

Clear All

Delete Selected

Movie Failure:



Login failed

Connection rejected by server.
again later.

Movie Setting:

Jabber Video



Sign-in Settings



- ☐ Start Jabber Video when I log on to my computer
- ☐ Sign in automatically

Servers

Internal Server

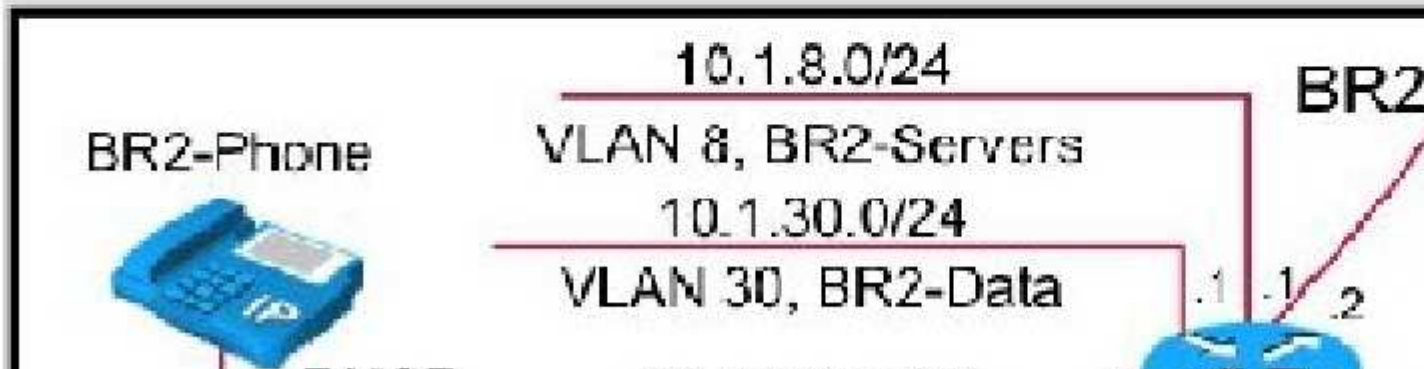
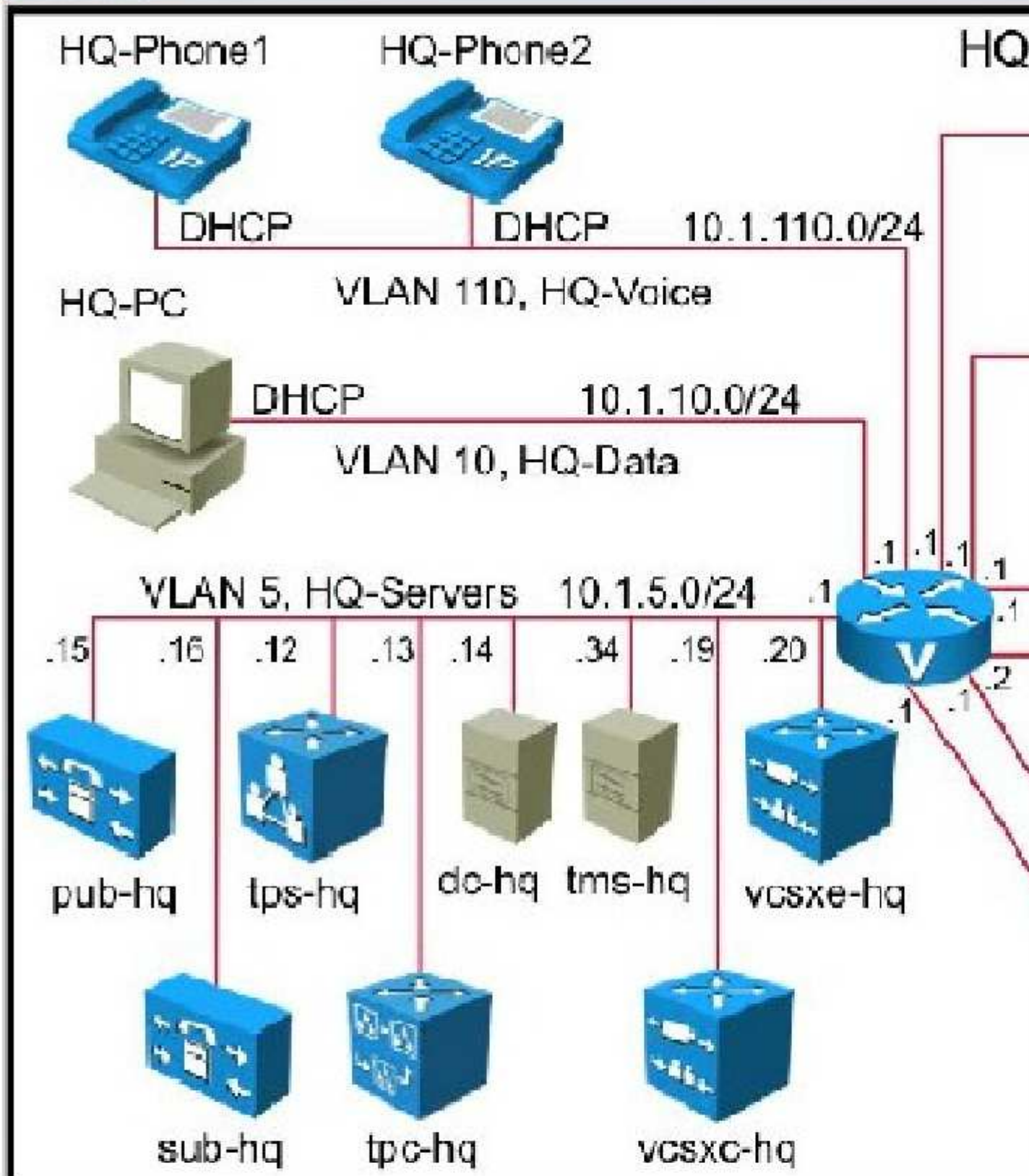
External Server

