400-051 378 Questions CCIE Collaboration

Question 1
Which two solutions resolve the issue?
Refer to the exhibit.

<table>
<thead>
<tr>
<th>Verify external database server reachability (pingable)</th>
<th>The following Cisco Unified IM Presence Service node to external database server connections fail</th>
</tr>
</thead>
<tbody>
<tr>
<td>[ ]</td>
<td>172.16.100.52 &gt;&gt; test (Persistent Chat)</td>
</tr>
<tr>
<td>Verify external database server connectivity (database connection check)</td>
<td></td>
</tr>
</tbody>
</table>

When enabling Group and Persistent Chat in an IM&P server, the administrator encountered the problem shown. Which two solutions resolve the issue? (Choose two)
A. Configure the external database to listen in the correct port.
B. Restart the Cisco Route Datastore service in the IM&P server.
C. Make sure the group chat system administrator has access.
D. Configure a new host under Group Chat Server Alias.
E. Fix the user permissions on the external database.

Correct Answer: AE
Explanation/Reference:

Question 2
Which two fields can be used to uniquely identify the same call in the Call Detail Records and the Call Management Records?
Which two fields can be used to uniquely identify the same call in the Call Detail Records and the Call Management Records? (Choose two)
A. nodeld
B. globalCallId_callId
C. callIdentifier
D. pkid
E. globalCallId_ClusterId
F. globalCallId_callManagerId
G. deviceName

Correct Answer: BF
Explanation/Reference:

Question 3
Which four configuration changes resolve this issue?
Refer to the exhibit.

voice translation-rule 5
rule 1 /91\([27]\..\)\(/\./\+1\"/\rule 2 /91\([567][456]\.).\(/\./\+0\"/\rule 3 /91\([7.\.)\(/\./\+1\"/\rule 4 /91\(.\./\+/0\"/

RTR1# test voice translation-rule 5 917765284569
Matched with rule 1
Original number: 917765284569 Translated number: +7765284569
Original number type: none Translated number type: none
Original number plan: none Translated number plan: none

A collaboration engineer is troubleshooting outgoing calls that do not work to a specific number. The PSTN provider is playing a prompt explaining that the dialed number is missing the "1" for long Distance calls. Which four configuration changes resolve this issue? (Choose four)
A. edit rule 4 and change /+1/ to /+112/
B. edit rule 1 and change ([27].) to (7+)
C. edit rule 3 and change /+112/ to /+112/
D. edit rule 2 and change /+2/ to /+112/
E. edit rule 2 and change ([567][456].) to ([5-7][4-6].)
Question 4
Which action resolves this issue?
During a Cisco Connection extension greeting, callers can press a single key to be transferred to a specific extension. However, callers report that the system does not process the call immediately after pressing the key. Which action resolves this issue?
A. Reduce Caller Input timeout in Cisco Unity Connection Service Parameters.
B. Lower the timer Wait for Additional Digits on the Caller input page.
C. Enable Ignore Additional Input on the Edit Caller input page for the selected key.
D. Enable Prepend Digits to Dialed Extensions and configure complete extension number on the Edit Caller input page for the selected key.
E. Reduce Caller input timeout in Cisco Unity Connection Enterprise Parameters.
Correct Answer: C
Explanation/Reference:

Question 5
Where the administrator can reset all database replication and initiate a broadcast of all tables on a Cisco Unified CM cluster running version 9.1?

A. Real Time Monitoring Tool
B. Cisco Unified Serviceability
C. Cisco Unified OS Administration
D. Cisco Unified CM CLI
E. Disaster Recovery System
Correct Answer: D
Explanation/Reference:

Question 6
Which two parameters, in the reply of an MGCP gateway to an Audit Endpoint message, indicate to a Cisco Unified CM that it has an active call on an endpoint? (Choose two)

A. Bearer Information
B. Call ID
C. Capabilities
D. Connection ID
E. Connection Parameters
F. Connection Mode
Correct Answer: AD
Explanation/Reference:

Question 7
Which two configuration changes allow this?
Users report that they are unable to control their Cisco 6941 desk phone from their Jabber client, but the Jabber client works as a soft phone. Which two configuration changes allow this? (Choose two)
A. Assign group “Standard CTI Allow Control of Phones supporting Connected Xfer and Conf” to the user.
B. Set the End User page to the Primary Extension on the desk phone.
C. Set the Owner User ID on the desk phone.
D. Assign group “Standard CTI Enabled User Group” to the user.
E. Assign group “Standard CTI Allow Control of Phones Supporting Rollover Mode” to the user.
Correct Answer: AE
Explanation/Reference:

Question 8
Which Cisco Unified CM service is responsible for detecting new Call Detail Records files and transferring them to the CDR Repository node?
A. Cisco CallManager
B. Cisco CDR Repository Manager
C. Cisco SOAP-CDRonDemand Service
D. Cisco Extended Functions
E. Cisco CDR Agent
Correct Answer: E
Explanation/Reference:

Question 9
Which two translation profiles create the required outcome?
A SIP carried delivers DIDs to a Cisco Unified Border Element in the form of +15567810XX, where the last two digits could be anything from 00 to 99. To match the internal dial plan, that number must be changed to 6785XXX, where the last two digits should be retained. Which two translation profiles create the required outcome? (Choose two)
A. rule 1 /555(.*).*(.*)/ /1502/
B. rule 1 /+ 1555(…).(…)$/ /152/
C. rule 1 /^+ 1555(678)10(..)$/ /1502/
D. rule 1 /15+678(… .)/6781/
E. rule 1 /.15+678?10?(..)/ /678501/

Correct Answer: DFGH
Explanation/Reference:
**Question 10**
Which two parameters are requested in an Audit Connection message from a Cisco Unified CM to endpoint on a MGCP gateway?

A. Call ID
B. Capabilities
C. Bearer Information
D. Connection Parameters
E. Connection Mode
F. Connection ID

Correct Answer: CF

**Question 11**
Which web administration page can you verify database replication health in a two-server Cisco Unified CM cluster running version 9.1?

A. Cisco Unified OS Administration
B. Cisco Unified CM Administration
C. Disaster Recovery System
D. Cisco Unified Reporting
E. Cisco Unified Serviceability.

Correct Answer: D

**Question 12**
Which two steps resolve this error?
A Collaboration engineer implemented Cisco EMCC between Cisco Unified CM clusters. The administrator has configured the bulk certificate management and exported the certificates to the SFTP server. After importing the certificates into each of the clusters, the administrator tested Cisco EMCC on a phone, but received “Login is unavailable (208)”. Which two steps resolve this error? (Choose two)

A. Update the cluster IDs so that they are unique in the EMCC network.
B. Enable the Allow Proxy service parameter on both clusters.
C. Restart the Cisco CallManager and Cisco Tomcat Servers.
D. Associate a user device profile for the user in the remote cluster.
E. Consolidate the exported certificates and reimport into each cluster.

Correct Answer: CE

**Question 13**
Which two options cause this problem?
Refer to the exhibit.

A Jabber for Windows application fails to connect to the voicemail server. Which two options cause this problem? (Choose two)

A. The jetty service has been disabled or is not running.
B. A voicemail user password configuration error exists.
C. A firewall is configured for blocking port 7080.
D. An SSL certificate has an encryption problem.
E. A company internal DNS server has a timeout problem.

Correct Answer: CE

**Explanation/Reference:**

http://www.aoowe.com/practice-400-051-3156.html
Question 14
Which three disconnect methods can be configured to fix this problem?
A user is troubleshooting an FXO line on a Cisco IOS router that remains connected even after the call ends. Which three disconnect methods can be configured to fix this problem? (Choose three)
A. Loop Current Feed Open Signaling Disconnect
B. Hookflash Duration Signaling Disconnect
C. CP-TONE Dual Supervisory Disconnect
D. Power Denial-based Supervisory Disconnect
E. Ground-start Signaling Disconnect
F. Tone-based Supervisory Disconnect
Correct Answer: DEF
Explanation/Reference:

Question 15
What happens to the USB e-token after the administrator fails to enter the correct password at the next attempt?
Refer to the exhibit.
A. The token is locked for five days, after which the retry counter resets.
B. The token is locked until unlocked by Cisco TAC.
C. The token is locked until Cisco CTL Client is uninstalled and reinstalled on the client PC.
D. The token cannot be used on the same client PC again. It can be used with another Cisco CTL Client on a different PC.
E. The token is locked forever.
Correct Answer: E
Explanation/Reference:

Question 16
Which two configuration charges solve this problem?
Refer to the exhibit.
In a Cisco Unified CM environment with default QoS configuration in the cluster, IP phone users report voice quality issues when they are downloading large files to their PC. Which two configuration changes solve this problem? (Choose two)
A. The srr-queue bandwidth share command must be changed to increase the weight of queue 1.
B. The global configuration of threshold 3 of queue 4 must be changed to mls qos srr-queue cos-map queue 4 threshold 3 0 5.
C. The srr-queue bandwidth shape command must be changed to increase the weight of queue 1.
D. The srr-queue bandwidth shape command must be removed from the interface configuration.
E. The priority-queue out command is missing from the interface configuration.

Correct Answer: CE

Explanation/Reference:

**Question 17**
Which service, available only on the publisher server in a Cisco Unified CM cluster, is needed to enable a mixed mode cluster?
A. Cisco Trust Verification
B. Cisco Transport Layer Security
C. Cisco CTL Provider
D. Cisco Certificate Expiry Monitor
E. Cisco Certificate Authority Proxy Function

Correct Answer: C

Explanation/Reference:

**Question 18**
Which two configurations changes fix this problem?
A Cisco Unified CM administrator configured the phone VPN for the remote users. The remote users cannot see their missed calls. Which two configurations changes fix this problem? (Choose two)
A. Enable Log Missed Calls in the phone line in Cisco Unified CM.
B. Configure a Log Server in the Common Phone profile in Cisco Unified CM.
C. Configure the webvpn-attributes in the Cisco ASA tunnel group.
D. Enable Password Persistence in the VPN profile in Cisco Unified CM.
E. Configure an Alternate TFTP in the remote phone.

Correct Answer: BE

Explanation/Reference:

**Question 19**

http://www.aoowe.com/practice-400-051-3156.html
Which two configurations meet the requirements?

An engineer is planning the voice call bandwidth requirements between two offices. The design requires a capacity of 25 concurrent audio calls using the G.729 codec and IPv6 transport. Which two configurations meet the requirements? (Choose two)

A. required bandwidth 850 kbps, given Ethernet transport, and a 30-byte payload.
B. required bandwidth 975 kbps, given Ethernet transport, and a 20-byte payload.
C. required bandwidth 400 kbps, given Ethernet transport, compressed RTP, and a 30-byte payload.
D. required bandwidth 300 kbps, given PPP transport, compressed RTP, and a 20-byte payload.
E. required bandwidth 250 kbps, given PPP transport, compressed RTP, and a 30-byte payload.
F. required bandwidth 650 kbps, given PPP transport, and a 20-byte payload.

Correct Answer: BD
Explanation/Reference:

Question 20
What time will it send the next NTFY message to the Cisco Unified CM?
Refer to the exhibit.

```
Jan 10 05:55:35.130: MGCP Packet sent to 10.1.1.2:2427
NTFY 217730192 *@MGCP-gateway.cisco.com MGCP 0.1
0:
0:
-
```

Jan 10 05:55:35.130: MGCP Packet received from 10.1.1.2:2427
```
200 217730192
```

The MGCP debugs were captured on a Cisco IOS MGCP PRI gateway registered to a Cisco Unified CM. Assume that this gateway had no active calls and will not take any new calls for the next 3 minutes. What time will it send the next NTFY message to the Cisco Unified CM?

A. Jan 10 05:56:35.130
B. Jan 10 05:55:45.130
C. Jan 10 05:55:50.130
D. Jan 10 05:56:05.130
E. Jan 10 05:55:40.130

Correct Answer: C
Explanation/Reference:

Question 21
Which three wireless security modes allow the user to enter user and password authentication on a Cisco 9971 IP Phone?
Refer to the exhibit.

```
Administrator Settings

Security Mode

- Open
- Open with WEP
- Shared Key
- LEAP
- EAP-FAST
- AKM
```

Which three wireless security modes allow the user to enter user and password authentication on a Cisco 9971 IP Phone?

A. Shared Key
B. EAP-FAST
C. Open with WEP
D. Open
E. LEAP
F. AKM

Correct Answer: BEF
Explanation/Reference:

Question 22
Which file does a Cisco IP phone with MAC address 1111.2222.3333 request from the TFTP server when TFTP configuration encryption is enabled?

Which file does a Cisco IP phone with MAC address 1111.2222.3333 request from the TFTP server when TFTP configuration encryption is enabled?

A. SEP111122223333.cnf.xml.enc
Question 23
What must be changed on the firewalls to fix this issue?
In a multicluster deployment model, a customer is using centralized TFTP service. Firewalls are being implemented between each of the clusters. After the installation of the firewalls, the centralized TFTP stopped serving files. What must be changed on the firewalls to fix this issue?
A. Open TCP port 22 between the leaf clusters and the central cluster.
B. Open TCP port 443 between the leaf cluster and the central cluster.
C. Open TCP port 8443 between the leaf cluster and the central cluster.
D. Open UDP port 69 between the leaf cluster and the central cluster.
E. Open TCP port 80 between the leaf cluster and the central cluster.
F. Open UDP port 500 to enable certificate synchronization between the TFTP servers.
Correct Answer: D
Explanation/Reference:

Question 24
Which option is the minimum Cisco Unified CM Service Parameter configuration that is needed to ensure compliance to this policy?
The Director of information Security of your company wants to log all calls when a user's phone goes off-hook and immediately back to on-hook in Call Detail Records. What option is the minimum Cisco Unified CM Service Parameter configuration that is needed to ensure compliance to this policy?
A. Set CDR Enabled Flag to True and set Call Diagnostics Enabled to Enable Only When CDR Enabled Flag is True.
B. Set CDR Enabled Flag to True and set Call Diagnostics Enabled to Enable Regardless of CDR Enabled Flag.
C. Set CDR Log Calls with Zero Duration Flag to True.
D. Set CDR Enabled Flag to True.
E. Set CDR Enabled Flag and CDR Log Calls with Zero Duration Flag to True.
Correct Answer: E
Explanation/Reference:

Question 25
Which option is the minimum Cisco Unified CM Service Parameter configuration that is needed to ensure compliance to this policy?
The Director of information Security of your company wants to log all calls when a user's phone speed dials to a busy PSTN destinations and hangs up in less than one second in Call Detail Records. Which option is the minimum Cisco Unified CM Service Parameter configuration that is needed to ensure compliance to this policy?
A. Set CDR Enabled Flag to True and set Call Diagnostics Enabled to Enable Only When CDR Enabled Flag is True.
B. Set CDR Enabled Flag to True and set Call Diagnostics Enabled to Enable Regardless of CDR Enabled Flag.
C. Set CDR Log Calls with Zero Duration Flag to True.
D. Set CDR Enabled Flag to True.
E. Set CDR Enabled Flag and CDR Log Calls with Zero Duration Flag to True.
Correct Answer: C
Explanation/Reference:

Question 26
Which configuration can the engineer apply to the Cisco Unified Border Element to accomplish this temporary fix?
A collaboration engineer is troubleshooting a delay in call completion on a SIP Cisco Unified Border Element gateway. The gateway is set up for dual stack IP with DNS as the dial peer target. In DNA, AAAA and A records are configured for the target. The engineer determines that IPv4 calls should remain IPv4, when possible, to temporarily resolve the issue. The company is testing some IPv6 applications so the engineer cannot disable IPv6 altogether. Which configuration can the engineer apply to the Cisco Unified Border Element to accomplish this temporary fix?
A. sip-ua
Protocol mode ipv4
B. voice service voip
Sip
No anat
C. voice service voip
No allow-connections ipv4 to ipv6
D. sip-ua
Protocol mode dual-stack preference ipv4
E. voice service voip
Sip
Preference ipv4
Correct Answer: D
Explanation/Reference:

Question 27
What must be changed about the configuration to allow the Unified CME to attach to the SAF network pass calls to and from the Unified CM network?
Refer to the exhibit.
router eigrp 1
  service-family ipv4 autonomous-system 1
  sf-interface loop0
  no split-horizon
  exit-sf-interface
topology base
  exit-sf-topology
  exit-service-family

client username cmcluster password 0 password12345
domain 1 default

voice service saf
profile trunk-route 1
  session protocol h323 interface Gig0/0 transport tcp port 1720

profile dn-block 1
pattern 1 type extension 3xxx

profile callcontrol 1
dn-service
trunk-route 1
dn-block 1

channel 1 vrouter 1 asystem 1
subscribe callcontrol wildcarded
publish callcontrol 1

dial-peer voice 2000 voip
session target saf
destination-pattern 2...

