Cisco.300-075.v2019-04-27.q271

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NEW QUESTION: 1
Your company's main number is 408-526-7209, and your employee's directory numbers are 4-digit numbers. Which option should be configured if you want outgoing calls from a 4-digit internal directory number to be presented as a 10-digit number?
A. AAR group
B. calling party transformation pattern
C. translation pattern
D. route pattern
Answer: (SHOW ANSWER)

NEW QUESTION: 2
When you configure a region to use the G.729 codec, which other codecs can be utilized in the region?
A. The region will use all of the codecs supported by Cisco Unified Communications Manager as long as a software Media Termination Point is available.
B. The region can use all the codecs supported by Cisco Unified Communications Manager.
C. The region will only use the codec configured in the region configuration field and any other codecs of equal or lower bandwidth.
D. The region will use the configured codec and the default codec as long as it does not exceed the configured bandwidth for the region.
Answer: (SHOW ANSWER)

NEW QUESTION: 3
Which bandwidth amounts are correct for configuring locations?
A. 8 kb/s for G.729, 64 kb/s for G.711, and 16 kb/s for G.722
B. 8 kb/s for G.729, 8 kb/s for G.711, and 8 kb/s for G.722
C. 8 kb/s for G.729, 64 kb/s for G.711, and 64 kb/s for G.722

D. 64 kb/s for G.729, 64 kb/s for G.711, and 64 kb/s for G.722

Answer: (SHOW ANSWER)

NEW QUESTION: 4

Refer to the exhibit.

IOS SAF Forwarder Config

```
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
 !
 topology base
  external-client HQ_SAF
 exit-service-family
!
!
service-family external-client listen ipv4 5050
 external-client HQ_SAF
 username SAFUSER
 password SAFPASSWORD
 keepalive 3600000
```

The exhibit shows a SAF Forwarder configuration attached to a Cisco Unified Communications Manager.

Which minimum configuration for a Cisco Unified Communications Manager Express SAF Forwarder is needed to establish a SAF neighbor relationship with this SAF Forwarder?

A. router eigrp SAF

!

service-family ipv4 autonomous-system 1

!

topology base exit-

sf-topology exit-

service-family

!

voice service saf

profile trunk-route 1

session protocol sip interface Loopback1 transport tcp port

5060 ! profile dn-block 1 alias-prefix 1972555

pattern 1 type extension 4xxx

!

profile

callcontrol 1 dn-

service trunk-

route 1 dn-block
1 dn-block 2
!
channel 1 vrouter SAF asystem 1
subscribe callcontrol wildcarded
publish callcontrol 1
!
B. router eigrp
SAF i
service-family ipv4 autonomous-system 1
!
topology base exit-
sf-topology exit-
service-family
voice service saf
profile trunkroute 1
session protocol sip interface Loopback1 transport tcp port 5060 !
C. None of above configurations contain sufficient information.
D. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
Answer: (SHOW ANSWER)

NEW QUESTION: 5
A voice-mail product that supports only the G.711 codec is installed in headquarters. Which action allows branch Cisco IP phones to function with voice mail while using only the G.729 codec over the WAN link to headquarters?
A. Configure transcoding within Cisco Unified Communications Manager.
B. Configure transcoding resources in Cisco IOS and assign to the MRGL of Cisco IP phones.
C. Configure transcoder resources in the branch Cisco IP phones.
D. Configure Cisco Unified Communications Manager regions.
E. 729 codec over the WAN link to headquarters?
Answer: (SHOW ANSWER)

NEW QUESTION: 6

In the case of a VCS cluster, which configuration element is recommended for endpoint H.323 registration?

A. hostname of the VCS cluster configuration master

B. hostname of the VCS cluster member peer

C. static IP addresses

D. DNS SRV records pointing to the VCS cluster name

Answer: (SHOW ANSWER)

NEW QUESTION: 7

Which two statements about SAF service identifier numbers are true? (Choose two.)

A. They are generated in the format telco.cisco.saf-forwader.db.replicate.data.local.

B. They are generated in the format data-source:sub-service:instance.matrix.fifty.saf.

C. They are 16-bit decimal identifiers.

D. They are 32-bit decimal identifiers.

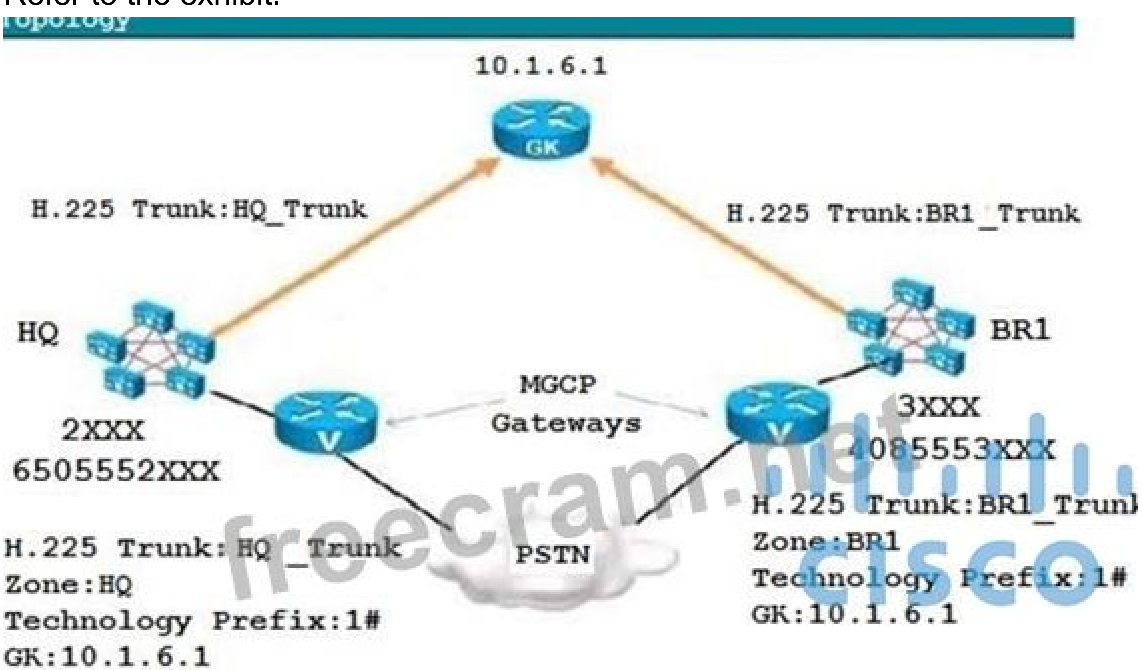
E. They are generated in the format service:sub- service:instance.instance.instance.instance.

F. They are generated in the format data.saf.cucm-publisher.asf@domain.local.

Answer: (SHOW ANSWER)

NEW QUESTION: 8

Refer to the exhibit.



Gatekeeper Config

```
gatekeeper
zone local HQ cisco.com 10.1.6.1
zone local BR1 cisco.com
zone prefix HQ2...
zone prefix BR1 3...
gw-type-prefix 1# default-technology
no shutdown
```

Assume that NANP is being used and 9 is used for PSTN access code Long distance national calls are preceded with 1.
How should the HQ Cisco Unified Communications Manager be configured for calls to 3XXX to be sent to the gatekeeper at 1 0 1 6 1 with PSTN backups?
A. Configure a route pattern for 3XXX Assign this route pattern to a route list that points to two route groups. The first route group contains the H 225 trunk. The second route group contains the MGCP gateway with prefix digits 1 408555 for the outgoing called number.
B. Configure a route pattern for 1#3XXX Assign this route pattern to a route list that points to a route group that lists the H 225 trunk as first choice and the MGCP gateway as a second choice.
C. Configure a route pattern for 3XXX Assign this route pattern to a route list that points to two route groups. The first route group contains the H 225 trunk. The second route group contains MGCP gateway with prefix digits 91 408554 for the called number.
D. Configure a route pattern for 4085543XXX. Assign this route pattern to a route list that points to two route groups. The first route group contains the H 226 trunk. The second route group contains MGCP gateway.
Answer: (SHOW ANSWER)

NEW QUESTION: 9
A presales engineer is working on a quote for a major customer and must evaluate how many Cisco VCS Expressway traversal call licenses for which to plan.
Calls to and from which three routes must the engineer include in the tally? (Choose three.)
A. gateway
B. Cisco 9971 Endpoint
C. border controllers
D. gatekeeper
E. SIP trunk
F. VCS
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/telepresence-video- communication-server-vcs/118872-technote-vcs-00.html

NEW QUESTION: 10
A. The SIP route patterns have not been properly configured.
B. The Tomcat certificates do not match.
C. The cluster ID does not match.
D. One cluster is using TLS certificate, and the other is using Password.
E. The Cisco Unified Resource Identifier service needs a restart.
F. The ILS authentication password does not match.
Answer: (SHOW ANSWER)

NEW QUESTION: 11

Refer to the exhibit.

```
voice service saf
 profile trunk-route1
 session protocol sip interface Loopback1 transport tcp port 506(
 !
profile dn-block 1 alias-prefix 1972555
 pattern 1 type extension 4XXX
 !
profile dn-block 2
 pattern 1 type global 14087071222
 !
profile callcontrol 1
 dn-service
  trunk-route 1
  dn-block 1
  dn-block 2
 !
!
!
channel 1 vrouter SAF asystem 1
 subscribe callcontrol wildcarded
 publick callcontrol 1
!
```

How does the Cisco Unified Communications Manager advertise dn-block 1?

A. 4XXX and the ToDID will 0:+ 1972555

B. 4XXX and the ToDID will 0:1972555

C. 4XXX and the ToDID will 0:

D. 4XXX

E. 19725554XXX

Answer: (SHOW ANSWER)

NEW QUESTION: 12

Which statement is true when considering a Cisco VoIP environment for regional configuration?

A. The default codec does not matter if you have defined a hardware MTP in your Cisco Unified Communications Manager environment.

B. G.729 requires 24K of bandwidth per call.

C. To deploy a Cisco H.323 gatekeeper, you must configure MTP resources on the gatekeeper and only use G.711 between regions.

D. G.711 requires 128K of bandwidth per call.

Answer: (SHOW ANSWER)

NEW QUESTION: 13

Which three items must you configure to enable SAF Call Control Discovery? (Choose three.)

A. hosted DN groups

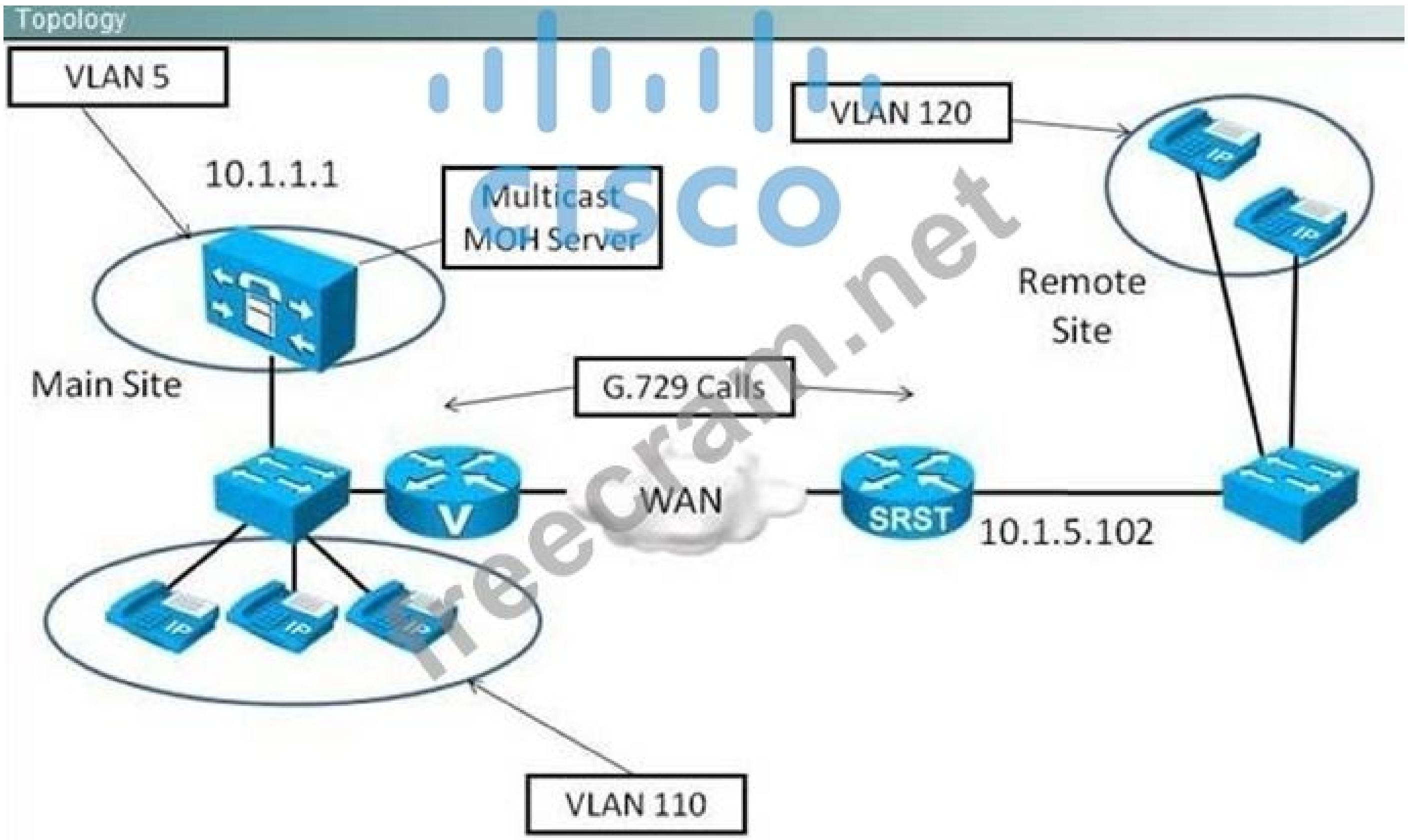
B. the SIP or H.323 trunk

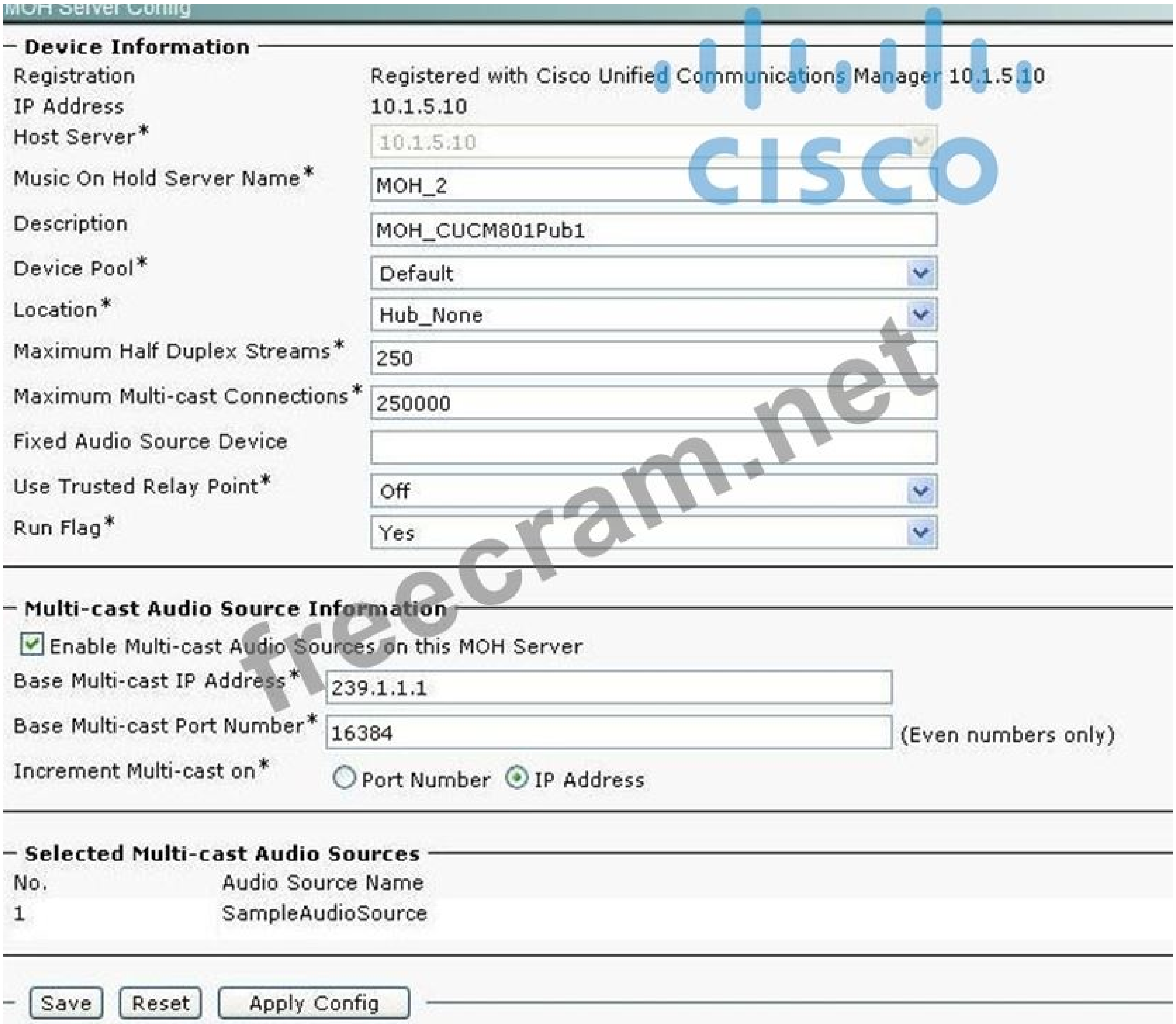
C. hosted DN patterns

D. a calling search space

E. translation patterns
F. route patterns
Answer: (SHOW ANSWER)

NEW QUESTION: 14
Refer to the exhibit.





Multicast MOH needs to be run from flash at the remote site. Which Cisco IOS configuration at the SRST router is correct?

A. !

call-manager-fallback

ip source-address 10.1.5.102 port

2000 max-ephones 2

max-dn 8 dual-line

moh music-on-hold au

multicast moh 239.1.1.1 port 163t

B. call-manager-fallback

ip source-address 10.1.120.1 port 2000 max-ephones 2 max-dn 8 moh music-on- hold.au

multicast moh 239.1.1.1 port 32767

C. !

call-manager-fallback

ip source-address 10.1.5.102 port

2000 max-ephones 2
max-dn 8 dual-line
moh music-on-hold au
multicast moh 239.1.1.1 port 16384
D. !
call-manager-fallback
ip source-address 10.1.5.102 port
2000 max-ephones 2
max-dn 8 dual-line
moh music-on-hold an
multicast moh 239.1.1.1 port 32767
Answer: (SHOW ANSWER)

NEW QUESTION: 15
An engineer is working on a Cisco VCS Control routing configuration and wants users to be able to dial ccnpcollab and have calls routed to ccnpcollab@cisco.com. Which option achieves this aim?
A. search rules
B. transforms
C. access rules
D. call policy
Answer: (SHOW ANSWER)
Explanation/Reference:
Explanation:
Although Call Policy could be used, typically this type of alias change is done using transforms (either pre- search transforms or search rule transforms). Call Policy could be justified as the correct answer based on how the question is interpreted. If this change is only intended for the ccnpcollab alias being dialed, and not for other aliases that might be dialed without a domain, then call policy makes since. Still, I would never use Call Policy in this capacity, nor does Cisco recommend this.

NEW QUESTION: 16
Which DNS SRV Records must be configured on the external DNS server in a mobile remote access scenario with Cisco Expressway?
A. _cuplogin._tcp.example.com
B. _cisco-uds._tcp.example.com
C. _collab-edge._tls.example.com
D. _collab-edge._udp.example.com
Answer: (SHOW ANSWER)

NEW QUESTION: 17
Which two statements about the functionality of a gatekeeper are true? (Choose two.)
A. Cisco Unified Communications Manager registers with a gatekeeper via SIP.
B. A gatekeeper can enable CAC and AAR.
C. Cisco Unified Communications Manager has gatekeeper functionality built in.
D. A gatekeeper can enable CAC, but not AAR.
E. Cisco Unified Communications Manager registers with a gatekeeper via H.323.
Answer: (SHOW ANSWER)

NEW QUESTION: 18
If your IP telephony administrator asks you to configure a local zone for your dial plan to control the volume of calls between two end points in a centralized multisite environment, which two types of Call Admission Control can be implemented? (Choose two.)
A. locations based
B. automated alternate routing
C. gatekeeper based
D. SRST
E. Cisco Unified Communications Manager based
Answer: A,B (LEAVE A REPLY)
Explanation/Reference:
Explanation:
Location-based call admission control (CAC) manages WAN link bandwidth in Cisco Unified Communications Manager. Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number.

NEW QUESTION: 19
Which statement about technology implementation strategy is true?
A. Cisco Unified Communications Manager Express can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.
B. SRST can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.
C. Cisco Unified Communications Manager Express in SRST mode can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.

D. SRST and MGCP fallback can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.
Answer: (SHOW ANSWER)

NEW QUESTION: 20
A. All the remaining Service Advertisement Framework forwarders are notified for their learned patterns.
B. The Service Advertisement Framework forwarder contacts all the remaining Service Advertisement Framework forwarders in the cluster.
C. All learned patterns are purged from the local cache after the Call Control Discovery PSTN Failover Duration parameter expires.
D. The Cisco Unified Communications Manager establishes a connection with the primary and secondary Service Advertisement Framework after the Learned Pattern IP Reachable Duration parameter expires.
E. Calls are routed to the PSTN gateway after the Call Control Discovery Learned Pattern IP Reachable Duration parameter expires.
F. Call Control Discovery immediately redirects all the calls to the PSTN gateway based on the learned patterns.
Answer: C,E (LEAVE A REPLY)

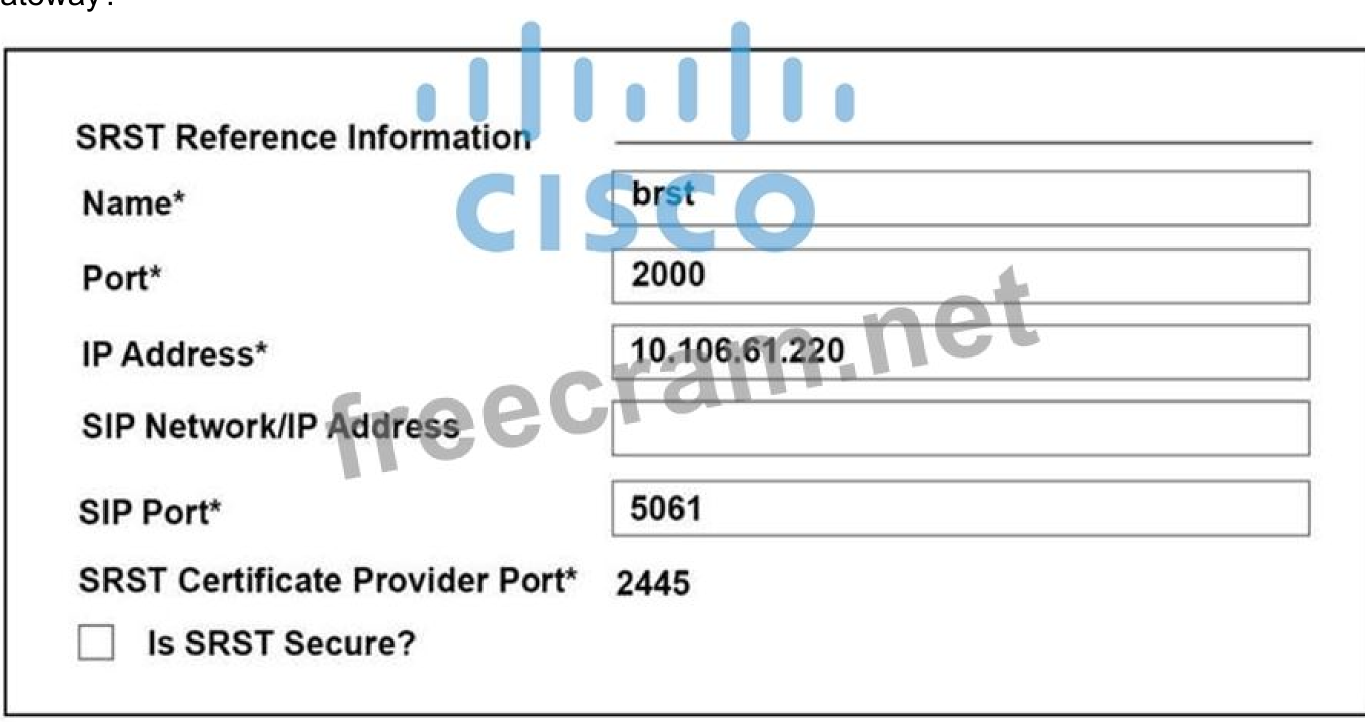
NEW QUESTION: 21
If the device pool in the phone record does not match the device pools in the matching subnet, what will the system consider the phone to be?
A. roaming
B. unknown
C. unregistered
D. new device
Answer: (SHOW ANSWER)

NEW QUESTION: 22
Cisco Unified border element is configured to support RSVP-based CAC. When is the RSVP path and reservation message sent and received?
A. Immediately after the call setup message is received and the reservation message is received after
B. The path and reservation messages are sent and received after the H.245 capabilities negotiation is completed.
C. The path is setup once the global command call rsvp-sync is configured.
D. 245 capabilities negotiation is completed.
E. The path and reservation messages are sent and received immediately after the call setup message is received.
Answer: (SHOW ANSWER)

NEW QUESTION: 23
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. DNS tracing
B. Dynamic DNS
C. OPTIONS ping
D. Continuous ping
Answer: (SHOW ANSWER)

NEW QUESTION: 24
Which zone will the VCS Control use to route calls to the VCS Expressway?
A. neighbor zone
B. DNS zone
C. ENUM zone
D. traversal client zone
Answer: (SHOW ANSWER)

NEW QUESTION: 25
Refer to the exhibit. Why are Cisco Unified IP Phones 9971 and 8961 not registering to the SRST gateway?



A. The SIP network or IP address is not provided.
B. Cisco Unified IP Phones 9971 and 8961 do not work in SRST.
C. The SRST reference is not configured on the phone device page.
D. The SIP port is configuring incorrectly in the reference.
Answer: (SHOW ANSWER)

NEW QUESTION: 26
When Cisco Extension Mobility is implemented, which CSS is used for calling privileges?
A. The user device profile line CSS combined with the device CSS of the physical phone used to log in the extension mobility user.
B. The combined line/device CSS of the user device profile.
C. The user device profile device CSS combined with the line CSS of the physical phone used to log in the extension mobility user.
D. The combined line/device CSS of the physical phone is used to log in the extension mobility user.
E. Only the user device profile device CSS is used.
Answer: A (LEAVE A REPLY)

NEW QUESTION: 27
Company X has deployed a VCS Control with a local zone and a traversal client zone. To facilitate external calls, VCS Expressway is deployed and traversal server zone is set up there. Video endpoints inside Company X have registered, but are unable to receive calls from outside endpoints.
Which option could be the cause of this issue?
A. The local zone on the VCS Control does not have a search rule configured.
B. When a traversal zone is set up on VCS Control only outbound calls are possible.
C. The traversal zone on the VCS Control does not have a search rule configured.
D. The access control list on the VCS Control must be updated with the IP for the external users.
Answer: (SHOW ANSWER)

NEW QUESTION: 28
When considering Cisco Unified Communications Manager failover, how many backup servers can be configured in a Cisco Unified Communications Manager Group?
A. 1
B. 5
C. 2
D. 4
E. 3
F. 6
Answer: (SHOW ANSWER)
Explanation/Reference:
Explanation:
A Cisco Unified Communications Manager group comprises a prioritized list of up to three Cisco Unified Communications Managers. Each group must contain a primary Cisco Unified Communications Manager, and it may contain one or two backup Cisco Unified Communications Managers.

Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/accm-861-cm/a02redun.html

NEW QUESTION: 29
Which two options must be selected in the SIP Trunk Security Profile configuration between Cisco Unified Communications Manager and Expressway? (Choose two.)
A. Accept out-of-dialog refer
B. Accept presence subscription
C. Enable application-level authorization
D. Accept replaces header
E. Transmit security status
F. Accept unsolicited notification
Answer: (SHOW ANSWER)

NEW QUESTION: 30
A. modify the enterprise parameter
B. modify the device pool
C. modify the line settings
D. modify the service parameter
Answer: (SHOW ANSWER)

NEW QUESTION: 31
An engineer has configured a Cisco EX60 to register with a Cisco VCS-C, but the device is not showing up as registered. During troubleshooting, which component will the engineer likely find missing in the configuration?
A. gatekeeper
B. TMS
C. MCU
D. default gateway
E. DNS
Answer: (SHOW ANSWER)

NEW QUESTION: 32
Which two statements about Cisco Unified Mobility are true? (Choose two.)
A. Mobile use of Cisco Jabber requires mobility.
B. Mobility is enabled by default.
C. Single number reach is a feature of mobility.
D. Mobility is used to support mobile phones in Cisco Unified Communications Manager.
E. Mobility must be enabled per user.
Answer: (SHOW ANSWER)

NEW QUESTION: 33
Which situation requires TCP port 443 to be open for packets that are sourced from the Internet with a destination in the corporate DMZ?
A. when video endpoints that reside on the Internet require administrative access to the Cisco Expressway Edge
B. when you require encrypted calls to endpoints on your corporate LAN
C. when you want to enable calls to web applications by using HTTP
D. when you require administrative access to the Cisco Expressway Edge from the Internet
Answer: (SHOW ANSWER)
Explanation/Reference:
Explanation:
VCS/Expressway servers only use port 443 for administrative access. No call session of any sort on a VCS or Expressway use port 443. Also, the "Unified Communications Mobile and Remote Access via Cisco VCS Deployment Guide" identified port 443 as the administrative access port from pubic internet to VCS Expressway.
Reference:
http://www.cisco.com/c/dam/en/us/td/docs/telepresence/infrastructure/vcs/config_guide/X8-1/Mobile-Remote-Access-via-VCS-Deployment-Guide-X8-1-1.pdf

NEW QUESTION: 34
Which two call scenarios are supported by AAR? (Choose two.)
A. calls incoming through a gateway device within one location which terminates at an IP phone within another location
B. calls from remote sites where SRST is activated
C. calls from Cisco Unified Communications Manager Extension Mobility users who roam to different sites
D. calls with CTI route points as the origin or destination of calls
E. calls originating from an IP phone within one location which terminates at an IP phone within another location

Answer: A,E (LEAVE A REPLY)

NEW QUESTION: 35

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 TMS server under Registrations > whitelist

B. on the voice-enabled firewall at the edge of the network

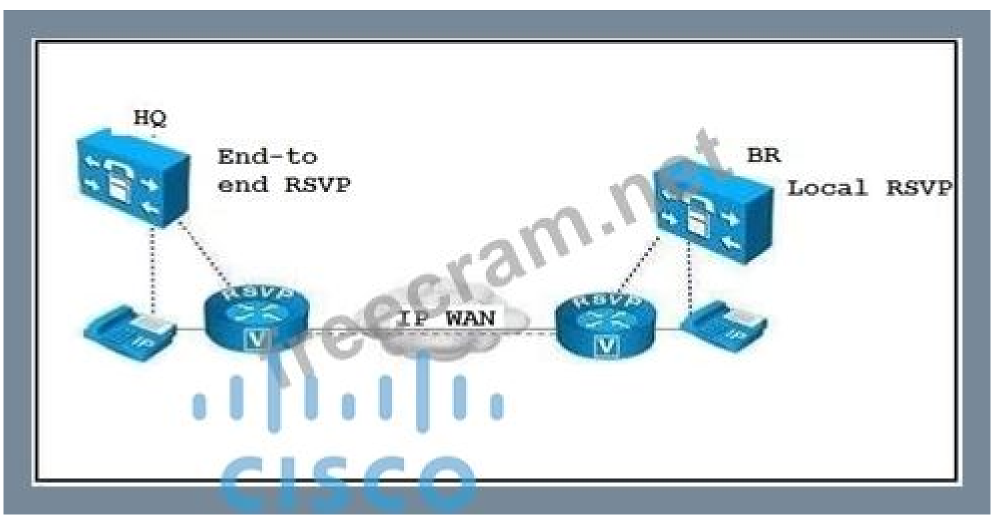
C. on the VCS under Configuration > registration > configuration

D. on the VCS under System > configuration > Registrations

Answer: (SHOW ANSWER)

NEW QUESTION: 36

Refer to the exhibit.



The HQ Cisco Unified Communications Manager has been configured for end-to-end RSVP.

The BR Cisco Unified Communications Manager has been configured for local RSVP. RSVP between the locations assigned to the IP phones and SIP trunks at each site are configured with mandatory RSVP.

When a call is placed from the IP phone at HQ to the BR phone at the BR site, which statement is true?

A. The Cisco Unified Communications Manager at HQ will fall back to local RSVP and place the call. No RSVP end-to-end will occur.

B. RSVP end-to-end will occur.

C. The Cisco Unified Communications Manager at HQ will use end-to-end RSVP. The BR Cisco Unified Communications Manager will use local RSVP.

D. The call will fail.

E. The call will proceed as a normal call with no RSVP reservation.

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:
A possible cause is that the same router is being used as the calling and called RSVP agents, and that router is not running the latest IOS version, which supports loopback on RSVP reservation. Make sure that the router is running the latest IOS version.

NEW QUESTION: 37
When configuring CAPF, user intervention is required when using which authentication mode?
A. by existing certificate (precedence to LSC)
B. by existing certificate (precedence to MIC)
C. user intervention never required
D. by authentication string
E. by null string
Answer: (SHOW ANSWER)

NEW QUESTION: 38
An engineer is configuring URI calling within the same cluster. Which four actions must be taken to accomplish this configuration? (Choose four.)
A. Configure SIP route patterns.
B. Configure the directory URI partition and calling search space.
C. Associate the directory URIs to directory numbers.
D. Activate the URI service in Cisco Unified Serviceability.
E. Configure SIP trunk.
F. Assign directory URIs to users.
G. Configure the SIP profile.
H. Configure the URI service parameters.
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BK_CD2F83FA_00_cucm-system-guide-90/CUCM_BK_CD2F83FA_00_system-guide_chapter_0101111.html

NEW QUESTION: 39
When implementing a Media Termination Point, what determines the number of sessions that is supported on each DSP?
A. the codecs that are used in universal transcoding mode
B. the number of full-duplex media streams
C. the Cisco Unified Communications Manager node setting
D. the size of the cluster that is being designed
Answer: (SHOW ANSWER)

NEW QUESTION: 40

A. The login will fail because only a single Cisco Extension Mobility profile is allowed.
B. The user must select the desired profile.
C. The user must login to both profiles in the order they are presented.
D. The user may login to both profiles in any order.
E. Login will only be allowed to multiple profiles if the service parameter Allow Multiple Logins is enabled.

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:

Users access Cisco Extension Mobility by pressing the Services or Applications button on a Cisco Unified IP Phone and then entering login information in the form of a Cisco Unified Communications Manager UserID and a Personal Identification Number (PIN). If a user has more than one user device profile, a prompt displays on the phone and asks the user to choose a device profile for use with Cisco Extension Mobility.

NEW QUESTION: 41

Refer to the exhibit.

```
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 8.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ-1_MTP
!
dspfarm profile 1 mtp
 codec pass-through
 rsvp
 maximum sessions software 20
 associate application SCCP
!
interface Serial10/1
 description IP-WAN
 ip address 10.1.4.101 255.255.255.0
 duplex auto
 speed auto
 ip rsvp bandwidth 64
!
```

How many calls are permitted by the RSVP configuration?

A. eight G.729 calls

B. two G.729 calls
C. one G.711 call
D. one G.729 call and one G.711 call
E. four G.729 calls
Answer: (SHOW ANSWER)

NEW QUESTION: 42
Which statement about RSVP is true?
A. MTP is typically used for RSVP agent configuration.
B. If the RSVP agent cannot reserve the required bandwidth on first effort, it enables low latency queuing and tries to connect the call again.
C. Cisco Unified Communications Manager uses the RSVP-configured bandwidth between sites as a method of determining if there is sufficient bandwidth for the call.
D. Cisco Unified Communications Manager uses an RSVP-enabled infrastructure and an RSVP-controlled agent to request a bandwidth reservation from the network in order to place a call.
Answer: (SHOW ANSWER)
Explanation/Reference:

NEW QUESTION: 43
An engineer is setting up a Cisco VCS Cluster with SIP endpoints only. While configuring the Cisco VCS peers, which signaling protocol is used between peers to determine the best route for calls?
A. SIP
B. H.323
C. SCCP
D. MGCP
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference:
http://www.cisco.com/c/dam/en/us/td/docs/telepresence/infrastructure/vcs/config_guide/X8-7/Cisco-VCS-Cluster-Creation-and-Maintenance-Deployment-Guide-X8-7.pdf (page 4, basic configuration is done, third point)

NEW QUESTION: 44
On which two call logs is the media encryption enforced in a Collaboration Edge design? (Choose two.)
A. Expressway-C to Expressway-E
B. Expressway-E to outside-located endpoint
C. Expressway-C to internal endpoint
D. Expressway-E to Cisco Unified Communications Manager
E. Expressway-C to Cisco Unified Communications Manager

Answer: (SHOW ANSWER)

NEW QUESTION: 45
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 voice saf dndball
B. show ip saf neighbors
C. show saf neighbors
D. show eigrp service-family ipv4 neighbors
E. show eigrp address-family ipv4 neighbors
Answer: (SHOW ANSWER)

NEW QUESTION: 46
Which option is a valid test scenario to verify that Cisco Unified Communications Manager failover works and endpoints register with the backup call manager?
A. During a predetermined maintenance window, stop the Cisco IP Phone Services service on the primary Unified CM. Devices should reregister with the backup Unified CM in the Cisco CallManager Group.
B. During a predetermined maintenance window, stop the Unified CM service on the primary call manager. Devices should reregister with the backup Unified CM in the CallManager Group.
C. 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.
D. 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.
Answer: (SHOW ANSWER)

NEW QUESTION: 47
In a Centralized Call processing architecture, you have deployed Extension Mobility (EM) feature. After the deployment of EM, when one of the end- users tries to login to the IP phone, the Error

25 is displayed on the screen. What three things should you do to resolve this issue? (Choose three.)

A. upgrade the firmware of the IP Phone to the latest version

B. subscribe device profile to EM phone service in case the enterprise subscription of EM Service is disabled

C. activate EM feature service under Cisco Unified Serviceability

D. associate EM Device profile with the end-user

E. update EM Phone Service URL to point to the publisher

F. subscribe the MAC address of the IP Phone to EM Service

Answer: (SHOW ANSWER)

NEW QUESTION: 48

For which VoIP protocol does a gatekeeper provide address translation and control access?

A. SIP

B. H.248

C. Skinny

D. H.323

Answer: (SHOW ANSWER)

NEW QUESTION: 49

When an incoming PSTN call arrives at an H.323 gateway, how does the calling number get normalized to a global E.164 number with + prefix in Cisco Unified Communications Manager?

A. Normalization is achieved by local route group that is assigned to the H.323 gateway.

B. Normalization is done using the gateway incoming called party prefixes based on number type.

C. Normalization is done using translation patterns.

D. Normalization is done using route patterns.

E. Normalization is done using the gateway incoming calling party prefixes based on number type.

Answer: (SHOW ANSWER)

NEW QUESTION: 50

A. AAR reroutes calls through the WAN when the PSTN connection fails.

B. AAR reroutes calls through the PSTN when Cisco Unified Communications Manager CAC blocks a call.

C. AAR reroutes calls through the PSTN upon WAN failure.

D. AAR reroutes calls to another phone number when a Cisco IP Phone unregisters.

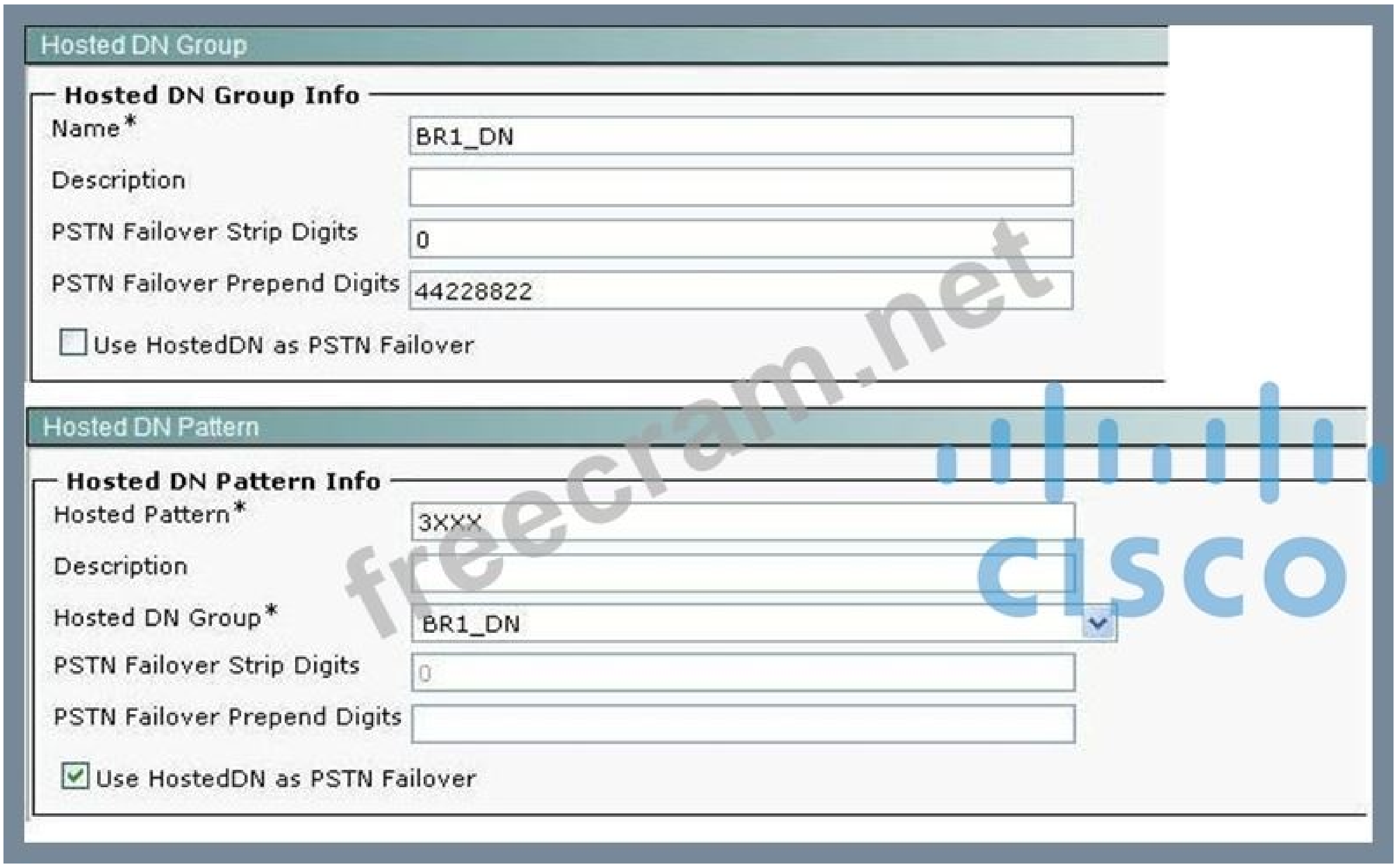
Answer: (SHOW ANSWER)

NEW QUESTION: 51

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. A TMS server is needed to allow the firewall traversal to occur between the VCS Expressway and the VCS Control servers.
B. The firewall at Company X must have all SIP ALG functions disabled.
C. The firewall at Company X requires a rule to allow all traffic from the DMZ to pass to the same network that the VCS Control is on.
D. The VCS Control should not be on the LAN, but it must be located in the DMZ with the Expressway.
Answer: (SHOW ANSWER)

NEW QUESTION: 52
How is the region assigned to a device such as an IP phone?
A. Regions are assigned directly in the device configuration page.
B. Regions can be assigned only through a device pool.
C. Regions can be assigned either directly on the device configuration page or through the device pool. If both configurations exist, the device pool region configuration takes precedence.
D. Regions can be assigned either directly on the device configuration page or through the device pool. If both configurations exist, the device region configuration takes precedence.
Answer: (SHOW ANSWER)

NEW QUESTION: 53
Refer to the exhibit.



Which pattern will be advertised try the Cisco Unified Communications Manager?

A. 3XXX and the ToDID will be 44228822.

B. 3XXX and the ToDID will be 0:+44228822.

C. 3XXX and the ToDID will be 0:+

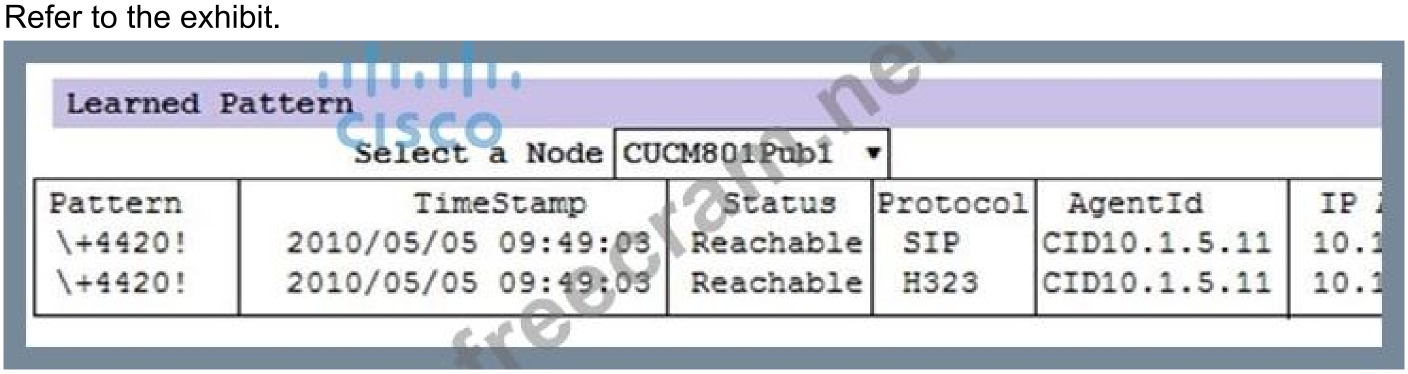
D. 3XXX and the ToDID will be 0:.

E. 3XXX and the TnOID will be 0:44228822.

Answer: (SHOW ANSWER)

NEW QUESTION: 54

Refer to the exhibit.



Pattern	TimeStamp	Status	Protocol	AgentId	IP A
\+4420!	2010/05/05 09:49:03	Reachable	SIP	CID10.1.5.11	10.1
\+4420!	2010/05/05 09:49:03	Reachable	H323	CID10.1.5.11	10.1

When a user presses a speed dial to +442079460255 when the SAF network is down, which event should occur?

A. The call will reroute via the PSTN with the constructed PSTN number as 00442079460255.

B. The call will reroute via the PSTN with the constructed PSTN number as +442079460255.

C. The call will fail because the called number will be 2079460255.

D. The call will reroute via the PSTN with the constructed PSTN number as 442079460255.

E. The call will fail because the ToDID is 0:.
Answer: (SHOW ANSWER)

NEW QUESTION: 55
Company X has a primary and a backup Cisco Unified Communications Manager instance. The administrator had to do maintenance on the primary node and did a shutdown, which resulted in a failover to the backup node. What happens when the primary node comes back online?
A. Nothing, the endpoints only failover when the node they lose connection to their registered node.
B. Endpoints detect that the primary is back and reregisters automatically.
C. The primary node becomes the backup node.
D. The backup node must be shut down first to allow the endpoints to realize that the primary node is online again.
Answer: (SHOW ANSWER)

NEW QUESTION: 56
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 Survivable Remote Site Telephony
B. MGCP fallback
C. Cisco Unified Communications Manager Express in SRST mode
D. Cisco Unified Communications Manager Express
E. SCCP fallback
Answer: (SHOW ANSWER)

NEW QUESTION: 57
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 calls should have dropped when the WAN failed so users would be instantly aware.
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 the phones should have started ringing the instant the WAN connection failed to signal the start of SRST mode.
Answer: (SHOW ANSWER)

NEW QUESTION: 58

You are deploying a remote office setup that connects with Cisco Unity Communications Manager at a hub location. You have an available dedicated bandwidth of 20% from the 2-Mb/s WAN circuit for VoIP that supports a maximum of 17 calls. Which codec do you configure in Cisco Unity Communications Manager to achieve this?

A. iSAC
B. G.722
C. G.729
D. G.711
E. iLBC
F. GSM-FR

Answer: (SHOW ANSWER)

NEW QUESTION: 59

What is the purpose of a SAF Client?

A. To reside in the Cisco IOS software, and to communicate with the SAF forwarder
B. To learn about and advertise or subscribe information about SAF network services
C. To decode address information and route calls to and from the end points
D. To pass IP information from the CUCM to the endpoint

Answer: (SHOW ANSWER)

NEW QUESTION: 60

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

Answer: C (LEAVE A REPLY)

NEW QUESTION: 61

Which statement describes locations based CAC?

A. It is a purely logical assignment and does not reflect the actual topology or the actual bandwidth available.
B. Locations based CAC is used to determine the codec used between sites.
C. Cisco Unified Communications Manager uses an arbitrary bandwidth value for each location.
D. Each device may have more than one location assigned.
E. The configured bandwidth limit is dependent on the destination location of the call.

Answer: (SHOW ANSWER)

NEW QUESTION: 62
An engineer must enable video desktop sharing between a Cisco Unified Communications Manager registered video endpoint and a Cisco VCS registered video endpoint.
Which protocol must be enabled in SIP profile for VCS SIP trunk on Cisco Unified Communications Manager?
A. RDP
B. H.264
C. H.224
D. H.263
E. BFCP
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/about/press/internet-protocol-journal/back-issues/table-contents-57/153-binary.html

NEW QUESTION: 63
When an incoming PSTN call arrives at an H.323 gateway, how does the called number get normalized to an internal directory number in Cisco Unified Communications Manager?
A. Normalization is done using the gateway incoming calling party prefixes based on number type.
B. Normalization is done by configuring the significant digits for inbound calls on the H.323 gateway configuration in Cisco Unified Communications Manager.
C. Normalization is done using route patterns.
D. Normalization is achieved by local route group that is assigned to the H.323 gateway.
Answer: (SHOW ANSWER)

NEW QUESTION: 64
Which symbol is required for globalized call routing?
A. /
B. %
C. *
D. +
E. ;

Answer: (SHOW ANSWER)

NEW QUESTION: 65
Which interface can you use to configure alarms and traces in Cisco Unified Communications Manager?
A. Cisco Unified OS Administration
B. Cisco Unified Communications Manager Administration
C. real-time monitoring
D. disaster recovery system
E. Cisco Unified Serviceability
F. Cisco Unified Reporting
Answer: (SHOW ANSWER)

NEW QUESTION: 66
A voice engineer is enabling video capabilities between H.323 and SIP endpoints. Which component allows for standardized caller addresses between the endpoints?
A. search rules
B. SIP route pattern
C. policy service
D. transform
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: http://www.cisco.com/en/US/tech/tk652/tk701/technologies_configuration_guide_chapter09186a00800eadee.html

NEW QUESTION: 67
In Cisco Unified Communications Manager, where do you configure the default bit rate for audio and video devices?
A. Enterprise Parameters
B. Region under Region Information
C. Cisco CallManager service under Service Parameter Configuration
D. Enterprise Phone Configuration
Answer: B (LEAVE A REPLY)
Explanation/Reference:
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_5_1/ccmcfg/bccm-851-cm/b02regio.html

NEW QUESTION: 68
Which parameter prevents the Cisco TelePresence Video Communication Server from responding to endpoint gatekeeper discovery requests?

A. On the VCS, navigate to Configuration, Protocols, H.323, and set Auto Registration to off.
B. On the VCS, navigate to Configuration, Registration, Configuration, and set Auto Registration to off.
C. On the VCS, navigate to Configuration, Registration, Allow List, and set Auto Registration to off.
D. On the VCS, navigate to Configuration, Protocols, H.323, and set Auto Discover to off.
Answer: D (LEAVE A REPLY)

NEW QUESTION: 69
Which action configures the registration of transcoder resources?
A. Cisco IOS software registers transcoder resources with SCCP.
B. Cisco IOS software does not register transcoder resources.
C. Cisco IOS software registers transcoder resources with H.323.
D. Cisco IOS software registers transcoder resources with SIP.
Answer: (SHOW ANSWER)

NEW QUESTION: 70
A. TLC queries for the time zone as part of configuration.
B. TLC adjusts the time change appropriately.
C. TLC uses its local time for all systems.
D. TLC produces an error and must be run remotely.
Answer: (SHOW ANSWER)

NEW QUESTION: 71
Which technologies provide remote-site redundancy for Cisco IP Phones during a WAN failure?
A. SRST and AAR
B. TEHO and MGCP fallback
C. SRST and MGCP fallback
D. SRST and TEHO
Answer: (SHOW ANSWER)

NEW QUESTION: 72
Refer to the exhibit.

```
voice-card 0
 dspfarm
 dsp services dspfarm

sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 6.0
sccp

sccp ccm group 1
 associate ccm 2 priority 1
 associate profile 1 register HQ_XCODER

dsp farm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 2
 associate application SCCP
 no shutdown
```

You have configured transcoder resources in both an IOS router and a Cisco Unified Communications Manager. When you review the configurations in both devices the IP addresses and transcoder names are correct, but the transcoder is failing to register with the Cisco Unified Communications Manager.

Which command needs to be edited to allow the transcoder to register properly?

A. The associate profile and dsp farm profile numbers need to match associate ccm 2 command.

B. The associate ccm 2 priority 1 command needs to be changed so the ccm value matches identifier 1 in the sccp ccm 10.1.1.1 command.

C. The sccp ccm group number needs to match the associate ccm 2 command.

D. The maximum sessions command must match the number of codecs configured under the dsp farm profile.

E. The sccp ccm group number must match the voice-card number.

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:

The value of the IP address should match the IP address in the ip source-address command.

NEW QUESTION: 73

The corporate WAN has been extended to two newly acquired sites and it includes gatekeeper support.

Each site has a Cisco CallManager and an H.323 gateway that allows connection to the PSTN.

Which connection method is best for these two new customers?

