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

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

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

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.)
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

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

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

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

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

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

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
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

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

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

QUESTION 14
Which two options are valid service parameter settings that are used to set up proper video QoS behavior
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

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

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

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

Explanation
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

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

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

QUESTION 24
How many active gatekeepers can you define in a local zone?
A. 1
B. 2
C. 5
D. 10
E. 15
F. unlimited

Correct Answer: A
Section: (none)
Explanation

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

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

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

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

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

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

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

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

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:**

---

**General**

<table>
<thead>
<tr>
<th>Product:</th>
<th>Cisco TelePresence SX20</th>
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</thead>
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<td>Last boot:</td>
<td>Last Wednesday at 21:45</td>
</tr>
<tr>
<td>Serial number:</td>
<td>ABCD12345678</td>
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<td>Software version:</td>
<td>TC7.3.0</td>
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<td>Installed options:</td>
<td>PremiumResolution</td>
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<td>IPv4:</td>
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<tr>
<td>MAC address:</td>
<td>01:23:45:67:89:AD</td>
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<tr>
<td>Temperature:</td>
<td>59.5°C / 137.3°F</td>
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</table>
DX650 Configuration:
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<th>Description</th>
</tr>
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<tr>
<td>1</td>
<td>Line [1] - 3304 in Devices</td>
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<tr>
<td>2</td>
<td>Line [2] - Add a new DN</td>
</tr>
<tr>
<td>3</td>
<td>Radial</td>
</tr>
<tr>
<td>4</td>
<td><a href="mailto:sx20-3@osl226.local">sx20-3@osl226.local</a></td>
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<td>5</td>
<td>Add a new SD</td>
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<td>6</td>
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<td>Add a new SD</td>
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<td>16</td>
<td>Add a new SD</td>
</tr>
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----- Unassigned Associated Items -----

Add a new SD
### MRGL:

**Status**

- **1 records found**

### Media Resource Group List (1 - 1 of 1)

Find Media Resource Group List where Name begins with

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<tr>
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<th>Select All</th>
<th>Clear All</th>
<th>Delete Selected</th>
</tr>
</thead>
</table>

### DP:
### Locations:

#### Locations (1 - 3 of 3)

Find Locations where Location begins with

<table>
<thead>
<tr>
<th>Location Name</th>
</tr>
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### Device Pool (1 - 3 of 3)

Find Device Pool where Device Pool Name

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<thead>
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<th>Name</th>
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<tr>
<td>Default</td>
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<tr>
<td>GSM</td>
</tr>
</tbody>
</table>

[Add New]  [Select All]  [Clear All]  [Delete Selected]
AARG:

Automated Alternate Routing Group

Find Automated Alternate Routing Group where Name begins with

CSS:

Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name begins with

Check boxes:

- All Devices
- All-Devices

Options:

Add New  Select All  Clear All  Delete Selected

Movil Failure:
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:

Locations:
### Locations

Find Locations where Location begins with

<p>| | | |</p>
<table>
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- Add New
- Select All
- Clear All
- Delete Selected

### CSS

Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name begins with

<p>| | |</p>
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</tbody>
</table>

- All Devices
- All-Devices

- Add New
- Select All
- Clear All
- Delete Selected

**Movie Failure:**
Login failed

Connection rejected by server. again later.

Movie Setting:
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]
Subzones:
Name: HQ

Authentication policy

Total bandwidth available

Total bandwidth available

Calls into or out of the site

Calls into or out of the site

Calls entirely within the site

Calls entirely within the site
Links:

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<thead>
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<tr>
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<td>DefaultSZtoTraversalsZ</td>
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<td>TraversalsSubZone</td>
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<td>YCS HQ - toHQ</td>
<td>HQ</td>
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</table>

Pipe:
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

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.
### Device Pool (1 - 3 of 3)

<table>
<thead>
<tr>
<th>Find Device Pool where</th>
<th>Device Pool Name</th>
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<td>Default</td>
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<td>Default</td>
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<td>GSM</td>
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</table>

Locations:
## Locations

Find Locations where Location begins with

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Add New    Select All    Clear All    Delete Selected

## CSS

Find Calling Search Space where CSS Name begins with

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</table>

Add New    Select All    Clear All    Delete Selected

**SRST:**
Configure support for outbound calls to SRST phones in Cisco Unified Communications Manager.

Configure SRST at the BR2 gateway allowing phone registration.
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
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/Reference:

http://www.gratisexam.com/