Question 1
When a call is placed from the IP phone at HQ to the BR phone at the BR site, which statement is true?

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: D
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.

Question 2
To permit three G.729 calls, what should the bandwidth value be for the ip rsvp bandwidth command?

Refer to the exhibit. 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. 88
E. 128

Answer: D

Question 3
How many calls are permitted by the RSVP configuration?

Refer to the exhibit. How many calls are permitted by the RSVP configuration?
A. one G.711 call  
B. two G.729 calls  
C. one G.729 call and one G.711 call  
D. eight G.729 calls  
E. four G.729 calls  

Answer: B

**Question 4**

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

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: D

**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.

**Question 5**

Which configuration should be implemented?

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_MRGL. 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_MRGL. The hardware conference bridge must be configured first.  
C. Assign the hardware conference bridge to HQ_MRGL. Configure a second HQ_MRGL_2 and assign the software conference bridge to it.  

Answer: A

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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: C  
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 ...  

Question 6  
When a call between two HQ users is being conferenced with a remote user at BR, which configuration is needed?  
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. The HQ_MRG must contain the transcoder device.  
The HQ_MRGL must be assigned to the HQ phones.  
C. A transcoder should be configured at the remote site and assigned to all remote phones through the BR_MRGL.  
D. The HQ_MRG must contain the transcoder device.  
The HQ_MRGL must be assigned to the software conference bridge.  
E. Enable the software conference bridge to support the G.711 and G.729 codecs in Cisco Unified Communications Manager Service Parameters.  

Answer: D  

Question 7  
What three things should you do to resolve this issue?  
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. activate EM feature service under Cisco Unified Serviceability  
C. associate EM Device profile with the end-user  
D. subscribe the MAC address of the IP Phone to EM Service  
E. update EM Phone Service URL to point to the publisher  
F. subscribe device profile to EM phone service in case the enterprise subscription of EM Service is disabled  

Answer: BCD  

Question 8  
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?  
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 takes on the default clusterwide device profile.  
B. The phone creates a new device profile automatically.  
C. The phone immediately logs the user off.  
D. The phone crashes and reboots.  

Answer: A  

Question 9  
Which two entities could be represented by device mobility groups?  
Which two entities could be represented by device mobility groups? (Choose two.)  
A. countries  
B. regions  
C. directory numbers  
D. transcoders  

Answer: AB
Question 10
How does the visiting cluster determine if the user is a local user or a remote user?
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 using a third-party automatic provisioning tool to verify user ID
B. by broadcasting a request to all clusters to verify the user type
C. from user IDs that are created by default when the user logs in
D. by using Extension Mobility Cross Cluster Session Initiation Protocol (SIP) trunks
E. by verifying against the local database
F. by verifying the visiting Trivial File Transfer Protocol
Answer: E

Question 11
Which CSS is responsible for ensuring that the correct partitions are accessed when calls are sent to the Enterprise user’s mobile phone?
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: D
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.

Question 12
When Cisco Extension Mobility is implemented, how is the audio source for the MOH selected?
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: A
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.

Question 13
When multiple Cisco Extension Mobility profiles exist, which actions take place when a user tries to log in to Cisco Extension Mobility?
When multiple Cisco Extension Mobility profiles exist, which actions take place when a user tries to log in to Cisco Extension Mobility?
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: B
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.

Question 14
When Cisco Extension Mobility is implemented, which CSS is used for calling privileges?
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 user device profile device CSS combined with the line CSS of the physical phone used to log in the extension mobility user.
C. Only the user device profile device CSS is used.
D. The combined line/device CSS of the physical phone is used to log in the extension mobility user.
E. The combined line/device CSS of the user device profile.
Answer: A

Question 15
Which two options for a Device Mobility-enabled IP phone are true?
Which two options for a Device Mobility-enabled IP phone are true? (Choose two.)
A. The phone configuration is not modified.
B. The roaming-sensitive parameters of the current (that is, the roaming) device pool are applied.
C. The user-specific settings determine which location-specific settings are downloaded from the Cisco Unified Communications Manager device pool.
D. If the DMGs are the same, the Device Mobility-related settings are also applied.
Answer: BD

Question 16
What is the purpose of a SAF Client?
What is the purpose of a SAF Client?
A. To decode address information and route calls to and from the end points
B. To pass IP information from the CUCM to the endpoint
C. To learn about and advertise or subscribe information about SAF network services
D. To reside in the Cisco IOS software, and to communicate with the SAF forwarder
Answer: C

Question 17
When a user presses a speed dial to +442079460255 when the SAF network is down, which event should occur?
Refer to the exhibit. 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 442079460255.
B. The call will reroute via the PSTN with the constructed PSTN number as +442079460255.
C. The call will reroute via the PSTN with the constructed PSTN number as 00442079460255.
D. The call will fail because the ToDID is 0.
E. The call will fail because the called number will be 2079460255.
Answer: B

Question 18
Which minimum configuration is needed for the SAF Internal Client to register with this SAF Forwarder?
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
   profile trunk-route 1
   session protocol sip interface Loopback1 transport tcp port 5060 i
   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 Loopback1 transport tcp port 5060 !
   profile dia-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
   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 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
! D. 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
E. router eigrp SAF
! service-family ipv4 autonomous-system 1
! topology base exit-sf-topology
exit-service-family i

Answer: A

Question 19
Which two actions are performed by the Call Control Discovery service after the local Cisco Unified Communications Manager loses its TCP connection with the primary and secondary Service Advertisement Framework?
Which two actions are performed by the Call Control Discovery service after the local Cisco Unified Communications Manager loses its TCP connection with the primary and secondary Service Advertisement Framework? (Choose two.)
A. Calls are routed to the PSTN gateway after the Call Control Discovery Learned Pattern IP Reachable Duration parameter expires.
B. All learned patterns are purged from the local cache after the Call Control Discovery PSTN Failover Duration parameter expires.
C. The Service Advertisement Framework forwarder contacts all the remaining Service Advertisement Framework forwarders in the cluster.
D. All the remaining Service Advertisement Framework forwarders are notified of their learned patterns.
E. 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.
F. Call Control Discovery immediately redirects all the calls to the PSTN gateway based on the learned patterns.

Answer: AB

Question 20
What is the maximum number of Service Advertisement Framework forwarders to which the Cisco Unified Communications Manager can connect? 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. 1
B. 2
C. 3
D. 4
E. 6
F. as many as are configured

Answer: B

Question 21
Which three characters should you avoid entering in the description? You are entering the description for the Service Advertisement Framework forwarder.
Which three characters should you avoid entering in the description? (Choose three.)
A. &
B. %
C. 
D. 
E. $
F. @

Answer: ABC

Question 22
What are the two tasks that you must perform to configure the Service Advertisement Framework forwarder in Cisco Unified Communications Manager? What are the two tasks that you must perform to configure the Service Advertisement Framework forwarder in Cisco Unified Communications Manager? (Choose two.)
A. create VPN groups
B. create VPN profiles
C. create a new Service Advertisement Framework security profile
D. set feature configuration parameters of Call Control Discovery
E. configure Service Advertisement Framework forwarder information
F. enable enterprise parameter for Service Advertisement Framework forwarder

Answer: CE

Question 23
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: E
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

Question 24
The exhibit shows a SAF Forwarder configuration attached to a Cisco Unified Communications Manager.
Refer to the exhibit. 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 trunkroute 1
   session protocol sip interface Loopback1 transport tcp port 5060
B. router eigrp SAF
   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
   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
C. router eigrp SAF
   service-family ipv4 autonomous-system 1
   topology base
   exit-sf-topology

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exit-service-family
D. None of above configurations contain sufficient information.
Answer: C

**Question 25**
When the Manager places a call to 3001 when the SAF network is down, what happens?
Refer to the exhibit. When the Manager places a call to 3001 when the SAF network is down, what happens?

**Answer:** A

**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.

**Question 26**
Which Cisco IOS SAF Forwarder configuration is correct?
Refer to the following exhibit. Which Cisco IOS SAF Forwarder configuration is correct?

**Answer:** A
Question 27
How does the Cisco Unified Communications Manager advertise dn-block 2?
Refer to the exhibit. How does the Cisco Unified Communications Manager advertise dn-block 2?
Question 28
How does the Cisco Unified Communications Manager advertise dn-block 1?
Refer to the exhibit. How does the Cisco Unified Communications Manager advertise dn-block 1?

A. 4XXX and the ToDID will 0:
B. 4XXX and the ToDID will 0:1972555
C. 4XXX
D. 4XXX and the ToDID will 0:+1972555
E. 19725554XXX

Answer: B

Question 29
Which CSS is used at the HQ Cisco Unified Communications Manager to reroute calls via the PSTN when the SAF network is unavailable?
Refer to the exhibit. Which CSS is used at the HQ Cisco Unified Communications Manager to reroute calls via the PSTN when the SAF network is unavailable?

A. 14087071222 with number type international
B. +14087071222 with number type international
C. +14087071222
D. 14087071222
Answer: C
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: C

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.

Question 31

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

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

Question 32
Which configuration elements must match for adjacent neighbors to establish a SAF neighbor relationship?
Refer to the exhibit. Which configuration elements must match for adjacent neighbors to establish a SAF neighbor relationship?

A. the label name specified in the router eigrp command
B. the autonomous-system number specified in the service-family ipv4 autonomous-system command
C. the sf-interface configuration
D. the topology base configurations
E. the label name specified in the router eigrp command and the autonomous-system number

Answer: B

Question 33
Which statement is true?
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 on the line configuration of the phones.
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 under the AAR service parameters.

Answer: B
Question 34
Which three reasons for this error are true?
When configuring intercluster URI dialing, an engineer gets the error message “Local cluster cannot connect to the ILS network”. Which three reasons for this error are true? (Choose three.)
A. The SIP route patterns have not been properly configured.
B. The Tomcat certificates do not match.
C. The Cisco Unified Resource Identifier service needs a restart.
D. The ILS authentication password does not match.
E. The cluster ID does not match.
F. One cluster is using TLS certificate, and the other is using Password.
Answer: BDF

Question 35
Where should the transcoder reside?
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_MRGL.
B. The transcoder should reside at the BR site and assigned to BR_MRGL.
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: B

Question 36
Which configuration should be implemented?
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_MRGL. 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_MRGL. The hardware conference bridge must be configured first.
C. Assign the hardware conference bridge to HQ_MRGL. Configure a second HQ_MRGL_2 and assign the software conference bridge to it.
D. Assign the hardware conference bridge to HQ_MRGL. Configure a second HQ_MRGL_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.

Question 37
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?
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 by configuring the significant digits for inbound calls on the H.323 gateway configuration in Cisco Unified Communications Manager.
B. Normalization is done using route patterns.
C. Normalization is done using the gateway incoming calling party prefixes based on number type.
D. Normalization is achieved by local route group that is assigned to the H.323 gateway.

Question 38
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?
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 done using translation patterns.
B. Normalization is done using route patterns.
C. Normalization is done using the gateway incoming called party prefixes based on number type.
D. Normalization is achieved by local route group that is assigned to the H.323 gateway.

Question 39
What is the difference between an MGCP gateway and a SIP gateway?
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. 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.
C. 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.
D. An MGCP gateway can register with Cisco Unified Communications Manager. A SIP gateway will show status of “Unknown”.
E. 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.

Question 40
What is the difference between an H.323 gateway and a SIP gateway?
What is the difference between an H.323 gateway and a SIP gateway?
A. An H.323 gateway requires that dial peers be configured before PSTN calls can be placed and received. The SIP gateway requires no dial peers.
B. The H.323 gateway can be added in Cisco Unified Communications Manager under gateway type as H.323 Gateway. The SIP gateway can connect to Cisco Unified Communications Manager only through a SIP trunk.
C. A SIP gateway requires a call agent for PSTN calls to be placed and received. An H.323 gateway does not require a call agent for PSTN calls to be placed and received.
D. An H.323 gateway can register with Cisco Unified Communications Manager. A SIP gateway will show status of “Unknown”.
E. The H.323 gateway must be configured in Cisco Unified Communications Manager using a valid IP address on the gateway. The SIP gateway must be configured in Cisco Unified Communications Manager using the domain name.

Question 41
Which set of implementations would best address the overlapping directory number extensions for intersite (WAN) calling between the HQ site and the BR site?
Refer to the exhibit. Assume that the HQ phones have access to the HQ partition, and BR phones have access to the BR partition. Which set of implementations would best address the overlapping directory number extensions for intersite (WAN) calling between the HQ site and the BR site?
A. Configure a route pattern 8222.[12]XXX for site HQ, and assign it to partition HQ. Configure the called party DDI of Predot.
   Configure a route pattern for site BR 8111.[1-3]XXX, and assign it to partition BR. Configure called party DDI Predot.
   Use the local gateway at each site. Prefix the appropriate site code for the calling number.