A. H.225 trunk (gatekeeper-controlled)

B. SIP trunk

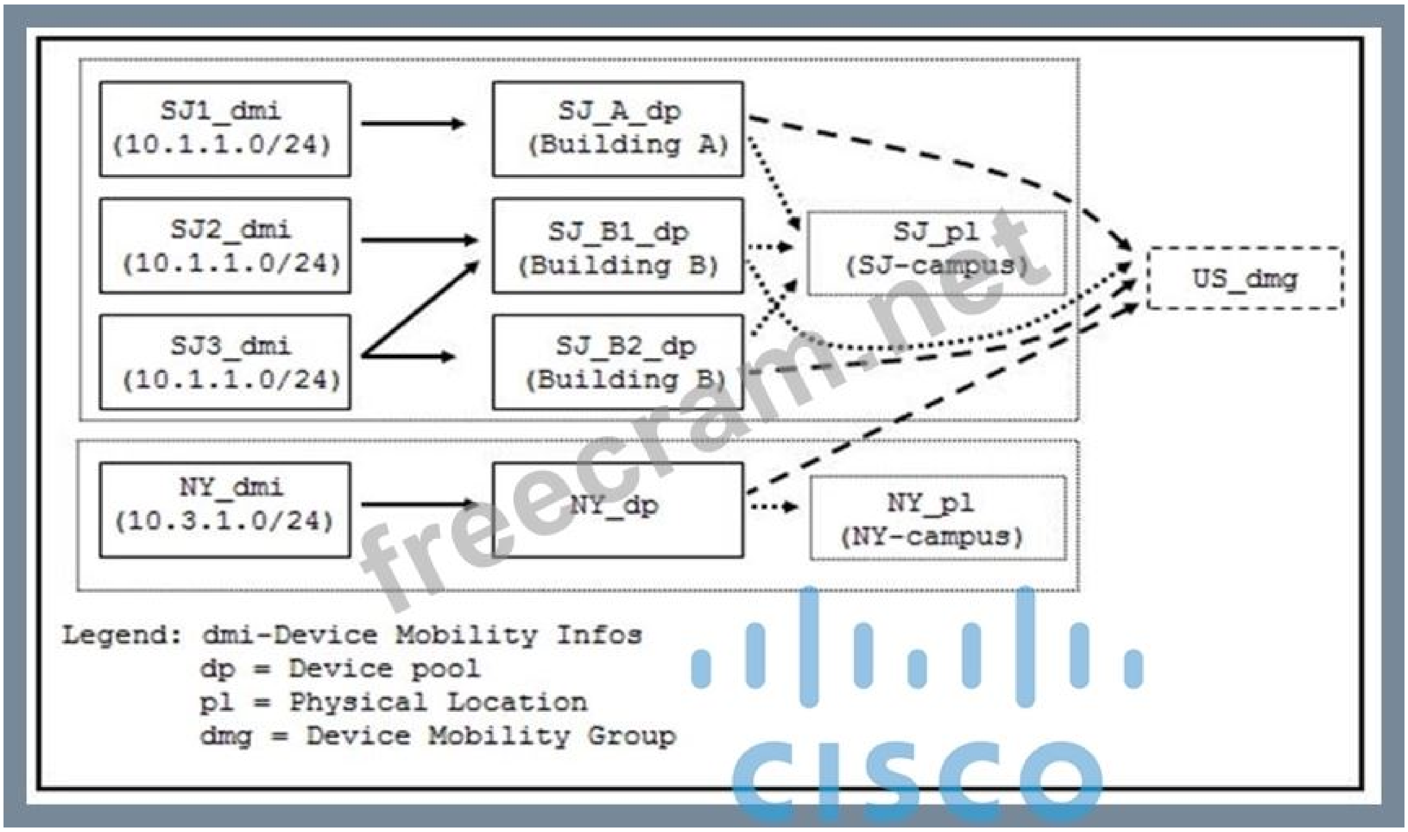
C. intercluster trunk (gatekeeper-controlled)

D. intercluster trunk (non-gatekeeper controlled)

Answer: C (LEAVE A REPLY)

NEW QUESTION: 74

Refer to the exhibit.



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

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

B. The 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.

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

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

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

Answer: (SHOW ANSWER)

NEW QUESTION: 75

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

A. Configure the publisher node only in the SAF Forwarder configuration page.

B. Configure the correct node in the EIGRP configuration of the gateway router that is associated with the Cisco Unified Communications Manager node.

C. Append an @ symbol at the end of the client label value in the SAF Forwarder configuration page.

D. Configure the SAF Security Profile Configuration to support only a single node.
Answer: (SHOW ANSWER)

NEW QUESTION: 76
Which two options are requirements for deploying an H.323 gateway with Cisco Unified Communications Manager? (Choose two.)
A. The Media Exchange Interface Capability Timer must be set to less than 20.
B. The Media Exchange Timer must be set to less than 20.
C. Cisco voicemail ports must be active.
D. The H.245TCSTimeout timer must be set to at least 25.
E. Cisco Unified Communications Manager and the H.323 gateway must be configured use the same TCP port for H.323 calls.
Answer: (SHOW ANSWER)

NEW QUESTION: 77
Company X wants to implement RSVP-based Call Admission Control and move away from the current location-based configuration.
Where does the administrator go to create a default profile?
A. System > Service Parameters > RSVP
B. System > Service Parameters > Call Manager > Clusterwide parameters > RSVP
C. on each MGCP gateway at all remote locations
D. System > Call Manager > Clusterwide > Service Parameters > RSVP
Answer: (SHOW ANSWER)

NEW QUESTION: 78
An engineer is configuring Global Dial Plan Replication. On one cluster, she would like to prevent the local cluster from routing calls to a specific pattern learned via ILS. What should be configured within the local CUCM cluster to accomplish this?
A. Create a block translation pattern
B. Create a block route pattern
C. Create a block transformation pattern
D. Create a block learned pattern

Answer: (SHOW ANSWER)

NEW QUESTION: 79
Which two statements about Cisco Unified Communications Manager Extension Mobility are true?
(Choose two.)
A. After an autogenerated device profile is created, you can associate it with one or more users.
B. An autogenerated device profiles can be loaded on a device at the same time as a user profile.
C. A device can adopt a user profile even when no user is logged in.
D. A device profile has most of the same attributes as a physical device.
E. Devices can be configured to allow more than one user to be logged in at the same time.
Answer: C,D (LEAVE A REPLY)
Explanation/Reference:
Reference: http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_administration_guide_chapter09186a0080153e60.html#wp1092734

NEW QUESTION: 80
A. enterprise parameters
B. enterprise phone configuration
C. service parameters
D. common phone profile
Answer: (SHOW ANSWER)

NEW QUESTION: 81
Which three characters should you avoid entering in the description? (Choose three.)
A. < >
B. %
C. #
D. $
E. @
F. &
Answer: (SHOW ANSWER)

NEW QUESTION: 82
Which three devices or applications support call preservation? (Choose three.)
A. a software conference bridge
B. Cisco Unified IP Phone 7962G
C. an annunciator
D. SIP trunks
E. JTAPI applications
F. TAPI applications

G. CTI applications
H. third-party H.323 endpoints
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/CUCM_BK_CD2F83FA_00_cucm-system-guide-90/CUCM_BK_CD2F83FA_00_system-guide_chapter_01011.html#CUCM_RF_C98194B0_00

NEW QUESTION: 83
Which statement is true when device mobility mode is enabled or disabled in the Phone Configuration window?
A. The combined service parameter settings and the device mobility mode phone settings will be used.
B. The service parameter settings take precedence over the device mobility mode phone settings.
C. The default settings will be used due to the conflicts.
D. The device mobility mode phone settings take precedence over the service parameter settings.
Answer: D (LEAVE A REPLY)

NEW QUESTION: 84
What is the purpose of a region?
A. to specify only the audio codec used within a site
B. to specify the audio and video codecs used between specific sites
C. to specify the range of codecs used between all other sites
D. to specify only the video codec used between other specific sites
Answer: B (LEAVE A REPLY)

NEW QUESTION: 85
Which two options are mandatory requirements for setting up Hardware MTP resource on Cisco IOS routers? (Choose two.)
A. one audio codec
B. T1 PRI card
C. available PVDMs or DSP resources on the router
D. RFC2833 for DTMF
E. LTI based local transcoding resources
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cminterop/configuration/15-mt/dia-15- mt-book/vc-enh-confr-vgr.html

NEW QUESTION: 86

Assume that the Cisco IOS SAF Forwarder is configured correctly. Which minimum configurations on Cisco Unified Communications Manager are needed for the SAF registration to take place?
A. SAF Trunk, CCD Requesting Service, and CCD Advertising Service
B. SAF Trunk, SAF Security Profile, SAF Forwarder, and CCD Advertising Service
C. SAF Trunk, SAF Security Profile, SAF Forwarder, and CCD Requesting Service
D. SAF Trunk, SAF Security Profile, and SAF Forwarder
E. SAF Trunk, SAF Security Profile, SAF Forwarder, CCD Requesting Service, and CCD Advertising Service
Answer: (SHOW ANSWER)

NEW QUESTION: 87
What is the default value for the Drop Ad Hoc Conference service parameter?
A. Never
B. Drop Ad Hoc Conference When Creator Leaves
C. When No On-Net Parties Remain in the Conference
D. When No Off-Net Parties Remain in the Conference
Answer: (SHOW ANSWER)

NEW QUESTION: 88
Which statement about SIP precondition is most correct?
A. When configuring SIP precondition, the IP phones and SIP trunk must have access to an RSVP agent.
B. When configuring SIP precondition, the SIP trunk must have access to an RSVP agent.
C. RSVP agents are only required for the IP phones. SIP trunks require RSVP agents only when fall back to local RSVP is configured.
D. When configuring SIP precondition, the IP phones must have access to an RSVP agent.
E. SIP trunk will always require RSVP agents regardless of what RSVP type is configured.
Answer: (SHOW ANSWER)

NEW QUESTION: 89
In which two locations can you verify that a phone has a standby Cisco Unified Communications Manager?
(Choose two.)
A. RTMT
B. phone menu
C. phone webpage
D. Cisco Unified Serviceability
Answer: B,C (LEAVE A REPLY)

NEW QUESTION: 90
A. The bandwidth settings of the site are fulfilling on-net call volume.

B. AAR is routing some of the calls.
C. The location-based CAC does not work properly.
D. The LBM service is malfunctioning.
Answer: (SHOW ANSWER)

NEW QUESTION: 91
When configuring a video ISDN gateway, which two actions are requirements for the Cisco Preferred Architecture for Enterprise Collaboration? (Choose two.)
A. Use an ! (exclamation point) at the end of each ISDN number, as a suffix.
B. Use SIP instead of H.323.
C. Use H.323 instead of SIP.
D. Perform dial string manipulation on Cisco Unified Communications Manager.
E. Use an * (asterisk) at the end of each ISDN number, as a suffix.
Answer: B,D (LEAVE A REPLY)

NEW QUESTION: 92
When you configure TEHO for long-distance calls and use the local PSTN gateways as fallback, how many route patterns do you require for a cluster with five sites that are located in different area codes?
A. 10 when not using a local route group
B. 5 when using a local route group
C. 6 when using a local route group
D. 15 when not using a local route group
Answer: (SHOW ANSWER)

NEW QUESTION: 93
How many nodes can a phone establish a connection to at the same time?
A. 3
B. 1
C. 4
D. 2
Answer: (SHOW ANSWER)

NEW QUESTION: 94
Which two statements about the use of the Intercluster Lookup Service in a multicluster environment are true? (Choose two.)
A. Cisco Unified Communications Manager uses the ILS to support intercluster URI dialing.
B. 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.
C. Directory URI replication does not need to be enabled individually for each cluster.
D. To enable URI replication in a cluster, check the Exchange Directory URIs with Remote Clusters check box that appears in the SIP trunk configuration menu.
E. 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.
Answer: (SHOW ANSWER)

NEW QUESTION: 95
Refer to the exhibit.

```
device 1:
interface 1
10.10.10.1
telephony-service
ip source-address 10.10.10.1 secondary 10.10.10.2

device 2:
interface 1
10.10.10.2
telephony-service
ip source-address 10.10.10.1 secondary 10.10.10.2
```


Which option describes the effect of this configuration?
A. It implements Cisco Unified CME redundancy.
B. It implements Cisco IOS redundancy.
C. It creates dial peers.
D. It configures failover.
E. It implements HSRP.
Answer: (SHOW ANSWER)

NEW QUESTION: 96
Which message can a Cisco VCS use to monitor the Presence status of an endpoint?
A. call-started
B. end-call
C. registration

D. start-call
Answer: (SHOW ANSWER)

NEW QUESTION: 97
What is the default DSCP/PHB for TelePresence video conferencing packets in Cisco Unified Communications Manager?
A. CS4/32
B. EF/46
C. CS3/24
D. CS6/48
E. AF41/34
Answer: (SHOW ANSWER)

NEW QUESTION: 98
While troubleshooting a connectivity issue between Cisco Unified Communications Manager, Expressway- C, and Expressway-E, an engineer sees this output in the Expressway-E logs:
Event="Authentication Failed" Service="SIP" Src-ip="10.50.2.1"
Src-port="25723" Detail="Incorrect authentication credential for user"
Protocol "TLS" Method="OPTIONS" Level="1"
What is the cause of this issue?
A. The Expressway-C Traversal Client username/password do not match the Expressway-E Traversal Server username/password.
B. The Expressway-C Traversal Server username/password do not match the Expressway-E Traversal Zone username/password.
C. The Expressway-C Traversal Server username/password do not match the Expressway-E Traversal Client username/password.
D. The Expressway-C Traversal Zone username/password do not match the Expressway-E Traversal Client username/password.
E. The Expressway-C Traversal Zone username/password do not match the Expressway-E Traversal Zone username/password.
Answer: (SHOW ANSWER)

NEW QUESTION: 99
Which two statements describe a transcoder? (Choose two.)
A. Transcoders convert lower bit rate streams to higher bit rate streams.
B. Transcoders may be combined with conferencing resources on a single DSP to conserve resources.
C. Transcoders may operate as a Media Termination Point when the codecs are of the same sampling rate but use different codec types.
D. Transcoders can convert an input stream of one codec to an output stream that uses a different codec type.

E. Transcoders can only connect two streams of the same codec type that use different sampling rates.
Answer: (SHOW ANSWER)

NEW QUESTION: 100
A. 3
B. 1
C. 5
D. 6
E. 2
F. 4
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmcfg/CUCM_BK_CDF59AFB_00_admin-guide-90/CUCM_BK_CDF59AFB_00_admin-guide_chapter_0100.html

NEW QUESTION: 101
How can the location setting be modified to resolve poor call quality?
A. No adjustment to location setting is needed
B. Decrease the audio bandwidth setting
C. Remove the audio bandwidth setting
D. Mark the bandwidth between the locations as unlimited
Answer: B (LEAVE A REPLY)

NEW QUESTION: 102
An engineer must resolve a call failure issue. When using RTMT, the engineer notices that the Location Bandwidth Manager-OutOfResources counter is showing a positive value. Which option is the cause of the call failure?
A. lack of transcoding resources
B. lack of conferencing resources
C. lack of audio or video bandwidth
D. lack of video bandwidth
E. lack of audio bandwidth
Answer: (SHOW ANSWER)

NEW QUESTION: 103
Which three steps configure Cisco Unified Survivable Remote Site Telephony for SIP phones? (Choose three.)
A. Configure voice register pool.
B. Configure voice register global dn.

C. Configure an SRST reference.
D. Configure a phone NTP reference.
E. Configure the SIP registrar.
F. Configure telephony service.
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/srst_setting_up_using_sip.html

NEW QUESTION: 104
You want to configure Cisco VCS SIP endpoints and H.323 endpoints so that they communicate with one another. To do this, which format must you use in the Search Rule?
A. name@domain
B. name@IP Address (192.168.100.0)
C. name@hostname
D. IP Address (192.168.100.0)
Answer: (SHOW ANSWER)

NEW QUESTION: 105
How many Cisco Unified Mobility destinations can be configured per user?
A. 6
B. 4
C. 1
D. 10
Answer: (SHOW ANSWER)

NEW QUESTION: 106
A video endpoint is configured with SIP only. What does a video endpoint use to register with the VCS Control?
A. IP address
B. system name
C. SIP URI
D. MAC address
Answer: (SHOW ANSWER)

NEW QUESTION: 107
Which Cisco IOS command is used for internal SAF Clients to check SAF learned routes?
A. show eigrp address-family ipv4 saf
B. show voice saf routes
C. show voice saf detail
D. show eigrp service-family ipv4 saf
E. show voice saf dndb all
Answer: (SHOW ANSWER)
Explanation/Reference:
Explanation:
Router# show voice saf dnDb all
Total no. of patterns in db/max allowed : 1/6000
Patterns classified under dialplans (private/global) : 0/1 Informational/Error stats - Patterns w/ invalid expr detected while add : 0 Patterns duplicated under the same instance : 0 Patterns rejected overall due to max capacity : 0 Attempts to delete a pattern which is invalid : 0 Last successful DB update @ 2009:12:14 15:42:45:967

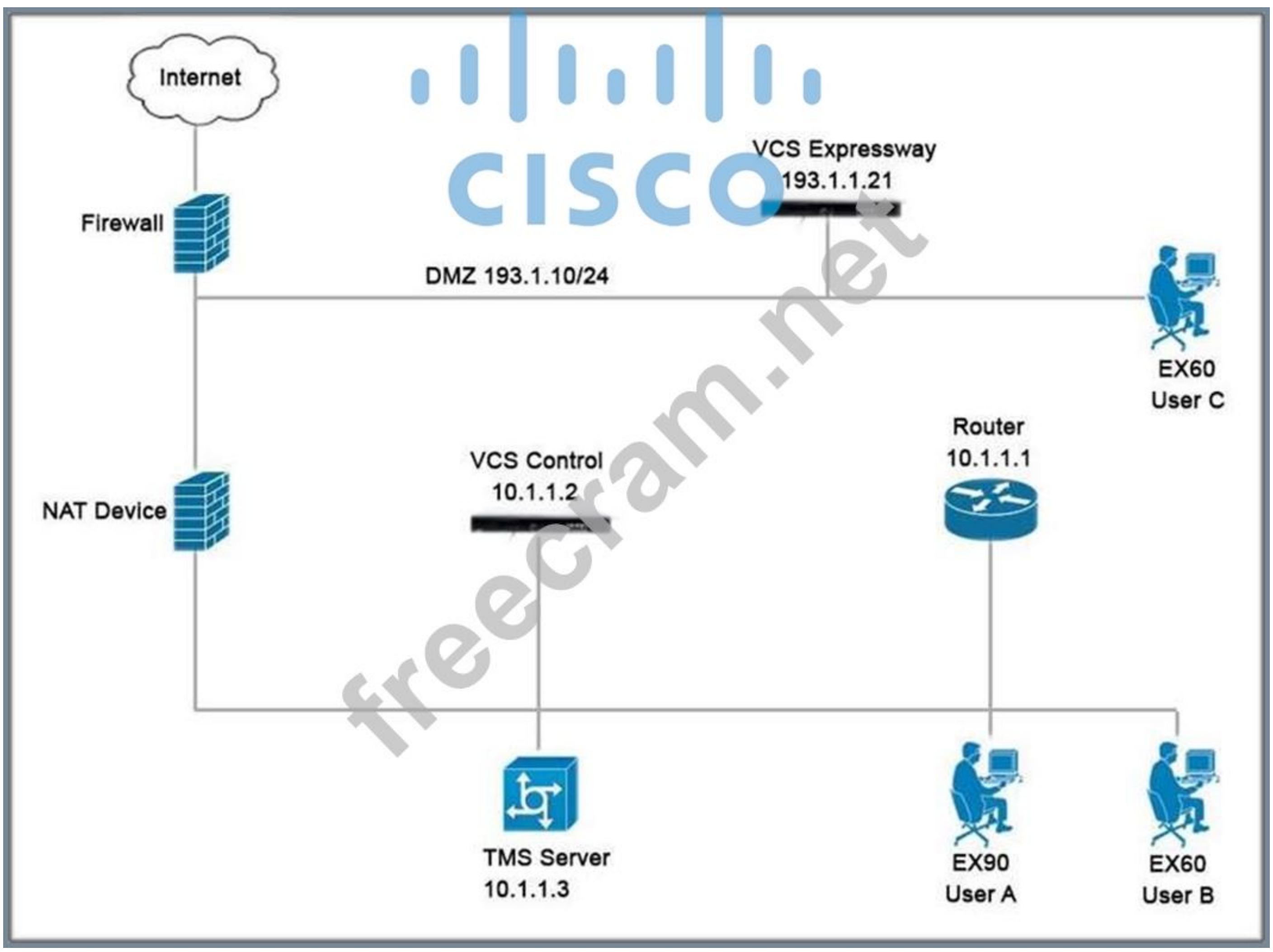
NEW QUESTION: 108
When an H.323 trunk is added for Call Control Discovery, which statement is true?
A. The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Enable SAF check box should be selected in the trunk configuration.
B. The H.323 trunk is added by selecting 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.
C. 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.
D. The H.323 trunk is added by selecting Call Control Discovery Trunk and then selecting H.323 as the protocol to be used.
Answer: (SHOW ANSWER)

NEW QUESTION: 109
Which two statements are true when considering a Cisco VoIP environment for regional configuration?
(Choose two.)
A. The default codec does not matter if you have defined a hardware MTP in your Cisco Unified Communications Manager environment.
B. G.711 requires 128K of bandwidth per call.

C. To deploy a Cisco H.323 gatekeeper, you must configure MTP resources on the gatekeeper and only use G.711 between regions.
D. G.729 requires 24K of bandwidth per call.
Answer: A,D (LEAVE A REPLY)

NEW QUESTION: 110
A. Automesh
B. Fullmesh
C. ILS updates
D. multicast
Answer: (SHOW ANSWER)

NEW QUESTION: 111
Refer to the exhibit.



An engineer receives a ticket to troubleshoot a one-way audio issue with these symptoms:

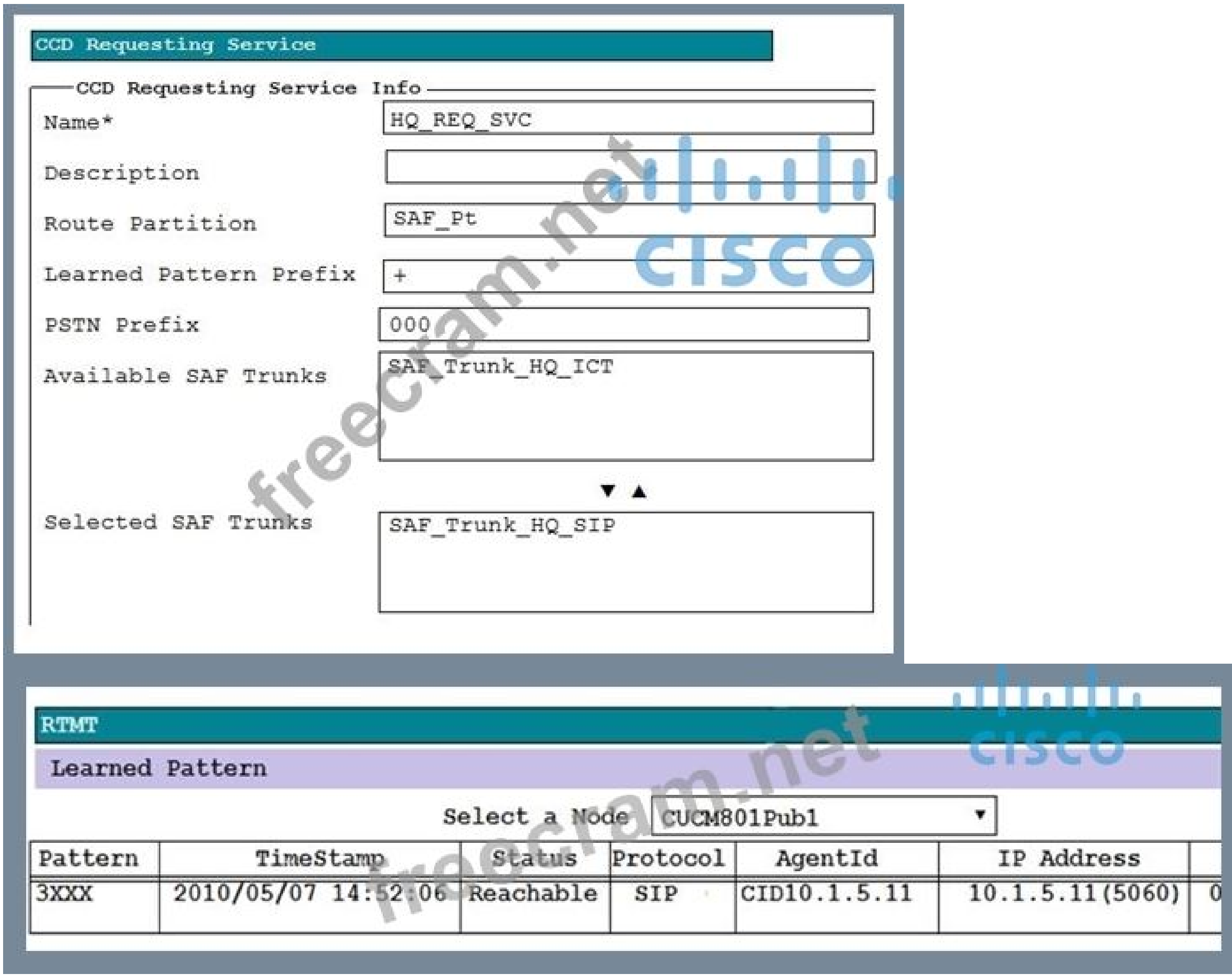
- User A can hear user B and vice versa.
- User A can hear user C, however user C cannot hear user A.
- User B can heat user C, however user C cannot hear user B.

Which two properties are the most likely reasons for this issue? (Choose two.)
A. The Cisco EX60 default gateway of user C is missing from the network configuration.
B. The NAT device is allowing only RTP/RTCP ports from the internal network to the DMZ.
C. The Cisco EX60 of user C is not responding to requests coming from the TMS server.
D. The Cisco VCS Expressway is not responding to the SIP INVITE coming from the Cisco VCS Control.
E. The router does not have a route back from the DMZ to the internal network.
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: http://www.cisco.com/c/en/us/support/docs/voice/voice-quality/5219-fix-1way-voice.html

NEW QUESTION: 112
In a cluster-wide deployment, what is the maximum number of Service Advertisement Framework forwarders to which the Cisco Unified Communications Manager can connect?
A. 4
B. as many as are configured
C. 3
D. 1
E. 6
F. 2
Answer: (SHOW ANSWER)

NEW QUESTION: 113
Which action configures CAC utilizing only Cisco Unified Communications Manager software?
A. Configure Cisco Unified Communications Manager RSVP-enabled locations.
B. Configure Cisco Unified Communications Manager regions.
C. Configure Cisco Unified Communications Manager MTPs.
D. Configure Cisco Unified Communications Manager locations.
Answer: (SHOW ANSWER)

NEW QUESTION: 114
Refer to the exhibit.



When the Manager places a call to 3001 when the SAF network is down, what happens?

A. The call fails.

B. The call is rerouted to the PSTN with the constructed PSTN number as +442288223001

C. The call is rerouted to the PSTN with the constructed PSTN number as 442288223001

D. The call is rerouted to the PSTN with the constructed PSTN number as 0002288223001

E. The call is rerouted to the PSTN with the constructed PSTN number as +0002288223001

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:

When the SAF forwarder loses network connection with its call-control entity, the SAF forwarder withdraws those learned patterns that were published by the call control entity. In this case, CCD requesting service marks those learned patterns as unreachable via IP, and the calls gets routed through the PSTN gateway.