A collaboration engineer was asked to attach a Cisco Unified CME to the Cisco UCM network of a client via SAF. The configuration was applied, but the Unified CME was not able to retrieve the dial plan. What must be changed about the configuration to allow the Unified CME to attach to the SAF network pass calls to and from the Unified CM network?

A. Split-horizon must be enabled under EIGRP.
B. The SAF EIGRP instance must be configured under a virtual instance name.
C. The session target on the dial peers should point to the next-hop SAF forwarder.
D. The Cisco Unified CME must be configured for SIP under voice service saf when communicating with Cisco Unified CM clusters over SAF.

Correct Answer: B
Explanation/Reference:

Question 28
Which two designs are valid?
A collaboration engineer is designing a Cisco Unity Connection network for a large client running 10 x. The client has 12 locations, each with their own Cisco Unity Connection cluster. Which two designs are valid? (Choose two)
A. full-mesh topology with HTTPS Networking
B. six clusters each in two full-mesh Unity Connection Digital Networks connected with VPIM
C. hub-and-spoke topology with Unity Connection Digital Networking
D. a 10-cluster Unity Connection Digital network connected to a 2-cluster HTTPS network
E. hub-and-spoke topology with HTTPS Networking
F. full-mesh topology with Unity Connection Digital Networking

Correct Answer: CF
Explanation/Reference:

Question 29
Which two configuration changes must be made to correct the audio quality issues?
Refer to the exhibit.
In a WAN environment with LLQ configured, audio quality issues are experienced during peak network traffic times. Which two configuration changes must be made to correct the audio quality issues? (Choose two)

A. The total configured percentage in the policy map must be decreased.
B. The class-map RTP must be changed to match-any.
C. The precedence 5 must be removed from the RTP class map.
D. The access-list mask must be changed to match the local subnet as the source.
E. The class-map control must be changed to match-any.

Correct Answer: BD

Explanation/Reference:

Question 30
Which parameter in an NTFY message is used by a Cisco IOS MGCP gateway to inform a Cisco Unified CM that an analog endpoint has gone off-hook?

A. Detect Events
B. Event States
C. Connection Mode
D. Observed Events
E. Requested Events

Correct Answer: A

Explanation/Reference:

Question 31
Which two Cisco Unified Border Element configuration changes will prevent this problem from happening again?

An outbound call is in progress through a Cisco Unified Border Element using G729r8 codec. And it is dropped after 60 minutes. Root cause analysis revealed that ITSP signaled a codec change to G711u. Which two Cisco Unified Border Element configuration changes will prevent this problem from happening again? (Choose two)

A. Configure the voice-class sip midcall-signaling block command on the outbound dial peer.
B. Configure the midcall-signaling preserve-codec under voice service voip.
C. Configure the voice class-codexp command with G711u and G729r8 codecs on the outbound dial peer.
D. Configure the voice-class sip midcall-signaling preserve-codec command on the outbound dial peer.
E. Configure the midcall-signaling preserve-codec command under each outbound ITSP dial peer.
F. Configure the midcall-signaling passthru media-change command under voice service voip.

Correct Answer: AD

Explanation/Reference:

Question 32
Which parameter in an RQNT message is used by a Cisco Unified CM to request a Cisco IOS MGCP gateway to report an on-hook event on an analog endpoint?
Which parameter in an RQNT message is used by a Cisco Unified CM to request a Cisco IOS MGCP gateway to report an on-hook event on an analog endpoint?
A. Detect Events  
B. Connection Mode  
C. Requested Events  
D. Event States  
E. Observed Events  
Correct Answer: C  
Explanation/Reference:

**Question 33**
Which two wireless security modes offer these configuration options on a Cisco 9971 IP Phone?  
Refer to the exhibit.

Which two wireless security modes offer these configuration options on a Cisco 9971 IP Phone? (Choose two)
A. Shared Key  
B. AKM  
C. EAP-FAST  
D. Open  
E. LEAP  
F. Open with WEP  
Correct Answer: AF  
Explanation/Reference:

**Question 34**
Which four types of licensing and database servers support this requirement?  
A Cisco Unified Contact Center Express manager wants to add database integration to the selfservice interactive voice response application. Which four types of licensing and database servers support this requirement? (Choose four.)
A. The server must have enhanced licensing.  
B. The server must have premium licensing.  
C. A server running Sybase Adaptive Server is required.  
D. A server running Oracle is required.  
E. A server running Postgres SQL is required.  
F. A server running SAP SQL server is required.  
G. A server running Microsoft SQL server is required.  
H. The server must have standard licensing.  
Correct Answer: BCDG  
Explanation/Reference:

**Question 35**
Which MGCP message does a Cisco IOS MGCP gateway send to the backup Cisco Unified CM server when two consecutive keep-alive exchanges failed with the primary Cisco Unified CM server?
Which MGCP message does a Cisco IOS MGCP gateway send to the backup Cisco Unified CM server when two consecutive keep-alive exchanges failed with the primary Cisco Unified CM server?  
A. AUEP  
B. DLCX  
C. NTFY  
D. RSIP  
E. AUCX  
Correct Answer: C  
Explanation/Reference:

**Question 36**
Which option describes what happens to the local copies of Call Detail Records files on the Cisco Unified CM subscribers after they are transferred to the publisher?
Which option describes what happens to the local copies of Call Detail Records files on the Cisco Unified CM subscribers after they are transferred to the publisher?
A. They will be compressed and backed up.
B. They will be deleted.
C. They will be deleted only after the subscriber received notification that the publisher has also deleted the correspondent files.
D. They will remain on the subscriber server until overwritten by new CDR files.
E. They will be compressed and then stored on the subscriber servers.

Correct Answer: B

Explanation/Reference:

**Question 37**

Which three softkeys can be offered on a Cisco IP Phone 7965, running SCCP firmware, when it is in Ring In state?

- A. iDivert
- B. DND
- C. Answer
- D. NewCall
- E. EndCall
- F. CallBack

Correct Answer: ABC

Explanation/Reference:

**Question 38**

What is the default data collection interval for Call Detail Records on Cisco Unified CM?

- A. 60 seconds
- B. 1 second
- C. 1440 seconds
- D. 600 seconds
- E. 3600 seconds

Correct Answer: A

Explanation/Reference:

**Question 39**

Which two phone VPN configurations meet this requirement?

A collaboration engineer is designing a phone VPN infrastructure and the company security team requires Active Directory for authentication. Which two phone VPN configurations meet this requirement? (Choose two)

- A. user ID and password authentication
- B. certificate-only authentication
- C. auto-network-detect authentication
- D. password-only authentication
- E. Cisco ASA Host ID check authentication
- F. Cisco Unified CM user ID and password authentication

Correct Answer: CE

Explanation/Reference:

**Question 40**

Which QoS command allows the engineer to use 70% of the link while maintaining a steady flow?

An engineer is configuring QoS for a 100 Mb WAN link. An ISP SLA was signed to support 70% of the link. Which QoS command allows the engineer to use 70% of the link while maintaining a steady flow?

- A. traffic-shape rate 100000000 70000000 70000000
- B. police cir 70000000 confirm-action transmit exceed-action drop
- C. police 70000000 13125000 confirm-action transmit exceed-action drop
- D. traffic-shape rate 70000000 8750000 8750000

Correct Answer: D

Explanation/Reference:

**Question 41**

Where can a Cisco Unified CM administrator define Billing Application Server(s) for Call Detail Records?

Where can a Cisco Unified CM administrator define Billing Application Server(s) for Call Detail Records?

- A. Cisco Unified Serviceability
- B. Service Parameters in Cisco Unified CM Administration.
- C. Enterprise Parameters in Cisco Unified CM Administration.
- D. Cisco Unified Reporting.
- E. Call Detail Records data collection internal is not a configurable parameter.

Correct Answer: A

Explanation/Reference:

**Question 42**

Which two SCCP call states support the CallBack softkey?

Which two SCCP call states support the CallBack softkey? (Choose two)

- A. On Hook
- B. Remote In Use
- C. Connected Transfer
- D. Ring In
- E. Off Hook
- F. Connected Conference
Correct Answer: AC
Explanation/Reference:

**Question 43**
Which two softkeys can be offered on a Cisco IP Phone 7965, running SCCP firmware, when it is in Connected Conference state?
Which two softkeys can be offered on a Cisco IP Phone 7965, running SCCP firmware, when it is in Connected Conference state? (Choose two)
A. EndCall
B. Transfer
C. Join
D. RmLstC
E. Select
F. Confirm

Correct Answer: AF
Explanation/Reference:

**Question 44**
What is the maximum number of configurable speed dial entries for a Cisco Unified 9971 IP Phone?
What is the maximum number of configurable speed dial entries for a Cisco Unified 9971 IP Phone?
A. 4
B. 199
C. 50
D. 3
E. 2

Correct Answer: B
Explanation/Reference:

**Question 45**
Which two power saving parameters are available on a Cisco 9971 IP Phone only when it is connected to a Cisco switch with the EnergyWise feature enabled?
Which two power saving parameters are available on a Cisco 9971 IP Phone only when it is connected to a Cisco switch with the EnergyWise feature enabled? (Choose two)
A. Enable Power Save Plus
B. Power Negotiation
C. Phone On Time
D. Display on Time
E. LLDP Power Priority
F. Day Display Not Active

Correct Answer: AC
Explanation/Reference:

**Question 46**
Which two statements describe negative effects of this event?
An engineer notices that two Cisco utility Connection servers in a cluster are in split-brain mode. The engineer corrects a network issue that allows the two servers to communicate again. Which two statements describe negative effects of this event? (Choose two)
A. A user calling in to check their voicemail during the recovery may be informed that their messages are not available.
B. Message waiting lights van become out of sync after the split-brain recovery. Forcing the administrator to run an MWI Synchronization.
C. The replication between the nodes becomes defunct, requiring the administrator to run utilities to re-establish intracluster.
D. A message left on the subscriber server during the outage may be lost during the cluster recovery.
E. The replication between the nodes becomes defunct, requiring the administrator to run utilities to re-establish intracluster communication.
F. The Unity Connection Database can become corrupted, causing the need to reinstall the subscriber server.

Correct Answer: AC
Explanation/Reference:

**Question 47**
DRAG DROP
Drag and drop the Cisco Unified CM database replication status values on the left to the correct replication status definition on the right.
Select and Place:

Correct Answer:
Question 48
Which three options are potential reasons the FXO port is not receiving Caller ID?
Refer to the exhibit.

A. "Enable Caller ID" was not configured on the Cisco Unified Communications Manager configuration.
B. The FXO port was configured to "loop-start" instead of "ground-start".
C. The Timing Guard-out parameter is incorrectly set to 1500 ms.
D. "Connection polar opx immediate" was used and does support caller ID.
E. Gateway with IOS 12.4(24T) was used and does not support this feature.
F. The NTFY message contains a Hung Up parameter.

Correct Answer: ACE
Explanation/Reference:

Question 49
Which SIP profile configuration satisfies this request?
An engineer received this requirement from a service provider:

Diversion header should match the network DID [email protected] for Call Forward and transfer scenarios back to PSTN.

Which SIP profile configuration satisfies this request?
A. voice class sip-profiles 200
   request INVITE sip-header Diversion modify "sip:(.*>)" 
   response 200 sip-header Diversion modify "sip:(.*>)" [email protected]

B. voice class sip-profiles 200
   request INVITE sdp-header Diversion modify "sip:(.*>)" 
   response 200 sdp-header Diversion modify "sip:(.*>)" [email protected]

C. voice class sip-profiles 200
   request REINVITE sip-header Diversion modify "sip:(.*>)" [email protected]
   response 200 sip-header Diversion modify "sip:(.*>)" [email protected]

D. voice class sip-profiles 200
   request REINVITE sdp-header Diversion modify "sip:(.*>)" [email protected]
   response 200 sdp-header Diversion modify "sip:(.*>)" [email protected]

Correct Answer: A
Explanation/Reference:

Question 50
Which configurations result in 915556781234 as the final called number?
Refer to the exhibit.

Explanation/Reference:
A call is received on a Cisco Unified Border Element gateway. The debug captures this SIP INVITE message. Which configurations result in 915556781234 as the final called number?

A. voice class uri 1 sip
host ipv4:10.41.11.10
  voice class uri 2 sip
  host ipv4:10.50.40.30
  dial-peer voice 1 voip
  incoming uri via 1
  translation-profile incoming 1
  dial-peer voice 2 voip
  incoming uri via 2
  translation-profile incoming 2
  dial-peer voice 3 voip
  incoming called-number +1T
  translation-profile incoming 3

B. voice class uri 1 sip
host dns:customer.com
  voice class uri 2 sip
  host ipv4:10.50.40.30
  dial-peer voice 1 voip
  incoming uri via 1
  translation-profile incoming 1
  dial-peer voice 2 voip
  incoming uri via 2
  translation-profile incoming 2
  dial-peer voice 3 voip
  incoming called-number +1T
  translation-profile incoming 3

C. voice class uri 1 sip
host ipv4:10.41.11.10
  voice class uri 2 sip
  host ipv4:10.50.40.30
  dial-peer voice 1 voip
  incoming called-number +1[2-9][2-9]…..
  translation-profile incoming 1
  dial-peer voice 2 voip
  incoming called-number +1[2-9][2-9]…..
  translation-profile incoming 2
  dial-peer voice 3 voip
  incoming called-number +1[2-9][2-9]…..
  translation-profile incoming 3

D. voice class uri 1 sip
host dns:customer.com
  voice class uri 2 sip
  host ipv4:10.50.40.30
  dial-peer voice 1 voip
  incoming uri via 1
  translation-profile incoming 1
  dial-peer voice 2 voip
  incoming uri via 2
  translation-profile incoming 2
  dial-peer voice 3 voip
  incoming called-number +1T
  translation-profile incoming 3

E. voice class uri 1 sip
host ipv4:10.41.11.10
  voice class uri 2 sip
  host ipv4:10.50.40.30
  voice class uri 3 sip
A Cisco Unified CME administrator is configuring SNR for a line and has these requirements:
- The remote phone should receive the call after the local phones ring for 10 seconds.
- The ANI displayed on the remote phones should be the local extension number.

Which two configuration commands complete these requirements? (Choose two.)
A. snr 92875421 delay 15 timeout 10
B. snr 92875421 delay 10 timeout 20
C. snr calling-number local
D. snr calling-number remote
E. snr answer-too-soon 10

Correct Answer: BC

Explanation/Reference:

Question 51
Which two configuration commands complete these requirements?
Refer to the exhibit.

ephone-dn 3 octo-line
  number 1645
  label 1645
  description John Doe
  name John Doe
  mobility
!
ephone-template 1
  softkeys idle Redial Newcall Mobility Cfwdall Pickup Dnd
  softkeys connected Endcall Hold Mobility
!
ephone 3
  device-security-mode none
  mac-address 0023.5EB7.2949
  ephone-template 1
  type 7962
  button 1:3

http://www.aoowe.com/practice-400-051-3156.html

http://www.aoowe.com/
A PSTN caller initiates an inbound call. Which two dial peers can be selected as inbound dial peers? (Choose two.)

A. dial-peer voice 100 pots
answer-address [2-9]..[2-9]…$
voice-port 0/0/0:23

B. dial-peer voice 200 pots
destination-pattern [2-9]..[2-9]..[2-9]…$
voice-port 0/1/0:15

C. dial-peer voice 300 pots
incoming called-number 704[2-9]…$
voice-port 0/1/0:15

D. dial-peer voice 400 pots
answer-address 672[2-9]…$
voice-port 0/0/0:15

E. dial-peer voice 500 pots
incoming called-number 6..[2345689]…$
voice-port 0/1/0:15

Correct Answer: BE
Explanation/Reference:

Question 53
Which two external databases can be used to support that functionality?
A collaboration engineer is designing an Cisco IM&P implementation to support instant messaging logging for compliance. Which two external databases can be used to support that functionality? (Choose two.)

A. Oracle database
B. MySQL database
C. Microsoft SQL database
D. PostgreSQL database
E. Informix SQL database

Correct Answer: AD
Explanation/Reference:

Question 54
What two methods are used to set up the integration between Active Directory, Cisco Unified CM, and IM&P?
A customer has a single Active Directory domain with users in various email domains. Each user is associated to only one email domain. The customer wants their users to federate to external organizations using their email addresses. What two methods are used to set up the integration between Active Directory, Cisco Unified CM, and IM&P? (Choose two.)

A. CUCM LDAP Attribute for User ID set to sAMAccountName, CUCM LDAP Directory URI set to mail, IM Address Scheme set to Directory URI
B. CUCM LDAP Attribute for User ID set to mail, IM Address Scheme set to User ID
C. CUCM LDAP Attribute for User ID set to sAMAccountName, CUCM LDAP Directory URI set to msRTCSIP-primaryuseraddress, IM Address Scheme set to Directory URI
D. CUCM LDAP Attribute for User ID set to mail, CUCM LDAP Directory URI set to mail, IM Address Scheme set to Directory URI
E. CUCM LDAP Attribute for User ID set to mail, IM Address Scheme set to mail

Correct Answer: AD
Explanation/Reference:

Question 55
Which two steps resolve this issue?
Multiple Jabber for Windows users are having problems logging into the voicemail server. The Cisco Unity Connection administrator has reset the password and emailed them the new credentials, as well as the instructions about how to reset them in Jabber. The users cannot see the Phone Accounts tab under Jabber settings to complete the instructions. Which two steps resolve this issue? (Choose two.)