B. Configure a single route pattern for both sites 8[12,12,12].[1-32]XXX. Use a route list that contains the local route group for each site. Prefix the appropriate site code for the calling number.

C. Configure a translation pattern 8222.[12]XXX for site HQ, and assign it to partition HQ.
   Use a CSS that contains the partitions for BR phones.
   Configure a translation pattern 8111.[1-3]XXX for site BR, and assign it to partition BR.
   Use a CSS that contains the partitions for HQ phones.
   For both translation patterns, configure the called party DDI of Predot.
   Prefix the appropriate site code for the calling number.

D. Configure a translation pattern 8222.[12]XXX for site HQ, and assign it to partition BR.
   Use a CSS that contains the partitions for HQ phones.
   Configure a translation pattern 8111.[1-3]XXX for site BR, and assign it to partition HQ.
   Use a CSS that contains the partitions for BR phones.
   For both translation patterns, configure the called party DDI of Predot.
   Prefix the appropriate site code for the calling number.

Question 42
Which two statements are true regarding the implementation of globalized call-routing in terms of localized call egress?

Which two statements are true regarding the implementation of globalized call-routing in terms of localized call egress? (Choose two.)
A. Calling-party numbers are routed from the gateway or trunks to phones.
B. Called-party numbers are routed from the gateway or trunks to phones.
C. Calling-party numbers of internal calls are routed from the gateway or trunks.
D. Calling-party calls are routed to the gateway and trunks.

Answer: AD

Question 43
Which E.164 transformation pattern represents phone numbers in Germany?

Which E.164 transformation pattern represents phone numbers in Germany?
A. +49.
B. 49.X
C. 49..
D. +49.X

Answer: A

Question 44
Which trunk should you use in an H.323 gatekeeper-controlled network?

Which trunk should you use in an H.323 gatekeeper-controlled network?
A. H.323
B. H.225
C. SIP
D. Intercluster
E. MGCP FXO trunk
F. MGCP T1/E1 trunk

Answer: B

Question 45
How can this be accomplished?

When an external call is placed from Ajax, they would like the ANI that is sent to the PSTN to be the main number, not the extension. For domestic calls, they would like 10 digits sent; for international calls, they would like to send the country code 1 and the 10 digits. How can this be accomplished?
A. Add a translation pattern to the dial peers in the gateway that adds the appropriate digits to the outgoing ANI.
B. In the external call route patterns, set the external phone number mask to the main number. Use 10 digits in the domestic route pattern and 1 followed by the main number digits in the international route patterns.
C. Use a calling party transform mask for each route group in the corresponding route list configuration. Set the explicit 10-digit main number for domestic calls and 1 followed by the main number for the international route patterns.
D. In the directory number configurations, set the prefix digits field to the country code and the 10 digits of the main number. This will be truncated to the 10-digit number for domestic calls and sent out in its entirety for international calls.

Answer: C
Question 46
When the Cisco Unified Communications Manager advertises the Hosted DN Pattern, which pattern would be advertised?
Refer to the exhibit When the Cisco Unified Communications Manager advertises the Hosted DN Pattern, which pattern would be advertised?

<table>
<thead>
<tr>
<th>Hosted DN Pattern Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
</tr>
<tr>
<td>Hosted DN Pattern</td>
</tr>
<tr>
<td>PSTN Failover Strip Digits</td>
</tr>
<tr>
<td>PSTN Failover Preceded Digits</td>
</tr>
</tbody>
</table>

A. 2XXX and the ToDID will be 0:+498950555
B. 2XXX and the ToDID will be 0:+498951 21
C. +498960552XXX and the ToDID will be 0:
D. +498960552XXX and the ToDID will be 0:
E. Both +4989505552XXX and +498951 21 2XXX will be advertised with ToDID of 0:

Answer: A

Question 47
When the IP phone at extension 3001 places a call to 9011 49403021 56001# what is the resulting called and calling number that is sent to the PSTN?
Refer to the following exhibit. The MGCP gateway has the following configurations:

called party transformationCSS HQ_cld_pty CSS (partition=HQ cld_pty.Pt) call.ng party transformation CSS HQ_clng_pty CSS (partition=HQ_clng_pty Pt)

A. The called number is 01 1 49403021 56001.
The calling number will be 5553001 and number type set to subscriber.
B. The called number is 011 49403021 56001.
The calling number will be 5215553001 and number type set to national.
C. The called number is 4940302156001 with number type set to international.
The calling number will be 5215553001 and number type set to national.
D. The called number is +4940302156001 with number type set to international.
The calling number will be 5215553001 and number type set to subscriber.

Answer: A
Explanation:
Check the check box “Use Calling Party’s External Phone Number Mask” if you want the full, external phone number to be used for calling line identification (CLID) on outgoing calls. You may also configure an External Phone Number Mask on all phone devices.

Question 48
Which statement about the emergency call is true?
Assume a centralized Cisco Unified Communications deployment with the headquarters in the U.K. and remote site in RTP. All route patterns are assigned a route list that points to the local route group. Local route groups have been configured on the U.K and RTP device pools. A U.K. user logs onto an RTP phone using the Cisco Extension Mobility feature and places an emergency call to 0000. Which statement about the emergency call is true?

A. The call will match the U.K_Emergency route pattern partition and will egress at the RTP gateway.
B. The call will match the U.K_Emergency route pattern partition and will egress at the U.K. gateway.
C. The call will match the RTP_Emergency route pattern partition and will egress at the U.K. gateway.
D. The call will match the RTP_Emergency route pattern partition and will egress at the RTP gateway.
E. The call will fail.

Answer: A

Question 49
How many route lists and route groups should be configured for AAR at a minimum?
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.
How many route lists and route groups should be configured for AAR at a minimum?

A. a single route list with a local route group for each site
B. two route lists and two route groups for each site
C. a single route list and four route groups for each site
D. None. The AAR CSS can point directly to the route pattern.

Answer: A

Question 50
Which partition should be configured in the AAR CSS applied at the phones?
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.

A. a single route list with a local route group for each site
B. two route lists and two route groups for each site
C. a single route list and four route groups for each site
D. None. The AAR CSS can point directly to the route pattern.

Answer: A
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 partition should be configured in the AAR CSS applied at the phones?

A. PSTN partition
B. LD partition
C. The HQ AAR CSS must include a partition assigned to route pattern 91408XXXXXXX. The BR1 AAR CSS must include a partition assigned to route pattern 91650XXXXXXX.
D. AAR CSS must contain translation pattern 9.1[2-9]XX[2-9]XXXXXX for each site that must be globalized. Otherwise the called numbers will not be localized at the egress gateway.

Answer: A

Question 51

What should the AAR group prefix be?

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.

What should the AAR group prefix be?

A. 9
B. 91
C. none
D. +
E. +1

Answer: C

Question 52

What should the Called Party Transformation Pattern at the U.S. gateway be configured as?

Refer to the exhibit. The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. Assuming the PSTN does not accept globalized numbers with + prefix.

What should the Called Party Transformation Pattern at the U.S. gateway be configured as?
A. +1. with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: None
B. +1. with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: None
C. +408. with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: 1
D. +1408. with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: None
E. +1.408! with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: None
Answer: D

Question 53
What should the TEHO-US route list configuration consist of?
Refer to the exhibit. The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. What should the TEHO-US route list configuration consist of?

A. First route group should point only to the U.K. gateway.
The second route group should point to the U.S. gateway.
B. First route group should be only the local route group.
The second route group should point to the U.S. gateway.
C. First route group should point only to the U.S. gateway.
The second route group should be the local route group.
D. The TEHO-US route list should contain only the local route group.
The globalized configuration means that the appropriate gateway will be selected automatically.
E. The +! route pattern should point directly to the U.S. gateway.
Answer: C
Explanation:
The route group points to one or more gateways and can choose the gateways for call routing based on preference. The route group can serve as a trunk group by directing all calls to the primary device and then using the secondary devices when the primary is unavailable. One or more route lists can point to the same route group.

Question 54
How should the translation pattern be configured?
Refer to the exhibit. The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. To match the US-TEHO pattern +!, how should the translation pattern be configured?
A. 9001.4085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +
B. 9.0014085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +1
C. 900.14085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +1
D. 900.14085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +
E. 001.4085551234 with the Called Party Transformation:
Prefix Digits Outgoing Calls: +
Answer: D
Explanation:
The PSTN access code for the UK is 9, International call code is 001, The international escape character, +, signifies the international access code in a complete E.164 number format.

Question 55
Which process can localize a global E.164 with + prefix calling numbers for inbound calls to an IP phone so that users see the calling number in a local format?
A. Calling number localization is done using translation patterns.
B. Calling number localization is done using route patterns.
C. Calling number localization is done by configuring a calling party transformation CSS at the phone.
D. Calling number localization is done by configuring a calling party transformation CSS at the gateway.
E. Calling number localization is done by configuring the phone directory number in a localized format.
Answer: C

Question 56
When an incoming PSTN call arrives at an MGCP gateway, how does the called number get normalized to an internal directory number in Cisco Unified Communications Manager?
A. Normalization is done by configuring the significant digits for inbound calls on the MGCP gateway.
B. Normalization is done using route patterns.
C. Normalization is done using the gateway incoming called party prefixes based on number type.
D. Normalization is done using the gateway incoming calling party prefixes based on number type.
E. Normalization is achieved by local route group that is assigned to the MGCP gateway.
Answer: A

Question 57
When an incoming PSTN call arrives at an MGCP gateway, how does the calling number get normalized to a global E.164 number with the + prefix in Cisco Unified Communications Manager?
A. Normalization is done using translation patterns.
B. Normalization is done using route patterns.
C. Normalization is done using the gateway incoming called party prefixes based on number type.
D. Normalization is done using the gateway incoming calling party prefixes based on number type.
E. Normalization is achieved by local route group that is assigned to the MGCP gateway.
Answer: D
Explanation:
Configuring calling party normalization alleviates issues with toll bypass where the call is routed to multiple locations over the IP WAN. In addition, it allows Cisco Unified Communications Manager to distinguish the origin of the call to globalize or localize the calling party number for the phone user.

Question 58
Which two features require or may require configuring a SIP trunk?
Which two features require or may require configuring a SIP trunk? (Choose two.)
A. SIP gateway
B. Call Control Discovery between a Cisco Unified Communications Manager and Cisco Unified Communications Manager Express
C. Cisco Device Mobility
D. Cisco Unified Mobility
E. registering a SIP phone
Answer: AB
Explanation:
All protocols require that either a signaling interface (trunk) or a gateway be created to accept and originate calls. Device mobility allows Cisco Unified Communications Manager to determine whether the phone is at its home location or at a roaming location. Cisco Unified Mobility gives users the ability to redirect incoming IP calls from Cisco Unified Communications Manager to different designated phones, such as cellular phones.

Question 59
Which trunks would be most suitable for Connection Y?
Refer to the exhibit. Which trunks would be most suitable for Connection Y?
A. an H.225 trunk (gatekeeper-controlled)
B. intercluster trunk (gatekeeper-controlled)
C. a SIP trunk on the U.S. cluster and an intercluster trunk on the remote cluster
D. intercluster trunk (nongatekeeper-controlled)
E. No extra connections are required. As long as IP connectivity exists, you need only configure a route pattern for each site. The Cisco Unified Communications Manager will automatically forward the calls over the WAN if the destination directory number is not registered locally.
Answer: D

Question 60
Which method can be used to address variable-length dial plans?
A. Overlap sending and receiving.
B. Add a prefix for all calls that are longer than 10-digits long
C. Use nested translation patterns to eliminate inter-digit timeout
D. Use the @macro on the route pattern
E. Use MGCP gateways, which support variable-length dial plans
Answer: A
Explanation:
If the dial plan contains overlapping patterns, Cisco Unified Communications Manager does not route the call until the interdigit timer expires (even if it is possible to dial a sequence of digits to choose a current match). Check this check box to interrupt interdigit timing when Cisco Unified Communications Manager must route a call immediately. By default, the Urgent Priority check box displays as checked. Unless your dial plan contains overlapping patterns or variable length patterns that contain!, Cisco recommends that you do not uncheck the check box.