SIP Domain

OK

Cancel

Topology:



Subzones:

- **Name: HQ**
- **Authentication policy**
- **Total bandwidth av**
- **Total bandwidth av**
- **Calls into or out of t**
- **Calls into or out of**
- **Calls entirely with**
- **Calls entirely with**

Links:

Links		
	Name ▼	Node 1
<input type="checkbox"/>	DefaultSZtoClusterSZ	DefaultSubZone
<input type="checkbox"/>	DefaultSZtoDefaultZ	DefaultSubZone
<input type="checkbox"/>	DefaultSZtoTraversalSZ	DefaultSubZone
<input type="checkbox"/>	SubZone001 ToDefaultSZ	HQ
<input type="checkbox"/>	SubZone001 ToTraversalSZ	HQ
<input type="checkbox"/>	TraversalSZtoDefaultZ	TraversalSubZone
<input type="checkbox"/>	YCS HQ - toHQ	HQ

Pipe:

Name: to_HQ_pipe

Total Bandwidth available

Total Bandwidth available

Calls through this pipe – B

Calls through this pipe – P

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?

- 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 44

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

DP:

DP

Status

 3 records found

Device Pool (1 - 3 of 3)

Find Device Pool where

<input type="checkbox"/>	Name 	
<input type="checkbox"/>	Default	Default
<input type="checkbox"/>	Default	Default
<input type="checkbox"/>	GSM	Default

Add New

Select All

Clear All

Delete Selected

Locations:

Locations

Locations (1 - 3 of 3)

Find Locations where Location begins with ▼

<input type="checkbox"/>	
<input type="checkbox"/>	H
<input type="checkbox"/>	P
<input type="checkbox"/>	S

Add New

Select All

Clear All

Delete Selected

CSS:

CSS

Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name ▼ begins w

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

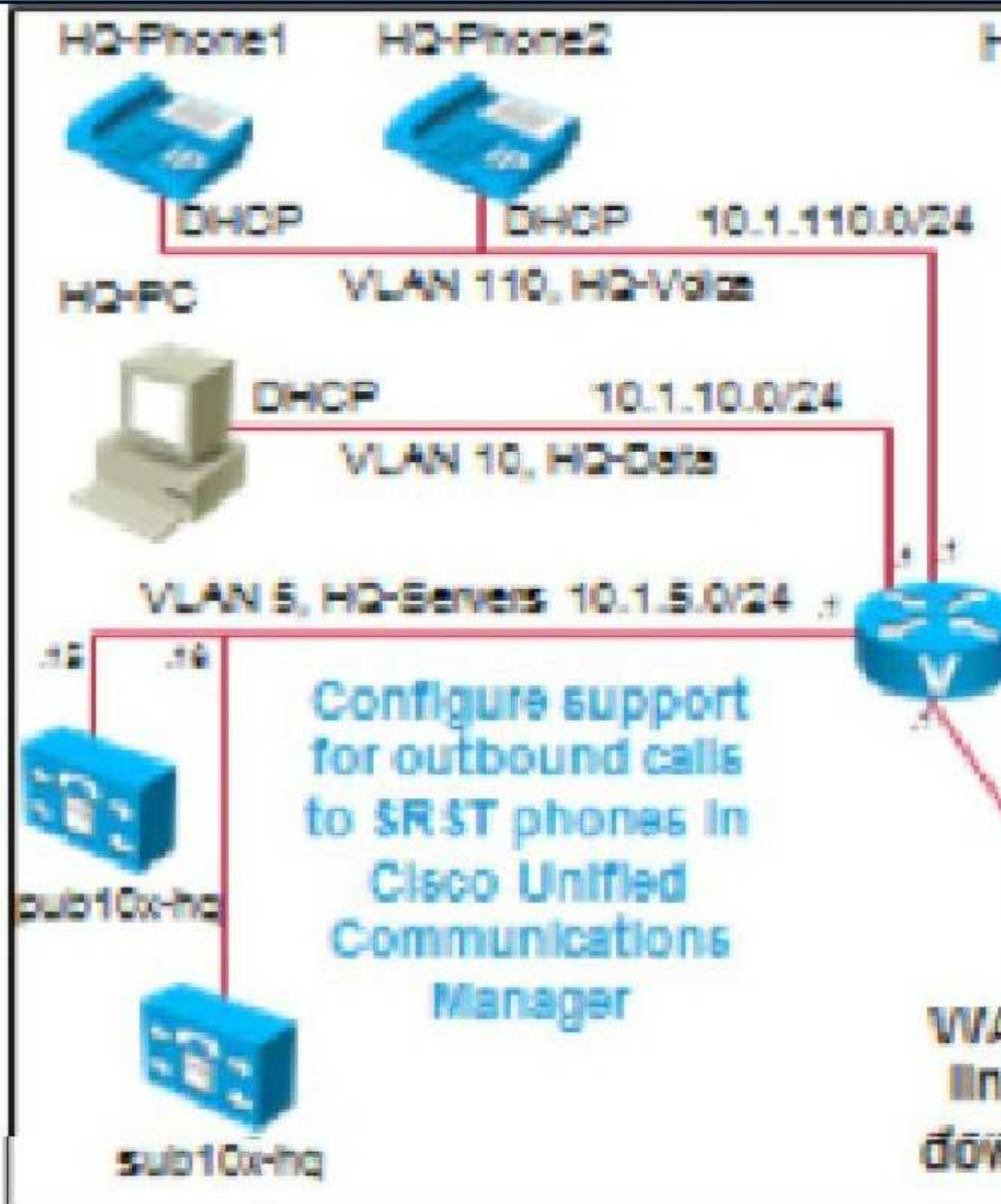
Add New

Select All

Clear All

Delete Selected

SRST:



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



```
voice service voip
    sip
        bind control source
        GigabitEthernet0/0/0
        bind media source-
        GigabitEthernet0/0/0
    registrar server
    !
voice register global
    max-dn 1
    max-pool 1
    !
voice register pool
```


At the HQ cluster, the CFUR
BR2 phone (+44228822400

- Forward Unregistered In
- Forward Unregistered In
- Forward Unregistered E
- Forward Unregistered E

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.)

- 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/rt228821....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/rt228822....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:**



<http://www.gratisexam.com/>