NEW QUESTION: 115

Your company's main number is 408-526-7209, and your employee's directory numbers are 4-digit numbers. Which option should be configured in CUCM if you want outgoing calls from a 4-digit internal directory number to be presented as a 10-digit number?

A. AAR group
B. calling party transformation in Route Pattern
C. route pattern filed in Route Pattern
D. IOS translation rules
Answer: (SHOW ANSWER)

NEW QUESTION: 116
Which device must be accessible from the public Internet in a Collaboration Edge environment?
A. VCS Control
B. Expressway-E
C. Expressway-C
D. Cisco IM and P
E. Cisco Unified Communications Manager
Answer: (SHOW ANSWER)

NEW QUESTION: 117
With Cisco Extension Mobility, when a user logs in to a phone type which has no user device profile, what will happen to the phone?
A. The phone immediately logs the user off.
B. The phone creates a new device profile automatically.
C. The phone crashes and reboots.
D. The phone takes on the default clusterwide device profile.
Answer: (SHOW ANSWER)

NEW QUESTION: 118
When implementing a dial plan for multisite deployments, what must be present for SRST to work successfully?
A. incoming and outgoing COR lists
B. configuration of the gateway as an MGCP gateway
C. dial peers that address all sites in the multisite cluster
D. translation patterns that apply to the local PSTN for each gateway
Answer: (SHOW ANSWER)

NEW QUESTION: 119
Which two commands verify Cisco IP Phone registration? (Choose two.)
A. show telephony-service ephone-dn
B. show ephone registered
C. show voice register session-server
D. show ccm-manager hosts
E. show sip-ua status registrar
Answer: (SHOW ANSWER)

NEW QUESTION: 120

Refer to the exhibit.

```
voice service saf
 profile trunk-route1
 session protocol sip interface Loopback1 transport tcp port 506(
 !
profile dn-block 1 alias-prefix 1972555
 pattern 1 type extension 4XXX
 !
profile dn-block 2
 pattern 1 type global 14087071222
 !
profile callcontrol 1
 dn-service
  trunk-route 1
  dn-block 1
  dn-block 2
 !
!
!
channel 1 vrouter SAF asystem 1
 subscribe callcontrol wildcarded
 publick callcontrol 1
!
```


Which CSS is used at the HQ Cisco Unified Communications Manager to reroute calls via the PSTN when the SAF network is unavailable?

A. the phone AAR CSS configured at the phone device

B. the phone line CSS

C. the phone line/device combined CSS

D. the phone device CSS

E. No special CSS is required. If SAF patterns are accessible, the PSTN reroute is automatic.

F. the SAF CSS configured on the CCD requesting service

Answer: (SHOW ANSWER)

NEW QUESTION: 121

Which two configurable options are available to enable Early Offer for calls over a Cisco Unified Communications Manager SIP trunk? (Choose two.)

A. Use Trusted Relay Point

B. Media Termination Point Required

C. Early Offer support for voice and video calls Mandatory (insert MTP if needed)

D. Accept Audio Codec Preferences in Received Offer

E. No Media Termination Point Required

Answer: (SHOW ANSWER)

NEW QUESTION: 122
Which solution is needed to enable presence and extension mobility to branch office phones during a WAN failure?
A. SRST without MGCP fallback
B. Cisco Unified Communications Manager Express in SRST mode
C. SRST with VoIP dial peers to Cisco Unified Communications Manager Express
D. SRST with MGCP fallback
Answer: (SHOW ANSWER)

NEW QUESTION: 123
What are the tasks required to route calls from H323 to SIP and viceversa? (Choose two.)
A. Config-protocols-Sip-Config-Mode-On
B. Config-protocols-Interworking-On
C. Config-protocols-Interworking-Off
D. Config-protocols-Interworking-registered only
E. Config-protocols-H323-H323Mode-On
Answer: (SHOW ANSWER)

NEW QUESTION: 124
Which three CLI commands are used when configuring H.323 call survivability for all calls? (Choose three.)
A. call preserve
B. telephony-service
C. transfer-system
D. voice service voip
E. call-router h323-annexg
F. h323
Answer: (SHOW ANSWER)

NEW QUESTION: 125
Which action configures transcoding resources in Cisco Unified Communications Manager to function with branch office Cisco IP Phones?
A. Configure the branch office IP phones with MRGs and MRGLs.

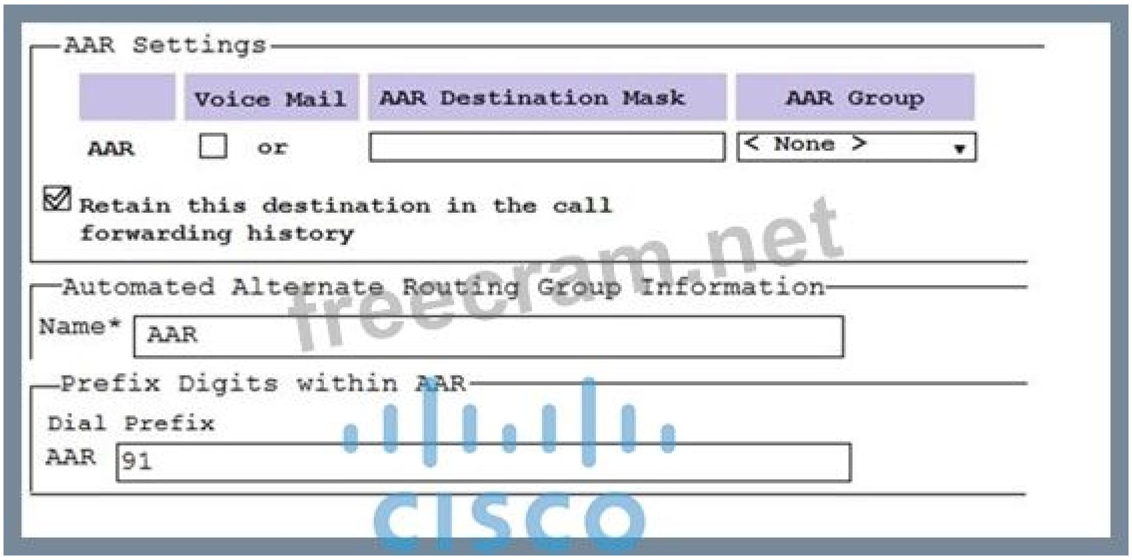
B. Configure the branch office IP phones with locations.
C. Configure the branch office IP phones with regions.
D. Configure the branch office IP phones with CSS and partitions.
Answer: (SHOW ANSWER)

NEW QUESTION: 126
Which service, when correctly configured in Enterprise Parameters, permits the phone configuration files to be encrypted?
A. Cisco IP Voice Media Streaming App Service
B. Cisco TFTP Service
C. Cisco CTL Client Service
D. Cisco CTIManager Service
E. Cisco Unified Communications Manager Service
Answer: (SHOW ANSWER)

NEW QUESTION: 127
Which two are gatekeeper-controlled trunk options that support gatekeeper call administration control?
(Choose two.)
A. H.323
B. H.225
C. intracluster
D. H.245
E. intercluster
Answer: B,E (LEAVE A REPLY)

NEW QUESTION: 128
Which statement about the function of a gatekeeper is true?
A. A gatekeeper can simplify the dial plan between many different Cisco Unified Communications Manager clusters.
B. A gatekeeper improves call routing between servers within a single Cisco Unified Communications Manager cluster.
C. Gatekeepers can be implemented to deploy RSVP-based CAC.
D. A gatekeeper can replace the dial plan of a Cisco Unified Communications Manager cluster.
Answer: (SHOW ANSWER)

NEW QUESTION: 129
Refer to the exhibit.



A user in RTP calls a phone in San Jose during congestion with Call Forward No Bandwidth (CFNB) configured to reach cell phone 4085550150. The user in RTP sees the message "Not Enough Bandwidth" on their phone and hears a fast busy tone. Which two conditions can correct this issue? (Choose two.)

A. The calling phone (RTP) needs to have AAR Group value of AAR under the AAR Settings.

B. The called phone (San Jose) needs to have AAR Group value of AAR under the AAR Settings.

C. The calling phone (RTP) needs to have the AAR destination mask of 914085550150 configured under the AAR Settings.

D. The calling phone (RTP) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.

E. The called phone (San Jose) needs to have the AAR destination mask of 914085550150 configured under the AAR Settings.

F. The called phone (San Jose) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:

Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth. With automated alternate routing, the caller does not need to hang up and redial the called party.

NEW QUESTION: 130

A. Service Parameters

B. Enterprise Parameters

C. Endpoint Configuration

D. Device Pool configuration

Answer: (SHOW ANSWER)

NEW QUESTION: 131

How do RSVP-enabled locations differ from Cisco Unified Communications Manager locations?

A. RSVP is configured in Cisco Unified Communications Manager independent of the ISR.

B. RSVP enables AAR within Cisco Unified Communications Manager.

C. RSVP is configured in the ISR independent of Cisco Unified Communications Manager.

D. RSVP is topology aware.

Answer: (SHOW ANSWER)

NEW QUESTION: 132

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

A. SRST supports more Cisco IP Phones than Cisco Unified Communications Manager Express in SRST mode.

B. MGCP fallback is required for ISDN call preservation.

C. Cisco Unified Communications Manager Express in SRST mode supports more Cisco IP Phones than SRST.

D. MGCP fallback functions with SRST.

Answer: (SHOW ANSWER)

NEW QUESTION: 133

Refer to the exhibit.

```
router eigrp SAF
 !
  sf-interface FastEthernet0/0
  topology base
  exit-sf-topology
 exit-service-family
```

Which configuration elements must match for adjacent neighbors to establish a SAF neighbor relationship?

A. the topology base configurations

B. the label name specified in the router eigrp command and the autonomous-system number

C. the sf-interface configuration

D. the label name specified in the router eigrp command

E. the autonomous-system number specified in the service-family ipv4 autonomous-system command

Answer: (SHOW ANSWER)

NEW QUESTION: 134

Which two features or functions can be implemented without utilizing MTP Resources? (Choose two.)

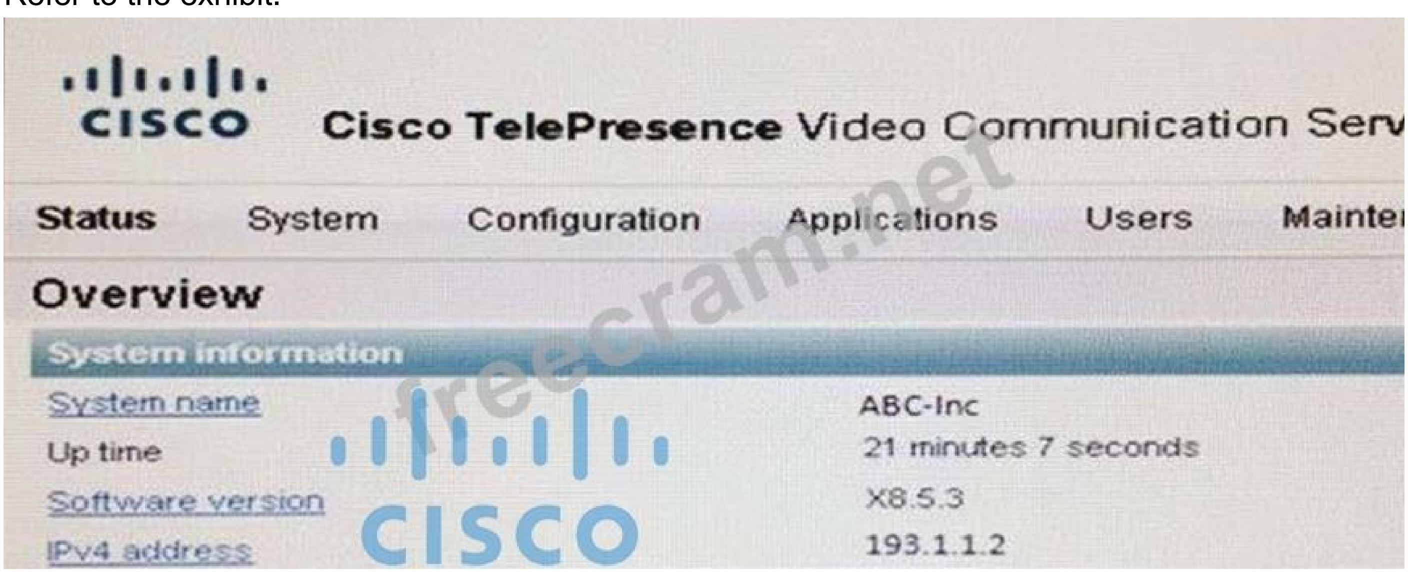
A. DTMF conversion from INBAND to RTP-NTE (rfc-2833)

B. IPV6 to IPV4 interworking

C. multicast music on hold

D. H.323 outbound fast start
E. SIP delayed offer
Answer: C,E (LEAVE A REPLY)
Explanation/Reference:
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/trunks.html

NEW QUESTION: 135
The network administrator at Enterprise X is creating the guidelines for a new IPT deployment consisting of a large number of remote offices. Every user within Enterprise X is assigned a directory number of 5 digits.
Which option might cause an issue in a multisite deployment?
A. The maximum number of IP phones are in use at each remote site.
B. All media streams are necessarily routed through the central office for calls to establish correctly.
C. Overlapping DID ranges are allocated to each site.
D. MoH cannot be provided for the remote sites.
Answer: (SHOW ANSWER)

NEW QUESTION: 136
Refer to the exhibit.


An engineer is deploying a new Cisco VCS Expressway for a company and has configured the IP address and the system name. After logging into the Cisco VCS Expressway admin page, the engineer sees this output. Which four options must be configured to complete the Cisco VCS Expressway system configuration? (Choose four.)
A. DHCP server
B. LDAP server
C. SIP URI
D. DNS server

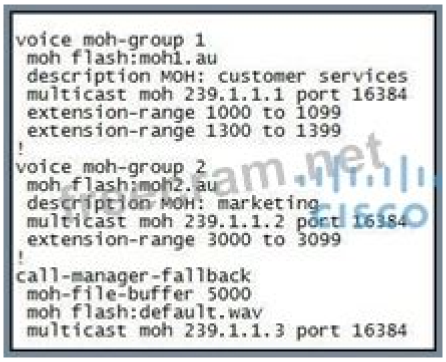
E. SIP server
F. Cisco Unified Communications Manager IP address
G. NTP server
H. security certificate
Answer: (SHOW ANSWER)

NEW QUESTION: 137
Which statement best describes a Media Termination Point?
A. In Cisco Unified Communications Manager 6.0, a Media Termination Point is required when two endpoints use a common method for sending DTMF between them such as for all calls using SIP trunks.
B. A Media Termination Point corrects a mismatch in DTMF transport types when using Delayed Offer for inbound SIP calls.
C. A hardware Media Termination Point can be used to transcode a-law to mu-law and vice versa, but both connections must utilize the same packetization periods.
D. A Media Termination Point is an entity that accepts two full-duplex G.711 streams. It bridges the media streams together and allows them to be set up and torn down independently.
E. When using H.323 Fast Start, for inbound and outbound Fast Start, a Media Termination Point is required.
Answer: (SHOW ANSWER)

NEW QUESTION: 138
Which two bandwidth management parameters are available during the configuration of Cisco Unified Communications Manager regions? (Choose two.)
A. Default Video Call Rate
B. Max Number of Video Sessions
C. Max Video Call Bit Rate (Includes Audio)
D. Default Audio Call Rate
E. Max Audio Bit Rate
Answer: (SHOW ANSWER)

NEW QUESTION: 139

Refer to the exhibit.



```
voice moh-group 1
 moh flash:moh1.au
 description MOH: customer services
 multicast moh 239.1.1.1 port 16384
 extension-range 1000 to 1099
 extension-range 1300 to 1399
!
voice moh-group 2
 moh flash:moh2.au
 description MOH: marketing
 multicast moh 239.1.1.2 port 16384
 extension-range 3000 to 3099
!
call-manager-fallback
 moh-file-buffer 5000
 moh flash:default.wav
 multicast moh 239.1.1.3 port 16384
```

What music on hold audio source will be heard if a user at extension 1372 places the user at extension
3041 on hold?

A. default.wav

B. moh1.au

C. moh1.wav

D. moh2.wav

E. moh2.au

Answer: (SHOW ANSWER)

NEW QUESTION: 140

A. Pool Profile

B. Group Profile

C. Device Profile

D. Specific Profile

Answer: (SHOW ANSWER)

NEW QUESTION: 141

An engineer is troubleshooting a dial path etc" and it gives a config snapshot.

There are three fields populated with "CCNPCOLAB", "Add Suffix" and "@Cisco.com" Mode : .....

Pattern type : Regex

Pattern string : @cisco.com

Pattern Behaviour :.......

Replace string : ccnpcollab

Target : .....

State : Disabled

A. Can not route call

B. Sent to Cisco.com

C. sent as CCNPCOLAB

D. Sent as CCNPCOLAB@cisco.com

Answer: (SHOW ANSWER)

NEW QUESTION: 142

An administrator is setting up analog phones that connect to a Cisco VG310. Which type of gateway or trunk on Cisco Unified Communication Manager for the Cisco VG310 must the administrator set up to allow the phones to have the call pickup feature?

A. H.323 gateway
B. SCCP gateway
C. H.225 trunk
D. MGCP gateway
E. SIP trunk

Answer: (SHOW ANSWER)

Explanation/Reference:
Reference:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_5_1/ccmsys/accm-851-cm/a08procl.html#wp1150402

NEW QUESTION: 143

Which three options describe the main functions of SAF Clients? (Choose three.)

A. starting Cisco Unified Communications Manager services throughout the cluster
B. registering the router as a client with the SAF network
C. registering Cisco Unified Communications Manager subscribers with the publisher
D. subscribing to SAF network services
E. providing publishing services to the SAF network
F. integrating with Cisco IM and Presence for additional services

Answer: (SHOW ANSWER)

NEW QUESTION: 144

Which two Cisco Extension Mobility attributes are available in the user device profile? (Choose two.)

A. regions
B. NTP information
C. phone button template
D. description

Answer: (SHOW ANSWER)

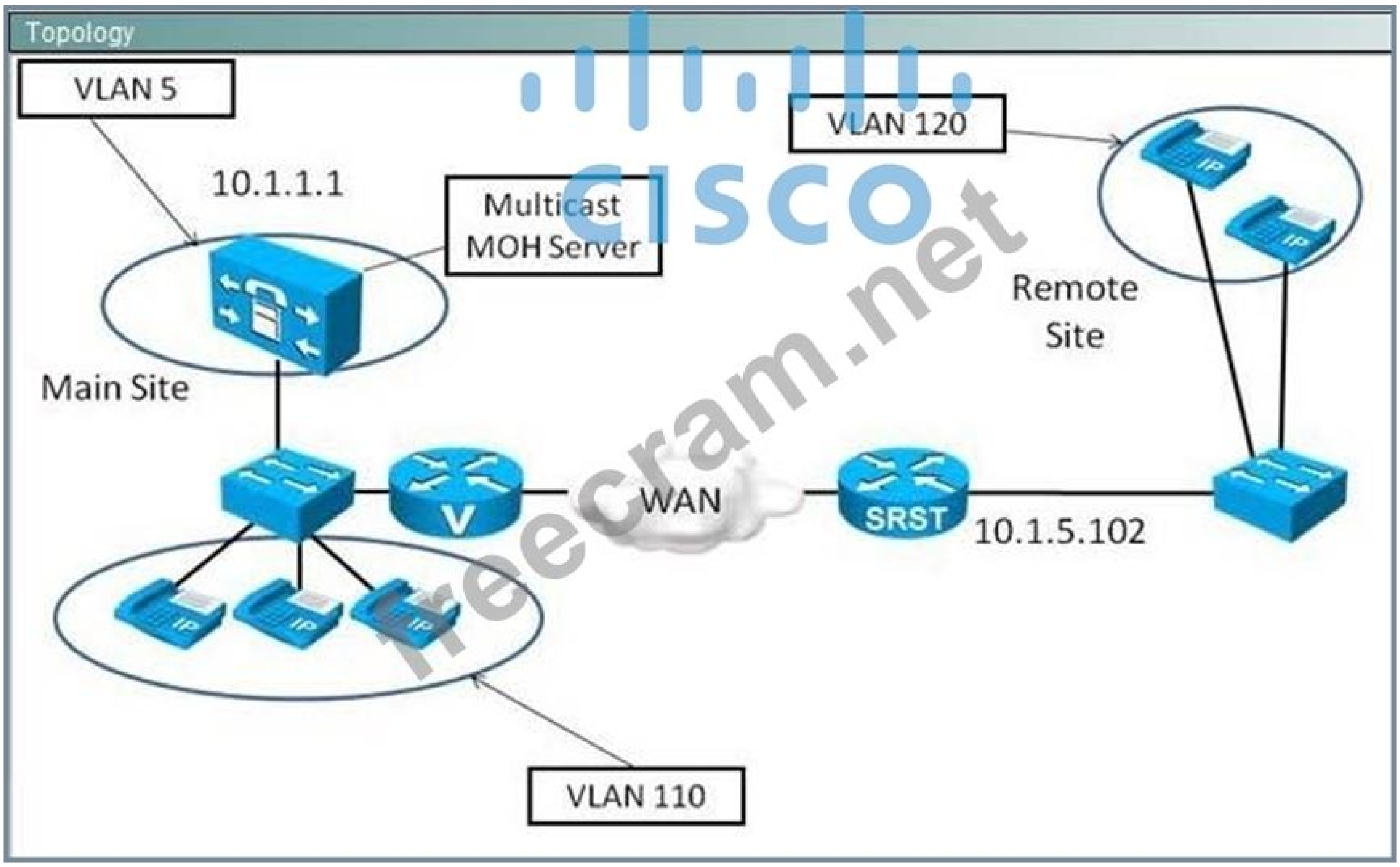
NEW QUESTION: 145

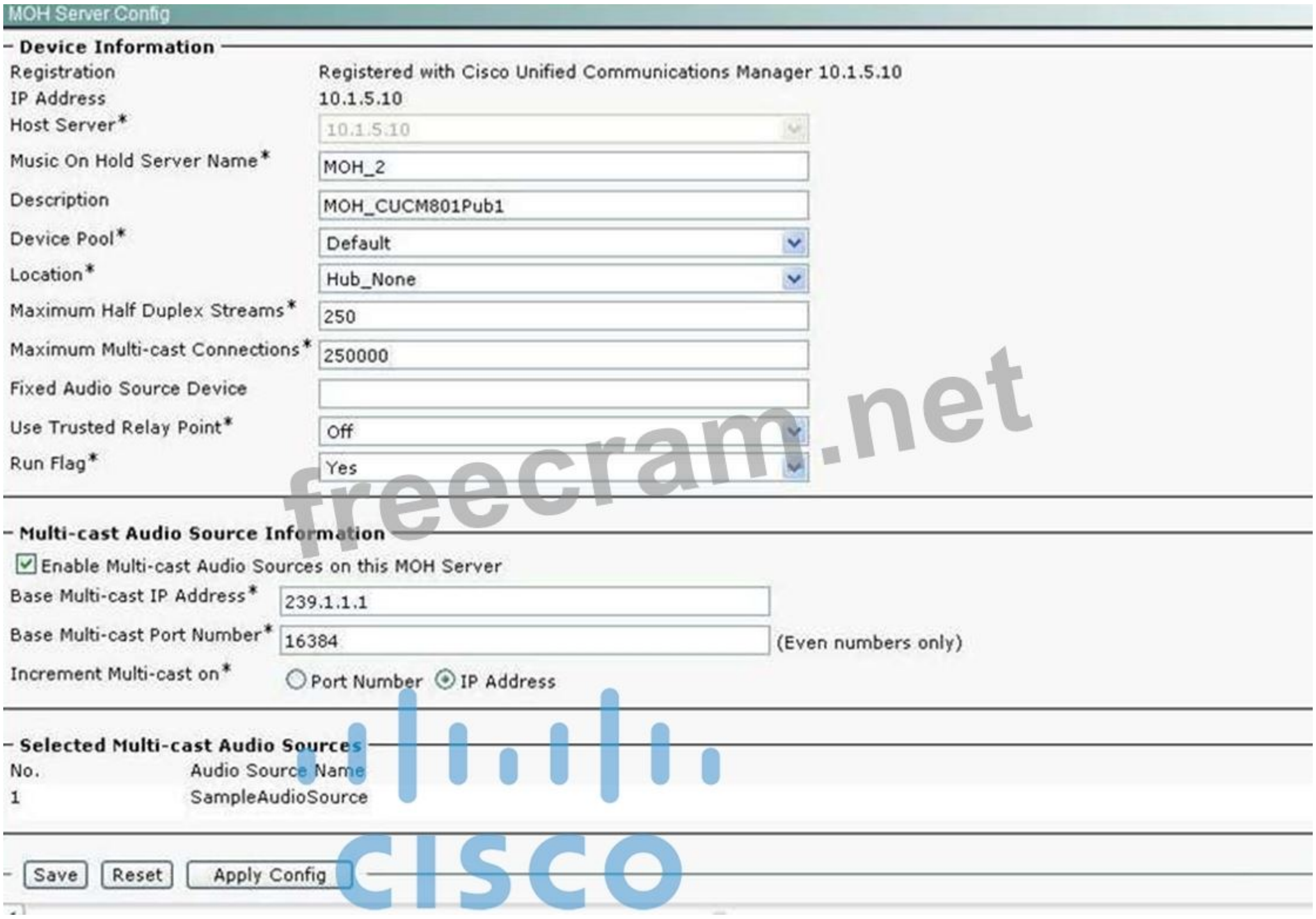
An engineer is performing an international multisite deployment and wants to create an effective backup method to access TEHO destinations in case the call limit triggers. Which technology provides this ability?

A. LRG
B. AAR
C. CFUR

D. SRST
Answer: (SHOW ANSWER)

NEW QUESTION: 146
Refer to the exhibit.





To stream multicast MOH to the remote site across the WAN, what should the minimum value for the Max Hops be configured as?

A. 3

B. 1

C. 2

D. 4

Answer: (SHOW ANSWER)

NEW QUESTION: 147

Which function can be implemented without MTP resources?

A. terminating a media stream that uses the same codec

B. music on hold

C. SIP early offer

D. DTMF relay conversion

Answer: (SHOW ANSWER)

NEW QUESTION: 148

In a Cisco Unified Communications Manager centralized call processing model, what is the best CAC method recommended for this type of deployment?