Question 61
Which three of the following are steps in configuring MGCP Fallback and Cisco Unified SRST?
A. Define the SRST reference for phones in the Device Pool configuration
B. Enable and configure the MGCP fallback and Cisco Unified SRST features on the IOS gateways.
C. Implement a simplified SRST dial plan on the remote-site-gateways to ensure connectivity for remote-site phones in SRST mode.
D. Enable SIP trunking between both remote and hub sites to provide mesh coverage.
E. Define the SRST reference in the configuration on the IP Phones.
F. Enable and configure the MGCP fallback on the IOS gateway but not Cisco Unified SRST since it is enabled automatically.
Answer: ABC

Question 62
Which of the following are two functions that ensure that the telephony capabilities stay operational in the remote location Cisco Unified SRST router?
A. Automatically detecting a failure in the network.
B. Initiating a process to provide call-processing backup redundancy.
C. Notifying the administrator of an issue for manual intervention.
D. Proactively repairing issues in the voice network.
Answer: AB
Which two configurations are needed to implement SRST in Cisco Unified Communications Manager?

A. SRST Gateway setting in Cisco Unified Communications Manager
B. SRST Reference configured in Cisco Unified Communications Manager
C. Device Pool SRST Reference setting
D. Call Manager Group setting
E. Cisco Unified Communications Locations setting

Answer: BC

Question 64
What component acts as a user agent for both ends of a SIP call while Cisco Unified SIP SRST is providing failover during a WAN outage?

A. B2BUA
B. SIP server
C. SIP proxy
D. SIP SRST router
E. SIP registrar

Answer: A

Question 65
Which ability does the Survivable Remote Site Telephony feature provide?

A. a means to allow the local site to continue to send and receive calls in the event of a WAN failure
B. a means to route calls on-net through other sites during high utilization periods
C. a method that allows for backup calls in the event that your gateway fails
D. the ability to force a call out of a certain trunk when the Cisco Unified Communications Manager is being upgraded

Answer: A

Question 66
This is the configuration on the voice gateway:

telephony-service
max-ephones 30
max-dn 60 preference 0
srst mode auto-provision all
srst dn line-mode dual
srst dn template 3
srst ephone description srst fallback auto-provision phone
srst ephone template 5
Which ephone-dn would be expected upon activation of SRST?

A. ephone-dn 1 dual-line
number 7001
description 7001
name 7001
ephone-dn-template 5
This DN is learned from srst fallback ephones
B. ephone-dn 1 dual-line
number 7001
description 7001
name 7001
ephone-dn-template 3
This DN is learned from srst fallback ephones
C. ephone-dn 1
number 7001
description 7001
name 7001
ephone-dn-template 5
This DN is learned from srst fallback ephones
D. ephone-dn 1
description 7001
name 7001
ephone-dn-template 3
This DN is learned from srst fallback ephones

Answer: A

Question 67
Which command can be used to manually send the MGCP gateway to register with the secondary Cisco Unified Communications Manager server?

A. ccm-manager switchover-to-backup
B. mgcp use backup
C. ccm-manager register backup
D. not supported

Answer: A

Question 68
A Cisco Unified Communications Manager cluster is installed in headquarters only.

A Cisco Unified Communications Manager cluster is installed in headquarters only.
How can international calls be blocked while national calls are allowed for branch office Cisco IP Phones during a WAN failure?
A. Configure CSS and partitions in Cisco Unified Communications Manager and apply the CSS and partitions to the SRST ISR.
B. Configure CSS and partitions in the SRST ISR.
C. Configure COR in the SRST ISR.
D. Configure voice translations in the SRST ISR.
Answer: C

Question 69
Which configuration command disables the secondary dial tone on the branch office ISR for users calling from the PSTN into the branch office during a WAN failure?
A. direct-inward-dial
B. voice translation-rule
C. incoming called-number
D. application
Answer: A

Question 70
Which option configures call preservation for H.323-based SRST mode?
A. voice service voip h323 call preserve
B. call preservation not possible with H.323
C. call-manager-fallback preserve-call
D. dial-peer voice 1 voip call preserve
Answer: A

Question 71
On which two factors would the number of IP phones and Directory Numbers that can register to the SRST router depend?
When you configure Cisco Unified Communications Manager, you need to configure the router for Survivable Remote Site Telephony in case the Cisco Unified Communications Manager stops working. On which two factors would the number of IP phones and Directory Numbers that can register to the SRST router depend? (Choose two.)
A. The protocol that is used in Cisco Unified Communications Manager
B. Cisco Unified Communications Manager version
C. Cisco IOS Software version
D. WAN link bandwidth
E. capacity of the Cisco Media Convergence Server
F. router platform
Answer: CF

Question 72
Which remote-site redundancy technology fails over to POTS dial peers from the Cisco Unified Communications Manager dial plan during a WAN failure?
Which remote-site redundancy technology fails over to POTS dial peers from the Cisco Unified Communications Manager dial plan during a WAN failure?
A. MGCP fallback
B. H.323 fallback
C. SCCP fallback
D. SIP fallback
Answer: A

Question 73
Which two configurations provide the best SIP trunk redundancy in Cisco Unified Communications Manager?
Which two configurations provide the best SIP trunk redundancy in Cisco Unified Communications Manager? (Choose two.)
A. Configure all SIP trunks with DNS SRV
B. Configure all SIP trunks with Cisco Unified Border Element
C. Configure all SIP trunks to point to a SIP gateway
D. Configure SIP trunks to be members of route groups and route lists
E. Configure all SIP trunks to allow TCP ports 5060
F. Configure all SIP trunks to point to a gatekeeper through SIP to H.323 gateway
Answer: AD
Explanation:
For SIP trunks, Cisco Unified Communications Manager supports up to 16 IP addresses for each DNS SRV and up to 10 IP addresses for each DNS host name. The order of the IP addresses depends on the DNS response and may be identical in each DNS query. The OPTIONS request may go to a different set of remote destinations each time if a DNS SRV record (configured on the SIP trunk) resolves to more than 16 IP addresses, or if a host name (configured on the SIP trunk) resolves to more than 10 IP addresses. Thus, the status of a SIP trunk may change because of a change in the way a DNS query gets resolved, not because of any change in the status of any of the remote destinations.

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Question 75
How to enable IP phones to establish calls to the PSTN when they have registered with the gateway?
You are the Cisco Unified Communications Manager in Certpaper.com.
You use a remote site MGCP gateway to provide redundancy when connectivity to the central Cisco Unified Communications Manager cluster is lost.
How to enable IP phones to establish calls to the PSTN when they have registered with the gateway? (Choose three.)
A. POTS dial peers must be added to the gateway to route calls from the IP phones to the PSTN.
B. The default service must be enabled globally.
C. The command ccm-manager mgcp-fallback must be configured.
D. COR needs to be configured to disallow outbound calls.
Answer: ABC

Question 76
Drag and Drop Question
Drag and Drop Question
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.

Answer:
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.

Question 77
What is the fastest way for an engineer to test the implementation of SRST in a production environment?
A. Shut down the Cisco Unified Communications Manager Servers.
B. Shut down the switch ports connected to the Cisco Unified Communications Manager Servers.
C. Use a remote site MGCP gateway to provide redundancy when connectivity to the central Cisco Unified Communications Manager cluster is lost.
Answer: B

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C. Add a null route to the publisher Cisco Unified Communications Manager at the remote router. Remove the null route when the operation is verified.
D. Unplug the IP phones from their switch ports.
E. Verification is not needed.

Question 78
Which two locations are the best locations that an end user can use to determine if an IP phone is working in SRST mode?
Which two locations are the best locations that an end user can use to determine if an IP phone is working in SRST mode? (Choose two.)
A. Cisco Unified Communications Manager Administration
B. IP phone display
C. Cisco Unified SRST Router
D. Cisco Unified MGCP Fallback Router
E. physical IP phone settings

Answer: BE
Explanation: IP Phone display and Physical phone IP settings are two locations were an end user can determine if an IP phone is working in SRST mode.

Question 79
To preserve analog calls in an MGCP switchback event, which three commands must be configured in the MGCP fallback router?
To preserve analog calls in an MGCP switchback event, which three commands must be configured in the MGCP fallback router? (Choose three.)
A. h323
B. mgcp-switchback-graceful
C. voice service voip
D. mgcp-graceful
E. preserve-h323
F. no h225 timeout keepalive

Answer: ACF

Question 80
When using Cisco Unified Communications Manager Express in SRST mode, how many multicast music on hold streams can be utilized by the system at any given time?
When using Cisco Unified Communications Manager Express in SRST mode, how many multicast music on hold streams can be utilized by the system at any given time?
A. 3
B. 6
C. 2
D. 4
E. 1
F. 5

Answer: B

Question 81
While operating in SRST, what is needed to route calls outside of the remote site location to the PSTN?
While operating in SRST, what is needed to route calls outside of the remote site location to the PSTN?
A. SIP trunk
B. CallManager route patterns
C. translation patterns
D. POTS dial peers
E. VOIP dial peers

Answer: D
Explanation: in time of srst configuration on router, please configure a dial-peer so that call flow in SRST mode.

Question 82
While configuring Call Survivability in Cisco Unified Communications Manager, what step is mandatory to reach remote sites while in SRST mode?
While configuring Call Survivability in Cisco Unified Communications Manager, what step is mandatory to reach remote sites while in SRST mode?
A. Enable Cisco Remote Site Reachability.
B. Configure CFUR.
C. Enable the SRST checkbox in the MGCP gateway.
D. Configure the H.323 gateway for SRST in Cisco Unified Communications Manager.
E. Enable the Failover Service parameter.

Answer: B
Explanation: Call Forward Unregistered (CFUR) functionality provides the automated rerouting of calls through the PSTN when an endpoint is considered unregistered due to a remote WAN link failure.

Question 83
In what Cisco solution is Simple Network-Enabled Auto Provision technology used?
In what Cisco solution is Simple Network-Enabled Auto Provision technology used?
A. Cisco Unified Gateway Duplication
B. Cisco Unified CallManager Redundancy
C. Cisco Unified SRST
D. Cisco Unified Call Survivability

Answer: C
Explanation:
When the system automatically detects a failure, Cisco Unified SRST uses Simple Network Auto-Provisioning (SNAP) technology to auto-configure a branch office router to provide call processing for the Cisco Unified IP phones that are registered with the router.

**Question 84**
What value should be entered into the gatekeeper to support this bandwidth?

Video calls using 384 kbps need to be supported across a gatekeeper-controlled trunk. What value should be entered into the gatekeeper to support this bandwidth?

A. 768 kbps  
B. 384 kbps  
C. 512 kbps  
D. 192 kbps

Answer: B  
Explanation:  
A 384-kb/s video call may comprise G.711 at 64 kb/s (for audio) plus 320 kb/s (for video). This sum does not include overhead. If the audio codec for a video call is G.729 (at 24 kb/s), the video rate increases to maintain a total bandwidth of 384 kb/s.

**Question 85**
What happens when the fourth call is placed from HO to BR?

Refer to the exhibit. Locations-based CAC has been configured between HQ and the BR site. Assume that the priority queue has been provisioned correctly for three G.729 calls. What happens when the fourth call is placed from HO to BR?

A. The call will get through via the WAN, but it will experience poor audio quality.  
B. The call will fail.  
C. The call will be queued until one of the existing calls drop.  
D. The call will get through without any issues.

Answer: B

**Question 86**
Which two statements about symmetric encryption are true?

Which two statements about symmetric encryption are true? (Choose two.)

A. With symmetric encryption, the encryption key equals the decryption key.  
B. Symmetric encryption is commonly used to sign asymmetric keys.  
C. Symmetric encryption is a good choice for real-time encryption of bulk data.  
D. Symmetric encryption uses asymmetric keys.

Answer: AC  
Explanation:

There are two basic techniques for encrypting information: symmetric encryption (also called secret key encryption) and asymmetric encryption (also called public key encryption.) In symmetric key As long as both sender and recipient know the secret key, they can encrypt and decrypt all messages that use this key. A public key is made freely available to anyone who might want to send you a message. A second, private key is kept secret, so that only you know it.

**Question 87**
Enabling authentication and encryption for CTI, JTAPI, and TAPI applications requires which two tasks?

Enabling authentication and encryption for CTI, JTAPI, and TAPI applications requires which two tasks? (Choose two.)

A. Enter the encryption key into the application.  
B. Set up an IPsec association between the application and Cisco Unified CallManager.  
C. Configure related security parameters in the CTI, JTAPI, and TAPI application.  
D. Add the application user or end users to the Standard CTI Secure Connection user group, Standard CTI Allow Reception of SRTP Key Material user group, and Standard CTI Enabled user group.

Answer: CD  
Explanation:  
You must also add the application users or the end users to the Standard CTI Secure Connection user group in Cisco Unified Communications Manager.
Administration to enable TLS for the application. After you add the user to this group and install the certificate, the application ensures that the user connects via the TLS port.