A. In the Cisco Unified CM Jabber Service Profile, change the Credentials source for voicemail service to "not set".
B. In Cisco Unified CM, create a MailStore service and assign it to the Jabber Service Profile as Primary.
C. In the IM&P server CCMCIP Profile, uncheck the “Make this the default CCMCIP Profile for the system”.
D. In the IM&P server Enterprise Parameters Configuration, enable the Phone Personalization parameter.
E. In the Cisco Unified CM Jabber Service Profile, uncheck “Make this the default service profile for the system”.

Correct Answer: AD
Explanation/Reference:
Correct Answer: A
Explanation/Reference:

**Question 56**
Which four types of licensing and database servers support this requirement?
A Cisco Unified Contact Center Express manager wants to add database integration to the self-service interactive voice response application. Which four types of licensing and database servers support this requirement? (Choose four.)
A. The server must have enhanced licensing.
B. The server must have premium licensing.
C. A server running Sybase Adaptive Server is required.
D. A server running Oracle is required.
E. A server running PostgreSQL is required.
F. A server running SAP SQL server is required.
G. A server running Microsoft SQL server is required.
H. The server must have standard licensing.
Correct Answer: BCDG
Explanation/Reference:

**Question 57**
Which three characteristics can be reported about the call distribution?
Refer to the exhibit.
A Cisco collaboration engineer is writing a report to summarize the call distribution characteristics in a Cisco Unified Contact Center Express queue. Which three characteristics can be reported about the call distribution? (Choose three.)
A. This queue will not work because no prompt has been selected.
B. Calls to this queue can be distributed in a round-robin manner between agents.
C. Agents that are answering calls for this queue can answer calls to other queues if available.
D. Agents in this queue are expected to finish (wrap-up) a call within 60 seconds.
E. Calls to this queue are handled in the order they were received unless prioritized by the script.
F. Changing the queue name from SupportQueue to Support01 requires updates to the script.
G. Agents logged in to this queue automatically receive calls without the need to do anything else (automatic work).
Correct Answer: CEF
Explanation/Reference:

**Question 58**
Which three steps remove the site from the network?
A company is decommissioning a site where a Cisco Unity Connection cluster resides. This cluster is part of a larger network of Unity Connection servers linked using HTTPS networking. Which three steps remove the site from the network? (Choose three.)
A. Determine if the Unity Connection cluster to be decommissioned sits between the hub and another Unity Connection site in the hub-and-spoke topology.
B. Remove the Unity Connection primary server from the HTTPS network on each node in the cluster.
C. Remove all servers in the Unity Connection cluster from the other clusters in the HTTPS network.
D. Update any downstream Unity Connection locations so that they link with a Unity Connection that will continue to have access to the hub location.
E. Remove the existing link to the remaining Unity Connection locations subtree and add new links to locations that will remain connected to the hub.
F. Update any remote call handlers and interview handlers that targeted the users on the location as well as any location downstream from the commissioned site to be removed.
Correct Answer: AEF
Explanation/Reference:

**Question 59**
How does the administrator grant these rights to the user?
A Cisco Unity Connection administrator receives a request from a user who wants the ability to change the caller input option 0 in their voicemail box as needed without calling for support. How does the administrator grant these rights to the user?

A. The administrator can set the caller input to “Transfer to alternate contact number” so the user can log into their voicemail account through the TUI and set their alternate contact number.
B. The administrator can set the caller input to “Transfer to alternate contact number” so the user can log into their voicemail account through their Cisco PCA page and set their alternate contact number.
C. The administrator can create a new call handler of which the user is an owner. The user controls the destination of that call handler by logging into the call handler via greetings administrator.
D. The administrator informs the user that this feature is a built-in option to the user Cisco PCA page under caller input.
E. The administrator informs the user that this feature is a built-in option for the user in the TUI under personal settings.

Correct Answer: A

Explanation/Reference:

Question 60
What does an outside caller hear when calling a user and forwarding to Cisco Unity Connection?
Refer to the exhibit.

A. The caller hears the Emergency greeting, followed by the voicemail greeting of the user they originally called.
B. The caller hears the message “Emergency Message is not available,” followed by the voicemail greeting of the user they originally called.
C. The caller hears the emergency greeting followed by the Opening Greeting message.
D. The caller hears the Main Message greeting and then the call is disconnected.

Correct Answer: B

Explanation/Reference:

Question 61
Which two pieces of information can be gathered about the calls traversing these border elements?
Refer to the exhibit.

A. The total number of calls is 150.
B. The number of nonnative calls is 70.
C. The number of native calls is 50.
D. The number of calls preserved is 220.
E. The total number of active calls is 100.

Correct Answer: AB

Explanation/Reference:

Question 62
Which two commands enable this functionality?

The Cisco Unified Border Element is configured using high availability with the Hot Standby Routing Protocol. Which two pieces of information can be gathered about the calls traversing these border elements? (Choose two.)
A. The total number of calls is 150.
B. The number of nonnative calls is 70.
C. The number of native calls is 50.
D. The number of calls preserved is 220.
E. The total number of active calls is 100.

Correct Answer: AB

Explanation/Reference:
A collaboration engineer has set up SAF on a Cisco IOS router to advertise and accept SAF information during a maintenance window. Which two commands enable this functionality? (Choose two.)
A. enroll callcontrol wildcarded
B. advertise callcontrol 1
C. subscribe callcontrol wildcarded
D. register callcontrol wildcarded
E. publish callcontrol 1
F. distribute callcontrol 1

Correct Answer: CE

**Explanation/Reference:**

**Question 63**
Which option describes the issue?
A collaboration engineer has just implemented SAF as a hub-and-spoke network. The hub uses its loopback interface for SAF advertisements. Updates are coming into the hub router, but are not being advertised out. Which option describes the issue?
A. Multicast is not enabled across the WAN.
B. SAF is set up on a VRF.
C. SAF username/password are incorrect.
D. The autonomous system is mismatched.
E. Split horizon is enabled.

Correct Answer: E

**Explanation/Reference:**

**Question 64**
Which option describes the security encryption status of this active call on a Cisco IP phone?
Refer to the exhibit.

Which option describes the security encryption status of this active call on a Cisco IP phone?
A. unencrypted call signaling and media
B. encrypted call signaling but unencrypted call media
C. encrypted call media but unencrypted call signaling
D. encrypted call signaling and media
E. Not enough information provided to answer this question.

Correct Answer: D

**Explanation/Reference:**

**Question 65**
Which two phone security functions are available to this Cisco IP phone? (Choose two.)
Refer to the exhibit.

Which two phone security functions are available to this Cisco IP phone? (Choose two.)
A. default Authentication of TFTP downloaded files using a signing key
B. encryption of TFTP Configuration files using a signing key
C. encrypted call signaling but unencrypted call media
D. encrypted call media but unencrypted call signaling
E. encrypted call signaling and media
F. local trust verification on the phone

Correct Answer: AB

**Explanation/Reference:**

**Question 66**
How many failed token password attempts have occurred on this Cisco CTL client?
Refer to the exhibit.
How many failed token password attempts have occurred on this Cisco CTL client?
A. 4
B. 9
C. 14
D. 19
E. 24
Correct Answer: C
Explanation/Reference:

Question 67
Which configuration file does a Cisco IP phone with MAC address 1111.2222.3333 request from the TFTP server when an Initial Trust List file is present?
A. SEP111122223333.cnf.xml
B. SEP111122223333.cnf
C. SEP111122223333.cnf.xml.sgn
D. SEPDefault.cnf.xml.sgn
E. SEP111122223333.cnf.xml.enc.sgn
Correct Answer: C
Explanation/Reference:

Question 68
Which certificate file contains the private key used to sign the TFTP configuration file for download authentication with Initial Trust List enabled IP phones? Refer to the exhibit.
A. PUB.pem tomcat-trust trust-cert
B. SUB.pem CallManager-trust trust-cert
C. CAPF.pem CAPF cert
D. TVS.pem TVS cert
E. CallManager.pem CallManager cert
Correct Answer: E
Explanation/Reference:
Question 69
Which four requirements are mandatory to enable a mixed mode Cisco Unified CM cluster?

Which four requirements are mandatory to enable a mixed mode Cisco Unified CM cluster? (Choose four.)
A. Cisco CTL Provider Service activated and enabled
B. Cisco Certificate Authority Proxy Function activated and enabled
C. Cisco Trust Verification activated and enabled
D. Cisco CTL client
E. a minimum of one USB e-token
F. a minimum of two USB e-token
G. a minimum of one soft e-token

Correct Answer: ABDF
Explanation/Reference:

Question 70
Which Cisco Unified CM service is installed by default and authenticates certificates on behalf of IP phones and other endpoints?

Which Cisco Unified CM service is installed by default and authenticates certificates on behalf of IP phones and other endpoints?
A. Cisco CTL Provider
B. Cisco Certificate Authority Proxy Function
C. Cisco Trust Verification
D. Cisco CallManager
E. Cisco TFTP

Correct Answer: C
Explanation/Reference:

Question 71
Which three parameters are requested in an Audit Endpoint message from a Cisco Unified CM to an endpoint on a MGCP gateway?

Which three parameters are requested in an Audit Endpoint message from a Cisco Unified CM to an endpoint on a MGCP gateway? (Choose three.)
A. Bearer Information
B. Call ID
C. Capabilities
D. Connection ID
E. Connection Mode
F. Connection Parameters
G. Request Identifier
H. Observed Events

Correct Answer: CDG
Explanation/Reference:

Question 72
Which RAS message is sent next by the H.323 gateway?

Refer to the exhibit.

Debug RAS output is logged on a H.323 gateway. Which RAS message is sent next by the H.323 gateway?
A. ARQ
B. BRQ
C. IRQ
D. LRQ
E. RRQ

Correct Answer: E
Explanation/Reference:

Question 73
Which two options are the correct Cisco IOS Gatekeeper configuration that could produce the output shown in the exhibit?

Refer to the exhibit.

<table>
<thead>
<tr>
<th>Zone</th>
<th>Gk</th>
<th>mcc</th>
<th>gw</th>
<th>10.1.1.1</th>
<th>10.1.1.2</th>
<th>eToken Prefix Table</th>
<th>gw-type-prefix 1 default-technology</th>
<th>no shutdown</th>
<th>zone local GK cciecollab.com</th>
<th>Zone remote HQGK_2 cciecollab.com 10.1.1.2</th>
</tr>
</thead>
</table>

10.1.1.1 and 10.1.1.2 are node IP addresses of a Cisco Unified CM cluster. Which two options are the correct Cisco IOS Gatekeeper configuration that could produce the output shown in the exhibit? (Choose two.)
A. gw-type-prefix 1 default-technology
B. no shutdown
C. zone local GK cciecollab.com
D. Zone remote HQGK_2 cciecollab.com 10.1.1.2

Correct Answer: ABC
Explanation/Reference:
Question 74
Which two statements describe the correct Gatekeeper Information parameters on Cisco Unified CM H.225 Trunk (Gatekeeper Controlled) configuration page that could produce the output shown in the exhibit?

Refer to the exhibit.

A. Default Technology Prefix is 1*.
B. Technology Prefix is 1.
C. H.323 IDs are HQGK_1 and HQGK_2.
D. H.323 ID is HQGK.
E. Technology Prefix is 1*.
F. Zone name is HQGK.

Correct Answer: BE
Explanation/Reference:

Question 75
Which statement describes the correct Cisco Unified CM configurations that produced the output shown in the exhibit?

Refer to the exhibit.

A. Device Name on the Cisco Unified CM Gatekeeper configuration page is HQGK.
B. Device Name on the Cisco Unified CM H.225 Trunk (Gatekeeper Controlled) configuration page is HQGK.
C. Device Name on the Cisco Unified CM H.225 Trunk (Gatekeeper Controlled) configuration page is HQGK_1,HQGK_2.
D. Device Name on the Cisco Unified CM Gatekeeper configuration page is HQGK_1,HQGK_2.
E. Not enough information has been provided to answer this question.

Correct Answer: B
Explanation/Reference:

Question 76
Which Cisco Unified CM service is responsible for periodically checking disk usage and deleting old Call Management Records files?

A. Cisco CallManager
B. Cisco CDR Agent
C. Cisco CDR Repository Manager
D. Cisco SOAP - CallRecord Service
E. Cisco Extended Functions

Correct Answer: C
Explanation/Reference:

Question 77
Which directory path on Cisco Unified CM publisher is used to temporarily store the Call Detail Records collected from other nodes until they are processed by the CDR Repository Manager?

A. car/yyyymmdd
B. preserve/yyyymmdd
C. cdr/yyyymmdd
D. collected/yyyymmdd
E. processed/yyyymmdd

Correct Answer: B
Explanation/Reference:
Question 78
Which Cisco Unified CM service is responsible for writing Call Management Records into the CDR Analysis and Reporting database?
A. Cisco CDR Agent
B. Cisco CAR DB
C. Cisco CDR Repository Manager
D. Cisco CAR Scheduler
E. Cisco Extended Functions

Correct Answer: D
Explanation/Reference:

Question 79
Which Cisco Unified CM service is responsible for writing Call Detail Records into flat files?
A. Cisco CallManager
B. Cisco CDR Agent
C. Cisco CDR Repository Manager
D. Cisco SOAP – CallRecord Service
E. Cisco Extended Functions

Correct Answer: A
Explanation/Reference:

Question 80
Where can a Cisco Unified CM administrator define Call Detail Records data collection interval?
A. Cisco Unified CM Administration Service Parameters
B. Cisco Unified CM Administration Enterprise Parameters
C. Cisco Unified Serviceability
D. Cisco Unified Reporting
E. Call Detail Records data collection interval is not a configurable parameter.

Correct Answer: B
Explanation/Reference:

Question 81
Which option is the minimum Cisco Unified CM Service Parameter configuration that is needed to ensure compliance to this policy?
The Information Technologies policy of your company mandates logging of all calls that last less than one second in Call Detail Records. Which option is the minimum Cisco Unified CM Service Parameter configuration that is needed to ensure compliance to this policy?
A. Set CDR Enabled Flag to True.
B. Set CDR Log Calls with Zero Duration Flag to True.
C. Set CDR Enabled Flag and CDR Log Calls with Zero Duration Flag to True.
D. Set CDR Enabled Flag to True and set Call Diagnostics Enabled to Enable Only When CDR Enabled Flag is True.
E. Leave CDR Enabled Flag and Call Diagnostics Enabled to their default settings.

Correct Answer: C
Explanation/Reference:

Question 82
Which option is the minimum Cisco Unified CM Service Parameter configuration that is needed to ensure compliance to this policy?
The Information Technologies policy of your company mandates logging of all unsuccessful calls that resulted in reorder tone in Call Detail Records. Which option is the minimum Cisco Unified CM Service Parameter configuration that is needed to ensure compliance to this policy?
A. Set CDR Enabled Flag to True.
B. Set CDR Log Calls with Zero Duration Flag to True.
C. Set CDR Enabled Flag and CDR Log Calls with Zero Duration Flag to True.
D. Set CDR Enabled Flag to True and set Call Diagnostics Enabled to Enable Regardless of CDR Enabled Flag.
E. Leave CDR Enabled Flag and Call Diagnostics Enabled to their default settings.

Correct Answer: A
Explanation/Reference:

Question 83
Which Cisco Unified CM service interfaces with Cisco IP Phones to allow users to report audio and other general problems on the phones?
Refer to the exhibit.
Which Cisco Unified CM service interfaces with Cisco IP Phones to allow users to report audio and other general problems on the phones?
A. Cisco Serviceability Reporter
B. Cisco Audit Event Service
C. Cisco CallManager Serviceability
D. Cisco Extended Functions
E. Cisco RTMT Reporter Servlet

Correct Answer: D
Explanation/Reference:

Question 84
Which statement about the dialing key strokes that allow the owner of this phone to reach directory number 2000 is true?
Refer to the exhibit.

A. Press the last button on the right hand side of the phone screen.
B. There is no way to speed dial to directory number 2000 because the speed dial entry is not assigned.
C. Press 7 on the phone keypad, followed by the Dial softkey.
D. Press 6 on the phone keypad, followed by the Dial softkey.
E. Press 5 on the phone keypad, followed by the AbbrDial softkey.

Correct Answer: E
Explanation/Reference:

Assume there are no classes of service restrictions and all numbers shown are reachable from this Cisco Unified IP 7965 Phone. Which statement about the dialing key strokes that allow the owner of this phone to reach directory number 2000 is true?
A. Press the last button on the right hand side of the phone screen.
B. There is no way to speed dial to directory number 2000 because the speed dial entry is not assigned.
C. Press 7 on the phone keypad, followed by the Dial softkey.
D. Press 6 on the phone keypad, followed by the Dial softkey.
E. Press 5 on the phone keypad, followed by the AbbrDial softkey.