A. gateway-based

B. QoS-based
C. region-based
D. location-based
E. RSVP-based
F. gatekeeper-based
Answer: (SHOW ANSWER)

NEW QUESTION: 149
Refer to the exhibit.

```
gatekeeper
zone local ClusterA lab.com 192.168.3.254
zone local ClusterB lab.com
zone prefix ClusterA 511*
zone prefix ClusterA 521*
zone prefix ClusterB 512*
zone prefix ClusterB 522*
bandwidth interzone default 64
bandwidth interzone zone ClusterB 48
gw-type-prefix 1#* default-technology
no shutdown
```

How many calls can be placed to Cluster B?
A. one G.711 call
B. three G.729 calls
C. one G.711 and three G.729 calls
D. There is no limit for incoming calls to Cluster B. Outgoing calls are limited to one G.711 and three
E. 729 calls.
Answer: (SHOW ANSWER)

NEW QUESTION: 150
A. 2500
B. 2000
C. 500
D. 1500
E. 1000
F. 5000
Answer: (SHOW ANSWER)

NEW QUESTION: 151

Which two configurations can you perform to allow Cisco Unified Communications Manager SIP trunks to send an offer in the INVITE? (Choose two.)
A. Select the Display IE Delivery check box in the gateway configuration.
B. Enable the Media Termination Point Required option on the SIP trunk.
C. Enable the Early Offer Support for Voice and Video Calls option on the SIP profile.
D. Execute the isdn switch-type primary-ni command globally.
E. Select the Enable Inbound FastStart check box on the Cisco Unified Communications Manager servers.
F. Select the SRTP Allowed check box on the SIP trunk.
Answer: (SHOW ANSWER)

NEW QUESTION: 152
Which statement best describes globalized call routing in Cisco Unified Communications Manager?
A. The CSS of all phones contain partitions assigned to route patterns that are in global format.
B. All called numbers sent out to the PSTN are in E.164 with the + prefix format.
C. All phone directory numbers are configured as an E.164 with the + prefix.
D. Call routing is based on numbers represented as an E.164 with the + prefix format.
E. All incoming calling numbers on the phones are displayed as an E.164 with the + prefix.
Answer: (SHOW ANSWER)

NEW QUESTION: 153
Which two steps must you take when implementing TEHO in your environment? (Choose two.)
A. Implement ICT trunks to remote locations.
B. Implement local failover.
C. Implement SIP to POTS.
D. Implement centralized failover.
E. Load-balance route lists within the cluster.
F. Load-balance PRI connections.
Answer: B,D (LEAVE A REPLY)

NEW QUESTION: 154

When you configure a globalized dial plan, in which three ways can you enable ingress gateways to process calls? (Choose three.)
A. Configure the called-party transformation settings for incoming calls on H.323 gateways.
B. Configure translation patterns in the partitions used by the gateway calling search space.
C. Configure SIP trunks between Cisco Unified Communications Manager clusters.
D. Configure a remote site device pool.
E. Configure a hunt group.
F. Configure the gateway with prefix digits to add necessary country and region codes.
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/8x/uc8x/dialplan.html#wp1153166

NEW QUESTION: 155
Which two options are functionalities of subzones in a Cisco VCS deployment? (Choose two.)
A. Apply registration, authentication, and media encryption policies.
B. Manage bandwidth to restrict standard definition endpoints from using more than 2 Mb of bandwidth.
C. Resolve names outside of the direct control of the Cisco VCS that exist on the public Internet.
D. Connect to another Cisco VCS on the same side of the firewall to extend dialing capabilities.
E. Traverse a firewall from a protected network to a public or DMZ network.
Answer: (SHOW ANSWER)

NEW QUESTION: 156
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: 157
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. The customer firewall must be configured with any rule for the IP address of the external Jabber client.
C. An additional interface must be enabled on the Cisco UCM and placed in the same subnet at the Expressway.
D. A SIP trunk has to be set up between the Expressway-C and Cisco UCM.
E. The Expressway server needs a neighbor zone created that points to Cisco UCM.
Answer: (SHOW ANSWER)

NEW QUESTION: 158
Refer to the exhibit.

IOS Config

```
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 8.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ-1_MTP
!
dspfarm profile 1 mtp
 codec pass-through
 rsvp
 maximum sessions software 20
 associate application SCCP
!
interface Serial0/1
 description IP-WAN
 ip address 10.1.4.101 255.255.255.0
 duplex auto
 speed auto
 ip rsvp bandwidth 40
!
```

What media resource should be configured in Cisco Unified Communications Manager?
A. Cisco IOS Media Termination Point
B. Cisco IOS Enhanced Media Termination Point Cisco
C. Cisco Media Termination Point Hardware
D. Cisco Media Termination Point Hardware (WS-SVC-CMM)
Answer: (SHOW ANSWER)

NEW QUESTION: 159

You have been asked to deploy Cisco Extension Mobility Cross Cluster for a distributed call processing environment. During the initial extension mobility login request, how does the visiting cluster determine if the user is a local user or a remote user?

A. by broadcasting a request to all clusters to verify the user type

B. by using Extension Mobility Cross Cluster Session Initiation Protocol (SIP) trunks

C. by verifying the visiting Trivial File Transfer Protocol

D. by verifying against the local database

E. by using a third-party automatic provisioning tool to verify user ID

F. from user IDs that are created by default when the user logs in

Answer: (SHOW ANSWER)

NEW QUESTION: 160

A. 0% Max Loss, 100 ms One-way Latency, 30 ms Jitter, 20% Overprovisioning

B. 1% Max Loss, 160 ms One-way Latency, 60 ms Jitter, 10% Overprovisioning

C. 1% Max Loss, 150 ms One-way Latency, 30 ms Jitter, 20% Overprovisioning

D. 5% Max Loss, 5 s One-way Latency, 30 ms Jitter, 20% Overprovisioning

Answer: (SHOW ANSWER)

Explanation/Reference:

Reference:

http://www.cisco.com/c/en/us/td/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND/QoS-SRND-Book/QoSIntro.html (interactive video)

NEW QUESTION: 161

Which two statements about international multisite dial plans are true? (Choose two.)

A. TEHO reduces WAN utilization.

B. TEHO is legal in all countries.

C. TEHO is legal in most countries, but illegal in others.

D. TEHO utilizes WAN links.

Answer: (SHOW ANSWER)

NEW QUESTION: 162

The administrator at Company X is trying to set up Extension Mobility and has done these steps:

- Set up end users accounts for the users who need to roam
- Set up a device profile for the type of phones users will be allowed to log in

Users have reported to the administrator that they are unable to log in to the phones designated for Extension Mobility. Which two options are the two reasons for this issue? (Choose two.)

A. The Extension Mobility service has not been enabled under the Cisco Unified Serviceability Page.

B. The username must be numeric only and must match the DN.

C. The user device profile is not associated to the correct end user.

D. The user must ensure that their main endpoint is online and registered, otherwise they cannot log in elsewhere.
E. Extension Mobility has not been enabled under Enterprise Parameters.
Answer: (SHOW ANSWER)

NEW QUESTION: 163
How are Cisco IP Phones directly configured to utilize local route groups?
A. with Cisco Unified Communications Manager CSS and partitions
B. with Cisco Unified Communications Manager device pools
C. with Cisco Unified Communications Manager locations
D. with Cisco Unified Communications Manager regions
E. with Cisco Unified Communications Manager AAR
Answer: (SHOW ANSWER)

NEW QUESTION: 164
What happens if location-based CAC is used and there is no bandwidth available when a remote caller is placed on hold?
A. Cisco Unified Communications Manager terminates the call.
B. Cisco Unified Communications Manager sends TOH rather than MOH.
C. Cisco Unified Communications Manager plays default MOH.
D. Cisco Unified Communications Manager attempts to reconnect the call immediately.
Answer: (SHOW ANSWER)

NEW QUESTION: 165
You are deploying a Cisco Unified Communications Manager solution with MGCP gateways at multiple locations. Which firewall and ACL configuration must you perform to allow the MCGP gateways to function correctly?
A. Block access to TCP ports 2427 and 2428.
B. Block TCP port 1720.
C. Allow access to TCP port 1720.
D. Allow access to TCP port 2428.
E. Open access to all TCP and UDP ports.
Answer: (SHOW ANSWER)

NEW QUESTION: 166
An engineer is configuring URI calling within the same cluster. Which two actions must be taken to accomplish this configuration? (Choose two.)
A. Configure SIP trunk.
B. Configure the SIP profile.
C. Activate the URI service in Cisco Unified Serviceability.
D. Configure SIP route patterns.

E. Assign directory URIs to users.

F. Configure the directory URI partition and calling search space.

Answer: (SHOW ANSWER)

Explanation/Reference:

Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BK_CD2F83FA_00_cucm-system-guide-90/CUCM_BK_CD2F83FA_00_system-guide_chapter_0101111.html

NEW QUESTION: 167

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

A. Device Information under Phone Configuration

B. ILS configuration under Advanced Features

C. +E.164 alternate number under Directory Number Settings

D. Route Pattern under Route/Hunt

Answer: C (LEAVE A REPLY)

NEW QUESTION: 168

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 AAR.

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

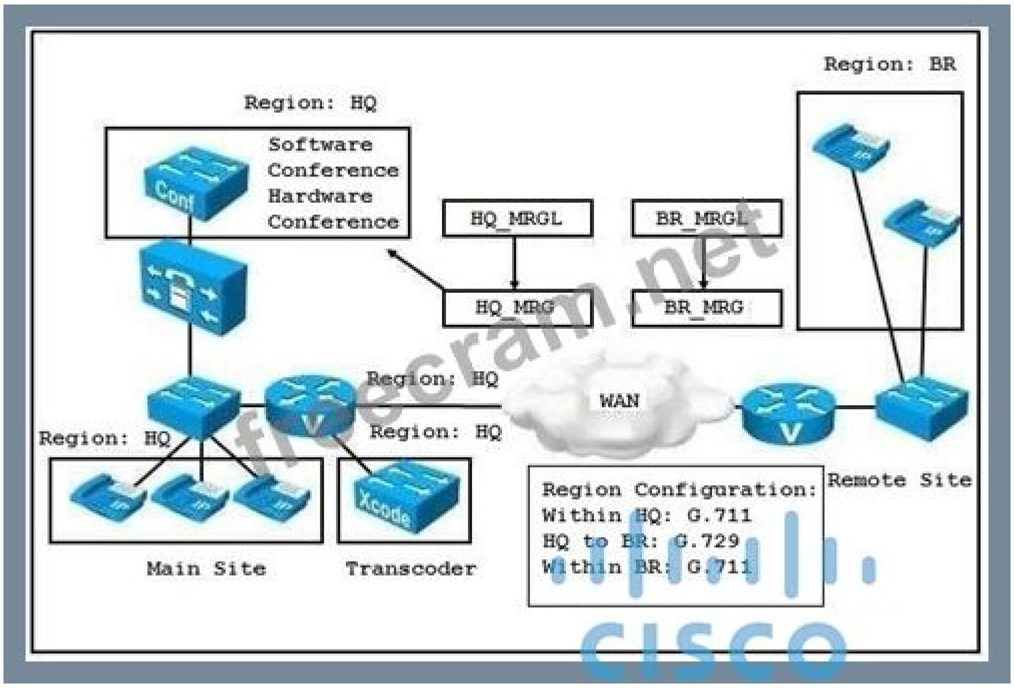
C. Configure Cisco Unified Communications Manager locations.

D. Configure an SIP trunk to the ISR.

Answer: A (LEAVE A REPLY)

NEW QUESTION: 169

Refer to the exhibit.



All HQ phones are configured to use HQ_MRGL and all BR phones are configured to use BR_MRGL. For the HQ phones always to use the hardware conference bridge as a first choice, which configuration should be implemented?

A. Ensure that both the hardware and software conference bridges are listed in the HQ_MRG. Ensure that the instance ID for the hardware conference bridge is 0.

B. Ensure that both the hardware and software conference bridges are listed in the HQ_MRG. The hardware conference bridge must be configured first.

C. Assign the hardware conference bridge to HQ_MRG.
Configure a second HQ_MRG_2 and assign the software conference bridge to it.
Add both the HQ_MRG and HQ_MRG_2 to the HQ_MRGL and list the HQ_MRG first.

D. Assign the hardware conference bridge to HQ_MRG.
Configure a second HQ_MRG_2 and assign the software conference bridge to it.
Configure an additional HQ_MRGL_2.
Add the HQ_MRG to HQ_MRGL. Add HQ_MRG_2 to HQ_MRGL_2.
The HQ_MRGL should be assigned to the HQ phones.
The HQ_MRGL_2 should be assigned to the HQ device pool.

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:

To ensure that the hardware bridge is utilized first with all its resources BEFORE the software bridge is used ... you need to have two separate MRG's and list the hardware MRG 1st in the MRGL ...

NEW QUESTION: 170

A. The Extension Mobility log in fails.

B. The device takes on the default device profile for its type.

C. The device uses the first device profile assigned to the user in Cisco Unified Communications Manager.
D. The user can log in but does not have access to any features, soft key templates, or button templates.
Answer: (SHOW ANSWER)

NEW QUESTION: 171
Refer to the exhibit.

```
controller T1 1/0
framing esf
linecode b8zs
pri-group timeslots 1-6,24

interface Serial 1/0:23
no ip address
encapsulation hdlc
isdn switch-type primary-ni
isdn incoming-voice voice
no cdp enable

interface Vlan120
ip address 10.2.120.1.255.255.255.0
h323-gateway voip bind scraddr 10.2.120.1

voice-port 1/0:23

dial-peer voice 123 pots
incoming called-number
direct inward-dial

dial-peer voice 23000 voip
destination-pattern 2...
session target ipv4:10.1.5.11
dtmf-relay h245-alphanumeric
codec g711ulaw
no vad
```

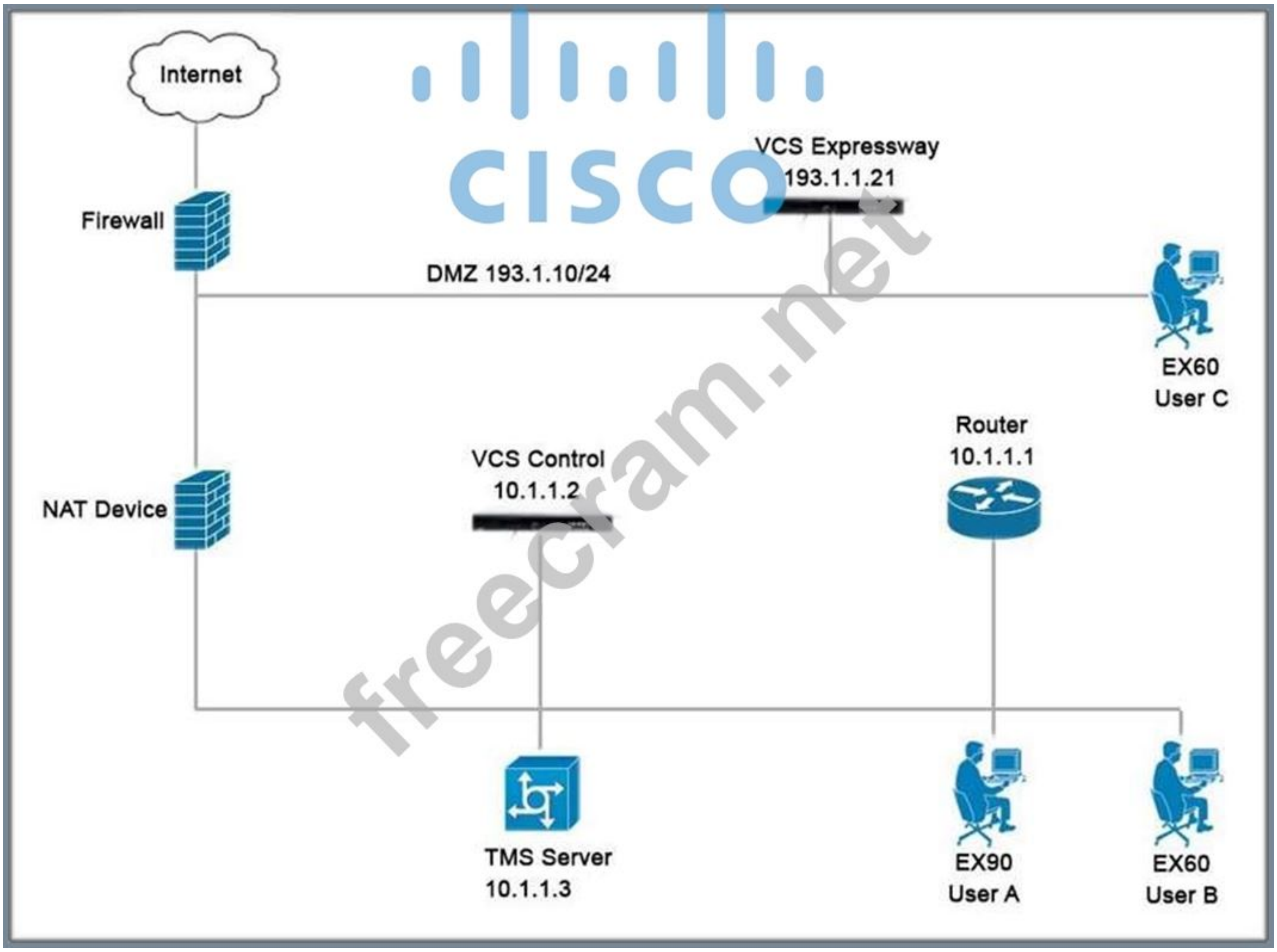

IT shows an H.323 gateway configuration in a Cisco Unified Communications Manager environment. An inbound PSTN call to this H.323 gateway fails to connect to IP phone extension 2001. The PSTN user hears a reorder tone. Debug isdn q931 on the H.323 gateway shows the correct called-party number as
5015552001.
Which two configuration changes can correct this issue? (Choose two.)
A. Add port 1/0:23 to dial-peer voice 123 pots.
B. Ensure that the Significant Digits for inbound calls on the H.323 gateway configuration is 4.

C. Add a voice translation profile to truncate the number from 10 digits to 4 digits. Apply the voice translation profile to the Voice-port. The configuration field "Significant Digits for inbound calls" is left at default (All).
D. Add the command h323-gateway voip id on interface vlan120.
E. Change the destination-pattern on the dial-peer voice 23000 VoIP to 501501? and change the Significant Digits for inbound calls to 4.
Answer: B,E (LEAVE A REPLY)
Explanation/Reference:
Explanation:
Choose the number of significant digits to collect, from 0 to 32. Cisco Unified Communications Manager counts significant digits from the right (last digit) of the number that is called.

NEW QUESTION: 172
When a call is made from a video endpoint to a Cisco TelePresence EX90 that is registered to a Cisco VCS Control, which portion of the destination URI is the is the first match that is attempted?
A. the directory number that is assigned to the Cisco TelePresence EX90
B. the full URI, including the domain portion
C. the destination alias, without the domain portion
D. the E.164 number that is assigned to the Cisco TelePresence EX90
Answer: (SHOW ANSWER)

NEW QUESTION: 173
What are the two tasks that you must perform to configure the Service Advertisement Framework forwarder in Cisco Unified Communications Manager? (Choose two.)
A. enable enterprise parameter for Service Advertisement Framework forwarder
B. create VPN profiles
C. create a new Service Advertisement Framework security profile
D. set feature configuration parameters of Call Control Discovery
E. create VPN groups
F. configure Service Advertisement Framework forwarder information
Answer: (SHOW ANSWER)

NEW QUESTION: 174
Refer to the exhibit.



Which three statements about when user A calls user C using SIP are true? (Choose three.)

A. Deploying a Cisco VCS Expressway behind a NAT mandates the use of the Advanced Networking option key.

B. SIP TCP/TLS ports must be opened from internal to DMZ and vice versa.

C. The NAT device must translate from 10.X.X X to 193.1.1.X and vice versa.

D. Deploying a Cisco VCS Control inside a NAT mandates the use of the Advanced Networking option key.

E. RTP and RTCP ports must be opened at the firewall from internal to DMZ and vice versa

F. Cisco VCS Control and Cisco VCS Expressway support static NAT.

Answer: (SHOW ANSWER)

NEW QUESTION: 175

In a distributed call processing network with locations-based CAC, calls are routed to and from intercluster trunks. Which trunk type is implemented in this network?

A. intercluster trunk with gatekeeper control

B. SIP trunk

C. intercluster trunk without gatekeeper control

D. h225 trunk

Answer: (SHOW ANSWER)

NEW QUESTION: 176

The following exhibit shows configs for H.323 gateway. You have been asked to implement TEHO from a remote branch office with area code 301 to the HQ office with area code 201 using Cisco Unified Communications Manager. The remote office has an MGCP gateway and the HQ office has an H.323 gateway. Once the call arrives at the HQ, it should break out to the PSTN as a seven-digit local call. Which statement about the route pattern is true?

```
dial-peer voice 7 pots
  destination-pattern 9[2-9]... ...
  port 1/0:23
!
dial-peer voice 11 pots
  destination-pattern 91[2-9]..[2-9]......
prefix 1
port 1/0:23
```

A. route pattern should be 91201.[2-9]XXXXXX with Discard Digit as Predot and Prefix 9

B. route pattern should be 91201.[2-9]XXXXXX with Discard Digit as Predot

C. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot and Prefix 9

D. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot

E. route pattern should be 91201.[2-9]XXXXXX

Answer: (SHOW ANSWER)

NEW QUESTION: 177

Which option describes a function of SIP preconditions?

A. SIP preconditions enable end-to-end RSVP for calls through the PSTN.

B. SIP preconditions enable end-to-end RSVP over an SIP trunk.

C. SIP preconditions enable RSVP between Cisco IP Phones.

D. SIP preconditions can be enabled in a gatekeeper.

Answer: (SHOW ANSWER)

NEW QUESTION: 178

Select two commands. One of which can be used to verify Cisco IP phone sip registration, and one of which can be used to verify Cisco IP phone sccp registration on Cisco Unified Communications Manger Express? (Choose two.)

A. show telephony-service ephone-dn

B. show voice register session-server

C. show ephone registered

D. show ccm-manager hosts

E. show sip-ua status registrar

Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/troubleshooting/guide/ts_phreg.html (see the steps)

NEW QUESTION: 179
Which option configures the secondary dial tone option for SRST mode to let the users hear the dial tone for PSTN calls?
A. dial-peer voice 1 pots
secondary dialtone 0
B. ccm-manager secondary dialtone 0
C. voice service voip
secondary dialtone 0
D. call-manager-fallback
secondary dialtone 0
Answer: (SHOW ANSWER)

NEW QUESTION: 180
A. video gateway
B. MGCP gateway
C. H.323 gatekeeper
D. MCU
Answer: C (LEAVE A REPLY)
Explanation/Reference:
Explanation:
As with H.323 MCUs, H.320 gateways are provisioned in Cisco Unified CallManager as H.323 gateways, and then route patterns are configured to extend calls to these devices.

NEW QUESTION: 181
When configuring Cisco Unified Mobility, which parameter controls the ability for a call to be extended to a configured remote destination number?
A. Calling Search Space under Phone Configuration
B. User Local under Remote Destination Profile Information
C. Calling Party Transformation Calling Search Space under Remote Destination Profile Information
D. Rerouting Calling Search Space under Remote Destination Profile information
E. Rerouting Calling Search Space under Remote Destination information
Answer: (SHOW ANSWER)

NEW QUESTION: 182
Refer to the exhibit.

```
Router(config)#class-map VIDEO
Router(config-cmap)#match dscp ?
  <0-63>    Differentiated services codepoint value
  af11      Match packets with AF11 dscp (001010)
  af12      Match packets with AF12 dscp (001100)
  af13      Match packets with AF13 dscp (001110)
  af21      Match packets with AF21 dscp (010010)
  af22      Match packets with AF22 dscp (010100)
  af23      Match packets with AF23 dscp (010110)
  af31      Match packets with AF31 dscp (011010)
  af32      Match packets with AF32 dscp (011100)
  af33      Match packets with AF33 dscp (011110)
  af41      Match packets with AF41 dscp (100010)
  af42      Match packets with AF42 dscp (100100)
  af43      Match packets with AF43 dscp (100110)
  cs1       Match packets with CS1(precedence 1) dscp (001000)
  cs2       Match packets with CS2(precedence 2) dscp (010000)
  cs3       Match packets with CS3(precedence 3) dscp (011000)
  cs4       Match packets with CS4(precedence 4) dscp (100000)
  cs5       Match packets with CS5(precedence 5) dscp (101000)
  cs6       Match packets with CS6(precedence 6) dscp (110000)
  cs7       Match packets with CS7(precedence 7) dscp (111000)
  default   Match packets with default dscp (000000)
  ef        Match packets with EF dscp (101110)

Router(config-cmap)#match dscp
```

The "DSCP for Video Calls" Cisco CallManager service parameter is set to 34. What is the correct DSCP value to use when configuring a class map in a Cisco IOS router?
A. cs4
B. af23
C. af41
D. ef
Answer: C (LEAVE A REPLY)

NEW QUESTION: 183
How is a SIP trunk in Cisco Unified Communications Manager configured for SIP precondition?
A. SIP precondition is configured by configuring a new SIP profile and selecting E2E for RSVP over SIP.
The new SIP profile must then be assigned to the SIP trunk.
B. The configuration is done by selecting a SIP precondition trunk for trunk type.
C. SIP precondition is configured by selecting E2E for RSVP over SIP on the default SIP profile assigned to the SIP trunk.
D. The configuration is automatically selected when RSVP is enabled for the location assigned to the trunk.
Answer: (SHOW ANSWER)

NEW QUESTION: 184

Which statement is correct about AAR?

A. The end users sees, "Network Congestion Rerouting?" but AAR is otherwise transparent to the end user and works without user intervention.

B. AAR will display "not enough bandwidth" on the IP phone while it reroutes the call.

C. AAR allows calls to be rerouted because of insufficient Cisco Unified Border Element controlled bandwidth to an ITSP.

D. AAR allows calls to be rerouted due to insufficient gatekeeper controlled IP WAN bandwidth.

Answer: A ([LEAVE A REPLY])

Explanation/Reference:

Explanation:

Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth. With automated alternate routing, the caller does not need to hang up and redial the called party.

NEW QUESTION: 185

An engineer is configuring Global Dial Plan Replication and wants to prevent the local cluster from routing the Vice President number 5555555555 to the remote cluster. Which action on the remote cluster accomplishes this task?

A. Create a block route pattern.

B. Create a block learned pattern.

C. Create a block transformation pattern.

D. Create a block translation pattern.

Answer: ([SHOW ANSWER])

Explanation/Reference:

Explanation:

GDPR using ILS allows for blocking learned patterns to be configured in the CUCM.

Block a Learned Pattern

If you want to prevent a local Cisco Unified Communications Manager cluster from routing calls to a learned alternate number or learned alternate number pattern, you can configure a local blocking rule on that cluster. Before routing a call to a learned number or learned pattern, ILS checks to see if a local blocking rule matches the dial string. If the blocking rule matches, Cisco Unified Communications Manager does not route the call.

Some additional characteristics of blocking rules:

- Blocking rules are applied only on the local cluster on which you configure them—ILS does not advertise blocking rules.
- Blocking rules are applied only to learned alternate numbers and learned patterns—Cisco Unified Communications Manager does not apply blocking rules to locally configured numbers or route patterns.

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/
CUCM_BK_F3AC1C0F_00_cucm-features-services-
guide-100/CUCM_BK_F3AC1C0F_00_cucm- features-services-
guide-100_chapter_011101.html#CUCM_TK_BDDD9061_00

NEW QUESTION: 186

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

A. on-net calling patterns

B. number of route patterns

C. caller ID

D. E911 calling

Answer: C,D (LEAVE A REPLY)

NEW QUESTION: 187

The Cisco Unified Communications system of a company has five types of devices:

* Cisco Jabber Desktop
* CP-7965
* DX-650
* EX-60
* MX-200

Which two types of devices are affected when an engineer changes the DSCP for Video Calls service parameter? (Choose two.)

A. MX-200

B. DX-650

C. CP-7965

D. EX-60

E. Cisco Jabber Desktop

Answer: (SHOW ANSWER)

NEW QUESTION: 188

Which codec is recommended by Cisco for multisite music on hold deployment that spans through the WAN links?