Question 88
Which statement is not true about GARP?
A. GARP attacks require access to the target LAN or VLAN.
B. GARP can be used for a man-in-the-middle attack.
C. GARP is normally used for HSRP.
D. GARP can be disabled at Cisco IP phones.
Answer: C
Explanation:
GARP (Gratuitous ARP) announce the presence of IP Phone on the network.

Question 89
An update of the configuration using the Cisco CTL client not needed when _______.
A. a Cisco Unified CallManager has been removed
B. an LSC of the IP phone is upgraded
C. a security token is added to the system
D. an IP address of the Cisco TFTP server has been changed
Answer: B

Question 90
Which statement about enrollment in the IP telephony PKI is true?
A. CAPF enrollment supports the use of authentication strings.
B. The CAPF itself has to enroll with the Cisco CTL client.
C. LSCs are issued by the Cisco CTL client or by the CAPF.
D. MICs are issued by the CAPF itself or by an external CA.
Answer: A

Question 91
What command is used to map internal extensions to the corresponding E.164 PSTN number when using Cisco Unified Communications Manager Express in SRST mode?
A. ephone-dn
B. dialplan-pattern
C. number
D. number-e.164
E. ephone-transnumber
Answer: B

Question 92
Which CAC configuration on a gatekeeper restricts to 10 G.711 audio calls?
A. Use the command bandwidth 10.
B. Use the command bandwidth 1280.
C. Use the command bandwidth 160.
D. Use the command bandwidth session 10.
Answer: B

Question 93
For which VoIP protocol does a gatekeeper provide address translation and control access?
A. H.323
B. SIP
C. Skinny
D. H.248
Answer: A

Question 94
Which statement about the function of a gatekeeper is true?
A. A gatekeeper improves call routing between servers within a single Cisco Unified Communications Manager cluster.
B. A gatekeeper can replace the dial plan of a Cisco Unified Communications Manager cluster.
C. A gatekeeper can simplify the dial plan between many different Cisco Unified Communications Manager clusters.
D. Gatekeepers can be implemented to deploy RSVP-based CAC.
Answer: C

Question 95
Which statement about the route pattern is true?
The following exhibit shows config 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?

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 91201[2-9]XXXXXX
D. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot
E. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot and Prefix 9

Answer: A

Question 96
Which device is needed to integrate H.320 into the Cisco video solution?
A. video gateway
B. MGCP gateway
C. H.323 gatekeeper
D. MCU

Answer: C
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.

Question 97
Which two configuration changes can correct this issue? Refer to the exhibit. 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: BE
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.

Question 98
How should the HQ Cisco Unified Communications Manager be configured for calls to 3XXX to be sent to the gatekeeper at 10161 with PSTN backups? Refer to the exhibit. 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 10161 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.

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 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.

D. 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.

Answer: A

Question 99
Which statement about SIP precondition is most correct?
A. When configuring SIP precondition, the SIP trunk must have access to an RSVP agent.
B. When configuring SIP precondition, the IP phones must have access to an RSVP agent.
C. When configuring SIP precondition, the IP phones and SIP trunk must have access to an RSVP agent.
D. RSVP agents are only required for the IP phones. SIP trunks require RSVP agents only when fall back to local RSVP is configured.
E. SIP trunk will always require RSVP agents regardless of what RSVP type is configured.

Answer: D

Question 100
How is a SIP trunk in Cisco Unified Communications Manager configured for SIP precondition?
A. The configuration is done by selecting a SIP precondition trunk for trunk type.
B. The configuration is automatically selected when RSVP is enabled for the location assigned to the trunk.
C. SIP precondition is configured by selecting E2E for RSVP over SIP on the default SIP profile assigned to the SIP trunk.
D. 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.

Answer: D

Question 101
What is the default DSCP/PHB for TelePresence video conferencing packets in Cisco Unified Communications Manager?
A. EF/46
B. CS6/48
C. AF41/34
D. CS3/24
E. CS4/32

Answer: E

Question 102
What is the correct DSCP value to use when configuring a class map in a Cisco IOS router?
Refer to the exhibit. 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?
Question 103
Which trunk type is implemented in this network?
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. intercluster trunk without gatekeeper control
C. SIP trunk
D. h225 trunk
Answer: B

Question 104
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 first match that is attempted?
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 first match that is attempted?
A. the full URI, including the domain portion
B. the destination alias, without the domain portion
C. the E.164 number that is assigned to the Cisco TelePresence EX90
D. the directory number that is assigned to the Cisco TelePresence EX90
Answer: B

Question 105
When you use the Query wizard to configure the trace and log central feature to collect install logs, if you have servers in a cluster in a different time zone, which time is used?
When you use the Query wizard to configure the trace and log central feature to collect install logs, if you have servers in a cluster in a different time zone, which time is used?
A. TLC adjusts the time change appropriately.
B. TLC uses its local time for all systems.
C. TLC queries for the time zone as part of configuration.
D. TLC produces an error and must be run remotely.
Answer: A

Question 106
Which statement is true when considering a Cisco VoIP environment for regional configuration?
Which statement is true when considering a Cisco VoIP environment for regional configuration?
A. G.711 requires 128K of bandwidth per call.
B. G.729 requires 24K of bandwidth per call.
C. The default codec does not matter if you have defined a hardware MTP in your Cisco Unified Communications Manager environment.
D. To deploy a Cisco H.323 gatekeeper, you must configure MTP resources on the gatekeeper and only use G.711 between regions.
Answer: B

Question 107
When implementing a Media Termination Point, what determines the number of sessions that is supported on each DSP?
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 size of the cluster that is being designed
C. the number of full-duplex media streams
D. the Cisco Unified Communications Manager node setting
Answer: A

Question 108
Which module is the minimum PVDM3 module needed to support video transcoding?
Which module is the minimum PVDM3 module needed to support video transcoding?

A. PVDM3-32
B. PVDM3-64
C. PVDM3-128
D. PVDM3-192

Answer: C

Question 109
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: C

Question 110
Which two bandwidth management parameters are available during the configuration of Cisco Unified Communications Manager regions?

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

Answer: BD

Question 111
When configuring Cisco Unified Mobility, which parameter defines the access control for a call that reaches out to a remote destination?

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

Answer: C

Question 112
How many Cisco Unified Mobility destinations can be configured per user?

A. 1
B. 10
C. 4
D. 6

Answer: B

Question 113
Which three steps are required when configuring extension mobility in Cisco Unified Communications Manager?

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

Answer: AEF

Question 114
What happens when a user logs in using the Cisco Extension Mobility Service on a device for which the user has no user device profile?

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

Answer: B

Question 115
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?

Refer to the exhibit. If an IP phone in San Jose roams to New York, which two IP phone settings will be modified by Device Mobility so that the phone can place and receive calls in New York? (Choose two.)
A. The physical locations are not different, so the configuration of the phone is not modified.
B. The physical locations are different, so the roaming-sensitive parameters of the roaming device pool are applied.
C. The device mobility groups are the same, so the Device Mobility-related settings are applied in addition to the roaming-sensitive parameters.
D. The Device Mobility information is associated with one or more device pools other than the home device pool of the phone, so one of the associated device pools is chosen based on a round-robin load-sharing algorithm.
E. The Device Mobility information is associated with the home device pool of the phone, so the phone is considered to be in its home location. Device Mobility will reconfigure the roaming-sensitive settings of the phone.

Answer: BC

Question 116
Which option is known as the location attribute that the global dialplan replication uses to advertise its dial plan information?

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

Answer: C

Question 117
Which functionality does ILS use to link all hub clusters in an ILS network?

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

Answer: B

Question 118
If you want to delete a SAF-enabled trunk from Cisco Unified Communications Manager Administration, what must you do first?

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

Answer: A

Question 119
Which statement about the SAF Client Control is correct?

A. The SAF Client Control is a configurable inherent component of Cisco Unified Communications Manager.
B. The SAF Client Control is a non-configurable inherent component of Cisco Unified Communications Manager.
C. The SAF Client Control is a non-configurable inherent component of the Cisco IOS Routers.
D. The SAF Client Control is a configurable inherent component of the Cisco IOS Routers.

Answer: B

Question 120
When using SAF, how do you prevent multiple nodes in a cluster from showing up in the Show Advance section of the SAF Forwarder configuration?

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

Answer: B

Question 121
Which code snippet is required for SAF to be initialized?

http://www.aoowe.com/practice-300-075-3138.html
A. Service Family  
B. External-Client  
C. router eigrp  
D. topology base  

Answer: C

**Question 122**

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

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

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

Answer: B

**Question 123**

Where do you configure the +E.164 number that is advertised globally via ILS?

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

A. ILS configuration under Advanced Features  
B. +E.164 alternate number under Directory Number Settings  
C. Device Information under Phone Configuration  
D. Route Pattern under Route/Hunt  

Answer: B

**Question 124**

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

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

A. Cisco Unified Communications Manager uses the ILS to support intercluster URI dialing.  
B. ILS contains an optional directory URI replication feature that allows the clusters in an ILS network to replicate their directory URIs to the other clusters in the ILS network.  
C. Directory URI replication does not need to be enabled individually for each cluster.  
D. To enable URI replication in a cluster, check the Exchange Directory URIs with Remote Clusters check box that appears in the SIP trunk configuration menu.  
E. If the ILS and directory URI replication feature is disabled on a cluster, this cluster still accepts ILS advertisements and directory URIs from other neighbor clusters; it just does not advertise its local directory URIs.  

Answer: AB

**Question 125**

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

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

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

Answer: BD

**Question 126**

Why did the users not realize the WAN was down and prevented access to their voicemail?

Company X has a Cisco Unified Communications Manager cluster and a Cisco Unity Connection cluster at its head office and implemented SRST for its branch offices. One Monday at 2:00 pm, the WAN connection to a branch office failed and stayed down for 45 minutes. That day the help desk received several calls from the branch saying their voicemail was not working but they were able to make and receive calls. Why did the users not realize the WAN was down and prevented access to their voicemail?

A. All the phones should have started ringing the instant the WAN connection failed to signal the start of SRST mode.  
B. All calls should have dropped when the WAN failed so users would be instantly aware.  
C. Although the phones were still working, the users should have noticed that the phone displays said “SRST Fallback Active”.  
D. The voice administrators at the head office did not call the users to notify them.  

Answer: C

**Question 127**

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

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

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

Answer: ABC

**Question 128**

Which option is a valid test scenario to verify that Cisco Unified Communications Manager failover works and endpoints register with the backup call manager? (Choose one.)

Which option is a valid test scenario to verify that Cisco Unified Communications Manager failover works and endpoints register with the backup call manager? (Choose one.)

A. During a predetermined maintenance window, stop the Cisco IP Phone Services service on the primary Unified CM. Devices should reregister with the backup Unified CM in the Cisco CallManager Group.  
B. During a predetermined maintenance window, stop the Unified CM service on the Publisher. Devices should reregister with the backup publisher in the Cisco CallManager Group.  

Answer: A
C. During a predetermined maintenance window, stop the TFTP service on the primary call manager. Devices should reregister with the backup Unified CM in the Cisco CallManager Group.
D. During a predetermined maintenance window, stop the Unified CM service on the primary call manager. Devices should reregister with the backup Unified CM in the CallManager Group.

Answer: D

Question 129
How long is the default keepalive period for SRST in Cisco IOS?
How long is the default keepalive period for SRST in Cisco IOS?
A. 45 sec
B. 30 sec
C. 60 sec
D. 120 sec

Answer: B

Question 130
When configuring Cisco Unified Survivable Remote Site Telephony, which CLI command enables this feature on the router?

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

Answer: A

Question 131
Which three CLI commands are used when configuring H.323 call survivability for all calls? (Choose three.)

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

Answer: ACD

Question 132
When considering Cisco Unified Communications Manager failover, how many backup servers can be configured in a Cisco Unified Communications Manager Group?
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: C

Question 133
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?
Which two options enable routers to provide basic call handling support for Cisco Unified IP Phones if they lose connection to all Cisco Unified Communications Manager systems? (Choose two.)
A. SCCP fallback
B. Cisco Unified Survivable Remote Site Telephony
C. Cisco Unified Communications Manager Express
D. MGCP fallback
E. Cisco Unified Communications Manager Express in SRST mode

Answer: BE

Question 134
Which two statements about remote survivability are true? (Choose two.)
A. SRST supports more Cisco IP Phones than Cisco Unified Communications Manager Express in SRST mode.
B. Cisco Unified Communications Manager Express in SRST mode supports more Cisco IP Phones than SRST.
C. MGCP fallback is required for ISDN call preservation.
D. MGCP fallback functions with SRST.