Correct Answer: E
Explanation/Reference:

Question 85
Which Cisco Unified IP Phone supports the most number of speed dial phone buttons?

A. Cisco Unified 7961
B. Cisco Unified 7965
C. Cisco Unified 7975
D. Cisco Unified 9951
E. Cisco Unified 9971

Correct Answer: C
Explanation/Reference:
Question 86
Which three softkeys can be offered on a Cisco IP Phone 7965, running SCCP firmware, when it is in Remote In Use state?
Which three softkeys can be offered on a Cisco IP Phone 7965, running SCCP firmware, when it is in Remote In Use state? (Choose three.)
A. Resume
B. EndCall
C. Select
D. Barge
E. NewCall
F. cBarge
G. Join
Correct Answer: DEF
Explanation/Reference:

Question 87
Which option is a mandatory softkey for a Cisco IP 7965, running SCCP firmware, in the Off Hook call state?
Which option is a mandatory softkey for a Cisco IP 7965, running SCCP firmware, in the Off Hook call state?
A. Redial
B. NewCall
C. EndCall
D. CfwdAll
E. There is no mandatory softkey in the Off Hook call state.
Correct Answer: E
Explanation/Reference:

Question 88
Which two SCCP call states support the MeetMe softkey?
Which two SCCP call states support the MeetMe softkey? (Choose two.)
A. On Hook
B. Connected
C. On Hold
D. Off Hook
E. Ring Out
F. Connected Conference
Correct Answer: AD
Explanation/Reference:

Question 89
Which three softkeys can be offered on a Cisco IP Phone 7965, running SCCP firmware, when it is in On Hold state?
Which three softkeys can be offered on a Cisco IP Phone 7965, running SCCP firmware, when it is in On Hold state? (Choose three.)
A. Select
B. Confrn
C. NewCall
D. EndCall
E. iDivert
F. Park
G. Hold
Correct Answer: ACE
Explanation/Reference:

Question 90
Which option is a mandatory LDAP attribute for a user to be synchronized to Cisco Unified Communications Manager?
Which option is a mandatory LDAP attribute for a user to be synchronized to Cisco Unified Communications Manager?
A. uid
B. telephoneNumber
C. employeeNumber
D. sn
E. mail
Correct Answer: D
Explanation/Reference:

Question 91
What is the status of this user in the Unified CM database at 1:00 am on March 2nd 2014?
Tom Lee is an active user in a Cisco Unified CM deployment with fully functional LDAP synchronization and authentication to an Active Directory. Daily resynchronization is set at 11:00 pm. At 8:00 am on March 1st 2014, this user was deleted from the AD. What is the status of this user in the Unified CM database at 1:00 am on March 2nd 2014?
A. active
B. inactive
C. delete pending
D. awaiting authorization
E. permanently deleted
Correct Answer: B
Explanation/Reference:

Question 92
What will Tom Lee experience when he attempts to log in Extension Mobility at an IP phone and then access his Unified CM User Options page on his PC, at 9:00 am on March 1st 2014?
Tom Lee is an active user in a Cisco Unified CM deployment with fully functional LDAP synchronization and authentication to an Active Directory. Daily resynchronization is set at 11:00 pm. At 8:00 am on March 1st 2014, this user was deleted from the AD.

What will Tom Lee experience when he attempts to log in Extension Mobility at an IP phone and then access his Unified CM User Options page on his PC, at 9:00 am on March 1st 2014?
A. Extension Mobility will not work, but the User Options page will work.
B. Extension Mobility and the User Options page will not work.
C. Extension Mobility will work, but the User Options page will not work.
D. Extension Mobility and the User Options page will work.
E. The information provided is insufficient to answer this question.

Correct Answer: C
Explanation/Reference:

Question 93
Which Cisco Unified CM Application user is created by default and used by Cisco Unified CM Extension Mobility?
A. CCMAdministrator
B. EMSysUser
C. TabSyncSysUser
D. CCMSysUser
E. CTIWUUser

Correct Answer: D
Explanation/Reference:

Question 94
Which two options are Power Save configuration parameters for Cisco 9971 IP Phones?
(A. Phone On Time
B. Phone Off Idle Timeout
C. Day Display Not Active
D. Enable Audio Alert
E. Enable Power Save Plus
F. Display on Duration

Correct Answer: CF
Explanation/Reference:

Question 95
Which three conditions will a Cisco 9971 IP Phone request the “xmldefault.cnf.xml” file from a TFTP server in a Cisco Unified CM cluster?
(A. The phone is registered to the CUCM cluster but need to update its firmware
B. The phone is attempting to register to the CUCM cluster for the first time
C. Auto-registration is disabled on CUCM cluster
D. The phone has not yet been defined in the CUCM database
E. The phone is attempting to change from SIP firmware to SCCP firmware
F. Auto-registration is enabled on CUCM cluster

Correct Answer: BDF
Explanation/Reference:

Question 96
What caused this message on a Cisco 9971 IP phone, connected to a Cisco 3750X PoE switch, when a mobile phone is plugged into the IP Phone’s back USB port?

Refer to the exhibit.

Correct Answer: BDF
Explanation/Reference:
What caused this message on a Cisco 9971 IP phone, connected to a Cisco 3750X PoE switch, when a mobile phone is plugged into the IP Phone’s back USB port?
A. The back USB port only supports Cisco USB devices such as a Cisco Unified Video camera
B. USB classes for this USB port are not properly configured
C. The USB port is not enabled by the administrator
D. The mobile phone is requesting more power than the USB port could provide
E. USB devices are not supported when the IP phone is powered by a Cisco PoE switch

Correct Answer: D
Explanation/Reference:

Question 97
Which three Ethernet Setup Administrator Settings are manually configurable locally on the Cisco 9971 IP phone?
Refer to the exhibit.

Which three Ethernet Setup Administrator Settings are manually configurable locally on the Cisco 9971 IP phone? (Choose three)
A. Operational VLAN Id
B. Admin VLAN Id
C. PC VLAN
D. SW Port Setup
E. PC Port Setup

Correct Answer: BDE
Explanation/Reference:

Question 98
Which out-of-dialog SIP OPTIONS ping response put dial-peer tag 1111 into its current operational state?
Refer to the exhibit.

Which out-of-dialog SIP OPTIONS ping response put dial-peer tag 1111 into its current operational state?
A. 401 Unauthorized
B. 505 Version Not Supported
C. 406 Not Acceptable
D. 482 Loop Detected
E. 500 Server Internal Error

Correct Answer: B
Explanation/Reference:

Question 99
Which statement describes a video conference viewing mode on a Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3 that is configured to work with Cisco Unified Communications Manager?

Which statement describes a video conference viewing mode on a Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3 that is configured to work with Cisco Unified Communications Manager?
A. Video of one participant is displayed to all other video capable participants in a round-robin manner.
B. Video of the loudest speaker is displayed across all video capable participants.
C. Video of one participant, except for those with mute enabled, is displayed to all other video capable participants in a round-robin manner.
D. The dedicated conference lecturer can one participant at a time, while all others can only see the lecturer.
E. Video of one participant is displayed to all other video capable participants in a random manner using an algorithm hard-coded in Cisco IOS.

Correct Answer: B
Explanation/Reference:

Question 100
Which statement describes how much of the DSP resources are reserved for video conference when voice-service dsp-reservation 40 is configured on a Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3?

Which statement describes how much of the DSP resources are reserved for video conference when voice-service dsp-reservation 40 is configured on a Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3?
A. 60% of the total available DSP resources
B. 40% of the total available DSP resources
C. 50% of the total available DSP resources
D. Video conferencing resources are reserved dynamically by Cisco IOS and cannot be changed.
E. This command is used for voice resource reservation only.

Correct Answer: B
Explanation/Reference:
Correct Answer: A
Explanation/Reference:

**Question 101**
Which Cisco IOS multipoint video conferencing profile reserves DSPs when it is created in the configuration?
A. flex mode video
B. guaranteed-audio
C. rendezvous
D. heterogeneous
E. guaranteed-video
Correct Answer: D
Explanation/Reference:

**Question 102**
Which Cisco IOS multipoint video conferencing profile attempts to reserve DSPs only when it is activated with an actual conference?
A. homogeneous
B. guaranteed-audio
C. rendezvous
D. heterogeneous
E. flex mode video
Correct Answer: B
Explanation/Reference:

**Question 103**
Which TFTP server address selection option has the highest precedence on Cisco SCCP IP phones using firmware release 8.0(6) or later?
A. a manually configured alternate TFTP option on the phone
B. the first Option 150 IP address received from the DHCP server
C. the first Option 66 dotted decimal IP address received from the DHCP server
D. the first IPv6 TFTP Server address received from the DHCP server
E. the value of next-server IP address in the boot-up process
Correct Answer: A
Explanation/Reference:

**Question 104**
Which two clock rates does Performance Monitor use to calculate RTP jitter values? (Choose two.)
A. PCMU (G.711 mu-law) , 8000 Hz
B. PCMU (G.711 mu-law) , 32000 Hz
C. PCMA (G.711 A-law) , 16000 Hz
D. H.263 , 90,000 Hz
E. H.263 , 64,000 Hz
Correct Answer: AD
Explanation/Reference:

**Question 105**
Which two QoS guidelines are recommended for provisioning interactive video traffic? (Choose two.)
A. Latency should be no more than 4 seconds.
B. Overprovision interactive video queues by 20% to accommodate bursts.
C. Loss should be no more than 5%.
D. Interactive video should be marked to DSCP CS4.
E. Jitter should be no more than 30 ms.
Correct Answer: BE
Explanation/Reference:

**Question 106**
Which two rules apply to MMOH in SRST? (Choose two.)
A. A maximum of three MOH groups are allowed.
B. Cisco Unified SRST voice gateway allows you to associate phones with different MOH groups on the basis of their IP address to receive different MOH media streams.
C. A maximum of five media streams are allowed.
D. Cisco Unified SRST voice gateway allows you to associate phones with different MOH groups on the basis of their MAC address to receive different MOH media streams.
E. Cisco Unified SRST voice gateway allows you to associate phones with different MOH groups on the basis of their extension numbers to receive different MOH media streams.
Correct Answer: CE
Explanation/Reference:

**Question 107**
Which two settings must be the same between the backup source and restore target with DRS in Cisco Unified Communications Manager? (Choose two.)
A. Server Hostname
B. Server IP Address
C. Cluster Security Password
D. NTP Servers
E. Domain Name
F. Certificate Information

Correct Answer: AB
Explanation/Reference:

Question 108
Which two steps must be taken when configuring EMCC?
Which two steps must be taken when configuring EMCC? (Choose two.)
A. An SFTP server that all clusters share must be set up.
B. Certificates from all remote clusters must be imported into each cluster.
C. Cross-cluster Enhanced Location CAC must be configured.
D. End users must be configured to only exist in their home cluster.
E. A device pool for EMCC phones must be configured.
F. Define MLPP domains.

Correct Answer: AB
Explanation/Reference:

Question 109
Which three components are required when configuring the Cisco Unified Communications Manager for time-of-day routing?
Which three components are required when configuring the Cisco Unified Communications Manager for time-of-day routing? (Choose three)
A. Partition
B. Time Period
C. Time Schedule
D. Time Zone
E. Date Time Group

Correct Answer: ABC
Explanation/Reference:

Question 110
Which three message types for RTCP are valid?
Which three message types for RTCP are valid? (Choose three.)
A. sender report
B. end of participation
C. source description
D. sender codec
E. receiver packets
F. average MOS

Correct Answer: ABC
Explanation/Reference:

Question 111
Which three locations does the TFTP server search when a device requests a configuration file from a TFTP server?
In a Cisco Unified Communications Manager system, which three locations does the TFTP server search when a device requests a configuration file from a TFTP server? (Choose three.)
A. internal caches
B. local disk
C. alternate file server
D. NFS server
E. FTP server
F. load server

Correct Answer: ABC
Explanation/Reference:

Question 112
How many bearer channels are available to carry voice traffic?
Refer to the exhibit.
From this NFAS-enabled T1 PRI configuration on a Cisco IOS router, how many bearer channels are available to carry voice traffic?

A. 91
B. 92
C. 93
D. 94
E. 95

Correct Answer: D
Explanation/Reference:

Question 113
When DSP oversubscription occurs on a Cisco IOS router using DSP modules that are based on the C5510 chipset, what will happen when an analog phone connected to a FXS port goes off-hook?

A. A fast busy tone will be played.
B. A slow busy tone will be played.
C. A network busy tone will be played.
D. A dial tone will be played, but digits will not be processed.
E. No tone will be played.

Correct Answer: E
Explanation/Reference:

Question 114
Which codec complexity mode, when deployed on Cisco IOS routers with DSPs using the C5510 chipset, supports the most G.711 calls per DSP?

A. Low
B. Medium
C. High
D. Secure
E. Flex

Correct Answer: E
Explanation/Reference:

Question 115
Which description of what will happen when the user of IP phone B presses the Transfer soft key is true?

Refer to the exhibit.

All displayed devices are registered to the same Cisco Unified Communications Manager server and the phones are engaged in an active call. Assume that the...
provided configurations exist at the phone line level and multicast MOH is disabled cluster wide.

Which description of what will happen when the user of IP phone B presses the Transfer soft key is true?
A. IP phone A user hears audio source 3 from MOH server A.
B. IP phone A user hears audio source 4 from MOH server B.
C. IP phone A user hears audio source 3 from MOH server B.
D. IP phone A user hears tone on-hold beep tones.
E. IP phone A user hears no on-hold music or beep tones.

Correct Answer: E
Explanation/Reference:

Question 116
Which statement about external media supportability is true for this migration?
Company ABC is planning to migrate from Cisco MCS-hosted Cisco Unified Communications Manager applications to Cisco UC on UCS B-Series servers. Which statement about external media supportability is true for this migration?
A. The Cisco Music on Hold USB audio sound card on the MCS servers will continue to work on the USB ports on the UCS server.
B. The Cisco Music on Hold USB audio sound card on the MCS servers will continue to work through the USB ports on the Cisco UCS server KVM dongle cable adaptor connected to the front of the UCS server.
C. The Cisco Music on Hold USB audio sound card on the MCS servers will not work on the UCS server.
D. The Cisco Music on Hold USB audio sound card can be mapped to a virtual USB port on a VMware virtual machine on the UCS server.
E. The Cisco Music on Hold USB audio sound card can be mapped to a virtual serial port on a VMware virtual machine on the UCS server.

Correct Answer: C
Explanation/Reference:

Question 117
Which option describes how you can show the same contacts in your Jabber for Windows on-premise client as you do on the corporate directory of your IP phone?
When users are inside the corporate firewall, the client can use either UDS or LDAP for contact resolution. If you deploy LDAP within the corporate firewall, Cisco recommends that you synchronize your LDAP directory server with Cisco Unified Communications Manager to allow the client to connect with UDS when users are outside the corporate firewall.

Correct Answer: A
Explanation/Reference:

Explanation/Reference:

Correct Answer: A
Explanation/Reference:

Question 118
Which three statements about configuring partitioned intradomain federation to Lync are true? (Choose three.)
Which three statements about configuring partitioned intradomain federation to Lync are true?
A. Intradomain federation to Lync is only possible using SIP.
B. IM&P and Lync should federate to any required presence domain.
C. You must update the URIs of any users migrated from Lync to IM&P to match the Cisco Unified Presence Server SIP URI format.
D. A static route must be added to point the local presence domain to the Lync server.
E. Microsoft RCC must be enabled.
F. The Enable use of Email Address when Federating option can be turned on if SIP URIs are different between IM&P and Lync.

Correct Answer: ACD
Explanation/Reference:

Explanation:

Please refer to the link for more information:

Question 119
Which three issues prevent a customer from seeing the presence status of a new contact in their Jabber contact list?
Which three issues prevent a customer from seeing the presence status of a new contact in their Jabber contact list?
A. incoming calling search space on SIP trunk to IM&P
B. IM&P incoming ACL blocking inbound status
C. subscribe calling search space on SIP trunk to IM&P
D. PC cannot resolve the FQDN of IM&P
E. Owner user ID is not set on device.
F. Primary DN is not set in end user configuration for that user.
G. Subscriber calling search space is not defined on user’s phone.

Correct Answer: BCD
Explanation/Reference:


Question 120
What is the appropriate next step?
Two Jabber clients are unable to pass instant messages between each other. What is the appropriate next step?
A. Review XCP router logs.
B. Open port 5060 on the firewalls between the PCs and the IM&P servers.
C. Review SIP proxy logs.
D. Review Help > Show Connection Status in each Jabber client, and pull logs as necessary.