A. G.728

B. G.729

C. G.711

D. G.725

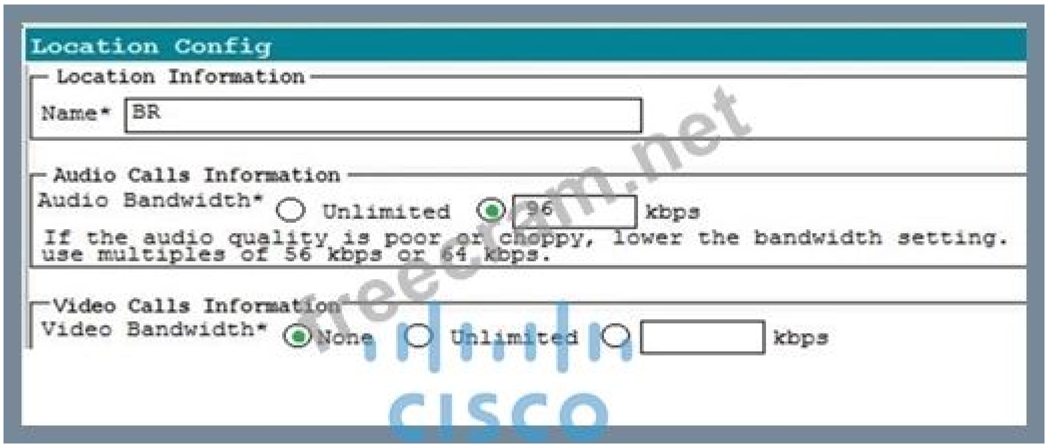
E. G.722

F. G.723

Answer: (SHOW ANSWER)

NEW QUESTION: 189

Refer to the exhibit.



Assuming the regions configuration to BR only permits G.729 codec, how many calls are allowed for the BR location?

A. Total of four calls; two incoming and two outgoing.

B. Total of two calls in either direction.

C. Total of four calls to the BR location. Outgoing calls are not impacted by the location configuration.

D. Total of four calls in either direction.

E. Two outgoing calls. Incoming calls are unlimited.

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:

In performing location bandwidth calculations for purposes of call admission control, Cisco Unified Communications Manager assumes that each G.729 call stream consumes 24 kb/s amount of bandwidth.

NEW QUESTION: 190

A. The secondary server keeps sending keepalive message to the primary server and when it succeeds, it unregisters the phones to force them to register to the primary.

B. The phones never re-register with the primary server until the active (secondary) one goes down.

C. When the primary server goes online, it sends out an "ALIVE" message via broadcast so that the phones re-register.

D. The phone sends keepalive messages to the primary server frequently and when it succeeds, the phone re-registers with it.

Answer: (SHOW ANSWER)

NEW QUESTION: 191

Where do you go to configure restrictions on audio bandwidth?

A. serviceability

B. region

C. partition

D. enterprise parameters

E. location
Answer: (SHOW ANSWER)

NEW QUESTION: 192
When configuring Cisco Unified Survivable Remote Site Telephony, which CLI command enables this feature on the router?
A. call-manager-fallback
B. ccm-manager sccp local
C. ccm-manager redundant-host
D. ccm-manager switchback
Answer: (SHOW ANSWER)

NEW QUESTION: 193
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.
Static routes are not supported.
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 only on SAF routers. Non-SAF routers act as an IP cloud.
D. SAF requires that the EIGRP be configured on all routers, including non-SAF routers.
Answer: (SHOW ANSWER)

NEW QUESTION: 194
Which three options are overlapping parameters for roaming when a device is configured for Device Mobility? (Choose three.)
A. device pool
B. location
C. network locale
D. codec
E. MRGL
F. extension
Answer: (SHOW ANSWER)
Explanation/Reference:
Explanation:
The overlapping parameters for roaming-sensitive settings are Media Resource Group List, Location, and Network Locale. The overlapping parameters for the Device Mobility-related settings are Calling Search Space (called Device Mobility Calling Search Space at the device pool), AAR Group, and AAR Calling Search Space. Overlapping parameters configured at the

phone have higher priority than settings at the home device pool and lower priority than settings at the roaming device pool.
Reference: https://supportforums.cisco.com/document/77096/device-mobility

NEW QUESTION: 195
Which three commands can be used to verify SRST fallback mode? (Choose three.)
A. show telephony-service voice-port
B. show telephony-service tftp-bindings
C. show telephony-service ephone
D. show telephony-service ephone-dn
E. show telephony-service all
Answer: C,D,E (LEAVE A REPLY)

NEW QUESTION: 196
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. create connection
B. restart in progress
C. modify endpoint
D. audit endpoint
E. delete notification
F. end connections
Answer: A,B,D (LEAVE A REPLY)

NEW QUESTION: 197
Which option best describes a service that assembles a network model from configured locations and link data in one or more clusters?
A. Shadow
B. Weight
C. LBM Hub
D. LBM

Answer: D (LEAVE A REPLY)

NEW QUESTION: 198

Which statement about TEHO is true?

A. The dial plan is simplified with local route groups.

B. Local route groups add complexity to the dial plan.

C. Toll charges can be reduced when TEHO is implemented with CAC.

D. Toll charges can be reduced when TEHO is implemented with MGCP fallback.

Answer: (SHOW ANSWER)

NEW QUESTION: 199

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

A. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Enable SAF check box should be selected.

B. The SIP trunk is added by selecting Call Control Discovery Trunk and then selecting SIP as the protocol to be used.

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)

NEW QUESTION: 200

A. dialplan-pattern

B. number-e.164

C. ephone-transnumber

D. number

E. ephone-dn

Answer: (SHOW ANSWER)

NEW QUESTION: 201

Which statement about H.323 Gatekeeper Call Admission Control is true?

A. Gatekeeper Call Admission Control applies to centralized Cisco Unified Communications deployments that use locations based Call Admission Control.

B. Gatekeeper Call Admission Control applies to distributed Cisco Unified Communications deployments.

C. Gatekeeper Call Admission Control applies only to distributed Cisco Unified Communications Express deployments.

D. Gatekeeper Call Admission Control setting is configured in Cisco Unified Communications Manager.

Answer: (SHOW ANSWER)

Explanation/Reference:
Explanation:
In distributed call processing deployments on a simple hub-and-spoke topology, you can implement call admission control with a Cisco IOS gatekeeper. In this design, the call processing agent (which could be a Unified CM cluster, Cisco Unified Communications Manager Express (Unified CME), or an H.323 gateway) registers with the Cisco IOS gatekeeper and queries it each time the agent wants to place an IP WAN call.

NEW QUESTION: 202
Which code snippet is required for SAF to be initialized?
A. External-Client
B. topology base
C. Service Family
D. router eigrp
Answer: (SHOW ANSWER)

NEW QUESTION: 203
Which device is used to connect to the H.323 gatekeeper?
A. H.323 trunk
B. SIP trunk
C. H.323 gateway
D. MGCP gateway
Answer: (SHOW ANSWER)

NEW QUESTION: 204
What is the purpose of configuring a hardware-based MTP when deploying Cisco Unified Communications Manager?
A. when you need the ability to grow support by using DSPs
B. to allow for supplementary services such as hold, transfer, and conferencing
C. when you need support for up to 24 MTP sessions on the same server and 48 on a separate server
D. when you want to only use Cisco Unified Communications Manager resources
Answer: (SHOW ANSWER)

NEW QUESTION: 205
You have deployed a centralized call processing solution with a multicast MOH server at your central site on a different VLAN from your Cisco Unified Communications Manager servers and IP phones. When central site users place calls on hold, dead air or silence is heard. Which two actions will resolve this issue? (Choose two.)
A. Enable multicast routing on all the routers at the central site.

B. Increase the TTL in the configuration of the MOH server to 2 so that packets can cross the VLAN boundary.
C. Decrease the TTL configuration in the Cisco Unified Communications Manager server to 0 so that the multicast packets only go to the VLAN that contains the Cisco Unified Communications Manager server and IP phones.
D. Configure ip-sparse mode on the router interfaces and increase the TTL on the routers to 2.
E. Keep the TTL at 1 for the MOH server and increase the TTL for IP multicast routing to 2 on router interfaces.
F. Enable multicast routing on only those router interfaces that connect the voice and MOH VLANs.
Answer: (SHOW ANSWER)

NEW QUESTION: 206
Refer to the exhibit.

```
device 1:
interface 1
10.10.10.1
telephony-service
ip source-address 10.10.10.1 secondary 10.10.10.2

device 2:
interface 1
10.10.10.2
telephony-service
ip source-address 10.10.10.1 secondary 10.10.10.2
```

Which option describes the effect of this configuration?
A. It configures failover.
B. It implements HSRP.
C. It implements Cisco Unified CME redundancy.
D. It configures a standby Cisco Unified CME.
E. It creates dial peers.
F. It implements Cisco IOS redundancy.
Answer: (SHOW ANSWER)

NEW QUESTION: 207
Company X has three locations connected via a low bandwidth WAN. Which two configurations are required in the Cisco Unified Communications Manager regions to provide the most suitable use of bandwidth while preserving the call quality? (Choose two.)
A. g729 codec for intraregion calling
B. g729 codec for all calling
C. g729 codec for interregion calling
D. g722/g711 codec for interregion calling
E. g722/g711 for intraregion calling
F. g722/g711 codec for all calling

Answer: (SHOW ANSWER)

NEW QUESTION: 208
Which device must be responsible for properly marking network traffic when a Cisco IP phone is connected to a Cisco switch?
A. branch router
B. core layer switch
C. Cisco IP phone
D. access layer switch
Answer: (SHOW ANSWER)

NEW QUESTION: 209
Company A has deployed a VCS Control and is attempting to register a third-party endpoint. The engineer has confirmed that no traffic is being blocked for the endpoint and it is receiving a valid IP address. Which option could be the cause of this registration failure?
A. Third-party endpoints are not compatible with VCS Control, only with VCS Expressway.
B. The VCS Control must be deployed together with VCS Expressway before endpoints can register to either one.
C. Cisco Unified Communications Manager is required in addition to the VCS Control.
D. An incorrect SIP domain is configured on the VCS Control for the endpoint.
Answer: (SHOW ANSWER)

NEW QUESTION: 210
A. Verify that the H.323 redundant connection is active in Cisco Unified Communications Manager.
B. Verify that HSRP is active on the Cisco Unified Communications Manager subscriber servers.
C. Verify that media resources fail over to a secondary subscriber server when the publisher fails.
D. Verify that Cisco Unified IP phones running SCCP go into SRST mode when the WAN connection is disconnected.
E. Verify that SCCP fallback is configured in Cisco Unified Communications Manager.
F. Verify that all phones are registered to a second subscriber server.
Answer: (SHOW ANSWER)

NEW QUESTION: 211
How long is the default keepalive period for SRST in Cisco IOS?
A. 60 sec
B. 30 sec
C. 45 sec
D. 120 sec
Answer: (SHOW ANSWER)

NEW QUESTION: 212
After forgetting to log out of his IP phone in the main office, an Extension Mobility user is unable to log in to a different IP phone at a remote office. Which option is a possible reason for the problem?
A. The device pool is misconfigured.
B. The user's Extension Mobility profile is misconfigured.
C. The user can log in to only one device at a time.
D. The phone at the remote location is a different model than the phone in the user's main office.
Answer: C (LEAVE A REPLY)

NEW QUESTION: 213
Which three statements are true in order for inbound PSTN calls to work in an H.323 gateway configured with Cisco Unified Communications Manager? (Choose three.)
A. A VoIP dial peer pointing to Cisco Unified Communications Manager should be configured.
B. The command h323-gateway voip tech-prefix should be configured on the H.323 interface.
C. The command h323-gateway voip id should be configured under the H.323 interface.
D. A pots dial peer with direct-inward-dial and incoming-called number should be configured.
E. The H.323 gateway should be registered with Cisco Unified Communications Manager.
F. The command h323-gateway voip bind srcaddr should be configured on the H.323 interface.
Answer: (SHOW ANSWER)

NEW QUESTION: 214
Which two statements about Cisco Unified Communications Manager Extension Mobility are true?
(Choose two.)
A. Devices can be configured to allow more than one user to be logged in at the same time.
B. If no one is logged in, the phone uses either an auto generated profile or a user defined profile.
C. After an autogenerated device profile is created, you can associate it with one or more users.
D. A device profile has most of the same attributes as a physical device.
E. An autogenerated device profiles can be loaded on a device at the same time as a user profile.
Answer: (SHOW ANSWER)
Explanation/Reference:

Reference: http://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications- manager-callmanager/18772-extension-mobility.html

NEW QUESTION: 215

Which two types of trunks can support Cisco Unified Communications Manager? (Choose two.)

A. switch port trunks

B. PIMG trunks

C. SIP trunks

D. H.225 trunks

E. CO trunks

F. POTS trunks

Answer: C,D (LEAVE A REPLY)

Explanation/Reference:

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_5_1/ccmsys/accm-851-cm/a08trnk.html#wp1098521

NEW QUESTION: 216

When device mobility mode is enabled or disabled for a cluster, to which does the cluster setting apply?

A. mobile phones in the cluster that are in default mode

B. all phones in the cluster that subscribed to device mobility

C. mobile phones in the cluster that support device mobility

D. all phones in the cluster that support device mobility

Answer: (SHOW ANSWER)

NEW QUESTION: 217

Refer to the exhibit.

```
Service-policy output: VOICE-VIDEO

  queue stats for all priority classes:

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

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

        Class-map: VIDEO (match-all)
        0 packets, 0 bytes
        5 minute offered rate 0 bps, drop rate 0 bps
        Match: dscp af32 (28)
        Queuing
        queue limit 64 packets
        queue depth/total drops/no-buffer drops)0/0/0
        (pkts output/bytes output) 10/560
        bandwidth 25% (384 kbps)

        Class map: TELEPRESENCE (match-all)
        0 packets, 0 bytes
        5 minute offered rate 0 bps, drop rate 0 bps
        Match: dscp af32 (28)
        Queuing
        queue limit 64 packets
        queue depth/total drops/no-buffer drops)0/0/0
        (pkts output/bytes output) 10/560
        bandwidth 25% (384 kbps)

        Class map: class-default (match-any)
        10 packets, 560 bytes
        5 minute offered rate 0 bps, drop rate 0 bps
        Match: any
        Queuing
        queue limit 64 packets
        queue depth/total drops/no-buffer drops/flowdrops) 0/0/0/0
        (pkts output/bytes output) 10/560
        bandwidth 25% (384 kbps)
        Fair-queue: per-flow queue limit 16
```


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

A. 34 (100010)

B. 41 (101001)

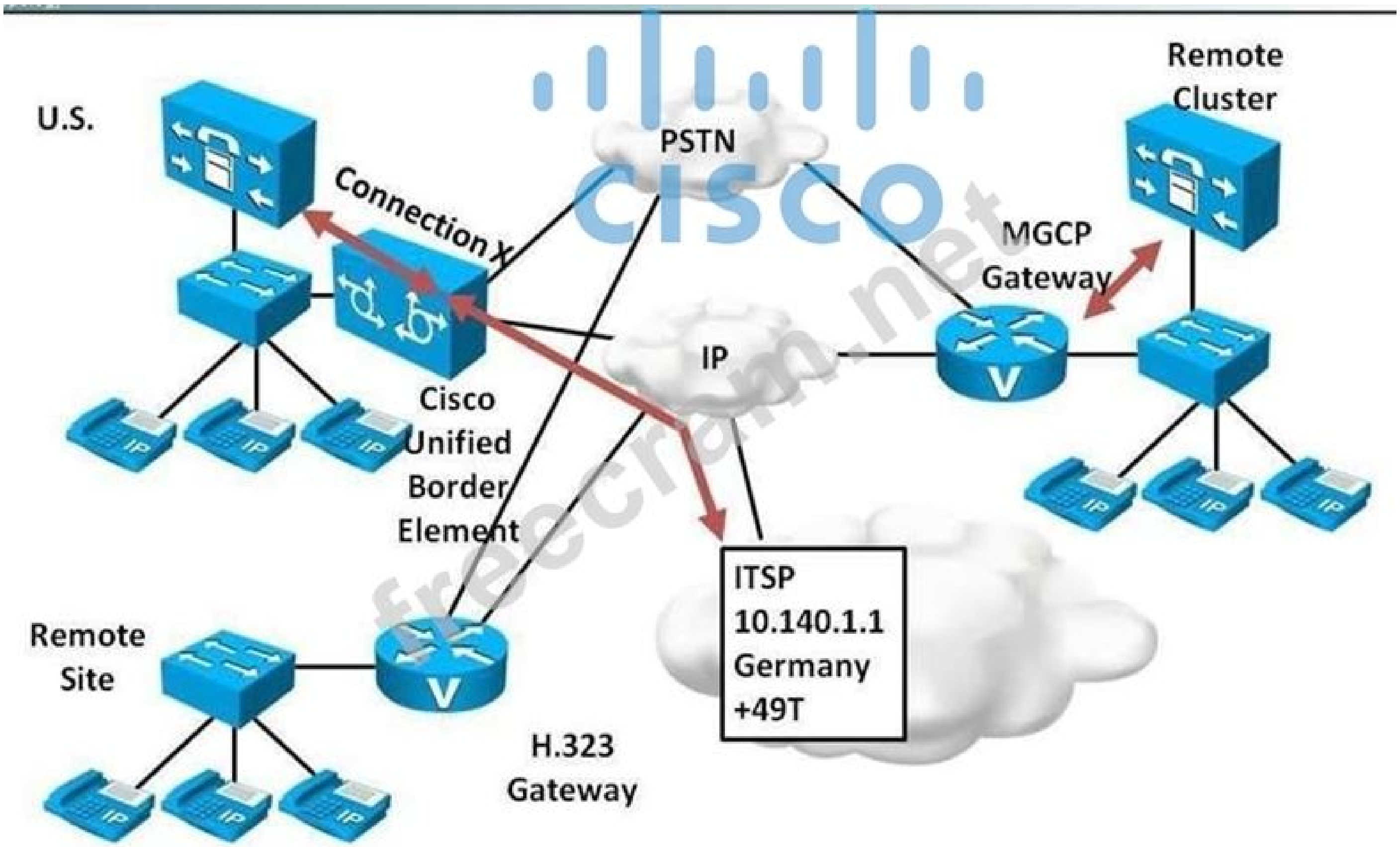
C. 28 (011100)

D. 46 (101110)

Answer: (SHOW ANSWER)

NEW QUESTION: 218

Refer to the following exhibits.


U.S.
Connection X
PSTN
Remote
Cluster
MGCP
Gateway
IP
Cisco
Unified
Border
Element
ITSP
10.140.1.1
Germany
+49T
Remote
Site
H.323
Gateway

CUBE Config

```
!
hostname HQ_gateway
!
card type e1 0 0
enable password cisco123
!
no aaa new-model
network-clock-participate wic 0
!
ip source-route
ip cef
!
isdn switch-type primary-net5
!
voice-card 0
!
voice service voip
 allow-connections sip to sip
!
controller E1 0/0/0
 pri-group timeslots 1-12,16 service mgcp
!
interface Loopback0
 ip address 10.1.111.1 255.255.255.0
!
interface GigabitEthernet0/0
 no ip address
 ip pim sparse-dense-mode
 duplex auto
 speed auto
 media-type rj45
!
interface GigabitEthernet0/0.5
 encapsulation dot1Q 5
 ip address 10.1.5.1 255.255.255.0
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.10
 encapsulation dot1Q 10
 ip address 10.1.10.1 255.255.255.0
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/0.110
 encapsulation dot1Q 110
 ip address 10.1.110.1 255.255.255.0
 ip helper-address 10.1.5.10
 ip pim sparse-dense-mode
!
interface GigabitEthernet0/1
 ip address 10.140.1.2 255.255.255.0
 duplex auto
 speed auto
 media-type rj45
!
interface Serial0/0/0:15
 no ip address
 encapsulation hdlc
 isdn switch-type primary-net5
 isdn incoming-voice voice
 no cdp enable
```

CUBE Config

```
!
interface Serial0/1/0
 no ip address
 ip pim sparse-dense-mode
 encapsulation frame-relay IETF
!
interface Serial0/1/0.101 point-to-point
 ip address 10.12.1.1 255.255.255.0
 ip pim sparse-dense-mode
 snmp trap link-status
 frame-relay interface-dlci 101
!
interface Serial0/1/0.102 point-to-point
 ip address 10.13.1.1 255.255.255.0
 snmp trap link-status
 frame-relay interface-dlci 102
!
router eigrp 10
 network 10.0.0.0
!
ip forward-protocol nd
!
voice-port 0/0/0:15
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server 10.1.5.10
!
mgcp
mgcp call-agent 10.1.5.10 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 ecm
mgcp rtp payload-type g726r16 static
mgcp behavior g729-variants static-pt
!
mgcp profile default
!
dial-peer voice 1111 voip
 session protocol sipv2
 incoming called-number .
!
dial-peer voice 222 voip
 session protocol sipv2
 destination-pattern +49T
 session target ipv4:10.140.1.1
!
```

```
!
gateway
!
gatekeeper
 shutdown
!
line con 0
line aux 0
line vty 0 4
!
end
```

Users in the U.S dial Germany by calling 9011 49 followed by the remaining digits. What would be the most suitable configuration for Connection X?

A. Configure a SIP trunk to 10.140.1.1 and a SIP route pattern +49T in Cisco Unified Communications Manager.

B. Configure a SIP trunk to the Cisco Unified Border Element and route pattern +49T in Cisco Unified Communications Manager.

C. Configure a SIP trunk to the Cisco Unified Border Element. Configure a translation pattern for 9011.49T using DDI Predot prefix + and CSS to point to a route pattern partition \+! which uses the SIP trunk.

D. Configure a SIP trunk to the ITSP. Configure a translation pattern for 9011.49T using DDI predot prefix
+ and CSS to point to a route pattern partition \+! which uses the SIP trunk.

Answer: (SHOW ANSWER)

NEW QUESTION: 219

Refer to the exhibit.

```
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 8.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ-1_MTP
!
dspfarm profile 1 mtp
 codec pass-through
 rsvp
 maximum sessions software 20
 associate application SCCP
!
interface Serial10/1
 description IP-WAN
 ip address 10.1.4.101 255.255.255.0
 duplex auto
 speed auto
 ip rsvp bandwidth 64
!
```

To permit three G.729 calls, what should the bandwidth value be for the ip rsvp bandwidth command?

A. 32
B. 48
C. 64
D. 128
E. 88

Answer: (SHOW ANSWER)

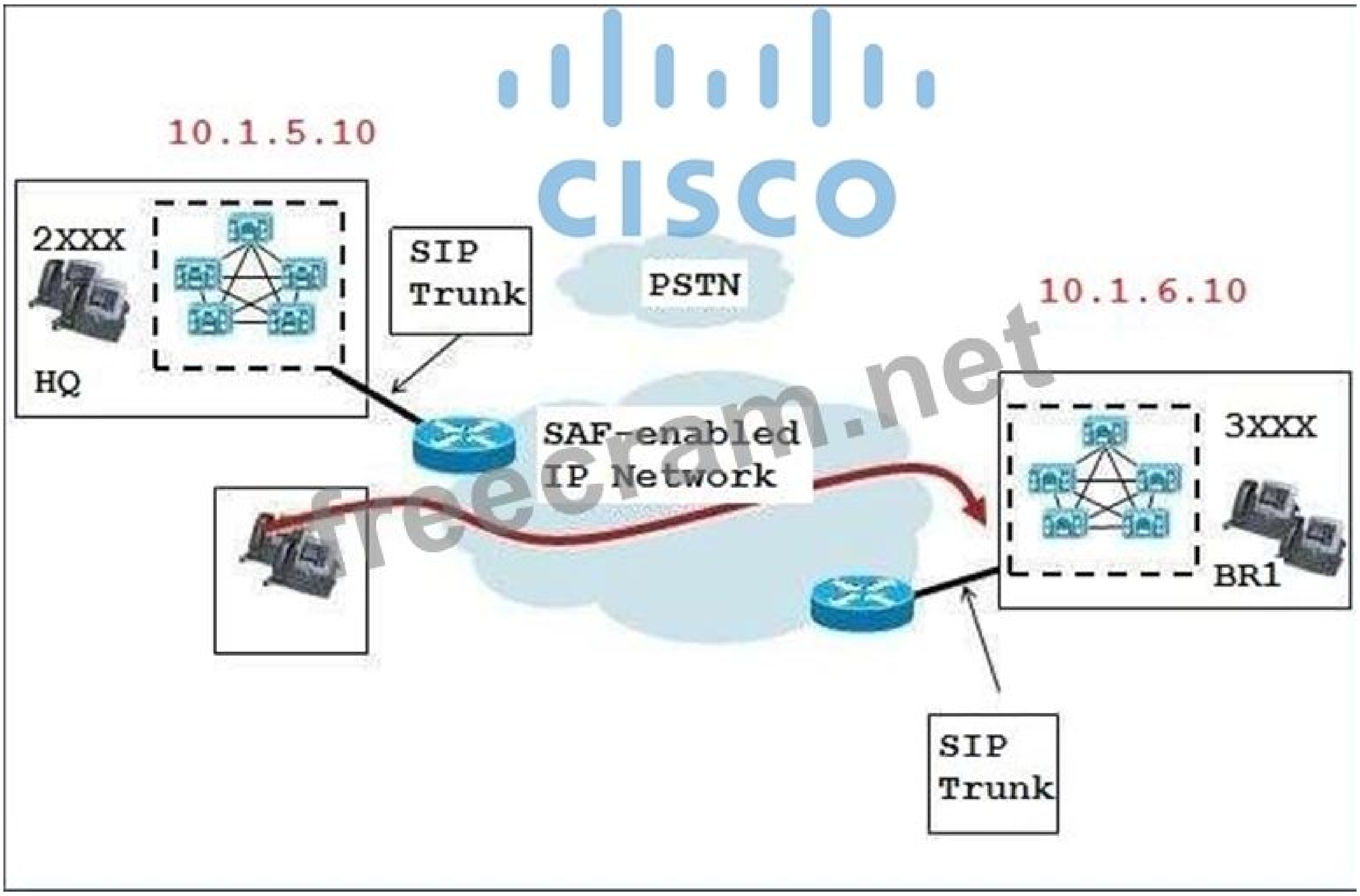
NEW QUESTION: 220

A. Configure Cisco Unified Communications Manager regions.
B. Configure a gatekeeper with an SIP trunk.
C. Configure Cisco Unified Communications Manager locations.
D. Configure a gatekeeper and a gatekeeper-controlled trunk in Cisco Unified Communications Manager with bandwidth control.

Answer: (SHOW ANSWER)

NEW QUESTION: 221

Refer to the exhibit.



What should the destination IP address be configured as on the HQ and BR1 SIP trunks?

A. The HQ SIP trunk destination IP address should be 10.1.6.10. The BR1 SIP trunk destination IP address should be 10.1.5.10.

B. The destination IP address is not configurable. Each cluster will learn about the remote trunk IP address through SAF learned routes.

C. The destination IP address will be learned automatically and configured on the SIP trunks after the Cisco Unified Communications Managers discover themselves.

D. The HQ SIP trunk destination IP address should be the HQ SAF Forwarder IP address. The BR1 SIP trunk destination IP address should be the BR1 SAF Forwarder IP address.

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:

The gatekeeper changes the IP address of this remote device dynamically to reflect the IP address of the remote device.

NEW QUESTION: 222

Which three configuration settings are included in a default region configuration? (Choose three.)