Answer: AD

Question 135
What is the correct value to use for the “DSCP for TelePresence Calls” Cisco CallManager service parameter?
Refer to the exhibit. What is the correct value to use for the “DSCP for TelePresence Calls” Cisco CallManager service parameter?
Question 136
Where can you change the clusterwide DSCP setting for Cisco Unified Communications Manager?
A. enterprise parameters
B. service parameters
C. enterprise phone configuration
D. Ethernet configuration

Answer: B

Question 137
Where does the administrator go to create a default profile?
Company X wants to implement RSVP-based Call Admission Control and move away from the current location-based configuration.

A. System > Call Manager > Clusterwide > Service Parameters > RSVP
B. System > Service Parameters > RSVP
C. System > Service Parameters > Call Manager > Clusterwide parameters > RSVP
D. on each MGCP gateway at all remote locations

Answer: C

Question 138
Which statement about configuring the Cisco VCS Control and Cisco VCS Expressway is true?
Which statement about configuring the Cisco VCS Control and Cisco VCS Expressway is true?
A. You do not need to configure search rules for traversal calls.
B. You need to configure the firewall to allow communication from the Cisco VCS Expressway to the Cisco VCS Control.
C. The username on the Cisco VCS Control and Cisco VCS Expressway are local and do not need to match.
D. The Cisco VCS Expressway is the Traversal Server.

Answer: D

Question 139
Where is this option enabled?
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.

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

Answer: B
Question 140
What must the administrator configure on the firewall to stabilize this deployment?
A new administrator at Company X has deployed a VCS Control on the LAN and VCS Expressway in the DMZ to facilitate VPN-less SIP calls with users outside of the network. However, the users report that calls via the VCS are erratic and not very consistent. What must the administrator configure on the firewall to stabilize this deployment?
A. The VCS Control should not be on the LAN, but it must be located in the DMZ with the Expressway.
B. The firewall at Company X must have all SIP ALG functions disabled.
C. The firewall at Company X requires a rule to allow all traffic from the DMZ to pass to the same network that the VCS Control is on.
D. A TMS server is needed to allow the firewall traversal to occur between the VCS Expressway and the VCS Control servers.
Answer: B

Question 141
Which two additional steps are needed to complete this deployment?
Company X currently uses a Cisco Unified Communications Manager, which has been configured for IP desk phones and Jabber soft phones. Users report however that whenever they are out of the office, a VPN must be set up before their Jabber client can be used. The administrator for Company X has deployed a Collaboration Expressway server at the edge of the network in an attempt to remove the need for VPN when doing voice. However, devices outside cannot register. Which two additional steps are needed to complete this deployment? (Choose two.)
A. A SIP trunk has to be set up between the Expressway-C and Cisco UCM.
B. An additional interface must be enabled on the Cisco UCM and placed in the same subnet at the Expressway.
C. The customer firewall must be configured with any rule for the IP address of the external Jabber client.
D. The Expressway server needs a neighbor zone created that points to Cisco UCM.
E. Jabber cannot connect to Cisco UCM unless it is on the same network or a VPN is set up from outside.
Answer: AD

Question 142
Which DNS SRV Records must be configured on the external DNS server in a mobile remote access scenario with Cisco Expressway?
A. _collab-edge._tls.example.com
B. _collab-edge._udp.example.com
C. _cisco-uds._tcp.example.com
D. _cuplogin._tcp.example.com
Answer: A

Question 143
When you connect a Cisco VCS Control to Cisco Unified Communications Manager by using a SIP trunk, which mechanism do you use to verify that the trunk has an active connection?
A. OPTIONS ping
B. DNS tracing
C. Continuous ping
D. Dynamic DNS
Answer: A

Question 144
To do this, which format must you use in the Search Rule?
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. [email protected]
B. IP Address (192.168.100.0)
C. [email protected]
D. [email protected] Address (192.168.100.0)
Answer: A

Question 145
Which zone will the VCS Control use to route calls to the VCS Expressway?
Which zone will the VCS Control use to route calls to the VCS Expressway?
A. neighbor zone
B. DNS zone
C. traversal client zone
D. ENUM zone
Answer: C

Question 146
What does a video endpoint use to register with the VCS Control?
A video endpoint is configured with SIP only.
What does a video endpoint use to register with the VCS Control?
A. IP address
B. SIP URI
C. MAC address
D. system name
Answer: B

Question 147
Which two options should be used to create a secure traversal zone between the Expressway-C and Expressway-E? (Choose two.)
A. Expressway-C and Expressway-E must trust each other’s server certificate.
B. One Cisco Unified Communications traversal zone for H.323 and SIP connections.
C. A separate pair of traversal zones must be configured if an H.323 connection is required and Interworking is disabled.
D. Enable username and password authentication verification on Expressway-E.
E. Create a set of username and password on each of the Expressway-C and Expressway-E to authenticate the neighboring peer.

Answer: AC

Question 148
Which command displays the detailed configuration of all the Cisco Unified IP phones, voice ports, and dial peers of the Cisco Unified SRST router?
A. show call-manager-fallback all
B. show dial-peer voice summary
C. show ephone summary
D. show voice port summary

Answer: A

Question 149
Where should the administrator look next to confirm that the correct DSCP markings are being set?
The administrator at Company X is getting user reports of inconsistent quality on video calls between endpoints registered to Cisco Unified Communications Manager. The administrator runs a wire trace while a video call is taking place and sees that the packets are not set to AF41 for desktop video as they should be.
Where should the administrator look next to confirm that the correct DSCP markings are being set?
A. on the MGCP router at the edge of both networks
B. the service parameters in the VCS Control
C. the QoS service parameter in Cisco Unified Communications Manager
D. on the actual Cisco phone itself because the DSCP setting is not part of its configuration file downloaded at registration.
E. The setting cannot be changed for video endpoints that are registered to Cisco Unified Communications Manager, but only when they are registered to the VCS Control.

Answer: C

Question 150
Which feature allows you to specify which endpoints ring when someone calls a user on a specific destination ID?
A. FindME
B. Extension Mobility
C. Speech Connect
D. Single Number Reach

Answer: A

Question 151
What is the default DSCP/PHB for video conferencing packets in Cisco Unified Communications Manager?
A. EF/46
B. CS6/48
C. AF41/34
D. CS3/24

Answer: C

Question 152
Which two Cisco Extension Mobility attributes are available in the user device profile? (Choose two.)
A. regions
B. description
C. phone button template
D. NTP information

Answer: BC

Question 153
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. unregistered
C. unknown
D. new device

Answer: A

Question 154
During device failover to the secondary Cisco Unified Communications Manager server, how does the phone recognize that the primary server is back?
During device failover to the secondary Cisco Unified Communications Manager server, how does the phone recognize that the primary server is back?
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. When the primary server goes online, it sends out an “ALIVE” message via broadcast so that the phones re-register.
C. The phones never re-register with the primary server until the active (secondary) one goes down.
D. The phone sends keepalive messages to the primary server frequently and when it succeeds, the phone re-registers with it.

Answer: D

**Question 155**
Which system configuration is used to set audio codecs?

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

Answer: A

**Question 156**
Which three statements about configuring an encrypted trunk between Cisco TelePresence Video Communication Server and Cisco Unified Communications Manager are true? (Choose three.)

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

Answer: ACE

**Question 157**
When an IP phone moves from one device mobility group to another, which two configuration components are not changed? Assume that local route groups are configured. When an IP phone moves from one device mobility group to another, which two configuration components are not changed? (Choose two.)

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

Answer: BE

Explanation:  
Although the phone may have moved from one subnet to another, the physical location and associated services have not changed.

**Question 158**
Which two are gatekeeper-controlled trunk options that support gatekeeper call administration control? (Choose two.)

A. H.323  
B. H.245  
C. H.225  
D. intercluster  
E. intracluster

Answer: CD

**Question 159**
Which connection method is best for these two new customers? 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. intercluster trunk (gatekeeper-controlled)  
C. SIP trunk  
D. intercluster trunk (gatekeeperscontrolled)

Answer: D

**Question 160**
Which device is used to connect to the H.323 gatekeeper?

A. H.323 gateway  
B. SIP trunk  
C. H.323 trunk  
D. MGCP gateway

Answer: C

**Question 161**
Which statement best describes globalized call routing in Cisco Unified Communications Manager?

A. All incoming calling numbers on the phones are displayed as an E.164 with the + prefix.  
B. Call routing is based on numbers represented as an E.164 with the + prefix format.  
C. All called numbers sent out to the PSTN are in E.164 with the + prefix format.  
D. The CSS of all phones contain partitions assigned to route patterns that are in global format.
All phone directory numbers are configured as an E.164 with the + prefix.

Answer: B

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

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

Answer: C

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

A. 1
B. 2
C. 5
D. 10
E. 15
F. unlimited

Answer: A

What does the Cisco Unified Communications Manager “LocationOutOfResources” counter indicate?

The network administrator has been investigating bandwidth issues between the central office and remote sites where location-based CAC is implemented. What does the Cisco Unified Communications Manager “LocationOutOfResources” counter indicate?

A. This counter represents the total number of times that a call on a particular Cisco Unified Communications Manager through the location failed due to lack of bandwidth.
B. This counter represents the total number of times that a call through locations failed due to the lack of bandwidth.
C. This counter represents the total number of failed video-stream requests (most likely due to lack of bandwidth) in the location where the person who initiated the video conference resides.
D. This counter represents the total number of times since the last restart of the Cisco IP Voice Streaming Application that a Cisco Unified Communications Manager connection was lost.

Answer: B

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

A. The Advanced Networking or Dual Network Interfaces option key has been installed.
B. VCS rewrites the Layer 3 source address of outbound SIP and H.323 packets to 209.165.200.230.
C. VCS applies 209.165.200.230 to outbound SIP and H.323 payload messages.
D. With static NAT enabled on the LAN2 interface, VCS applies 209.165.200.230 to outbound H.323 and SIP payload traffic exiting the LAN1 interface.

Answer: AC

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

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

Answer: B

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

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

Answer: D

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

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

Answer: D
Question 169
Which statement is true when device mobility mode is enabled or disabled in the Phone Configuration window?
A. The device mobility mode phone settings take precedence over the service parameter settings.
B. The service parameter settings take precedence over the device mobility mode phone settings.
C. The combined service parameter settings and the device mobility mode phone settings will be used.
D. The default settings will be used due to the conflicts.
Answer: A

Question 170
When device mobility mode is enabled or disabled for a cluster, to which does the cluster setting apply?
A. all phones in the cluster that support device mobility
B. all phones in the cluster that subscribed to device mobility
C. mobile phones in the cluster that support device mobility
D. mobile phones in the cluster that are in default mode
Answer: A

Question 171
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. Device Mobility settings modify the AAR group and the AAR CSS, the roaming-sensitive settings modify the device CSS.
D. 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.
Answer: D

Question 172
Which two statements about international multisite dial plans are true?
A. TEHO is legal in all countries.
B. TEHO is legal in most countries, but illegal in others.
C. TEHO reduces WAN utilization.
D. TEHO utilizes WAN links.
Answer: BD

Question 173
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. 15 when not using a local route group
B. 6 when using a local route group
C. 5 when using a local route group
D. 10 when not using a local route group
Answer: B

Question 174
How are Cisco IP Phones directly configured to utilize local route groups?
A. with Cisco Unified Communications Manager device pools
B. with Cisco Unified Communications Manager CSS and partitions
C. with Cisco Unified Communications Manager regions
D. with Cisco Unified Communications Manager AAR
Answer: A

Question 175
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: A

Question 176
What are two important considerations when implementing TEHO to reduce long-distance cost?
A. on-net calling patterns
B. E911 calling
C. number of route patterns
D. caller ID
Answer: A
Question 177
Which technologies provide remote-site redundancy for Cisco IP Phones during a WAN failure?
A. SRST and MGCP fallback
B. SRST and TEHO
C. TEHO and MGCP fallback
D. SRST and AAR

Answer: A

Question 178
Which statement is true regarding the configuration of SAF Forwarder?
A. In a multisite dial plan, SAF Forwarders may exist in multiple autonomous systems.
B. The client label that is configured in Cisco Unified Communications Manager must match the configuration on the SAF Forwarder router.
C. There should not be multiple nodes of Cisco Unified Communications Manager clusters acting as SAF clients.
D. The destination IP address must match the loopback address of the SAF router.

Answer: A

Question 179
Which configuration change is needed to enable NANP international dialing during MGCP fallback?
A. Change the dial peer to dial-peer voice 901 voip.
B. Change the dial peer to dial-peer voice 9011 pots.
C. Add the command prefix 011 to the dial peer.
D. Add the command prefix 9011 to the dial peer.