Correct Answer: C
Explanation/Reference:

Correct Answer: A
Explanation/Reference:
Explanation:
The XCP Router is the core communication functionality on the Cisco Unified Presence server. It provides XMPP-based routing functionality on Cisco Unified Presence: it routes XMPP data to the other active XCP services on Cisco Unified Presence, and it accesses SDNS to allow the system to route XMPP data to Cisco Unified Presence users. The XCP router manages XMPP sessions for users, and routes XMPP messages to and from these sessions.

Question 121
Which three services must be stopped to change the IM & Presence service default domain setting of DOMAIN.NOT.SET?
Which three services must be stopped to change the IM & Presence service default domain setting of DOMAIN.NOT.SET? (Choose three.)
A. Cisco XCP Router
B. Cisco Intercluster Sync Agent
C. Cisco XCP Authentication Service
D. Cisco SIP Proxy
E. Cisco Presence Engine
F. Cisco AXL Web service
Correct Answer: ADE
Explanation/Reference:
Explanation:
Change the Domain Value
Follow this procedure if you want to change the domain value (from one valid domain value to another valid IP proxy domain value). This procedure is applicable if you have a DNS or non-DNS deployment.
Procedure
Step 1 Stop the Cisco SIP Proxy, Presence Engine and XCP Router services on IM and Presence on all nodes in your cluster.
Step 2 On the publisher node, perform the following steps to configure the new domain value:
   a. Select IM and Presence Administration > System > Cluster Topology.
   b. In the right pane, select Settings.
   c. Configure the Domain Name value with the new domain.
   a. Select IM and Presence Administration > System > Service Parameters, and select the Cisco SIP Proxy service.
   b. Configure the Federation Routing IM and Presence FQDN with the new domain.
   c. You will be prompted to confirm these configuration changes. Select OK for both prompts, and then select Save.
Step 3 On all nodes in the cluster, use this CLI command to set the new domain:
   set network domain
This CLI command invokes a reboot of the servers.
Step 4 On all nodes in the cluster, manually start the Cisco Presence Engine and Cisco XCP Router services after the reboot is complete (if required).
Step 5 Manually regenerate all certificates on each node in the cluster.

Question 122
Which two settings should be configured on the SIP Trunk Security Profile for the IM & Presence Service SIP Trunk?
Which two settings should be configured on the SIP Trunk Security Profile for the IM & Presence Service SIP Trunk? (Choose two.)
A. Check to enable Accept Presence Subscription.
B. Verify that the setting for Incoming Transport Type is TCP+UDP.
C. Configure Device Security Mode to Encrypted.
D. Check to enable Application Level Authorization.
E. Configure the Outgoing Transport Type to TLS.
Correct Answer: AB
Explanation/Reference:
Explanation:
Configure SIP Trunk Security Profile for IM and Presence Service
Procedure
Configure SIP Trunk Security Profile for IM and Presence Service
Procedure
Procedure
Step 1 Choose Cisco Unified CM Administration > System > Security > SIP Trunk Security Profile.
Step 2 Click Find.
Step 3 Click Non Secure SIP Trunk Profile.
Step 4 Click Copy and enter DIP Trunk in the Name field.
Step 5 Verify that the setting for Device Security Mode is Non Secure.
Step 6 Verify that the setting for Incoming Transport Type is TCP+UDP.
Step 7 Verify that the setting for Outgoing Transport Type is TCP.
Step 8 Check or enable the following:
   • Accept Presence Subscription
   • Accept Outgoing REFER
   • Accept Unregistered Redirect
   • Accept Redirected Header
Step 9 Click Save.
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/9_0/CUP0_BK_CFF5B189_00_config-adminguide-imp-90/CUP0_BK_CFF5B189_00_config-admin-guide-imp-90_chapter_0101.html

Question 123
Which Cisco IM and Presence service is responsible for logging all IM traffic that passes through the IM and Presence server to an external database for IM compliance?
Which Cisco IM and Presence service is responsible for logging all IM traffic that passes through the IM and Presence server to an external database for IM compliance?
A. Cisco Presence Engine
B. Cisco Serviceability Reporter
C. Cisco Sync Agent
D. Cisco XCP Connection Manager
E. Cisco XCP Message Archiver
Correct Answer: E
Explanation/Reference:
Explanation:
The Cisco Unified Presence XCP Message Archiver service supports the IM Compliance feature. The IM Compliance feature logs all messages sent to and from the Cisco Unified Presence server, including point-to-point messages, and messages from ad-hoc (temporary) and permanent chat rooms for the Chat feature. Messages are logged to an external Cisco-supported database.

Question 124
Which external database software is required for the Cisco IM and Presence compliance feature?
Which external database software is required for the Cisco IM and Presence compliance feature?
A. MySQL
B. EnterpriseDB
C. MSSQL
D. SQLite
E. PostgreSQL
Correct Answer: E
Explanation/Reference:
The following Cisco Unified Presence features require an external database:
- Permanent Group Chat feature – Cisco Unified Presence supports two types of group chat, temporary (ad-hoc) chat and permanent chat. You do not require an external database for temporary chat to work. However, if you require permanent chat rooms on Cisco Unified Presence, you must configure an external database.
- Instant Messaging Compliance – If you deploy the native Message Archiver (MA) component on Cisco Unified Presence for compliance logging, you require an external database.

Requirements for Configuring an External Database
- Hardware requirements:
  A remote server on which you install the PostgreSQL database(s).
- Software requirements:
  – Cisco Unified Presence, release 8.x.
  – PostgreSQL database, versions 8.3.x through 9.1.1
  – You can install the PostgreSQL database on either a Linux or a Windows operating system. See the PostgreSQL documentation for details on the supported operating systems and platform requirements.

Question 125
Which statement describes the external database requirement for the Cisco IM and Presence permanent group chat feature?
Which statement describes the external database requirement for the Cisco IM and Presence permanent group chat feature?
A. All nodes in a Cisco IM and Presence cluster can share a physical external database.
B. All nodes in a Cisco IM and Presence cluster can share a logical external database.
C. Each node in a Cisco IM and Presence cluster must have its own physical external database.
D. Each node in a Cisco IM and Presence cluster must have its own logical external database.
E. An external database is not mandatory.
Correct Answer: D
Explanation/Reference:
When you configure an external database entry on IM and Presence, you assign the external database to a node, or nodes, in your cluster as follows:
- For the Compliance feature, you require at least one external database per cluster. Depending on your deployment requirements, you can also configure a separate external database per node.
- For the Permanent Group Chat feature, you require a unique external database per node. Configure and assign a unique external database for each node in your cluster.
- If you deploy both the Permanent Group Chat and Compliance features on an IM and Presence node, you can assign the same external database to both features.

Question 126
Which two enterprise presence domains can federate with Cisco IM and Presence by using SIP?
Which two enterprise presence domains can federate with Cisco IM and Presence by using SIP? (Choose two.)
A. AOL
B. Microsoft OCS
C. IBM Sametime
D. Cisco WebEx Connect
E. Google Talk
F. Cisco Unified Presence 8.X Releases
Correct Answer: AB
Explanation/Reference:
Microsoft Lync and OCS support presence services with sip as well as AOL so to sip is easy to troubleshoot and feasible for signaling that's why cisco federate these with sip.

Question 127
Which server should the fourth user be assigned?
Refer to the exhibit.
In this high-availability Cisco IM and Presence deployment with three subclusters, the first user is assigned to server 1A; the second user is assigned to server 2A; and the third user is assigned to server 3A. Assume that the Cisco IM and Presence is set to Active/Standby mode, to which server should the fourth user be assigned?

A. 1A  
B. 3B  
C. 1B  
D. 2A  
E. 3A

Correct Answer: A

Explanation/Reference:
This deployment model provides the same level of redundancy and high availability as outlined in the “Balanced Redundant High-Availability Deployment” section in this chapter.

The only difference is that users are not deployed in a balanced fashion, but rather all reside on the primary server in the subcluster, and the backup server is there as a standby option if a node failure occurs.

Question 128
Which protocol that is used between Cisco IM and Presence and Cisco Unified Communications Manager is responsible for the exchange of phone state presence information?

A. AXL/SOAP  
B. CTI/QBE  
C. SIP/SIMPLE  
D. LDAP  
E. XMPP

Correct Answer: C

Explanation/Reference:
To provide interoperability between communications systems, SIP is the protocol leveraged. Enterprise Presence solutions need to provide for a uniform definition of the main communication services such as IM, voice, video, e-mail, web calendaring, and so on, while SIP delivers the necessary features.

Question 129
Which protocol does the Cisco Jabber client use, in conjunction with Cisco IM and Presence, to deliver enterprise-class instant messaging services?

A. SIP  
B. CTI/QBE  
C. XMPP  
D. IRC  
E. ICQ

Correct Answer: C

Explanation/Reference:
Many federated IM networks communicate using an open standard, such as Jabber, that leverages the Extensible Messaging and Presence Protocol (XMPP). Networks using XMPP provide open communications with other XMPP-based networks.

Question 130
Which statement about high availability for XMPP federation in Cisco IM and Presence is true?

A. A maximum of two Cisco IM and Presence nodes can be enabled for XMPP federation.  
B. Cisco IM and Presence load balances outbound requests across all nodes that are enabled for XMPP federation.  
C. Cisco IM and Presence load balances outbound requests across both nodes that are enabled for XMPP federation in a subcluster.  
D. The XMPP federation-enabled nodes should have different priorities and weights on the published DNS SRV for proper inbound request node selection.  
E. A single DNS SRV record that resolves to an XMPP federation-enabled node must be published on a public DNS server for inbound request routing.

Correct Answer: B

Explanation/Reference:
High availability for XMPP federation differs from the high availability model for other IM and Presence Service features because it is not tied to the two node subcluster model. To provide high availability for XMPP federation, you must enable two or more IM and Presence Service nodes in your cluster for XMPP federation; having multiple nodes enabled for XMPP federation not only adds scale but it also provides redundancy in the event that any node fails.
Question 131
Which server will the fourth user be automatically assigned?
Refer to the exhibit.

![Diagram of Cisco IM and Presence Service with subclusters]

In this high-availability Cisco IM and Presence deployment with three subclusters, the first user is assigned to server 1A; the second user is assigned to server 2A; and the third user is assigned to server 3A. Assume that Cisco IM and Presence is set to active-active mode. To which server will the fourth user be automatically assigned?
A. 1A
B. 3B
C. 1B
D. 2A
E. 3A

Correct Answer: C
Explanation/Reference:
You can achieve a balanced mode High Availability deployment by evenly balancing users across all nodes in the subcluster, but only using up to 35% of the CPU of each IM and Presence node. The balanced mode High Availability deployment option in a redundant mode supports up to fifteen thousand users per cluster. For example, if you have six IM and Presence nodes in your deployment, and fifteen thousand users, you assign 2.5 thousand users to each IM and Presence node. When you use the balanced mode High Availability deployment option in a redundant mode, compared to a nonredundant mode, only half the number of users are assigned to each node. However, if one node fails, the other node will handle the full load of the additional 50% of users in the subcluster, even at peak traffic. In order to support this failover protection, you must turn on High Availability in each of the subclusters in your deployment.

Question 132
Which protocol is used by presence-enabled users in Cisco IM and Presence to control phones that are registered to Cisco Unified Communications Manager?

A. AXL/SOAP
B. CTI/QBE
C. SIP/SIMPLE
D. LDAP
E. XMPP

Correct Answer: B
Explanation/Reference:
The CTI gateway provides desk phone control when users are configured for phone association mode. Proper installation calls upon information to specify CTI gateway server names, addresses, ports, and protocols on CUPS. Configured correctly, the CTI gateway enables users logging in to CU/PC to reach the CTI gateway.

Question 133
Which agent will receive the call when a select resource step is triggered in the script for the Customer Service CSQ?
Refer to the exhibit.
Assume that all shown agents are available to take a call. Which agent will receive the call when a select resource step is triggered in the script for the Customer Service CSQ?

A. s1dispatch-PA
B. s1dispatch-OH
C. the agent that has been idle the longest
D. the agent with the shortest handled time

Correct Answer: A

Explanation/Reference:
The Contact Service Queue (CSQ) controls incoming calls by determining where an incoming call should be placed in the queue and to which agent the call is sent. After you assign an agent to a resource group and assign skills, you need to configure the CSQ. You assign agents to a CSQ by associating a resource group or by associating all skills of a particular CSQ. Agents in the selected resource group or who have all the selected skills are assigned to the CSQ. Skills within the CSQ can be ordered. This means that when resources are selected, a comparison is done based on the competency level (highest for “most skilled” and lowest for “least skilled”) of the first skill in the list. If there is a “tie”, the next skill within the order is used, and so on.

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/contact_center/contact_center_administration/guide/UCCX_BK_W1AF9DDD_00_ucx-admin-guide-10-0/UCCX_BK_W1AF9DDD_00_ucx-admin-guide-10-0_chapter_0111.html#UCCX_TP_C6155D52_00

Question 134
Which two guidelines are recommended when configuring agent phones for Cisco Unified CCX agents?

Which two guidelines are recommended when configuring agent phones for Cisco Unified CCX agents? (Choose two.)

A. In the Multiple Call/Call Waiting Settings section, set the Maximum Number of Calls to 2.
B. In the Multiple Call/Call Waiting Settings section, set the Busy Trigger value to 2.
C. The Unified CCX extension for the agent must be listed within the top four extensions on the device profile.
D. In the Multiple Call/Call Waiting Settings section, set the Maximum Number of Calls to at least 3.
E. Always enable SRTP when configuring an agent phone.

Correct Answer: AC

Explanation/Reference:
Follow these guidelines when configuring agent phones for Unified CCX agents:

1. In the Multiple Call/Call Waiting Settings section, set the Maximum Number of Calls to 2. (Choose two.)
2. In the Multiple Call/Call Waiting Settings section, set the Busy Trigger value to 2.
3. The Unified CCX extension for the agent must be listed within the top four extensions on the device profile.
4. In the Multiple Call/Call Waiting Settings section, set the Maximum Number of Calls to at least 3.
5. Always enable SRTP when configuring an agent phone.

Explanation/Reference:
Always disable (turn off) Secure Real-Time Transport Protocol (SRTP) when configuring a Cisco Unified Communications product. You can disable SRTP for a specified device or for the entire Unified Communications Manager.

For a specified device — Choose Device > Phone. In the Find and List Phones page, select the required phone device. In the Phone Configuration page for the selected phone, scroll down to the Protocol Specific Information section. To turn off SRTP on the phone device, select any one of the Non Secure SCCP Profile auth by choices from the drop-down list in SCCP Phone Security Profile or SCCP Device Security Profile field.

For the entire Unified Communications Manager cluster — Choose System > Enterprise Parameters. In the Enterprise Parameters Configuration page, scroll down to the Security Parameters section, to verify that the corresponding value for the Cluster Security Mode field is 0. This parameter indicates the security mode of the cluster. A value of 0 indicates that phones will register in nonsecure mode (no security).

The Unified CCX extension for the agent must be listed within the top 4 extensions on the device profile. Listing the extension from position 5 on will cause Unified
CCX to fail to monitor the device, so the agent will not be able to log in.

A. Do not forward any Unified Communications Manager device to the Unified CCX extension of an agent.
B. Do not forward any Unified Communications Manager device to a Unified CCX route point.
C. Do not configure the Unified CCX extension of an agent to forward to a Unified CCX route point.
D. Do not configure any Unified Communications Manager device to the Unified CCX extension of an agent.
E. Do not configure the same Unified CCX extension in more than one device or device profile (Configuring a Unified CCX extension in one device or device profile is supported.)
F. Do not configure any Unified Communications Manager device to the Unified CCX extension of an agent to forward to a Unified CCX route point.

Question 135

Which two statements describe the remote supervisory monitoring feature in Cisco Unified Contact Center Express?
A. It is supported on Cisco Unified CCX Enhanced and Premium editions.
B. It requires a Cisco Supervisor Desktop or any data network connectivity.
C. Agents are aware that they are being silently monitored.
D. Calls can be silently monitored from a PSTN phone.
E. It works with SPAN port monitoring only.
F. It works with JTAPI port monitoring.

Correct Answer: BD
Explanation/Reference:
Agents use the Cisco Agent Desktop (commonly referred to as CAD) to login to the Unified CCX server and control their ACD state, control incoming and outgoing calls, chat with supervisors and other agents on their team, view their own real-time statistics, and view their own recent call activity. Supervisors use the CSD to view real-time queue and agent statistics, view recent call activity for agents, change agent states, chat with agents, and send marquee messages to all agents on the selected team. With the Enhanced or Premium packages, the supervisor can also barge-in or intercept ACD calls, silently monitor agents, and record agent calls.

Question 136

Which statement describes DTMF processing on Cisco Unified Contact Center Express with supported SIP-based agent IP phones that are registered to Cisco Unified Communications Manager?
A. Cisco Unified CCX receives the DTMF digits via SIP NOTIFY messages.
B. Cisco Unified CCX receives the DTMF digits via SIP INFO messages.
C. Cisco Unified CCX receives the DTMF digits via JTAPI messages.
D. Cisco Unified CCX receives the DTMF digits via SIP INFO messages.
E. Cisco Unified CCX receives the DTMF digits as part of the audio encoding in the RTP stream.