A. Immersive Bandwidth

B. Video Call Bandwidth

C. Audio Codec

D. Link Loss Type

E. Real Time Protocol

F. Location Description

Answer: B,C,D (LEAVE A REPLY)

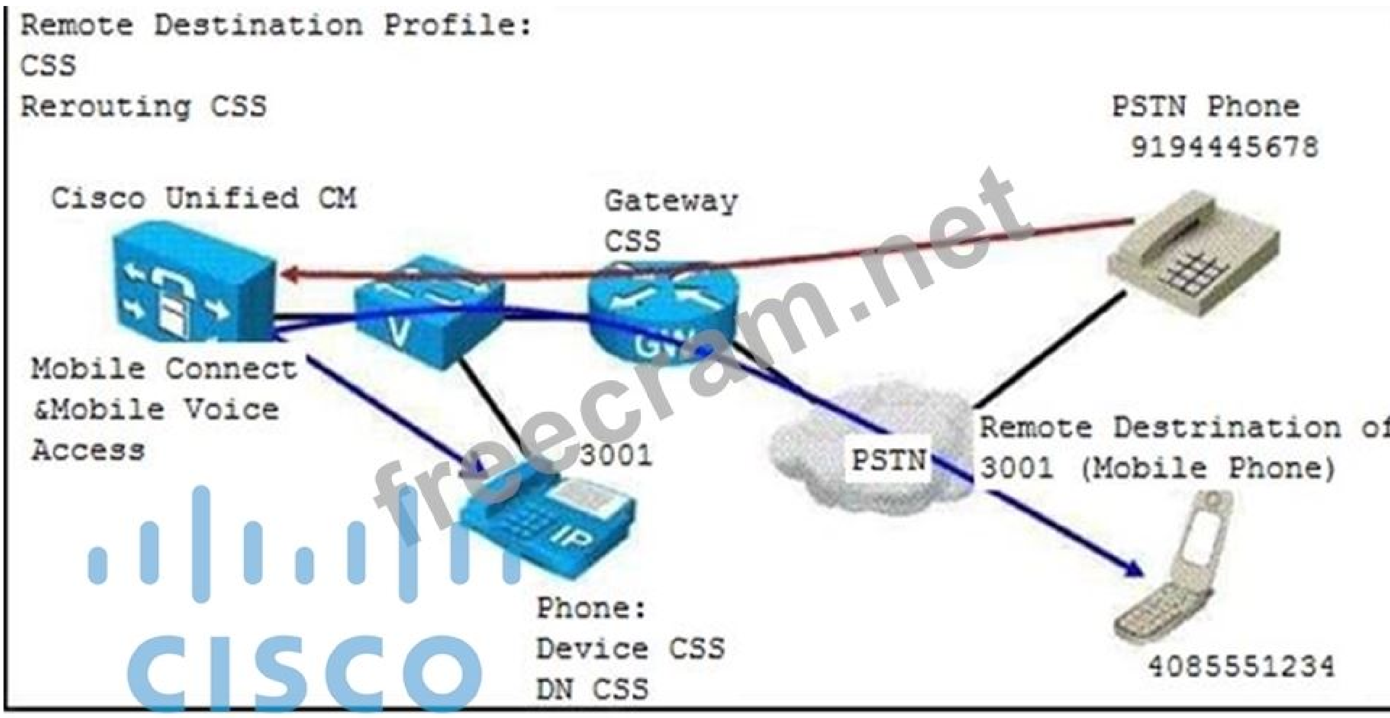
Explanation/Reference:

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_0_1/ccmcfg/bccm-801-cm/b02regio.html#wp1077135

NEW QUESTION: 223

Refer to the exhibit.



With the Mobile Connect feature configured, when the PSTN phone calls the Enterprise user at extension

3001, the Enterprise user's mobile phone does not ring. Which CSS is responsible for ensuring that the correct partitions are accessed when calls are sent to the Enterprise user's mobile phone?

A. the gateway CSS

B. the Phone Device CSS

C. the Remote Destination Profile CSS

D. the Remote Destination Profile Rerouting CSS

E. the Phone Line (DN)CSS

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:

Ensure that the gateway that is configured for routing mobile calls is assigned to the partition that belongs to the Rerouting Calling Search Space. Cisco Unified Communications Manager determines how to route calls based on the remote destination number and the Rerouting Calling Search Space.

NEW QUESTION: 224

You recently implemented call redundancy at a new remote site, and users report that calls are dropped when the remote site supposedly is in SRST.

Which two actions must you take to troubleshoot the problem? (Choose two.)

A. Restart Cisco Unified Communications Manager services to confirm that the server is working correctly.

B. Check the Region settings in Cisco Unified Communications Manager.

C. Confirm that a calling search space is assigned to the voice gateway in Cisco Unified Communications Manager.

D. Confirm that the site devices are associated with a Cisco Unified Communications Manager group and that four Cisco Unified Communications Manager servers are available.

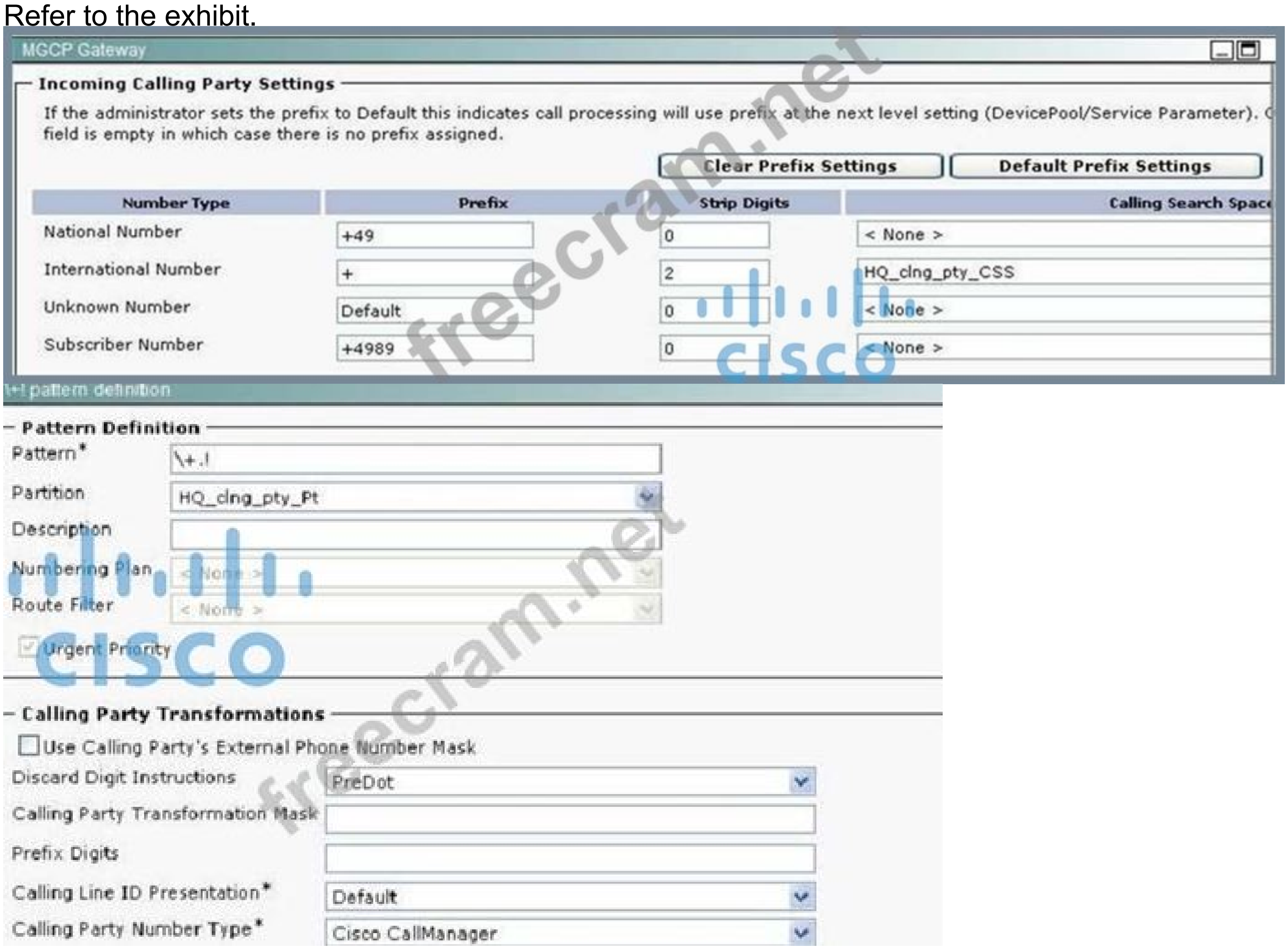
E. Confirm that the site has an SRST reference that is correctly associated with the Cisco Unified Communications Manager group.

F. Confirm that SRST is configured on the voice gateway.

Answer: (SHOW ANSWER)

NEW QUESTION: 225

Refer to the exhibit.



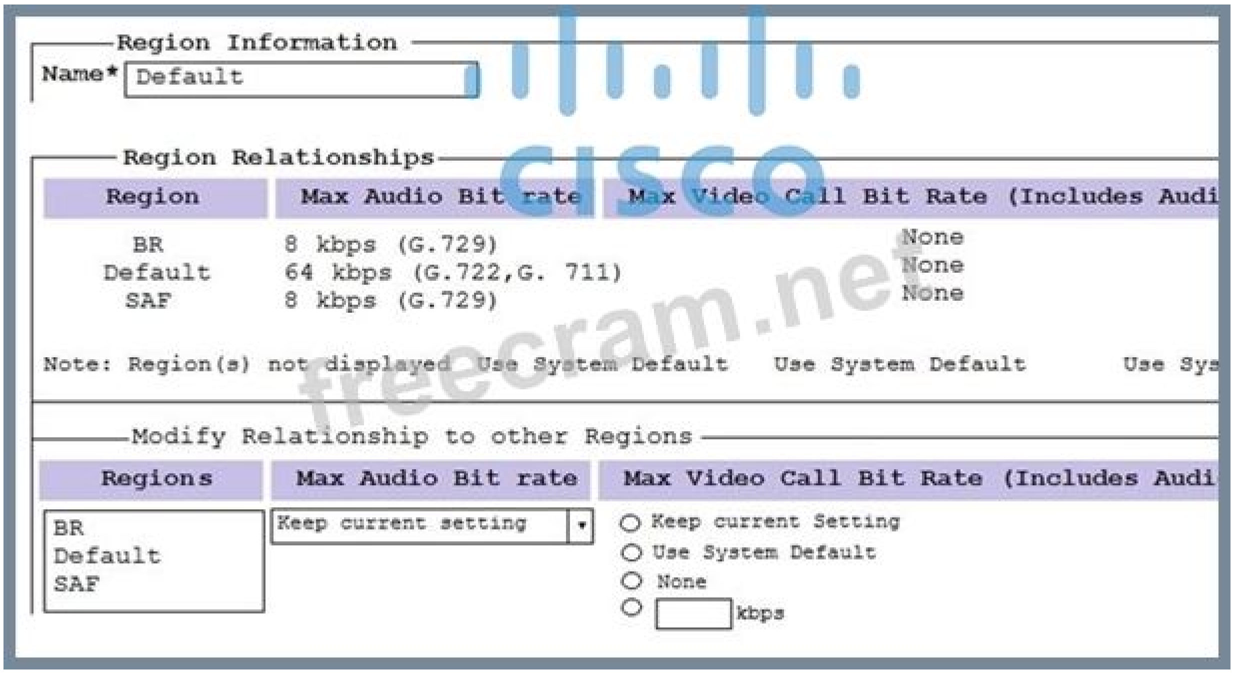
A PSTN call arrived at the MGCP gateway. The calling number was received as 14087071222 with number set to type international. The HQ_clng__pty_CSS contains the HQ_clng_pty__Pt

partition. Which caller ID is displayed on the IP phone?
A. 087071222
B. +087071222
C. 14087071222
D. 4087071222
E. 14087071222
Answer: (SHOW ANSWER)

NEW QUESTION: 226
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 saf registration
C. show voice saf dndb all
D. show eigrp service-family ipv4 clients
E. show eigrp address-family ipv4 clients
Answer: (SHOW ANSWER)

NEW QUESTION: 227
What impact do roaming-sensitive settings and Device Mobility settings have on call routing?
A. Device Mobility settings have no impact on call routing, but roaming-sensitive settings modify the AAR group, AAR CSS, and device CSS.
B. Device Mobility settings modify the device CSS and the roaming-sensitive settings modify the AAR group and AAR CSS.
C. Roaming-sensitive settings are settings that do not have an impact on call routing. Device Mobility settings, on the other hand, may have an impact on call routing because they modify the device CSS, AAR group, and AAR CSS.
D. Device Mobility settings modify the AAR group and the AAR CSS, the roaming-sensitive settings modify the device CSS.
Answer: C (LEAVE A REPLY)

NEW QUESTION: 228
Refer to the exhibit.



Which statement about the configuration between the Default and BR regions is true?

A. Only 64 kbps will be used between the two regions because the link is "lossy".

B. Calls between the two regions can use only the G.729 codec.

C. Both codecs can be used depending on the loss statistics of the link. When lossy conditions are high, the G.711 codec will be used.

D. Calls between the two regions can use either 64 kbps or 8 kbps.

Answer: (SHOW ANSWER)

NEW QUESTION: 229

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

A. URI

B. route pattern

C. route string

D. location controller

Answer: (SHOW ANSWER)

NEW QUESTION: 230

A. MX-200

B. a Cisco Jabber Desktop

C. CP-7965

D. EX-60

E. DX-650

Answer: (SHOW ANSWER)

NEW QUESTION: 231

Which three configuration settings are included in a default region configuration in CUCM 10x? (Choose three.)

A. Audio Codec

B. Immersive Bandwidth
C. Link Loss Type
D. Location Description
E. Video Call Bandwidth
F. Real Time Protocol
Answer: (SHOW ANSWER)

NEW QUESTION: 232
What is required to effectively use the Cisco CTL client to activate security in an IP telephony network?
A. Security Tokens
B. H.323 gateway
C. Secure SRST
D. IPSEC
Answer: A (LEAVE A REPLY)

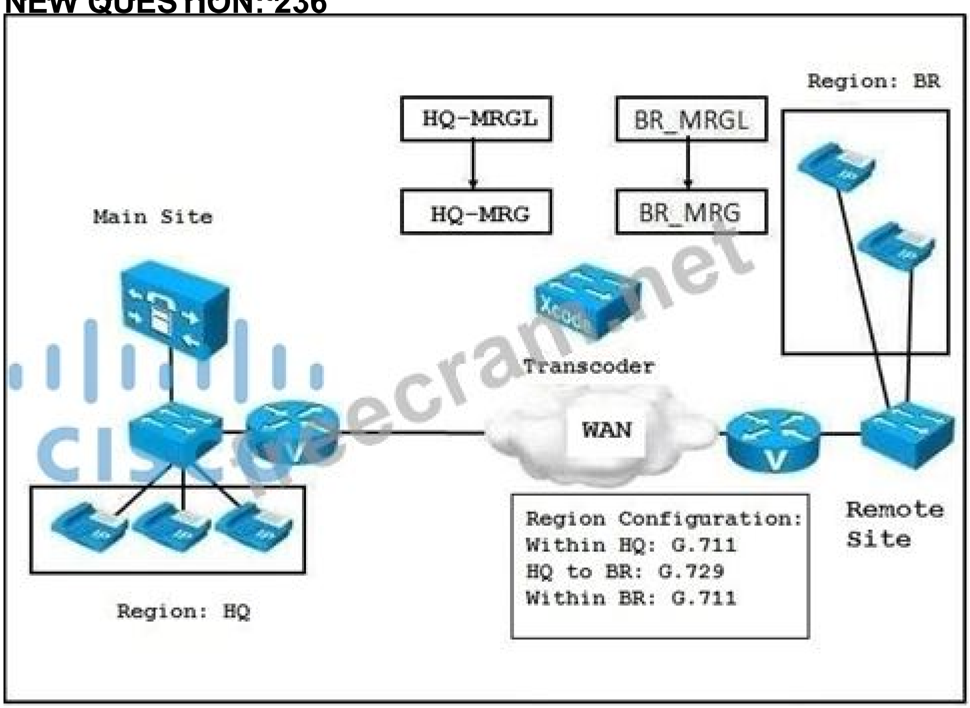
NEW QUESTION: 233


An engineer is deploying a new Cisco VCS Expressway for a company and has configured the IP address and the system name. After logging into the Cisco VCS Expressway admin page, the engineer sees this output. Which option must be configured to complete the Cisco VCS Expressway system configuration?
A. LDAP server
B. DHCP server
C. Cisco Unified Communications Manager IP address
D. DNS server
Answer: (SHOW ANSWER)

NEW QUESTION: 234
Which module is the minimum PVDM3 module needed to support video transcoding?
A. PVDM3-128

B. PVDM3-64
C. PVDM3-32
D. PVDM3-192
Answer: A (LEAVE A REPLY)

NEW QUESTION: 235
Up to how many Cisco unified Communications Manager nodes can a skinny controlled phone simultaneously establish an SCCP connection with?
A. 2
B. 3
C. 4
D. 1
Answer: (SHOW ANSWER)

NEW QUESTION: 236
Refer to the exhibit.


HQ_MRGL is assigned to the HQ IP phones. BR_MRGL is assigned to the BR IP phones. The remote site BR IP phones support only the G.711 codec. Where should the transcoder reside?
A. The transcoder should reside at the HQ site and assigned to HQ_MRG.
B. The transcoder should reside at the BR site and assigned to BR_MRG.
C. The transcoder should be assigned to its own MRG, which should then be assigned to the default device pool.
D. A transcoder is not needed. The HQ phones will automatically change over to the G.711 codec.
Answer: (SHOW ANSWER)

NEW QUESTION: 237
When Cisco Extension Mobility is implemented, how is the audio source for the MOH selected?

A. The audio source that is configured at the user device profile is selected.
B. The audio source that is configured at the home phone of the user is selected.
C. The audio source that is configured at the physical phone used for the Cisco Extension Mobility login is selected.
D. The audio source that is configured in the IP Voice Media Streaming parameters is selected.
Answer: (SHOW ANSWER)
Explanation/Reference:
Explanation:
To specify the audio source that plays when a user initiates a hold action, choose an audio source from the User Hold MOH Audio Source drop-down list box from device profile configuration settings.

NEW QUESTION: 238
Which two have to be defined in the Forward All field? (Choose two)
A. destination
B. calling search space
C. partition
D. hunt list
Answer: (SHOW ANSWER)

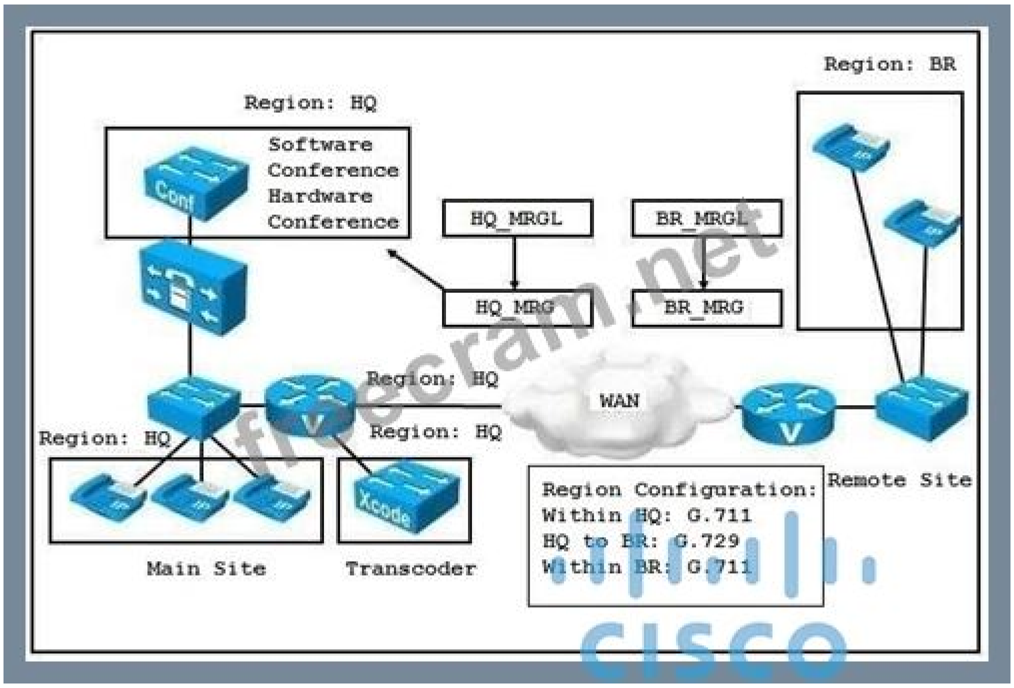
NEW QUESTION: 239
On which Cisco Unified Communications Manager configuration parameter does the CODEC that a Cisco IP Phone uses for a call depend?
A. physical location
B. region
C. enterprise parameters
D. location
E. media resources
Answer: (SHOW ANSWER)

NEW QUESTION: 240
A. start-call
B. in-call
C. end-call
D. call-ended
E. call-started
F. registration
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference: http://www.cisco.com/c/en/us/td/docs/telepresence/infrastructure/articles/
vcs_monitors_presence_status_endpoints_kb_186.html

NEW QUESTION: 241
An engineer is configuring a SIP profile for Cisco VCS SIP trunk on Cisco Unified Communications Manager and enables the option "Use Fully Qualified Domain Name" in SIP Requests. Which result is achieved by enabling this option?
A. Resolve FQDN using DNS type SRV record.
B. Resolve FQDN using DNS type A record.
C. Ensure FQDN is used in SIP Identity header.
D. Ensure FQDN is used in SIP Request header.
Answer: (SHOW ANSWER)
Explanation/Reference:
Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_6_1/ccmcfg/bccm-861-cm/b06siprf.html

NEW QUESTION: 242
Which two statements describe RSVP-enabled locations-based CAC? (Choose two.)
A. RSVP can be enabled selectively between pairs of locations.
B. Using RSVP for CAC simply allows admitting or denying calls based on a logical configuration that is ignoring the physical topology.
C. RSVP is topology aware, but only works with full mesh networks.
D. An RSVP agent is a Media Termination Point that the call has to flow through.
E. RSVP and RTP are used between the two endpoints.
Answer: (SHOW ANSWER)
Explanation/Reference:
Explanation:
The RSVP policy that is configured for a location pair overrides the default interlocation RSVP policy that configure in the Service Parameter Configuration window. RSVP supports audio, video, and data pass- through. Video data pass-through allows video and data packets to flow through RSVP agent and media termination point devices

NEW QUESTION: 243
Refer to the exhibit.



When a call between two HQ users is being conferenced with a remote user at BR, which configuration is needed?

A. The BR_MRG must contain the transcoder device. The BR_MRGL must be assigned to the BR phones.

B. Enable the software conference bridge to support the G.711 and G.729 codecs in Cisco Unified Communications Manager Service Parameters.

C. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the software conference bridge.

D. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the HQ phones.

E. A transcoder should be configured at the remote site and assigned to all remote phones through the BR_MRGL.

Answer: (SHOW ANSWER)

NEW QUESTION: 244

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

A. You need to configure the firewall to allow communication from the Cisco VCS Expressway to the Cisco VCS Control.

B. The Cisco VCS Expressway is the Traversal Server.

C. You do not need to configure search rules for traversal calls.

D. The username on the Cisco VCS Control and Cisco VCS Expressway are local and do not need to match.

Answer: (SHOW ANSWER)

NEW QUESTION: 245

You have deployed a Cisco 2821 ISR to perform as an SRST voice gateway at a remote site. During a network failure between the remote site and the central office, some of the phones

located at the remote site are unable to make phone calls. Which two options are potential causes of the problem? (Choose two.)
A. The site has exceeded the number of simultaneous calls allowed in SRST mode.
B. The ccm-manager fallback-mgcp command is configured incorrectly on the voice gateway.
C. The ccm-manager fallback command is configured incorrectly on the voice gateway.
D. The site has exceeded the number of SRST endpoints supported by the voice gateway.
E. Phones at the remote site are assigned to the incorrect device pool.
Answer: (SHOW ANSWER)

NEW QUESTION: 246
When configuring Cisco Unified Mobility, which parameter defines the access control for a call that reaches out to a remote destination?
A. Rerouting Calling Search Space under Remote Destination information
B. Rerouting Calling Search Space under Remote Destination Profile Information
C. User Local under Remote Destination Profile Information
D. Calling Party Transformation Calling Search Space under Remote Destination Profile Information
E. Calling Search Space under Phone Configuration
Answer: (SHOW ANSWER)

NEW QUESTION: 247
Which action configures PSTN backup for calls that are rejected by the gatekeeper CAC?
A. Configure a route pattern to a gateway in Cisco Unified Communications Manager.
B. Configure AAR in Cisco Unified Communications Manager.
C. Configure a route pattern, a route list, and route groups to a trunk and a gateway in Cisco Unified Communications Manager.
D. Configure CFUR in Cisco Unified Communications Manager.
Answer: (SHOW ANSWER)

NEW QUESTION: 248
Which option describes the reason that transcoding resources are added in Cisco Unified Communications Manager?
A. to enable transcoding resources in a Cisco Unified Communications Manager server
B. to enable Cisco Unified Communications Manager to select the optimal single codec for end-to-end calls
C. to provide transcoding resources in Cisco IOS gateways to Cisco Unified Communications Manager
D. to enable transcoding resources in Cisco IP Phones
Answer: (SHOW ANSWER)

NEW QUESTION: 249

When considering Extension Mobility, what happens if a user logs into a phone for which the user does not have a user device profile?

A. Another user device profile is loaded.

B. The phone reboots with an error.

C. If a default device profile for this phone has been configured, it is loaded.

D. The user cannot log in.

Answer: (SHOW ANSWER)

NEW QUESTION: 250

A. Cisco Unified OS Administration > Software Upgrades > TFTP File Management

B. Cisco Unified OS Administration > Settings > Version

C. Cisco Unified Reporting > System Reports

D. Cisco Unified CM Administration > Device Defaults Configuration

E. Cisco Unified CM Administration > Firmware Load Information

F. Cisco Unified CM Administration > Device > Phone

Answer: (SHOW ANSWER)

NEW QUESTION: 251

What is a prerequisite of AAR deployment?

A. You must have a single distributed call processing deployment.

B. Calls must be manually rerouted through the PSTN or other networks when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth.