Answer: C

Question 180
Which two of the following configurations if applied to the router, would remedy this situation?
Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 9971 Video IP Phone. The Cisco VCS and TMS control the the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows DP.

Answer: BD
SRST:

**Calling Search Space (1 - 2 of 2)**

Find Calling Search Space where CSS Name begins with

<table>
<thead>
<tr>
<th>CSS Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>A11-Devices</td>
</tr>
<tr>
<td>All-Devices</td>
</tr>
</tbody>
</table>

| Add New | Select All | Clear All | Delete Selected |

---

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

voice service voip
sip
    bind control source-interface GigabitEthernet0/0/0.130
    bind media source-interface GigabitEthernet0/0/0.130
registrar server
!
voice register global
    max-dn 1
    max-pool 1
!
voice register pool 1
    id network 10.1.130.0 mask 255.255.255.255
call-manager-fallback
    ip source-address 10.1.130.1
    max-dn 1 dual-line
    max-ephones 1
At the HQ cluster, the CFUR for the directory number BR2 phone (+442288224001) has been configured:

- **Forward Unregistered Internal Destination:** +442288224001
- **Forward Unregistered Internal Calling Search Space**
- **Forward Unregistered External Destination:** +442288224001
- **Forward Unregistered External Calling Search Space**

After configuring the CFUR for the directory number that is applied to BR2 phone (+442288224001), the calls fail from the PSTN. Which two of the following configurations if applied to the router, would remedy this situation? (Choose two.)

A. dial-peer voice 1 pots
   incoming called-number 228822….
   direct-inward-dial
   port 0/0/0:15

B. dial-peer voice 1 pots
   incoming called-number 228822….
   direct-inward-dial
   port 0/0/0:13

C. voice translation-rule 1
   rule 1/r228821…./S/+44&/
   exit ! voice translation-profile pstn-in
   translate called 1
   !
   voice-port 0/0/0:15
   translation-profile incoming pstn-in

D. voice translation-rule 1
   rule 1/r228822…./S/+44&/
   exit ! voice translation-profile pstn-in
   translate called 1
   !
   voice-port 0/0/0:15

E. The router does not need to be configured.

**Answer:** AD

**Question 181**
Which two steps must the administrator take to add the SIP trunk?

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

A. Set up a SIP trunk on Cisco UCM with the option Device-Trunk with destination address of the VCS server.
B. Set up a subzone on Cisco UCM with the peer address to the VCS cluster.
C. Set up a neighbor zone on the VCS server with the location of Cisco UCM using the menu option VCS Configuration > Zones > zone.
D. Set up a SIP trunk on the VCS server with the destination address of the Cisco UCM and Transport set to TCP.
E. Set up a traversal subzone on the VCS server to allow endpoints that are registered on Cisco UCM to communicate.

**Answer:** AC

**Question 182**
How many active gatekeepers can you define in a local zone?

A. 1  
B. 2  
C. 5  
D. 10  
E. 15  
F. unlimited

**Answer:** A

**Question 183**
Which task must you perform before deleting a transcoder?
Which task must you perform before deleting a transcoder?
A. Delete the dependency records.
B. Unassign it from a media resource group.
C. Use the Reset option.
D. Remove the device pool.
E. Remove the subunit.
F. Delete the common device configuration.

Answer: B

Question 184
With Media Gateway Control Protocol configuration on the voice gateway, which three types of messages are involved in the call flow between the call agent and the voice gateway? (Choose three.)
A. audit endpoint
B. modify endpoint
C. create connection
D. delete notification
E. restart in progress
F. end connections

Answer: ACE

Question 185
Which three configuration tasks need to be completed on the router to support the registration of Cisco Jabber clients?

Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phones. The Cisco VCS and TMS control the Cisco Telepresence Conductor, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.

DNS Server:
Device Pool:

http://www.aoowe.com/practice-300-075-3138.html
ip host _collab-edge._tls.hq.cisco.com SRV 1 1 8443 vcsxe8x-hq.hq.collab10x.cisco.com
ip host vcsxe8x-hq.hq.collab10x.cisco.com 10.1.5.20

Speed Dial:
<table>
<thead>
<tr>
<th>Phone</th>
<th>Speed Dial Button</th>
<th>Speed Dial Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td>HQ phone 1</td>
<td>1</td>
<td>hq2@ci</td>
</tr>
<tr>
<td>HQ phone 1</td>
<td>2</td>
<td>br1@ci</td>
</tr>
<tr>
<td>HQ phone 1</td>
<td>3</td>
<td>bb@ci</td>
</tr>
<tr>
<td>HQ phone 2</td>
<td>1</td>
<td>hq1@ci</td>
</tr>
<tr>
<td>HQ phone 2</td>
<td>2</td>
<td>br1@ci</td>
</tr>
<tr>
<td>HQ phone 2</td>
<td>3</td>
<td>bb@ci</td>
</tr>
<tr>
<td>BR1 phone 1</td>
<td>1</td>
<td>hq1@ci</td>
</tr>
<tr>
<td>BR1 phone 1</td>
<td>2</td>
<td>hq2@ci</td>
</tr>
<tr>
<td>BR1 phone 1</td>
<td>3</td>
<td>bb@ci</td>
</tr>
<tr>
<td>BB phone</td>
<td>1</td>
<td>hq1@ci</td>
</tr>
<tr>
<td>BB phone</td>
<td>2</td>
<td>hq2@ci</td>
</tr>
</tbody>
</table>
Which three configuration tasks need to be completed on the router to support the registration of Cisco Jabber clients? (Choose three.)
A. The DNS server has the wrong IP address.
B. The internal DNS Service (SRV) records need to be updated on the DNS Server.
C. Flush the DNS Cache on the client.
D. The DNS AOR records are wrong.
E. Add the appropriate DNS SRV for the Internet entries on the DNS Server.

Answer: BCE

Question 186
When you configure QoS on VCS, which settings do you apply if traffic through the VCS should be tagged with DSCP AF41?
A. Set QoS mode to DiffServ and tag value 32.
B. Set QoS mode to IntServ and tag value 34.
C. Set QoS mode to DiffServ and tag value 34.
D. Set QoS mode to IntServ and tag value 32.
E. Set QoS mode to ToS and tag value 32.

Answer: C

Question 187
Which command is needed to utilize local dial peers on an MGCP-controlled ISR during an SRST failover?
A. ccm-manager fallback-mgcp
B. telephony service
C. dialplan-pattern
D. isdn overlap-receiving
E. voice-translation-rule

Answer: A

Question 188
Which component is needed to set up SAF CCD?
A. SAF-enabled H.323 intercluster (gatekeeper controlled) trunk
B. SAF forwards on Cisco routers
C. Cisco Unified Communications cluster
D. SAF-enabled H.225 trunk

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

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

Answer: A

Question 190
Which setting is recommended?
You want to avoid unnecessary interworking in Cisco TelePresence Video Communication Server, such as where a call between two H.323 endpoints is made over SIP, or vice versa. Which setting is recommended?

A. H.323 – SIP interworking mode: Reject
B. H.323 – SIP interworking mode: On
C. H.323 – SIP interworking mode: Registered only
D. H.323 – SIP interworking mode: Off
E. H.323 – SIP interworking mode: Variable

Answer: C

Question 191
Which statement is true?
Refer to the exhibit. Which statement is true?

A. Endpoints can make calls to unknown IP addresses without the VCS querying any neighbors.
B. If the VCS receives a call to an unknown IP address, it queries its neighbors for the remote address and if permitted, routes the call through the neighbor.
C. Endpoints that are registered directly to the VCS can call only an IP address of a system that is also registered directly to that VCS.
D. Dialing by IP address is not supported on VCS.

Answer: A

Question 192
Which action configures phones in site A to use G.711 to site B, but uses G.729 to site C?

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

Answer: A

Question 193
Which gateway does the Cisco Unified Communications Manager control all call activity?

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

Answer: B

Question 194
Which command should you use?
You want to perform Media Gateway Control Protocol gateway maintenance. For this purpose, you disable Media Control Gateway Protocol gateway using the no mgcp command. After you perform the maintenance, you want to enable the Media Control Gateway Protocol gateway. Which command should you use?

A. enable mgcp
B. mgcp
c. mgcp enable
D. mgcp set
E. activate mgcp
F. mgcp active

Answer: B

Question 195
What is the standard Layer 3 DSCP media packet value that should be set for Cisco TelePresence endpoints?

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

Answer: D

Question 196

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

Answer: C

Question 197
After reviewing the exhibits, which of the following reasons could be causing this failure?

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

Status

Device Pool (1 - 3 of 3)

Locations

Locations (1 - 3 of 3)

CSS:

Calling Search Space (4 - 2 of 2)

SRST:
**SRST-BR2-Config:**

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

---

http://www.aoowe.com/practice-300-075-3138.html
voice service voip
  sip
    bind control source-interface GigabitEthernet0/0/0.130
    bind media source-interface GigabitEthernet0/0/0.130
  registrar server 
!
voice register global
  max-dn 1
  max-pool 1
!
voice register pool 1
  id network 10.1.130.0 mask 255.255.255.255
  call-manager-fallback
    ip source-address 10.1.130.1
    max-dn 1 dual-line
    max-ephones 1
At the HQ cluster, the CFUR for the directory number BR2 phone (+442288224001) has been configured:

- Forward Unregistered Internal Destination: +442288224001
- Forward Unregistered Internal Calling Search Space
- Forward Unregistered External Destination: +442288224001
- Forward Unregistered External Calling Search Space

After adding SRST functionality the SRST does not work. After reviewing the exhibits, which of the following reasons could be causing this failure?
A. Device Pool cannot be default.
B. The Cisco UCM is pointing to the wrong IPv4 address of the BR router.
C. The router does not support SRST.
D. The SRST enabled router is not configured correctly.

Answer: A

Question 198
Which action allows branch Cisco IP phones to function with voicemail while using only the G.729 codec over the WAN link to headquarters?
A. Configure Cisco Unified Communications Manager regions.
B. Configure transcoding within Cisco Unified Communications Manager.
C. Configure transcoding resources in Cisco IOS and assign to the MRGL of Cisco IP phones.
D. Configure transcoding resources in the branch Cisco IP phones.

Answer: C

Question 199
Which device configuration option will allow an administrator to control bandwidth between calls placed between branches?

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

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

Find Locations where Location begins with

- Hub_None
- Phantom
- Shadow

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

### CSS:

### Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name begins with

- All-Devices
- All-Devices

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

SRST:

http://www.aoowe.com/practice-300-075-3138.html
- **Name:** BR2
- **Port:** 2000
- **IP Address:** 10.1.5.15
- **SIP Network/IP Address:** 10.1.5.15
voice service voip
  sip
    bind control source-interface GigabitEthernet0/0/0.130
    bind media source-interface GigabitEthernet0/0/0.130
  registrar server
  !
voice register global
  max-dn 1
  max-pool 1
!
voice register pool 1
  id network 10.1.130.0 mask 255.255.255.255
call-manager-fallback
  ip source-address 10.1.130.1
  max-dn 1 dual-line
  max-ephones 1

At the HQ cluster, the CFUR for the directory number BR2 phone (+442288224001) has been configured:

- Forward Unregistered Internal Destination: +442288224001
- Forward Unregistered Internal Calling Search Space
- Forward Unregistered External Destination: +442288224001
- Forward Unregistered External Calling Search Space

Which device configuration option will allow an administrator to control bandwidth between calls placed between branches?
A. Media Resource Group List
B. Device Pool
C. Location
D. AAR Group
E. Regions

Answer: C

Question 200
Which two statements about configuring mobile and remote access on Cisco TelePresence Video Communication Server Expressway are true? (Choose two.)
A. The traversal server zone on Expressway-C must have a TLS verify subject name configured.
B. The traversal client zone and the traversal server zone Media encryption mode must be set to Force encrypted.
C. The traversal client zone and the traversal server zone Media encryption mode must be set to Auto.
D. The traversal client zone on Expressway-C Media encryption mode must be set to Auto.
E. The traversal client zone and the traversal server zone must be set to SIP TLS with TLS verify mode set to On.

Answer: BE

Question 201
Which parameter should be set to prevent H.323 endpoints from registering to Cisco TelePresence Video Communication Server automatically?
A. On the VCS, navigate to Configuration, Protocols, H.323, and set Auto Discover to off.
B. On the VCS, navigate to Configuration, Protocols, H.323, 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, Registration, Configuration, and set Auto Registration to off.