Correct Answer: C
Explanation/Reference:
Unified CCX CTI ports are notified of caller-entered digits (DTMF input) via JTAPI messages from Unified CM. Unified CCX does not support any mechanism to detect in-band DTMF digits where DTMF digits are sent with voice packets. In deployments with voice gateways or SIP phones that only support in-band DTMF, it is recommended to use the previous MTP solutions.

Question 137

Which mechanism enables the Cisco Unified CCX Cisco Agent Desktop application to obtain a copy of the RTP packet stream directly from a supported IP phone?
A. SPAN port monitoring
B. desktop monitoring
C. remote SPAN monitoring
D. reflector port monitoring
E. ESPAN monitoring

Correct Answer: B
Explanation/Reference:
Desktop monitoring provides a mechanism for the CAD application to obtain a copy of the RTP packet streams directly from the phone and therefore removes the need for a Monitoring component connected to the SPAN port on the Catalyst switch. A Cisco phone supporting desktop monitoring is required and the agent workstation running the CAD must be connected to the data port on the back of the agent phone. The Cisco IP Communicator also supports using desktop monitoring for silent monitoring and recording.

Question 138

How many RTP streams exist on the network when a Cisco Unified Contact Center Express agent is engaged in a call that is being silently monitored and recorded?
A. 3
B. 4
C. 5
D. 6
E. 8

Correct Answer: D
Explanation/Reference:
Desktop monitoring provides a mechanism for the CAD application to obtain a copy of the RTP packet streams directly from the phone and therefore removes the need for a Monitoring component connected to the SPAN port on the Catalyst switch. A Cisco phone supporting desktop monitoring is required and the agent workstation running the CAD must be connected to the data port on the back of the agent phone. The Cisco IP Communicator also supports using desktop monitoring for silent monitoring and recording.

Question 139

How many RTP streams exist on the network when a Cisco Unified Contact Center Express agent is engaged in a call that is being silently monitored?
A. 3
B. 4
C. 5
D. 6
E. 8

Correct Answer: B
Explanation/Reference:
Desktop monitoring provides a mechanism for the CAD application to obtain a copy of the RTP packet streams directly from the phone and therefore removes the need for a Monitoring component connected to the SPAN port on the Catalyst switch. A Cisco phone supporting desktop monitoring is required and the agent workstation running the CAD must be connected to the data port on the back of the agent phone. The Cisco IP Communicator also supports using desktop monitoring for silent monitoring and recording.

http://www.aoowe.com/practice-400-051-3156.html
Question 139
Which Cisco Unified Contact Center Express core system software component communicates with Cisco Agent Desktop for agent state control and call control?
A. Unified CCX Engine  
B. Database  
C. Monitoring  
D. Recording  
E. RmCm
Correct Answer: A
Explanation/Reference:
Explanation:
The Unified CCX Engine enables you to run multiple applications to handle Unified CM Telephony calls or HTTP requests. The Unified CGX Engine uses the Unified CM Telephony subsystem to request and receive services from the Computer Telephony Interface (CTI) manager that controls Unified CM clusters. The Unified CCX Engine is implemented as a service that supports multiple applications. You can use a web browser to administer the Unified CCX Engine and your Unified CCX applications from any computer on the network. Unified CCX provides you the following two web interfaces:
. Unified CCX Serviceability web interface: Used to view alarm and trace definitions for Unified CCX services, start and stop the Unified CCX Engine, monitor Unified CCX Engine activity, and so on.
. Unified CCX Serviceability web interface: Used to configure system parameters, subsystems, view real-time reports that include total system activity and application statistics, and so on.

Question 140
Which Cisco Unified Contact Center Express data store contains CSQ information?
A. configuration data store  
B. repository data store  
C. agent data store  
D. historical data store  
E. script data store
Correct Answer: A
Explanation/Reference:
Explanation:
The Database component is required for any Unified CCX deployment and manages access to the database. The Unified CCX Database contains four data stores. They are as follows:
. Configuration data store  
. Repository data store  
. Agent data store  
. Historical data store
The configuration data store contains Unified CCX configuration information such as resources (agents), skills, resource groups, teams, and CSQ information. The repository data store contains user prompts, grammars, and documents. The agent data store contains agent logs, statistics, and pointers to the recording files. The historical data store contains Contact Call Detail Records (CCDRs).

Question 141
Which statement describes the call recording operation on Cisco Unified Contact Center Express call agents that use Cisco IPPA?
A. Recording is facilitated via desktop monitoring on supported IP phones.  
B. Automatic recording is supported.  
C. Only G.711 codec is supported.  
D. Only SPAN port monitoring is supported.  
E. Call recording is not supported on Cisco Unified CCX call agents that use Cisco IPPA.
Correct Answer: D
Explanation/Reference:
Explanation:
On-demand recording of active agent calls, available in Enhanced and Premium versions, improves customer service and encourages appropriate and consistent agent behavior and it is a feature of Cisco Agent Desktop.

Question 142
Which call recording operation can be used to satisfy this requirement?
A company that is using the Cisco Unified Contact Center Express Enhanced version requires that selected types of agent calls are automatically recorded. Which call recording operation can be used to satisfy this requirement?
A. Recording is facilitated via desktop monitoring on supported IP phones.  
B. Automatic recording is supported.  
C. Only G.711 codec is supported.  
D. Only SPAN port monitoring is supported.  
E. Recording is not supported on the Cisco Unified CCX Enhanced version. It is supported only on the Premium version.
Correct Answer: D
Explanation/Reference:
Explanation:
On-demand recording of active agent calls, available in Enhanced and Premium versions, improves customer service and encourages appropriate and consistent agent behavior and it is a feature of Cisco Agent Desktop.

Question 143
Which Cisco Unified Contact Center Express script media step can invoke a VXML application to retrieve and play prompts on-demand from an off-box location?
Which Cisco Unified Contact Center Express script media step can invoke a VXML application to retrieve and play prompts on-demand from an off-box location?
A. Play Prompt step  
B. Voice Browser step  
C. Menu step
Correct Answer: A
Explanation/Reference:
Explanation:
On-demand recording of active agent calls, available in Enhanced and Premium versions, improves customer service and encourages appropriate and consistent agent behavior and it is a feature of Cisco Agent Desktop.
D. Recording step
E. Simple Recognition step

Correct Answer: B

Explanation/Reference:
CRA Voice Browser is fully integrated with the CRA Engine. You can use scripts designed in the CRA Editor to extend VoiceXML applications by providing ICD (Integrated Contact Distribution) call control and resource management. For example, you can use VoiceXML to build a speech dialog as a front end to collect information from the caller. You can then pass this information to the CRA script, and when the agent receives the call, the information collected by VoiceXML will be available. You use the Voice Browser step in the Media palette of the CRA Editor to invoke a VoiceXML application. You can use the bundled voicebrowser.aef script as an example for creating scripts that invoke VoiceXML. (You can create custom scripts to execute other steps in addition to VoiceXML.)

Question 144

Which Cisco Unified Contact Center Express data store contains user scripts, grammars, and documents?
A. configuration data store
B. repository data store
C. agent data store
D. historical data store
E. script data store

Correct Answer: B

Explanation/Reference:
Unified CCX applications might use auxiliary files that interact with callery, such as scripts, pre-recorded prompts, grammars, and custom Java classes. Depending on each implementation, Unified CCX applications use some or all of the following file types. The Unified CCX Server’s local disk prompt, grammar, and document files are synchronized with the central repository during Unified CCX engine startup and during run-time when the Repository datastore is modified.

Question 145

Which protocols connect the locations and servers together for messaging and replication?

Refer to the exhibit.

A. 1 – SMTP
2 – HTTP/HTTPS, SMTP
3 – None
B. 1 – SMTP
2 – SMTP
3 – SMTP
C. 1 – HTTP/HTTPS, SMTP
2 – HTTP/HTTPS, SMTP
3 – HTTP/HTTPS, SMTP
D. 1 – HTTP/HTTPS, SMTP
2 – SMTP
3 – None

Correct Answer: A

Explanation/Reference:
Cisco Unity Connection Site A has two locations. Cisco Unity Connection Site B has one location. Which protocols connect the locations and servers together for messaging and replication?

A. 1 – SMTP
2 – HTTP/HTTPS, SMTP
3 – None
B. 1 – SMTP
2 – SMTP
3 – SMTP
C. 1 – HTTP/HTTPS, SMTP
2 – HTTP/HTTPS, SMTP
3 – HTTP/HTTPS, SMTP
D. 1 – HTTP/HTTPS, SMTP
2 – SMTP
3 – None

Correct Answer: A

Explanation/Reference:
You can join two or more Connection servers or clusters (up to a maximum of ten) to form a well-connected network, referred to as a Connection site. The servers that are joined to the site are referred to as locations. (When a Connection cluster is configured, the cluster counts as one location in the site.) Within a site, each location uses SMTP to exchange directory synchronization information and messages directly with every other location. Each location is said to be linked to every other location in the site via an intrasite link.

When you link two Cisco Unity Connection sites with an intersite link, the gateway for each site is responsible for collecting information about all changes to the local site directory, and for polling the remote site gateway periodically to obtain information about updates to the remote site directory. The gateways use the HTTP or HTTPS protocol to exchange directory synchronization updates.
Question 146
Which two search scope options are removed from a directory handler when you check the “voice enabled” check box?
Which two search scope options are removed from a directory handler when you check the “voice enabled” check box? (Choose two.)
A. Class of Service
B. System Distribution List
C. Search Space
D. Partition
E. Phone System
Correct Answer: AB
Explanation/Reference:
You can configure the scope of a directory handler to define the objects that callers who reach the directory handler can find or hear. For phone directory handlers, you can set the scope to the entire server, to a particular class of service, to a system distribution list, or to a search space (either inherited from the call or specified for the directory handler). For voice-enabled directory handlers, you can set the scope to the entire server or to a search space (either inherited from the call or specified for the directory handler).
When callers search a directory handler for a particular name, if the scope of the directory handler is set to a search space, Cisco Unity Connection searches each partition in the search space and returns a list of all of the objects that match the name.

Question 147
Which three configuration dialog boxes can a user assign a search space?
In Cisco Unity Connection, to which three configuration dialog boxes can a user assign a search space? (Choose three.)
A. Routing Rule
B. Call Handler
C. Interview Handler
D. Contacts
E. Users
F. Port
G. Phone System
Correct Answer: ABE
Explanation/Reference:
In unity connection, user can assign a search space in:
Users
Call Routing Rules
System Distribution Lists
System Call Handlers
Directory Handlers
Interview Handlers
Digital Networking
VPIM Locations
Administrator-Defined Contacts

Question 148
Which option is a possible solution for this problem?
Refer to the exhibit.
The public key infrastructure debugs are generated on a Cisco IOS VPN router for a failed certification validation on an incoming connection from an IP phone client.
Which option is a possible solution for this problem?
A. Define a matching Certification Revocation List on the Cisco IOS VPN router.
B. Define a Certification Revocation List in the IP phone certificate.
C. Disable revocation check for the trustpoint.
D. Define an enrollment URL for the trustpoint.
E. Define a matching Certification Revocation List on the Cisco Unified Communications Manager.
Correct Answer: C
Explanation/Reference:
When a certificate is issued, it is valid for a fixed period of time. Sometimes a CA revokes a certificate before this time period expires; for example, due to security concerns or a change of name or association. A CA periodically issues a signed list of revoked certificates. Enabling revocation checking forces the IOS router to check that the CA has not revoked a certificate every time it uses that certificate for authentication.
When you enable revocation checking during the PKI certificate validation process, the router checks certificate revocation status. It can use either CRL checking or Online Certificate Status Protocol or both, with the second method you set in effect only when the first method returns an error, for example, that the server is unavailable.
With CRL checking, the router retrieves, parses, and caches Certificate Revocation Lists, which provide a complete list of revoked certificates. OCSP offers a more scalable method of checking revocation status in that it localizes certificate status on a Validation Authority, which it queries for the status of a specific certificate.

Question 149
Which two statements about virtual SNR in Cisco Unified Communications Manager Express are true?
Which two statements about virtual SNR in Cisco Unified Communications Manager Express are true? (Choose two.)
A. The SNR DN must be configured as SCCP.
B. Calls cannot be pulled back from the phone associated with the DN.
C. Ephone hunt groups are supported.
D. The virtual SNR DN must be assigned to an ephone.
E. Music on hold is supported for trunk and line side calls.
Correct Answer: AB
Explanation/Reference:
To configure a virtual SNR DN on Cisco Unified SCCP IP phones, perform the following steps:
Prerequisites
Cisco Unified CME 9.0 or a later version.
Restrictions
Virtual SNR DN only supports Cisco Unified SCCP IP phone DNs.
Virtual SNR DN provides no mid-call support.
Mid-calls are either of the following:
– Calls that arrive before the DN is associated with a registered phone and is still present after the DN is associated with the phone.
– Calls that arrive for a registered DN that changes state from registered to virtual and back to registered.
– Mid-calls cannot be pulled back, unanswered, or terminated from the phone associated with the DN.
– State of the virtual DN transitions from ringing to hold or remains on hold as a registered DN.

Question 150
Which two categories are state-based greetings on Cisco Unity Express?
Which two categories are state-based greetings on Cisco Unity Express? (Choose two.)
A. Meeting
B. Vacation
C. Internal
D. Closed
E. Alternate
F. Extended Absence
Correct Answer: CD
Explanation/Reference:
Beginning in version 7.1, you can configure multiple greetings. These greetings fall into the following three categories:
– Standard greetings
  – Alternate greetings
This category includes the following types of greetings:
  – Alternate
  – Meeting
  – Vacation
  – Extended absence
– State-based greetings:
This category includes the following types of greetings:
  – Busy
  – Closed
  – Internal

Question 151
Which two trigger types can invoke applications on Cisco Unity Express?
In addition to SIP triggers, which two trigger types can invoke applications on Cisco Unity Express? (Choose two.)
A. HTTP
B. IMAP
C. VoiceView
D. JTAPI
E. Cisco Unified CM telephony
F. voice mail
Correct Answer: AD
Explanation/Reference:
Triggers are incoming events that invoke application which in turn starts executing the script associated with that application. For example, the incoming event can be an incoming call or an incoming HTTP request.
After you have created and configured your application, you need to create a trigger on the Cisco Unity Express module to point to that application. Cisco Unity Express supports three types of triggers:
SIP triggers — Use this type of trigger to invoke applications in Cisco Unified CME and Cisco SRST mode. This type of trigger is identified by the phonenum which is dialed to invoke the desired application.
JTAPI triggers — Use this type of trigger to invoke applications in Cisco Unified Communications Manager mode. This type of trigger is identified by the phonenum which is dialed to invoke the desired application.
HTTP triggers — Use this type of trigger to invoke applications using an incoming HTTP request. Such a trigger is identified by the URL suffix of the incoming HTTP request. This type of trigger can only be used if an ITR license has been purchased and installed on the system.

Question 152
Which statement describes how the digit zero is handled in the predefined restriction tables in Cisco Unity Connection?
Which statement describes how the digit zero is handled in the predefined restriction tables in Cisco Unity Connection?
A. Zero is listed in the Default Out-Dial Restriction table.
B. Zero is listed in the Default System Transfer Restriction table.
C. Zero is listed in the Default Transfer Restriction table.
D. Zero is listed in the User-Defined and Automatically Added Alternate Extensions Restriction table.
E. Zero is not listed in any default restriction table configuration.

Correct Answer: E
Explanation/Reference:
When user dials "0", by default Unity Connection treats it as an operator call and does not block "0" by any restriction table configuration. Only the operator can modify transfer extension associated with operator call.

Question 153
When Cisco Unity Connection users attempt to connect using Web Inbox and receive a Site Is Unavailable error message, which service status should be verified?
A. Tomcat
B. Connection Exchange Notification Web Service
C. Connection Voicemail Web Service
D. Connection Administration
E. Secured Web Server

Correct Answer: A
Explanation/Reference:
Cisco Tomcat service, as the name suggests, is used by the Web Server of CUCM and helps display the administration, operating system, disaster recovery, and other GUI interfaces of CUCM. The service leverages a built-in CA for Tomcat in that it redirects the incoming HTTP requests to HTTPS using the default self-signed certificate.

Question 154
Which statement about accessing secure Cisco Unity Connection voice messages in an Exchange mailbox in a Single Inbox deployment is true?
A. Users can listen to a secure voice message if they use the Outlook email client.
B. Users can listen to a secure voice message if they use the Outlook email client with the ViewMail add-in.
C. Users can listen to a secure voice message with email clients other than Outlook if they have installed the ViewMail add-in.
D. Users cannot listen to a secure message in Exchange because it is not supported in Single Inbox.
E. Secure voice messages are stored on the Cisco Unity Connection server and the Exchange server.