C. Clustering must be implemented over the WAN.

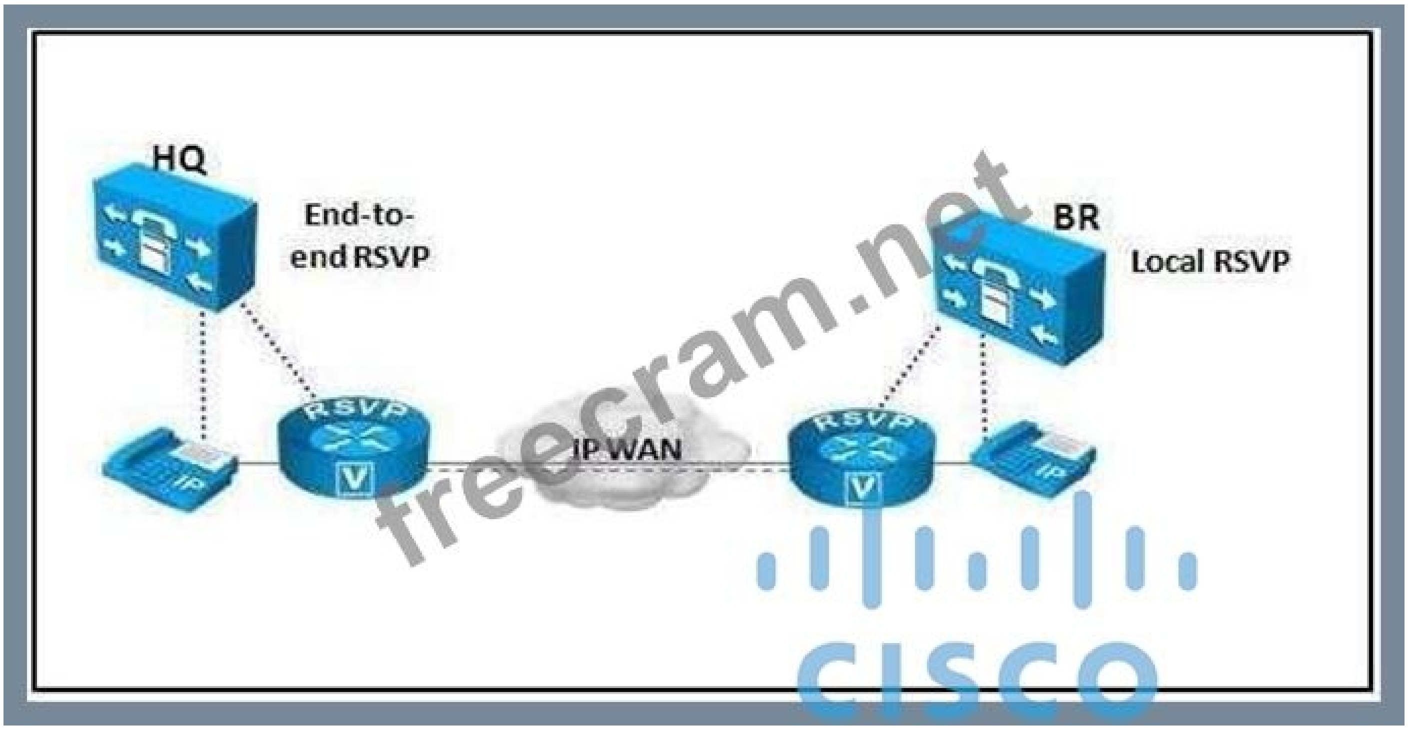
D. Calls must be automatically rerouted through the PSTN or other networks when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth.

E. You must have a centralized call processing deployment.

Answer: (SHOW ANSWER)

NEW QUESTION: 252

Refer to the exhibit.



The Cisco Unified Communications Manager at HQ has been configured for end-to-end RSVP. The Cisco Unified Communications Manager at BR has been configured for local RSVP. RSVP between the locations assigned to the IP phones and SIP trunks at each site are configured with mandatory RSVP. When a call is placed from the IP phone at the BR site to the IP phone at the HQ site, which statement is true?

A. The call will fail.

B. The Cisco Unified Communications Manager at BR will use local RSVP. The HQ Cisco Unified Communications Manager will use end-to-end RSVP.

C. The call will proceed as a normal call with no RSVP reservation.

D. The Cisco Unified Communications Manager at BR will fall back to local RSVP and place the call. No RSVP end-to-end will occur.

E. RSVP end-to-end will occur.

Answer: (SHOW ANSWER)

NEW QUESTION: 253

Which minimum configuration is needed for the SAF Internal Client to register with this SAF Forwarder?

A. router eigrp SAF

!

service-family ipv4 autonomous-system 1

!

topology base

exit-sf-topology

exit-service-family

!

voice service saf

!
channel 1 vrouter SAF asystem 1
B. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
voice service saf
profile trunk-route 1
session protocol sip interface Loopbackl transport tcp port 5060 ! profile dn-block 1 alias-prefix 1972555 pattern 1 type extension 4xxx
!
profile
callcontrol 1 dn-
service trunk-
route 1 dn-block
1 dn-block 2
!
channel 1 vrouter SAF asystem
1 subscribe callcontrol
wildcarded publish callcontrol 1
i
C. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
voice service saf
profile trunk-route 1
session protocol sip interface Loopbackl transport tcp port 5060 ! profile dn-block 1 alias-prefix 1972555 pattern 1 type extension 4xxx
!
profile callcontrol 1 dn-
service trunk-
route 1 dn-block

1 dn-block 2

i

D. router eigrp SAF

!

service-family ipv4 autonomous-system 1

!

topology base

exit-sf-topology

exit-service-family

!

voice service saf

profile trunk-route 1

session protocol sip interface Loopbackl transport tcp port 5060 i

E. router eigrp SAF

!

service-family ipv4 autonomous-system 1

!

topology base exit-sf-

topology exit-service-family i

Answer: D (LEAVE A REPLY)

NEW QUESTION: 254

Which two options are requirements for hardware MTP on Cisco IOS routers? (Choose two.)

A. DSP resources

B. the same audio codec on both legs of the call

C. a hardware transcoder

D. a binding IP address

E. a T1 card

F. an FXO card

Answer: (SHOW ANSWER)

NEW QUESTION: 255

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

A. Check the Enable Extension Mobility checkbox on the Directory Number Configuration page.

B. Create a user Device Profile.

C. Unsubscribe all other services from the Cisco IP Phone.

D. Check the Home Cluster checkbox on the End User Configuration page.

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: 256

Which three globalization dialing functions are enhanced in Cisco Unified Communications Manager 7.x and later? (Choose three.)

A. MGRL

B. TEHO

C. CER

D. AAR

E. SAF

F. click-to-call

Answer: (SHOW ANSWER)

Explanation/Reference:

Explanation:

TEHO stands for Tail End Hop Off, CER stands for Cisco Emergency Responder and AAR stands for Automated Alternate Routing. These are the correct answers according to the Cisco Collaboration System SRND.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/9x/uc9x/dialplan.html (benefits of new design approach)

NEW QUESTION: 257

Which CAC configuration on a gatekeeper restricts to 10 G.711 audio calls?

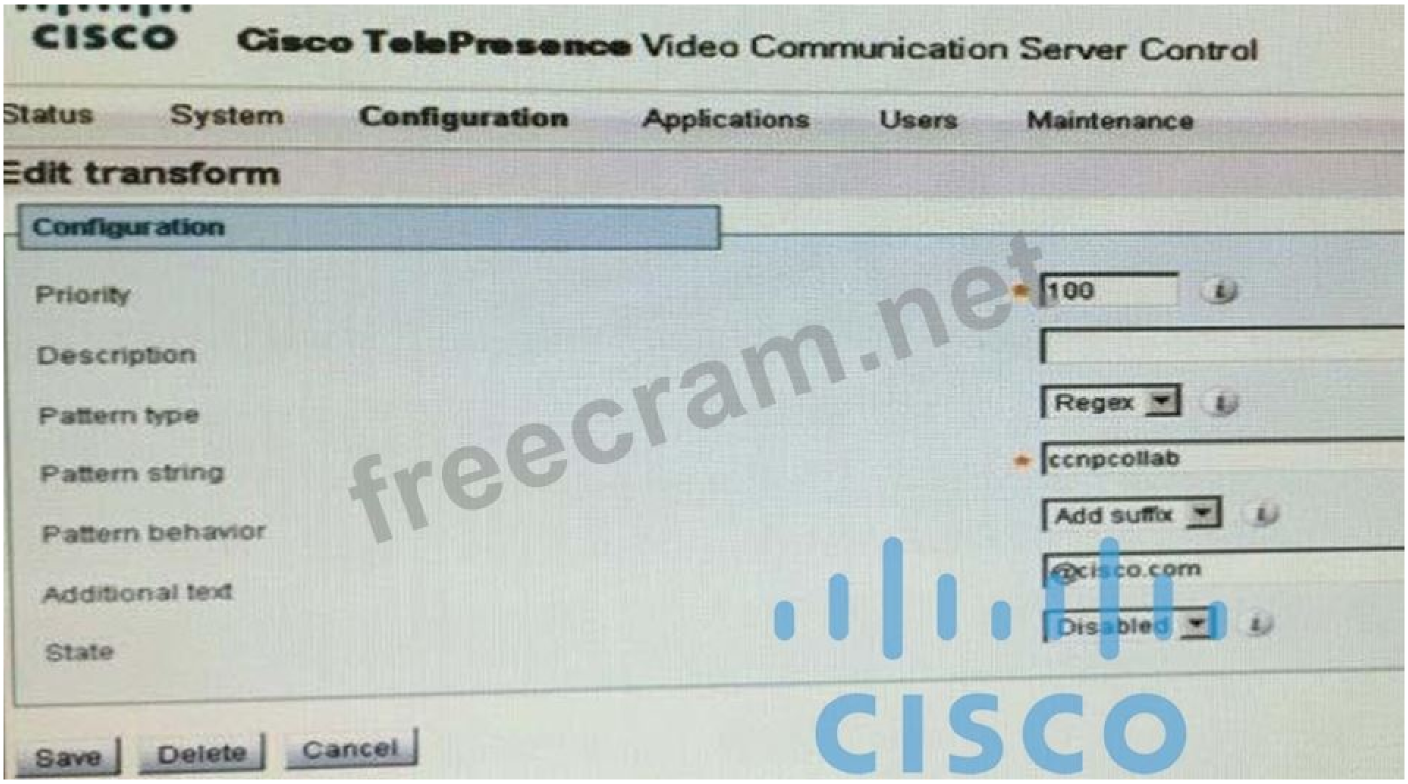
A. Use the command bandwidth 1280.

B. Use the command bandwidth 160.

C. Use the command bandwidth 10.

D. Use the command bandwidth session 10.

Answer: (SHOW ANSWER)



Refer to the exhibit. An engineer is troubleshooting a Cisco VCS Control call-routing issue. What happens when ccnpcollab is dialed?

A. The call is routed as ccnpcollab.

B. The call is not routed.

C. The call is routed to cisco.com as ccnpcollab

D. The call is routed as ccnpcollab@cisco.com

Answer: (SHOW ANSWER)

NEW QUESTION: 259

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

A. enterprise phone configuration

B. Ethernet configuration

C. service parameters

D. enterprise parameters

Answer: C (LEAVE A REPLY)

NEW QUESTION: 260

A. G.722

B. G.723

C. Advanced Audio Codec

D. G.711

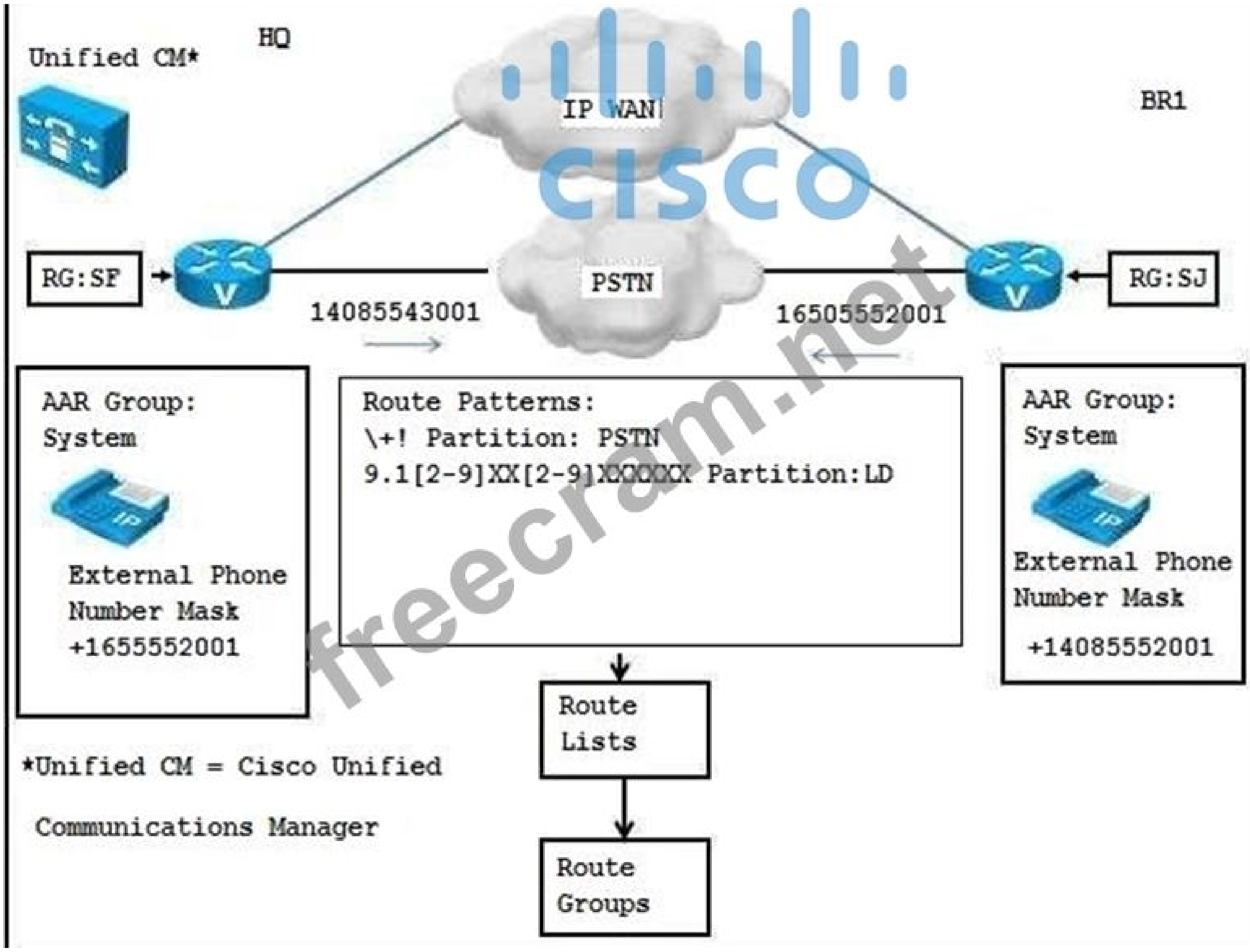
E. G.728

F. G.729

Answer: (SHOW ANSWER)

NEW QUESTION: 261

Refer to the exhibit.



The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number. Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit. Which statement is true?

A. The AAR group system must be configured on the device configuration of the phones.

B. The AAR group system must be configured under the AAR service parameters.

C. The single AAR group system cannot be used. A second AAR group must be configured in order to have source and destination AAR groups.

D. The AAR group system must be configured on the line configuration of the phones.

Answer: (SHOW ANSWER)

NEW QUESTION: 262

Which command should you use on the gatekeeper to specify the address of a Cisco Unified Communications Manager IP address?

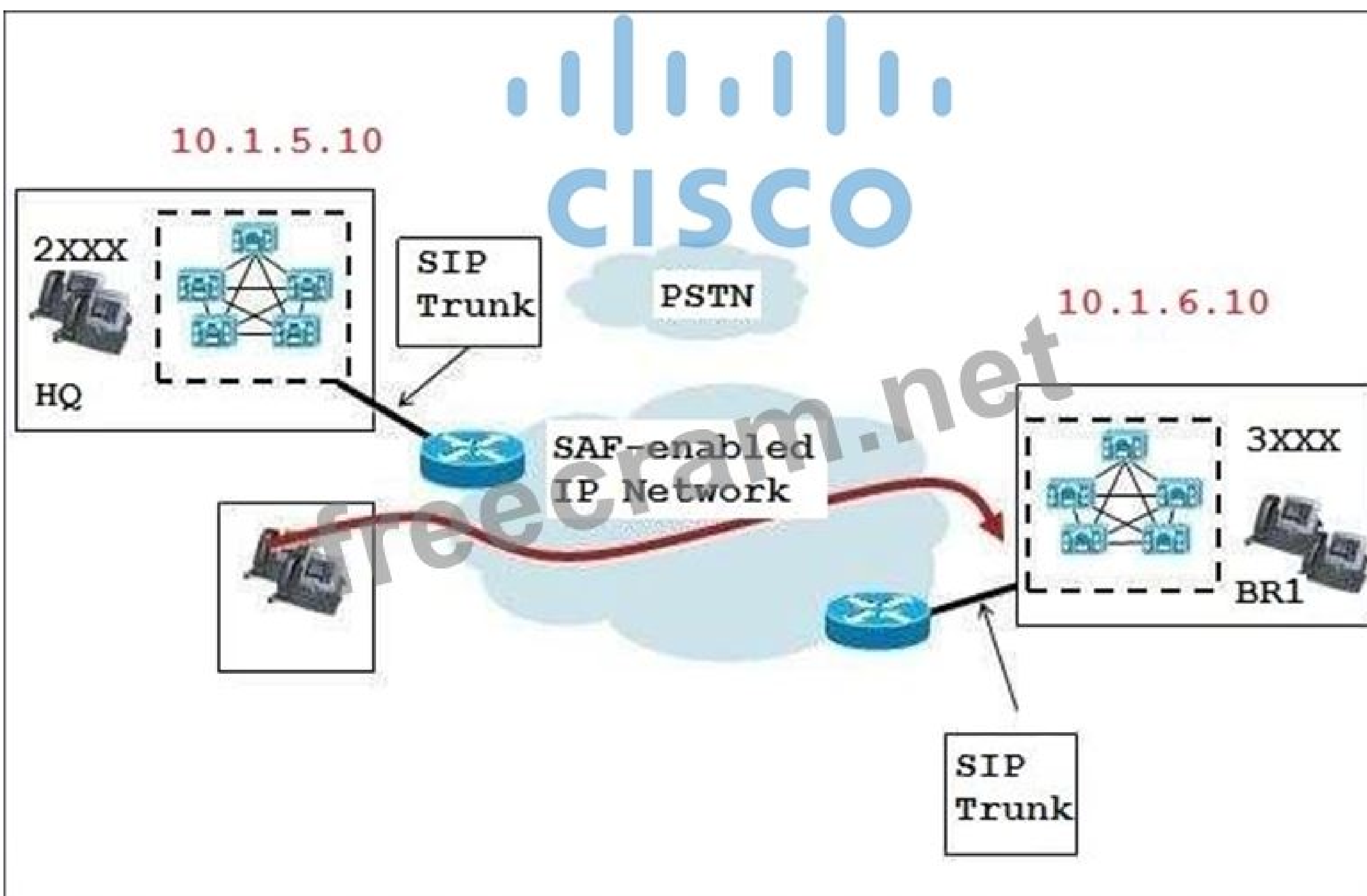
A: gw-prefix-ip

B: gw-ip-cm

C. gw-type-prefix-ip
D. gw-type-prefix
E. gw-type-ip
F. gw-ip-type
Answer: (SHOW ANSWER)

NEW QUESTION: 263
Which statement about the SAF Client Control is correct?
A. The SAF Client Control is a non-configurable inherent component of the Cisco IOS Routers.
B. The SAF Client Control is a non-configurable inherent component of Cisco Unified Communications Manager.
C. The SAF Client Control is a configurable inherent component of the Cisco IOS Routers.
D. The SAF Client Control is a configurable inherent component of Cisco Unified Communications Manager.
Answer: (SHOW ANSWER)

NEW QUESTION: 264
Refer to the exhibit.



What must be configured on the HQ Cisco Unified Communications Manager to allow HQ users to dial the SAF learned directory number pattern 3XXX?
A. Route pattern 3XXX should be configured and made available to HQ users through the phone CSS.

B. Route pattern 3XXX should be configured and made available to HQ phone users through the phone AAR CSS.
C. The SAF partition assigned to the SAF learned patterns must be available to the HQ phone users through the phone CSS.
D. The SAF partition assigned to the SAF learned patterns must be available to the HQ phone users through the phone AAR CSS.
E. The SAF directory number pattern 3XXX will be made available to all users automatically as soon as the SAF partition is selected.
Answer: (SHOW ANSWER)
Explanation/Reference:
Explanation:
By adopting the SAF network service, the call control discovery feature allows Cisco Unified Communications Manager to advertise itself along with other key attributes.

NEW QUESTION: 265
Which three options are supplementary services that are affected by MTP? (Choose three.)
A. Call Pickup
B. Call Back
C. Call Park
D. Call Hold
E. Call Transfer
F. Speed Dial
Answer: (SHOW ANSWER)

NEW QUESTION: 266
Which action configures AAR to route the calls that have been rejected by the gatekeeper CAC through the PSTN?
A. Configure AAR to work with SRST.
B. Configure Cisco IP Phones for AAR.
C. Configure AAR to work with CTI route points.
D. This configuration is not possible using AAR.
Answer: (SHOW ANSWER)

NEW QUESTION: 267
What is the difference between an MGCP gateway and a SIP gateway?
A. An MGCP gateway that dial peers be configured before PSTN calls can be placed and received. The SIP gateway requires no dial peers.
B. A SIP gateway requires a call agent for PSTN calls to be placed and received. An MGCP gateway does not require a call agent for PSTN calls to be placed and received.
C. The SIP gateway must be configured in Cisco Unified Communications Manager using a valid IP address on the gateway. The MGCP gateway must be configured in Cisco Unified

Communications Manager using the domain name.
D. An MGCP gateway can register with Cisco Unified Communications Manager. A SIP gateway will show status of "Unknown".
E. An MGCP gateway can be added in Cisco Unified Communications Manager under the Gateway Type field using the gateway model. The SIP gateway can connect to Cisco Unified Communications Manager only through a SIP trunk.
Answer: (SHOW ANSWER)

NEW QUESTION: 268
What are two requirements for configuring an Intercluster Trunk (gatekeeper-controlled) in Cisco Unified Communications Manager? (Choose two.)
A. The assigned name must be unique within the cluster.
B. The IP addresses of Cisco Unified Communications Manager systems in the remote cluster must be specified.
C. The Cisco Unified Communications Manager group in the assigned device pool will determine which Cisco Unified Communications Manager systems register with the gatekeeper.
D. RSVP must be enabled to provide CAC between clusters.
E. The gatekeeper must be defined in Cisco Unified Communications Manager before the intercluster trunk is added.
F. The Intercluster Trunk must have the same name in both clusters.
Answer: C,E (LEAVE A REPLY)

NEW QUESTION: 269
Which two entities could be represented by device mobility groups? (Choose two.)
A. regions
B. transcoders
C. countries
D. directory numbers
Answer: A,C (LEAVE A REPLY)

NEW QUESTION: 270
Refer to the exhibit.

```
voice service saf
 profile trunk-route1
 session protocol sip interface Loopback1 transport tcp port 506(
 !
profile dn-block 1 alias-prefix 1972555
 pattern 1 type extension 4XXX
 !
profile dn-block 2
 pattern 1 type global 14087071222
 !
profile callcontrol 1
 dn-service
  trunk-route 1
  dn-block 1
  dn-block 2
 !
!
!
channel 1 vrouter SAF asystem 1
 subscribe callcontrol wildcarded
 publick callcontrol 1
!
```

How does the Cisco Unified Communications Manager advertise dn-block 2?

A. +14087071222 with number type international

B. 14087071222

C. 14087071222 with number type international

D. +14087071222

Answer: (SHOW ANSWER)

NEW QUESTION: 271

DRAG DROP

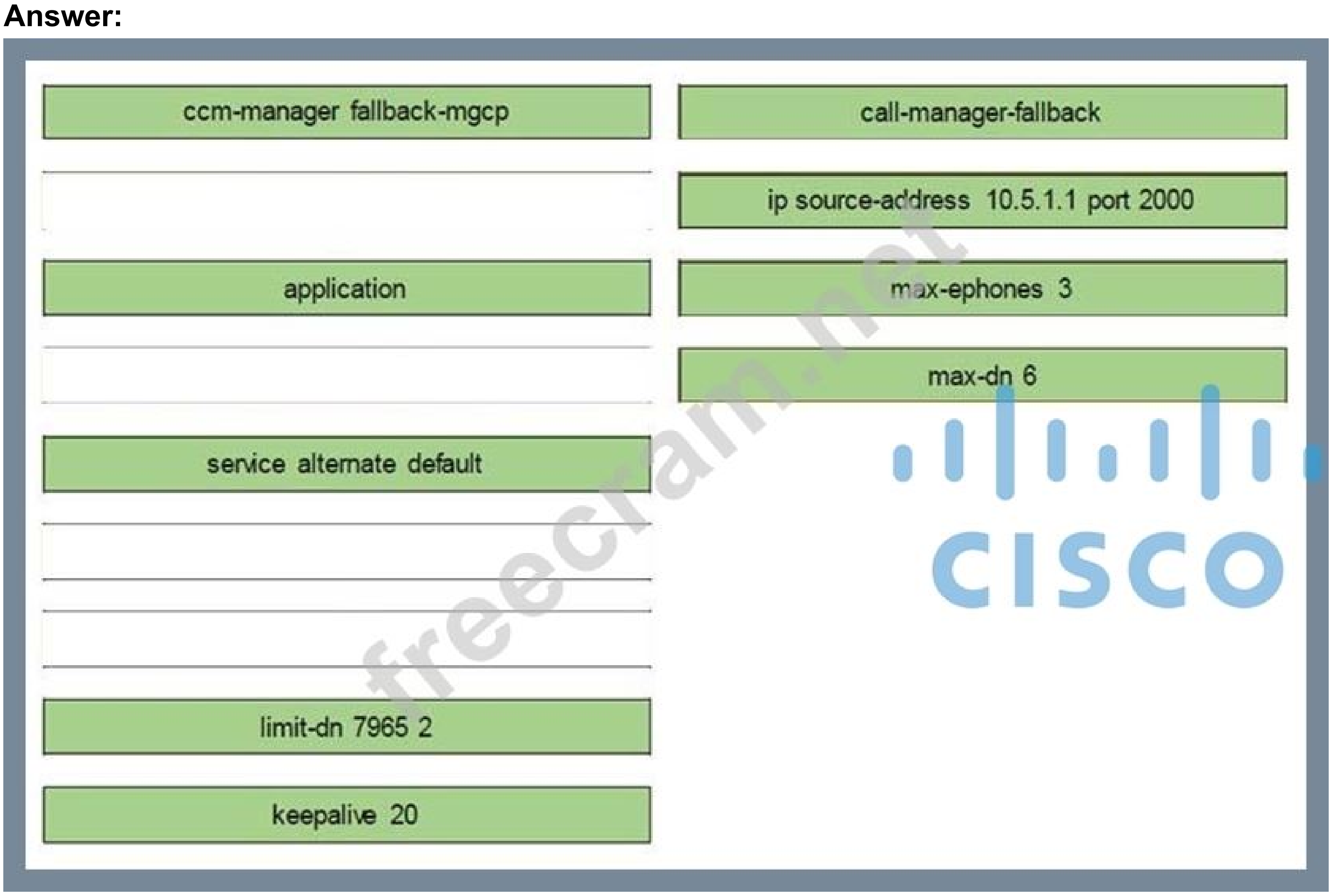
Click and drag the minimum Cisco Unified SRST configuration steps on the left to the spaces on the right.

Not all spaces on the right are used.

Select and Place:


ccm-manager fallback-mgcp
call-manager-fallback
application
ip source-address 10.5.1.1 port 2000
service alternate default
max-ephones 3
max-dn 6
limit-dn 7965 2
keepalive 20

Answer:


ccm-manager fallback-mgcp
application
service alternate default
limit-dn 7965 2
keepalive 20
call-manager-fallback
ip source-address 10.5.1.1 port 2000
max-ephones 3
max-dn 6

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