Answer: A

Question 202
Which three commands are mandatory to implement SRST for five Cisco IP Phones? (Choose three.)
A. call-manager-fallback
B. max-ephones
C. keepalive
D. limit-dn
E. ip source-address

Answer: ABE

Question 203
Which action is performed by the Media Gateway Control Protocol gateway with SRST configured, when it loses connectivity with the primary and backup Cisco Unified Communications Manager servers?
A. The gateway continues to make an attempt to connect to the backup Cisco Unified Communications Manager server.
B. The gateway backs fall back to the H.323 protocol for further call processing.
C. The gateway continues with the MGCP call processing without any interruption.
D. The gateway waits for the primary Cisco Unified Communications Manager server to come alive.
E. All MGCP call processing is interrupted until the Cisco Unified Communications Manager servers are online.
F. The MGCP calls are queued up until the Cisco Unified Communications Manager servers are online.

Answer: B
Question 204
Which option configures the secondary dial tone option for SRST mode to let the users hear the dial tone for PSTN calls?
A. voice service voip
   secondary dialtone 0
B. call-manager-fallback
   secondary dialtone 0
C. dial-peer voice 1 pots
   secondary dialtone 0
D. ccm-manager secondary dialtone 0
Answer: B

Question 205
Which solution is needed to enable presence and extension mobility to branch office phones during a WAN failure?
A. SRST with MGCP fallback
B. SRST without MGCP fallback
C. Cisco Unified Communications Manager Express in SRST mode
D. SRST with VoIP dial peers to Cisco Unified Communications Manager Express
Answer: C

Question 206
Which action routes the 11th call through the PSTN?
Cisco Unified Communications Manager is configured with CAC for a maximum of 10 voice calls. Which action routes the 11th call through the PSTN?
A. Configure an SIP trunk to the ISR.
B. Configure Cisco Unified Communications Manager AAR.
C. Configure Cisco Unified Communications Manager RSVP-enabled locations.
D. Configure Cisco Unified Communications Manager locations.
Answer: B

Question 207
Which action configures PSTN backup for calls that are rejected by the gatekeeper CAC?
Which action configures PSTN backup for calls that are rejected by the gatekeeper CAC?
A. Configure AAR in Cisco Unified Communications Manager.
B. Configure CFUR in Cisco Unified Communications Manager.
C. Configure a route pattern, a route list, and route groups to a trunk and a gateway in Cisco Unified Communications Manager.
D. Configure a route pattern to a gateway in Cisco Unified Communications Manager.
Answer: C

Question 208
What is the purpose of the local route group?
What is the purpose of the local route group?
A. minimize PSTN costs
B. help in the selection of the PSTN egress gateway
C. eliminate the need for a route list
D. allow manipulation of digits at the cost point to egress
Answer: B

Question 209
Which two options are configuration steps on Cisco Unified Communications Manager that are used when integrating with VCS Expressway servers?
Which two options are configuration steps on Cisco Unified Communications Manager that are used when integrating with VCS Expressway servers? (Choose two.)
A. allowing numeric dialing from Cisco phones to Expressway
B. configuring a device pool with video feature enabled
C. allowing dialing to Expressway domain from Cisco phones
D. creating an application user on Cisco Unified Communications Manager with assigned privileges
E. adding the Expressway servers to the Application Servers list
Answer: AC

Question 210
Which two options should be selected in the SIP trunk security profile that affect the SIP trunk pointing to the VCS?
Which two options should be selected in the SIP trunk security profile that affect the SIP trunk pointing to the VCS? (Choose two.)
A. Accept Unsolicited Notification
B. Enable Application Level Authorization
C. Accept Out-of-Dialog REFER
D. Accept Replaces Header
E. Accept Presence Subscription
Answer: AD

Question 211
What is the reason why this would happen?
Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phones. The Cisco VCS and TMS control the Cisco Telepresence Conductor, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows.
DNS Server:

```
ip dns server
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.1
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.1
ip host pub10x-hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.collab10x.cisco.com 10.1.5.16
ip host pub10x-hq.hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.hq.collab10x.cisco.com 10.1.5.16
ip host hq.cisco.com 10.1.5.1
```
Configure an ILS network including the HQ, BR1, and BB clusters.
Enable GDPR for URLs.
Enable GDPR for directory numbers.
Implement PSTN backup.

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

Speed Dial:
<table>
<thead>
<tr>
<th>Phone</th>
<th>Speed Dial Button</th>
<th>Speed Dial Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td>HQ phone 1</td>
<td>1</td>
<td>hq2@cis</td>
</tr>
<tr>
<td>HQ phone 1</td>
<td>2</td>
<td>br1@cis</td>
</tr>
<tr>
<td>HQ phone 1</td>
<td>3</td>
<td>bb@cis</td>
</tr>
<tr>
<td>HQ phone 2</td>
<td>1</td>
<td>hq1@cis</td>
</tr>
<tr>
<td>HQ phone 2</td>
<td>2</td>
<td>br1@cis</td>
</tr>
<tr>
<td>HQ phone 2</td>
<td>3</td>
<td>bb@cis</td>
</tr>
<tr>
<td>BR1 phone 1</td>
<td>1</td>
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<td>BR1 phone 1</td>
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<td>hq2@cis</td>
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<td>bb@cis</td>
</tr>
<tr>
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<td>1</td>
<td>hq1@cis</td>
</tr>
<tr>
<td>BB phone</td>
<td>2</td>
<td>hq2@cis</td>
</tr>
</tbody>
</table>
The intercluster URI call routing no longer allows calls between sites. What is the reason why this would happen?
A. Wrong SIP domain configured.
B. User is not associated with the device.
C. IP or DNS name resolution issue.
D. No SIP route patterns for cisco.lab exist.

Answer: C

Question 212
Which of the following reasons could be causing this failure?
Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phone. The Cisco VCS is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows DP.

Locations:
Subzones:

- 300-075 237 Questions Implementing Cisco IP Telephony and Video, Part 2 (CIPTV2)

http://www.aoowe.com/practice-300-075-3138.html
- **Name:** HQ
- **Authentication policy:** Treat as authenticated
- **Total bandwidth available** - Bandwidth restriction
- **Total bandwidth available** - Total bandwidth
- **Calls into or out of this subzone** - Bandwidth
- **Calls into or out of this subzone** - Total bandwidth
- **Calls entirely within this subzone** - Bandwidth
- **Calls entirely within this subzone** - Total bandwidth
A third collaboration call fails between the backbone site and the HQ site. After reviewing the exhibits, which of the following reasons could be causing this failure?

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

Answer: A

Question 213

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

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

CSS:
Subzones:

HQ-Phone1
HQ-Phone2
HQ-PC
VLAN 110, HQ-Voice
VLAN 10, HQ-Data
VLAN 5, HQ-Servers
pub-hq
tps-hq
dc-hq
tms-hq
vcsxe-hq
sub-hq
tpc-hq
vcsxe-hq

BR2-Phone
VLAN 8, BR2-Servers
VLAN 30, BR2-Data
DHCP
VLAN 130, BR2-Voice

BB
10.1.150.0/24
VLAN 150,
10.1.50.0/24
VLAN 50,
10.1.6.0/24
VLAN 6, BB-

BR1
10.1
VLA
10
VL

PSTN, WAN
WAN
HQ-BR2
10.12.1.0/24
10.13.1.0/24

SUBNETS:
- **Name:** HQ
- **Authentication policy:** Treat as authenticated
- **Total bandwidth available** - Bandwidth restriction
- **Calls into or out of this subzone** - Bandwidth
- **Calls entirely within this subzone** - Total bandwidth
**Question 214**
Which configuration does Cisco recommend for the peer address on the Expressway-C secure traversal zone when the Expressway-E has one NIC enabled?

A. Expressway-E internal IP address  
B. Expressway-E external IP address  
C. Expressway-E internal FQDN  
D. Expressway-E external FQDN

Answer: D

**Question 215**
Which local zone search rule configuration allows SIP registered endpoints to connect to H.323 endpoints that register with an H.323 E.164 number only?

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

A.  
B.  
C.  
D.  

Answer: C

---

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

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

Answer: BD

---

**Name: to HQ_pipe**

**Total Bandwidth available – Bandwidth restriction:**

**Total Bandwidth available – Total bandwidth limit:**

**Calls through this pipe – Bandwidth restriction:**

**Calls through this pipe – Per call bandwidth limit:**

---

http://www.aoowe.com/practice-300-075-3138.html
Answer: D

**Question 216**
Which device configuration option will allow an administrator to assign a device to a specific rights for making calls to specific DNs?

**Scenario:**
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 9971 Video IP Phone. The Cisco VCS is controlling the SX20, the Cisco Telepresence MCU, and the Cisco Jabber Telepresence for Windows.

**SX20 System information:**

![System Information](image)

**DX650 Configuration:**
| 1 | Line [1] - 3304 in Devices |
| 2 | Line [2] - Add a new DN |
| 3 | Redial |
| 4 | sx20-3@csl226.local |
| 5 | Add a new SD |
| 6 | Add a new SD |
| 7 | Add a new SD |
| 8 | Add a new SD |
| 9 | Add a new SD |
| 10 | Add a new SD |
| 11 | Add a new SD |
| 12 | Add a new SD |
| 13 | Add a new SD |
| 14 | Add a new SD |
| 15 | Add a new SD |
| 16 | Add a new SD |

--- Unassigned Associated Items ---

Product Type: Cisco DX650
Device Protocol: SIP

Real-time Device Status
Registration: Unregistered
IPv4 Address: 172.18.32.119
Active Load ID: sipdx650.10-2-3-26
Inactive Load ID: sipdx650.10-1-1-78
Download Status: None

Device Information
- Device is Active
- Device is trusted
- MAC Address: D0C789:
- Description: DX65 P
- Device Pool: Default
- Common Device Configuration: < None:
- Phone Button Template: Cisco DX
- Softkey Template: < None:
- Common Phone Profile: < None:
- Calling Search Space: All-Device
- AAR Calling Search Space: All-Device

MRGL:

Status
1 records found

Media Resource Group List (1 of 1)

Find Media Resource Group List where Name begins with: MRGL

Add New | Select All | Clear All | Delete Selected
### Device Pool (1 - 3 of 3)

Find Device Pool where Device Pool Name **begins with**

<table>
<thead>
<tr>
<th>Name</th>
<th>Data Unified CM Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Default</td>
</tr>
<tr>
<td>Default</td>
<td>Default</td>
</tr>
<tr>
<td>GSM</td>
<td>GSM</td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected

### Locations (1 - 3 of 3)

Find Locations where Location **begins with**

- Hub_None
- Phantom
- Shadow

Add New | Select All | Clear All | Delete Selected

### Automated Alternate Routing Group

Find Automated Alternate Routing Group where Name **begins with**

No active query. Please enter your search.

### Calling Search Space (1 - 2 of 2)

Find Calling Search Space where CSS Name **begins with**

<table>
<thead>
<tr>
<th>CSS Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>All-Devices</td>
</tr>
<tr>
<td>All-Devices</td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected

Movi Failure:
Movie Settings:
Which device configuration option will allow an administrator to assign a device to a specific rights for making calls to specific DN's?
A. Media Resource Group List
B. Device Pool
C. Location
D. AAR Group
E. Calling Search Space

Answer: E

Question 217
Which CPL configuration accomplishes this goal?
A local gateway is registered to Cisco TelePresence Video Communication Server with a prefix of 7. The administrator wants to stop calls from outside the organization being routed through it.

Which CPL configuration accomplishes this goal?