Correct Answer: B
Explanation/Reference:
Users can listen to a secure voice message if they use the Outlook email client with the ViewMail add-in. Because in this integration Outlook integrates with unity as secrmsmapclient.

Question 155
Which statement describes the supported integration method when Cisco Unity Connection and Cisco Unified Communications Manager are installed on the same server as Cisco Unified Communications Manager Business Edition?
A. Only SCCP integration is supported.
B. Only SIP integration is supported.
C. Both SCCP or SIP integration are supported, but you must choose one or the other.
D. Q-Sig integration is supported through a voice-enabled Cisco ISR router.
E. Circuit-switched integration is supported through PIMG.

Correct Answer: A
Explanation/Reference:
When installed on the same server there is no way to create trunk that is why scp is the only way Cisco Unity Connection and Cisco Unified Communications Manager are installed on the same server.

Question 156
Which statement about system broadcast messages in Cisco Unity Connection is true?
A. The user can skip a system broadcast message to listen to new messages first.
B. The user can forward a system broadcast message only if it has been played in its entirety.
C. System broadcast messages are synchronized between Cisco Unity Connection and Exchange when Single Inbox is configured.
D. System broadcast messages do not trigger MWI.
E. System broadcast messages are played immediately after users sign in and listen to message counts for new and saved messages.

Correct Answer: D
Explanation/Reference:
System broadcast messages are played immediately after users log on to Cisco Unity Connection by phone even before they hear message counts for new and saved messages. After logging on, users hear how many system broadcast messages they have and Connection begins playing them.

Question 157
What is the default treatment of a message that is left in the opening greeting default call handler in Cisco Unity Connection?
What is the default treatment of a message that is left in the opening greeting default call handler in Cisco Unity Connection?
A. It will be sent to the mailbox for the Operator user.
B. It will be sent to the Undeliverable Messages distribution list.
C. It will be sent to the mailbox of the system administrator.
D. It will be sent to the All Voicemail Users distribution list.
E. It will be sent to the General Delivery Mailbox.

Correct Answer: B
Default call handler is selected when we don't assign any call handler to a user and with this default call handler no specific user assigned so it doesn't go to any specific mailbox and goes to it will be sent to the Undeliverable Messages distribution list.

**Question 158**
Which Cisco Unity Connection call handler message is played when a caller enters a string of digits that is not found in the search scope?

- A. error
- B. closed
- C. internal
- D. busy
- E. alternate

Correct Answer: A

**Explanation/Reference:**
As soon as unity finds the unexpected behavior it prompts the error message to the user.

**Question 159**
Which three Cisco Unity Connection call handler greetings can be overridden by the internal greeting?

- A. holiday
- B. alternate
- C. error
- D. busy
- E. closed
- F. standard

Correct Answer: AEF

**Explanation/Reference:**
This greeting overrides the Standard, Closed, and Holiday greetings but only for internal callers or users defined in Cisco Unity Connection because the mentioned three greetings are defined for externals users.

**Question 160**
Which Cisco Unity Connection call handler greeting, when enabled, overrides all other greetings?

- A. holiday
- B. closed
- C. internal
- D. busy
- E. alternate

Correct Answer: E

**Explanation/Reference:**
An Alternate greeting might be enabled to override the Standard Greeting during certain times, because it is a personal greeting used for specific purpose.

**Question 161**
When Single Inbox is configured, what will happen to an email message that was moved from any Outlook folder to the Voice Outbox folder?

- A. The email message will be delivered to Cisco Unity Connection.
- B. The email message will be kept in the Voice Outbox folder.
- C. The move will fail because the operation is not supported.
- D. The email message will be moved to the Deleted Items folder.
- E. The email message will be permanently deleted and will not be retrievable.

Correct Answer: D

**Explanation/Reference:**
Voice messages queue for delivery in the Voice Outbox folder that is why it shows in Deleted Items folder.

**Question 162**
Which message-handling behavior describes how Cisco Unity Connection Single Inbox works for Outlook users who do not have ViewMail installed?

- A. Cisco Unity Connection voice messages are treated as emails without a WAV file attachment.
- B. Cisco Unity Connection voice messages are treated as voice messages.
- C. Cisco Unity Connection voice messages are treated as emails with a WAV file attachment.
- D. Cisco Unity Connection adds a Voice Outbox folder to the Outlook mailbox.
- E. Replies to Cisco Unity Connection voice messages are sent to Exchange as well as the Cisco Unity Connection mailbox for the recipient.

Correct Answer: C

**Explanation/Reference:**
Cisco unity here acts as an IMAP server for the outlook user who don't have view mail installed so user send their request as an IMAP client and unity will revert back with email and wav file attached to play.

**Question 163**
What should the administrator do to execute this change?

A Cisco Unity Connection administrator receives a name change request from a voice-mail user, whose Cisco Unity Connection user account was imported from Cisco Unified Communications Manager. What should the administrator do to execute this change?
A. Change the user data in the Cisco Unity Connection administration page, then use the Synch User page in Cisco Unity Connection administration to push the change to Cisco Unified Communications Manager.
B. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unity Connection administration to pull the changes from Cisco Unified CM.
C. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unified CM administration to push the change to Cisco Unity Connection.
D. Change the user profile from Imported to Local on Cisco Unity Connection Administration, then edit the data locally on Cisco Unity Connection.
E. Change the user data in Cisco Unity Connection and Cisco Unified Communications Manager separately.

Correct Answer: B

Explanation/Reference:
As we can see user are getting synch from call manager so we first have to change the details of user on call manager so that user will synch the changes from call manager.

Question 164
Which option is the default Cisco Wireless Unified Communications endpoints marking for video media traffic or video RTP traffic?
Which option is the default Cisco Wireless Unified Communications endpoints marking for video media traffic or video RTP traffic?
A. DSCP 8
B. DSCP 24
C. DSCP 34
D. DSCP 46

Correct Answer: C

Explanation/Reference:
When configuring network-level quality of service (QoS), Cisco video endpoints (including Cisco Unified IP Phone 8900 and 9900 Series and Cisco TelePresence System EX Series devices) generally mark traffic at Layer 3 according to Cisco general QoS guidelines related to voice and video packet marking (video media as DSCP 34 or PHB AF41; call signaling as DSCP 24 or PHB CS3) and therefore these devices can be trusted.

Question 165
Which statement about application inspection of SAF network services on an adaptive security appliance is true?
Which statement about application inspection of SAF network services on an adaptive security appliance is true?
A. The adaptive security appliance can inspect and learn the ephemeral port numbers that are used by H.225 and H.245 on SAF-enabled H.323 trunks.
B. An explicit ACL must be configured on the adaptive security appliance for SAF-enabled SIP trunks.
C. An explicit ACL must be configured on the adaptive security appliance for SAF-enabled H.323 trunks to account for ephemeral port numbers that are used by H.225 and H.245.
D. The adaptive security appliance can inspect and learn the ephemeral port numbers that are used by H.225 on SAF-enabled H.323 trunks, but H.245 ports must be explicitly defined.
E. The adaptive security appliance provides full application inspection for SAF network services.

Correct Answer: C

Explanation/Reference:
The Adaptive Security Appliances do not have application inspection for the SAF network service. When Unified CM uses a SAF-enabled H.323 trunk to place a call, the ASA cannot inspect the SAF packet to learn the ephemeral port numbers used in the H.225 signaling. Therefore, in scenarios where call traffic from SAF-enabled H.323 trunks traverses the ASAs, ACLs must be configured on the ASA to allow this signaling traffic. The ACL configuration must account for all the ports used by the H.225 and H.245 signaling.
Reference: Cisco Collaboration 9.x Solution Reference Network Design (SRND) page 4-34

Question 166
Which two security services are provided by the Phone Proxy function on a Cisco ASA appliance?
Which two security services are provided by the Phone Proxy function on a Cisco ASA appliance? (Choose two.)
A. It provides interworking to ensure that external IP phone traffic is encrypted, as long as the Cisco Unified Communications Manager cluster runs in secure mode.
B. It only applies to encrypted voice calls where both parties utilize encryption.
C. It manipulates the call signaling to ensure that all media is routed via the adaptive security appliance.
D. It supports encrypted TFTP operation of IP phone configuration files.
E. It intercepts and authenticates soft clients before they reach Cisco Unified Communications Manager clusters.
F. It requires a remote routing device with an IPsec VPN tunnel.

Correct Answer: CE

Explanation/Reference:
When using the Phone Proxy, the Cisco ASA appliance is inserted between the phones and Cisco Unified Communications Manager. The phones will now establish a TLS session with the ASA appliance. The appliance will, in turn, establish a proxy TLS connection with Cisco Unified Communications Manager on the phone’s behalf. This function generates two TLS sessions.

Question 167
What is the bandwidth that is guaranteed for voice signaling traffic with a DSCP value of CS3?
Refer to the exhibit.
Assume that the serial interface link bandwidth is full T1. What is the bandwidth that is guaranteed for voice signaling traffic with a DSCP value of CS3?

A. 33 percent of 1.544 Mb/s  
B. 5 percent of 1.544 Mb/s  
C. 38 percent of 1.544 Mb/s  
D. 62 percent of 1.544 Mb/s  
E. 0 percent of 1.544 Mb/s

Correct Answer: B

Explanation/Reference:
Under the policy map VOIP the CS3 value falls under the signal class-map, which has been allocated 5 percent of the bandwidth.

**Question 168**
Which Cisco enterprise medianet application class does Cisco Unified Personal Communicator belong?
To which Cisco enterprise medianet application class does Cisco Unified Personal Communicator belong?

A. VoIP Telephony  
B. Real-time Interactive  
C. Multimedia Conferencing  
D. Broadcast Video  
E. Low Latency Data

Correct Answer: C

Explanation/Reference:
Enterprise Medianet QoS Recommendations


**Question 169**
How much bandwidth should be allocated to the strict priority queue for eight VoIP calls that use a G.729 codec over a multilink PPP link with cRTP enabled?
Assume a 30-millisecond voice payload, 6 bytes for the Layer 2 header, 1 byte for the end-of-frame flag, and the IP, UDP, and RTP headers are compressed to 2 bytes, how much bandwidth should be allocated to the strict priority queue for eight VoIP calls that use a G.729 codec over a multilink PPP link with cRTP enabled?

A. 121.6 kb/s  
B. 92.8 kb/s  
C. 88.4 kb/s  
D. 83.2 kb/s  
E. 78.4 kb/s

Correct Answer: D

Explanation/Reference:

**Question 170**
How much bandwidth should be allocated to the strict priority queue for three VoIP calls that use a G.722 codec over an Ethernet network?
Assume 18 bytes for the Layer 2 header and a 10-millisecond voice payload, how much bandwidth should be allocated to the strict priority queue for three VoIP calls that use a G.722 codec over an Ethernet network?

A. 331.2 kb/s  
B. 347.8 kb/s  
C. 261.6 kb/s  
D. 303.2 kb/s  
E. 313.6 kb/s

Correct Answer: D
Question 171
Which entity signs a Cisco IP phone LSC?
A. Godaddy.com Enrollment Server
B. Manufacturer Certificate Authority
C. Registration Authority
D. Certificate Authority Proxy Function
E. Cisco Certificate Authority
Correct Answer: D
Explanation/Reference:

Question 172
Which two statements describe security services that are provided by the Phone Proxy function on a Cisco ASA appliance?
A. It is supported only on phones that use SCCP.
B. It is supported on an adaptive security appliance that runs in transparent mode.
C. It provides interworking to ensure that the external IP phone traffic is encrypted, as long as the Cisco Unified Communications Manager cluster runs in secure mode.
D. It provides a proxy of phone signaling, with optional use of NAT, to hide the Cisco Unified Communications Manager IP address from the public Internet.
E. It proxies phone media so that internal phones are not directly exposed to the Internet.
F. It supports IP phones that send phone proxy traffic through a VPN tunnel.
Correct Answer: DE
Explanation/Reference:

Question 173
Which statement describes the key security service that is provided by the TLS Proxy function on a Cisco ASA appliance?
A. It provides interworking to ensure that external IP phone traffic is encrypted, even if the rest of the system is unencrypted.
B. It only applies to encrypted voice calls where both parties utilize encryption.
C. It manipulates the call signaling to ensure that all media is routed via the adaptive security appliance.
D. It enables internal phones to communicate with external phones without encryption.
E. It protects Cisco Unified Communications Manager from rogue soft clients and attackers on the data VLAN.
Correct Answer: B
Explanation/Reference:

Question 174
What is the maximum amount of bandwidth allowed for priority queuing of RTP packets with a DSCP value of EF?
Refer to the exhibit.

Assume that the serial interface link bandwidth is full T1. What is the maximum amount of bandwidth allowed for priority queuing of RTP packets with a DSCP value of EF?
A. 33% of 1.544 Mb/s
B. 5% of 1.544 Mb/s
C. 38% of 1.544 Mb/s
D. 62% of 1.544 Mb/s
E. 0% of 1.544 Mb/s

Correct Answer: E

Explanation/Reference:
Since the use of the "priority" keyword was not used in this example 0% is the correct answer.

Question 175
Which Cisco enterprise medianet application class does Cisco TelePresence belong?

To which Cisco enterprise medianet application class does Cisco TelePresence belong?
A. VoIP Telephony
B. Real-time Interactive
C. Multimedia Conferencing
D. Broadcast Video
E. Low Latency Data

Correct Answer: B

Explanation/Reference:
Telepresence is used for video conferencing which can be done in Real-time so it is Real-time Interactive.

Question 176
Which statement describes the Cisco best practice recommendation about priority queue bandwidth allocation in relationship to the total link bandwidth when multiple strict priority LLQs are configured on the same router interface?

Which statement describes the Cisco best practice recommendation about priority queue bandwidth allocation in relationship to the total link bandwidth when multiple strict priority LLQs are configured on the same router interface?
A. Each LLQ should be limited to one-third of the link bandwidth capacity.
B. The sum of all LLQs should be limited to two-thirds of the link bandwidth capacity.
C. The sum of all LLQs should be limited to one-half of the link bandwidth capacity.
D. The sum of all LLQs should be limited to one-third of the link bandwidth capacity.
E. Cisco does not recommend more than one strict priority LLQ per interface.

Correct Answer: D

Explanation/Reference:
Cisco Technical Marketing testing has shown a significant decrease in data application response times when Real-Time traffic exceeds one-third of a link’s bandwidth capacity. Cisco IOS Software allows the abstraction (and, thus, configuration) of multiple LLQs. Extensive testing and production network customer deployments have shown that limiting the sum of all LLQs to 33 percent is a conservative and safe design ratio for merging real-time applications with data applications.

Question 177
Which algorithm is used to presort traffic going into the default queue?

In Cisco IOS routers that use low latency queuing, which algorithm is used to presort traffic going into the default queue?
A. first-in, first-out
B. last-in, first-out
C. weighted round robin
D. fair queuing
E. random processing

Correct Answer: D

Explanation/Reference:
WFQ is a flow-based queuing algorithm used in Quality of Service (QoS) that does two things simultaneously: It schedules interactive traffic to the front of the queue to reduce response time, and it fairly shares the remaining bandwidth between high bandwidth flows. A stream of packets within a single session of a single application is known as flow or conversation. WFQ is a flow-based method that sees over the network and ensures packet transmission efficiency which is critical to the interactive traffic. This method automatically stabilizes network congestion between individual packet transmission flows.

Question 178
How are queues serviced in Cisco IOS routers with the CBWFQ algorithm?

How are queues serviced in Cisco IOS routers with the CBWFQ algorithm?
A. first-in, first-out
B. weighted round robin based on assigned bandwidth
C. strict priority based on assigned priority
D. last-in, first-out
E. weighted round robin based on assigned priority

Correct Answer: B

Explanation/Reference:
Class Based Weighted Fair queuing is an advanced form of WFQ that supports user defined traffic classes i.e. one can define traffic classes based on match criteria like protocols, access control lists (ACLs), and input interfaces. A flow satisfying the match criteria for a class contributes the traffic for that particular defined class. A queue is allocated for each class, and the traffic belonging to that class is directed to the queue for that class.

Question 179
Which QoS tool category does compressed RTP belong?

To which QoS tool category does compressed RTP belong?
A. classification
B. marking
C. link efficiency
D. queuing
E. prioritization

Correct Answer: C
Explanation/Reference:
LLQ is a feature that provides a strict PQ to CBWFQ. LLQ enables a single strict PQ within CBWFQ at the class level. With LLQ, delay-sensitive data (in the PQ) is dequeued and sent first. In a VoIP with LLQ implementation, voice traffic is placed in the strict PQ.