1. <complexType name="IP" encoding="UTF-8" xmlns="http://www.tandberg.net/cpi-extensions">
   <complexContent>
     <extension base="cpi:configurable-options">
       <attribute name="origination-zone" type="cpi:destination-string"/>
       <attribute name="originating-zone" type="cpi:destination-string"/>
     </extension>
   </complexContent>
 </complexType>

2. <complexType name="IP" encoding="UTF-8" xmlns="http://www.tandberg.net/cpi-extensions">
   <complexContent>
     <restriction base="cpi:configurable-options">
       <attribute name="origination-zone" type="cpi:destination-string"/>
       <attribute name="originating-zone" type="cpi:destination-string"/>
     </restriction>
   </complexContent>
 </complexType>

3. <complexType name="IP" encoding="UTF-8" xmlns="http://www.tandberg.net/cpi-extensions">
   <complexContent>
     <restriction base="cpi:configurable-options">
       <attribute name="origination-zone" type="cpi:destination-string"/>
       <attribute name="originating-zone" type="cpi:destination-string"/>
     </restriction>
   </complexContent>
 </complexType>

4. <complexType name="IP" encoding="UTF-8" xmlns="http://www.tandberg.net/cpi-extensions">
   <complexContent>
     <restriction base="cpi:configurable-options">
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       <attribute name="originating-zone" type="cpi:destination-string"/>
     </restriction>
   </complexContent>
 </complexType>

5. <complexType name="IP" encoding="UTF-8" xmlns="http://www.tandberg.net/cpi-extensions">
   <complexContent>
     <restriction base="cpi:configurable-options">
       <attribute name="origination-zone" type="cpi:destination-string"/>
       <attribute name="originating-zone" type="cpi:destination-string"/>
     </restriction>
   </complexContent>
 </complexType>
Question 218
If delegated credentials checking has been enabled and remote workers can register to the VCS Expressway, which statement is true?
A. H.323 message credential checks are delegated.
B. SIP registration proxy mode is set to On in the VCS Expressway.
C. A secure neighbor zone has been configured between the VCS Expressway and the VCS Control.
D. SIP registration proxy mode is set to Off in the VCS Expressway.
Answer: D

Question 219
Which three commands are necessary to override the default CoS to DSCP mapping on interface Fastethernet0/1? (Choose three.)
A. mls qos map cos-dscp 0 10 18 26 34 46 48 56
B. mls qos map dscp-cos 8 10 to 2
C. mls qos
D. interface Fastethernet0/1
E. interface Fastethernet0/1
F. interface Fastethernet0/2
Answer: ACD

Question 220
Where do you specify the device mobility group and physical location after they have been configured?
A. phones
B. DMI
C. device mobility CSS
D. device pool
E. MRGL
F. locale
Answer: D

Explanation:
Before you configure a device pool, you must configure the following items if you want to choose them for the device pool, Cisco Unified Communications Manager group (required), Date/time group (required). Region (required), SRST reference (optional), Media resource group list (optional), Calling search space for auto-registration (optional). Reverted call focus priority (optional), Device mobility group (optional), Device mobility calling search space, Physical location (optional). Location, AAR group. AAR calling search space.

Question 221
Which two actions ensure that the call load from Cisco TelePresence Video Communication Server to a Cisco Unified Communications Manager cluster is shared across Unified CM nodes?
A. Create a neighbor zone in VCS with the Unified CM nodes listed as location peer addresses.
B. Create a single traversal client zone in VCS with the Unified CM nodes listed as location peer addresses.
C. Create one neighbor zone in VCS for each Unified CM node.
D. Create a VCS DNS zone and configure one DNS SRV record per Unified CM node.
E. In VCS set Unified Communications mode to Mobile and remote access and configure each
Question 222
What is the maximum number of Service Advertisement Framework forwarders that you can assign to a specific node?
In a node-specific Service Advertisement Framework forwarder deployment model, what is the maximum number of Service Advertisement Framework forwarders that you can assign to a specific node?
A. 1
B. 2
C. 3
D. 4
E. 5
F. 6
Answer: B

Question 223
What is causing this failure?
Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 9971 Video IP Phone. The Cisco VCS is controlling the SX20, the Cisco Telepresence MCU, and the Cisco Jabber Telepresence for Windows.
SX20 System information:

DX650 Configuration:
MRGL:

DP:

http://www.aoowe.com/practice-300-075-3138.html
### Device Pool (1 - 3 of 3)

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

### Locations (1 - 3 of 3)

<table>
<thead>
<tr>
<th>Location</th>
<th>AARG</th>
<th>CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hub_None</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Phantom</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Shadow</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Calling Search Space (1 - 2 of 2)

<table>
<thead>
<tr>
<th>CSS Name</th>
<th>AARG</th>
<th>CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>All-Devices</td>
<td></td>
<td></td>
</tr>
<tr>
<td>All-Devices</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Login failed

Connection rejected by server. Try logging in again later.

### Movie Settings:

Movie Settings:
A new DX650 IP phone with MAC address D0C7.8914.132D. IP address is 172.18-32.119 has been added to the Cisco Unified Communications Manager, but is not registering properly. What is causing this failure?
A. Device Pool cannot be default.
B. The DX650 is the incorrect calling search space.
C. The DX650 Phones does not support SIP.
D. The location Hub_None has not been activated.
E. The DX650’s MAC address is incorrect in the Cisco UCM.
Answer: E

Question 224
Which type of search message appears in the Cisco TelePresence Video Communication Server search history page when it receives a H.323 call from a RAS-enabled endpoint that originates from an external zone?
A. ARQ
B. SETUP
C. LRQ
D. INVITE
E. OPTIONS
Answer: C

Question 225
Which three devices support the SAF Call Control Discovery protocol? (Choose three.)
A. Cisco Unified Border Element
B. Cisco Unity Connection
C. Cisco IOS Gatekeeper
D. Cisco Catalyst Switch
E. Cisco IOS Gateway
F. Cisco Unified Communications Manager
Answer: AEF

Question 226
Which statement about Service Advertisement Framework is true? Which statement about Service Advertisement Framework is true?
A. SAF requires that the EIGRP be configured on all routers, including non-SAF routers.
B. SAF requires that the EIGRP be configured only on SAF routers.
C. SAF has no dependency on the underlying routing protocol, as long as it is a dynamic routing protocol. Static routes are not supported.
D. SAF operates on any dynamic or static IP routing configuration. SAF is totally independent of the underlying routing protocol.
Answer: D

Question 227
Which Cisco IOS command is used to verify that the Cisco Unified Communications Manager Express has registered with the SAF Forwarder?
A. show eigrp service-family ipv4 clients
B. show eigrp address-family ipv4 clients
C. show voice saf dndb all
D. show saf registration
E. show ip saf registration
Answer: A

Question 228
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? Which Cisco IOS command is used to verify that a SAF Forwarder that is registered with Cisco Unified Communications Manager has established neighbor relations with an adjacent SAF Forwarder?
A. show eigrp service-family ipv4 neighbors
B. show eigrp address-family ipv4 neighbors
C. show voice saf dndball
D. show saf neighbors
E. show ip saf neighbors

Answer: A

Question 229
When an H.323 trunk is added for Call Control Discovery, which statement is true?
When an H.323 trunk is added for Call Control Discovery, which statement is true?
A. The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Enable SAF check box should be selected in the trunk configuration.
B. The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Trunk Service Type should be Call Control Discovery.
C. The H.323 trunk is added by selecting Call Control Discovery Trunk and then selecting H.323 as the protocol to be used.
D. The H.323 trunk is added by selecting H.323 Trunk, and selecting Inter-Cluster Trunk as the Device Protocol. The destination IP address field is configured as ‘SAF’ to indicate that this trunk is used for SAF.

Answer: A

Question 230
When a SIP trunk is added for Call Control Discovery, which statement is true?
When a SIP trunk is added for Call Control Discovery, which statement is true?
A. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Enable SAF check box should be selected.
B. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Trunk Service Type should be Call Control Discovery.
C. The SIP trunk is added by selecting Call Control Discovery Trunk and then selecting SIP as the protocol to be used.
D. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The destination IP address field is configured as ‘SAF’ to indicate that this trunk is used for SAF.

Answer: B

Question 231
Which two options for a Device Mobility-enabled IP phone are true?
Which two options for a Device Mobility-enabled IP phone are true? (Choose two.)
A. The phone configuration is not modified.
B. The roaming-sensitive parameters of the current (that is, the roaming) device pool are applied.
C. The user-specific settings determine which location-specific settings are downloaded from the Cisco Unified Communications Manager device pool.
D. If the DMGs are the same, the Device Mobility-related settings are also applied.

Answer: BD

Question 232
Which of the following reasons could be causing this failure?
Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phone. The Cisco VCS is controlling the SX20, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows DP.

Locations:

http://www.aoowe.com/practice-300-075-3138.html
Movie Failure:

Login failed
Connection rejected by server. Try logging in again later.

OK

Movie Setting:
Topology:
- Name: HQ
- Authentication policy: Treat as authenticated
- Total bandwidth available - Bandwidth restriction
- Total bandwidth available - Total bandwidth
- Calls into or out of this subzone - Bandwidth
- Calls into or out of this subzone - Total bandwidth
- Calls entirely within this subzone - Bandwidth
- Calls entirely within this subzone - Total bandwidth
Both of the Cisco Telepresence Video for Windows clients are able to log into the server but can’t make any calls. After reviewing the exhibits, which of the following reasons could be causing this failure?
A. Wrong username and/or password.
B. Wrong SIP domain name.
C. The TMSPE is not working.
D. The bandwidth is incorrectly configured.

Answer: D

Question 233
What two tasks must be completed in order to support calls between the VCS controlled endpoints and the Cisco Unified CM endpoints?

Scenario:
There are two call control systems in this item. The Cisco UCM is controlling the DX650, the Cisco Jabber for Windows Client, and the 7965 and 9971 Video IP Phones. The Cisco VCS and TMS control the Cisco Telepresence Conductor, the Cisco TelePresence MCU, and the Cisco Jabber TelePresence for Windows

DNS Server:

```
ip dns server
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.1
ip host _cisco-uds._tcp.hq.cisco.com srv 1 1 8443 10.1.5.1
ip host pub10x-hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.collab10x.cisco.com 10.1.5.15
ip host pub10x-hq.hq.collab10x.cisco.com 10.1.5.15
ip host sub10x-hq.hq.collab10x.cisco.com 10.1.5.15
ip host hq.cisco.com 10.1.5.1
```
Configure an ILS network including the HQ, BR1, and BB clusters.

Enable GDPR for URLs.
Enable GDPR for directory numbers.
Implement PSTN backup.

ip host _collab-edge._tls.hq.cisco.com SRV 1 1 8443 vcsxe8x-hq.hq.collab10x.cisco.com

ip host vcsxe8x-hq.hq.collab10x.cisco.com 10.1.5.20

Speed Dial:
<table>
<thead>
<tr>
<th>Phone</th>
<th>Speed Dial Button</th>
<th>Speed Dial Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td>HQ phone 1</td>
<td>1</td>
<td>hq2@cisco</td>
</tr>
<tr>
<td>HQ phone 1</td>
<td>2</td>
<td>br1@cisco</td>
</tr>
<tr>
<td>HQ phone 1</td>
<td>3</td>
<td>bb@cisco</td>
</tr>
<tr>
<td>HQ phone 2</td>
<td>1</td>
<td>hq1@cisco</td>
</tr>
<tr>
<td>HQ phone 2</td>
<td>2</td>
<td>br1@cisco</td>
</tr>
<tr>
<td>HQ phone 2</td>
<td>3</td>
<td>bb@cisco</td>
</tr>
<tr>
<td>BR1 phone 1</td>
<td>1</td>
<td>hq1@cisco</td>
</tr>
<tr>
<td>BR1 phone 1</td>
<td>2</td>
<td>hq2@cisco</td>
</tr>
<tr>
<td>BR1 phone 1</td>
<td>3</td>
<td>bb@cisco</td>
</tr>
<tr>
<td>BB phone</td>
<td>1</td>
<td>hq1@cisco</td>
</tr>
<tr>
<td>BB phone</td>
<td>2</td>
<td>hq2@cisco</td>
</tr>
</tbody>
</table>
What two tasks must be completed in order to support calls between the VCS controlled endpoints and the Cisco Unified CM endpoints? (Choose two.)
A. Media Resource Group List.
B. Configure a SIP trunk on the Cisco Unified CM to point to the Cisco VCS.
C. Configure a neighbor zone on the Cisco Unified CM to point to the Cisco VCS.
D. Configure a SIP trunk on the Cisco VCS to point to the Cisco Unified CM.
E. Configure a neighbor zone on the Cisco VCS to point to the Cisco Unified CM.

Answer: BE

Question 234
Which sign is prefixed to the number in global call routing?
A. –
B. +
C. #
D. @
E. *

Answer: B

Question 235
Which two options are valid service parameter settings that are used to set up proper video QoS behavior across the Cisco Unified Communications Manager infrastructure? (Choose two.)
A. DSCP for Video Calls when RSVP Fails
B. Default Interregion Min Video Call Bit Rate (Includes Audio)
C. Default Interregion Max Video Call Bit Rate (Includes Audio)
D. DSCP for Video Signaling
E. DSCP for Video Signaling when RSVP Fails

Answer: AC

Question 236
What is an advantage of TEHO?
A. TEHO implemented with ISRs eliminates PSTN toll charges.
B. TEHO implemented with ISRs can reduce PSTN toll charges.
C. TEHO implemented with AAR reduces toll charges.
D. TEHO implemented with CFUR reroutes calls.

Answer: B
Question 237
Which component of Cisco Unified Communications Manager is responsible for sending keepalive messages to the Service Advertisement Framework forwarder?
A. Call Control Discovery requesting service
B. Hosted DNs service
C. Service Advertisement Framework client control
D. Cisco Unified Communications Manager database
E. Service Advertisement Framework-enabled trunk
F. gatekeeper

Answer: C