Question 180
How much bandwidth should be allocated to the strict priority queue for six VoIP calls that use a G.729 codec over a multilink PPP link with cRTP enabled? Assume 20 bytes of voice payload, 6 bytes for the Layer 2 header, 1 byte for the end-of-frame flag, and the IP, UDP, and RTP headers are compressed to 2 bytes, how much bandwidth should be allocated to the strict priority queue for six VoIP calls that use a G.729 codec over a multilink PPP link with cRTP enabled?
A. 80.4 kb/s
B. 91.2 kb/s
C. 78.4 kb/s
D. 69.6 kb/s
E. 62.4 kb/s

Correct Answer: D
Explanation/Reference:
Voice payloads are encapsulated by RTP, then by UDP, then by IP. A Layer 2 header of the correct format is applied; the type obviously depends on the link technology in use by each router interface: A single voice call generates two one-way RTP/UDP/IP packet streams. UDP provides multiplexing and checksum capability; RTP provides payload identification, timestamps, and sequence numbering.

Question 181
How much bandwidth should be allocated to the strict priority queue for five VoIP calls that use a G.729 codec over a multilink PPP link?
Assume 6 bytes for the Layer 2 header, 1 byte for the end-of-frame flag, and a 40-millisecond voice payload, how much bandwidth should be allocated to the strict priority queue for five VoIP calls that use a G.729 codec over a multilink PPP link?
A. 87 kb/s
B. 134 kb/s
C. 102.6 kb/s
D. 77.6 kb/s
E. 71.3 kb/s

Correct Answer: A
Explanation/Reference:
Voice payloads are encapsulated by RTP, then by UDP, then by IP. A Layer 2 header of the correct format is applied; the type obviously depends on the link technology in use by each router interface: A single voice call generates two one-way RTP/UDP/IP packet streams. UDP provides multiplexing and checksum capability; RTP provides payload identification, timestamps, and sequence numbering.

Question 182
What is its codec bit rate in kilobits per second?
The iLBC codec operates at 38 bytes per sample per 20-millisecond interval. What is its codec bit rate in kilobits per second?
A. 6.3
B. 13.3
C. 15.2
D. 16
E. 24

Correct Answer: C
Explanation/Reference:
The internet Low Bit Rate Codec (iLBC) is designed for narrow band speech and results in a payload bit rate of 13.33 kbits per second for 30-millisecond (ms) frames and 15.20 kbits per second for 20 ms frames. When the codec operates at block lengths of 20 ms, it produces 364 bits per block, which is packetized as defined in RFC 3952. Similarly, for block lengths of 30 ms it produces 400 bits per block, which is packetized as defined in RFC 3952. The iLBC has built-in error correction functionality to provide better performance in networks with higher packet loss.

Question 183
Which option describes how this Cisco IOS SIP gateway, with an analog phone attached to its FXS port, handles an incoming informational SIP 180 response message with SDP?
Refer to the exhibit.
Which option describes how this Cisco IOS SIP gateway, with an analog phone attached to its FXS port, handles an incoming informational SIP 180 response message with SDP?

A. It will enable early media cut-through.
B. It will generate local ring back.
C. It will do nothing because the message is informational.
D. It will terminate the call because this is an unsupported message format.
E. It will take the FXS port offhook.

Correct Answer: B

Explanation/Reference:

Which two statements about calls that match dial-peer voice 7 voip are true?

Refer to the exhibit.

A. All calls that match dial-peer voice 7 use G.711.
B. All calls that match dial-peer voice 7 have the Diversion header removed from SIP Invites.
C. All calls that match dial-peer voice 7 use NOTIFY-based, out-of-band DTMF relay.
D. All calls that match dial-peer voice 7 are marked with DSCP 32.
E. All calls that match dial-peer voice 7 are marked with DSCP 34.

Correct Answer: BE

Explanation/Reference:

Dial peer 7 refers to SIP profile 102, which we can see is configured to have the Diversion header removed from SIP Invites. Dial peer 7 marks traffic with AF41, which is equivalent to DSCP 34.

Which ephone-dn can join the hunt group whenever a wild card slot becomes available?

Refer to the exhibit.
Which ephone-dn can join the hunt group whenever a wild card slot becomes available?
A. ephone-dn 1  
B. ephone-dn 2  
C. ephone-dn 3  
D. ephone-dn 4  
E. ephone-dn 6

Correct Answer: C
Explanation/Reference:

Question 186
How many calls, inbound and outbound combined, are supported on the IP phone?
Refer to the exhibit.

How many calls, inbound and outbound combined, are supported on the IP phone?
A. 1  
B. 2  
C. 8  
D. 12  
E. 50

Correct Answer: E
Explanation/Reference:
Explanation: Output incomplete to figure out the answer

Question 187
Which statement about a virtual SNR DN-configured Cisco Unified Communications Manager Express-enabled Cisco IOS router is true?

Which statement about a virtual SNR DN-configured Cisco Unified Communications Manager Express-enabled Cisco IOS router is true?
A. Virtual SNR DN supports either SCCP or SIP IP phone DNs.
B. A virtual SNR DN is a DN that is associated with multiple registered IP phones.
C. Calls in progress can be pulled back from the phone that is associated with the virtual SNR DN.
D. The SNR feature can only be invoked if the virtual SNR DN is associated with at least one registered IP phone.
E. A call that arrives before a virtual SNR DN is associated with a registered phone, and still exists after association is made, but cannot be answered from the phone.

Correct Answer: E
Explanation/Reference:
Virtual SNR DN only supports Cisco Unified SCCP IP phone DNs.
Virtual SNR DN provides no mid-call support.
Mid-calls are either of the following:
– Calls that arrive before the DN is associated with a registered phone and is still present after the DN is associated with the phone.
– Calls that arrive for a registered DN that changes state from registered to virtual and back to registered.
– Calls that cannot be pulled back, answered, or terminated from the phone associated with the DN.
State of the virtual DN transitions from ringing to hold or remains on hold as a registered DN.

Question 188
Which two statements the network components are true?
Refer to the exhibit.
The exhibit shows the Cisco IOS CLI output of debug ip dhcp packet, which was captured on a router that is located at a branch office where a single IP phone is located. There is a standalone Cisco Unified Communications Manager server at the central site, which also provides DHCP services to the IP phone at the branch office. You are troubleshooting a problem where the IP phone received an IP address in the correct subnet and with a correct subnet mask from the DHCP server, but never completed registration with Cisco Unified CM. Assuming the IP phone is correctly defined on Unified CM, which two statements the network components are true? (Choose two.)

A. The MAC address of the IP phone is 01ec44761e3e7d.
B. The IP address of the DHCP server is 10.101.15.1.
C. The MAC address of the VLAN 101 interface is 01ec44761e3e7d.
D. The MAC address of the IP phone is ec44761e3e7d.
E. There is no IP connectivity between the VLAN 101 interface of the branch router and the ip-helper address that is configured on this interface.
F. Based on the information provided, we cannot conclude if there is IP connectivity between the IP phone and Cisco Unified CM.

Correct Answer: DF

Explanation/Reference:
In the logs the only information that we get is about the mac address of the IP phone because the IP phone is raising the boot request.

Question 189
Which enrollment method does a Cisco IOS VPN router trustpoint use to install a Certificate Authority Proxy Function certificate for LSC validation of a Cisco IP phone client?

Which enrollment method does a Cisco IOS VPN router trustpoint use to install a Certificate Authority Proxy Function certificate for LSC validation of a Cisco IP phone client?

A. HTTP proxy server
B. certificate authority server URL
C. terminal
D. self-signed
E. registration authority

Correct Answer: C

Explanation/Reference:
Router(config)#crypto pki trustpoint CAPF
enrollment terminal
authorization username subjectname commonname
revocation-check none

Router(config)#crypto pki authenticate CAPF

Things to Note:
- The enrollment method is terminal because the certificate has to be manually installed on the Router.


Question 190
Which two are characteristics of jitter buffers?

Which two are characteristics of jitter buffers? (Choose two.)

A. Jitter buffers are used to change asynchronous packet arrivals into a synchronous stream by turning variable network delays into constant delays at the destination end systems.
B. Jitter buffers are used to change asynchronous packet arrivals into a synchronous stream by turning variable network delays into constant delays at the sending systems.
C. The role of the jitter buffer is to balance the delay and the probability of interrupted playout due to late packets.
D. The role of the jitter buffer is to queue late packets and reorder out-of-order packets.
E. Jitter buffers are used to change asynchronous packet arrivals into a synchronous stream by queuing packets into constant delays at the sending systems.

Correct Answer: AC

Explanation/Reference:
Jitter buffers are used to remove the effects of jitter so that asynchronous packet arrivals are changed to a synchronous stream. The jitter buffer trades off between delay and the probability of interrupted playout because of late packets (discard).

Reference: http://www.appneta.com/blog/jitter-voip/

Question 191
Which two statements about the network components are true?

Refer to the exhibit.

The exhibit shows the Cisco IOS CLI output of debug ip dhcp packet, which was captured on a router that is located at a branch office where a single IP phone is located. There is a standalone Cisco Unified Communications Manager server at the central site, which also provides DHCP services to the IP phone at the branch office. You are troubleshooting a problem where the IP phone could not register to Cisco Unified Communications Manager. You have confirmed that the IP phone received an IP address in the correct subnet and with a correct subnet mask from the DHCP server. Assuming the IP phone is correctly defined on Unified CM, which
Question 188
Which statement about what happens to a Cisco IOS SIP VoIP dial-peer that never received any responses to its out-of-dialog OPTIONS ping is true?

A. Its admin state will be up but operational state will be down.
B. Its admin and operational state will remain up.
C. Its admin and operational state will be "busy-out".
D. Its admin state will be up but operational state will be "busy-out".

Correct Answer: A

Explanation/Reference:
You can check the validity of your dial peer configuration by performing the following tasks:

- If you have relatively few dial peers configured, you can use the `show dial-peer voice` command to verify that the configuration is correct. To display a specific dial peer or to display all configured dial peers, use this command. The following is sample output from the `show dial-peer voice` command for a specific VoIP dial peer:

```
router# show dial-peer voice 10
VoiceOverIpPeer10
  tag = 10, dest-pat = Q',
  incall-number = Q+14087',
  group = 0, Admin state is up, Operation state is down
  Permission is Answer,
  type = voip, session-target = Q',
  sess,proto = cisco, req-qos = bestEffort,
  acc-qos = bestEffort,
  fax-rate = voice, codec = g729r8,
  Expect factor = 10, Keep = 30, VAD = disabled, Poor QOV Trap = disabled,
  Connect Time = 0, Charged Units = 0
  Successful Calls = 0, Failed Calls = 0
  Last Disconnect Cause is ""
  Last Disconnect Text is ""
  Last Setup Time = 0
  To show the dial peer that matches a particular number (destination pattern), use the `show dial-plan number` command. The following example displays the VoIP dial peer associated with the destination pattern 51234:

```
router# show dial-plan number 51234
Macro Exp.: 14085551234
VoiceOverIpPeer1004
  tag = 1004, destination-pattern = Q+1408555...',
  answer-address = Q',
  group = 1004, Admin state is up, Operation state is up
  type = voip, session-target = Qipv4:1.13.24.0
  ip precedence: 0 UDP checksum = disabled
  session-protocol = cisco, req-qos = best-effort,
  acc-qos = best-effort,
  fax-rate = voice, codec = g729r8,
  Expect factor = 10, Keep = 30, VAD = enabled, Poor QOV Trap = disabled,
  Connect Time = 0, Charged Units = 0
  Successful Calls = 0, Failed Calls = 0
  Last Disconnect Cause is ""
  Last Disconnect Text is ""
  Last Setup Time = 0
  Matched: +14085551234 Digits: 7
  Target: ipv4:172.13.24.0
```

Question 193
Which out-of-dialog SIP OPTIONS ping response put dial-peer tag 1111 into its current operational state?

Refer to the exhibit.

```
Router#show dial-peer voice sum
dial-peer voice sum
  dial-peer: 1111
  tag: 1111
  type: voip
  min: 1000
  prefix: 10.1.1.0
  dest-pattern: 10.1.1.1
  sess-target: Qipv4:10.1.1.1
  req-qos: best-effort
  acc-qos: best-effort
  fax-rate: voice
  codec: g729r8
  keep: 30
  expect: 10
  vad: enabled
  poor qov trap: disabled
  connect time: 0
  charged units: 0
  successful calls: 0
  failed calls: 0
  last disconnect cause: ""
  last disconnect text: ""
  last setup time: 0
  matched: +14085551234 digits: 7
  target: ipv4:172.13.24.0
```

Which out-of-dialog SIP OPTIONS ping response put dial-peer tag 1111 into its current operational state?
A. 501 Not Implemented
B. 504 Server Time-out
C. 408 Request Timeout
Question 194
Which three options are valid per-session video conference participants supported on the Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3? (Choose three.)
A. 3
B. 4
C. 6
D. 8
E. 9
F. 12
G. 16

Correct Answer: BDG

Explanation: The integrated video conferencing services use the same DSP resources on PVDM3s that are used for widely deployed ISR G2 voice capabilities. These modules, in conjunction with Cisco IOS Software, perform audio and video mixing, video transcoding for certain resolutions, and other functions for video endpoints. PVDM3 modules support flexible media resources and conference profile management to maximize capacity with predictable enduser experiences. Both homogenous and heterogeneous video conferences are supported. A homogenous conference refers to one in which participants connect to the ISR G2 with devices that support the same video format attributes (for example, the same codec, resolution, frame rate, and bit rate). A heterogeneous conference refers to one in which participants can connect to a conference bridge with devices that support different video format attributes. Each conference allows 4, 8, or 16-party participants.

Question 195
Which Cisco packet voice and video digital signal processor 3 can be used for video mixing on a Cisco Integrated Router Generation 2?
A. PVDM3-16
B. PVDM3-32
C. PVDM3-64
D. PVDM3-128

Correct Answer: D

Explanation: All the PVDM3 types (that is, PVDM3-16, PVDM3-32, PVDM3-64, PVDM3-128, PVDM3-192, and PVDM3-256) support switched-only video conferences. Only PVDM3-128 and higher modules support video conferencing with video mixing, transcoding and transrating.

Question 196
Which Cisco IOS multipoint video conferencing profile is also known as best-effort video on the Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3?
A. homogeneous
B. guaranteed-audio
C. rendezvous
D. heterogeneous
E. flex mode video

Correct Answer: B

Explanation: Three types of video profiles are supported: homogeneous conferences (video switching), heterogeneous conferences (video mixing), and guaranteed audio conferences (best-effort video).
As the name suggests, when Guaranteed Audio Conferences is configured, the system attempts to display video for all participants; however, it does not guarantee that the video of all participants is displayed. For those participants whose video is not displayed, participants are downgraded to audio-only and the profile guarantees preservation of the audio portion of the call. This option gives you added flexibility because the DSPs are not all reserved when the profile is created; the system attempts to reserve them when this profile is activated with an actual conference. For example:
dispfarm profile 1 conference video guaranteed-audio
codec h264 vmp

codec h264 4cf


Question 197
Which two Cisco IOS multipoint video conferencing profiles are supported on the Cisco Integrated Router Generation 2 with packet voice and video digital signal processor 3? (Choose two.)
A. homogeneous
B. rendezvous
C. guaranteed-audio
D. scheduled
E. guaranteed-video
F. ad-hoc
Question 198
Which two analog telephony signaling methods are most vulnerable to glare conditions?
Which two analog telephony signaling methods are most vulnerable to glare conditions? (Choose two.)
A. FXS Loop-start
B. FXO Ground-start
C. E&M Wink-start
D. E&M Delay-dial
E. E&M Immediate-start
F. E&M Feature Group D

Correct Answer: AE

Explanation:
The loop start signaling method is more common and is typically used by residential phone lines. When a voice port is configured with loop start signaling, the device (telephone) closes the circuit loop that signals the CO voice port to provide dial tone; an incoming call is signaled on the CO by supplying a predefined voltage on the line. The loop start signaling method has one main disadvantage in that it has no method of preventing both sides of the connection from attempting to seize the line at the same time; this condition is referred to as glare. Because of this, loop start signaling is typically not used on high demand circuits.

With immediate-start, the calling side of the connection seizes the line by going off hook on the E-lead and address information is sent using dual-tone multifrequency (DTMF) digits. Immediate start signaling is vulnerable to glare just like loop-start signaling.

Question 199
Which option describes the method used by Cisco IOS gateways to tunnel QSIG signaling messages in H.323 protocol?
Refer to the exhibit.

Which option describes the method used by Cisco IOS gateways to tunnel QSIG signaling messages in H.323 protocol?
A. H.323 Annex M1
B. H.323 Annex M2
C. H.323 Annex A
D. ISDN Generic Transparency Descriptor
E. H.450.1

Correct Answer: D

Explanation:
H.323 is an umbrella recommendation that encompasses various ITU-T recommendations, primarily recommendations H.225.0 and H.245 (basic communication capabilities) and recommendation H.450.1 (generic functional protocol for the support of supplementary services). Tunneling QSIG over H.323 is specified in H.323 Annex-M1. However, Cisco IOS Generic Software H.323 QSIG tunneling does not implement Annex-M1 (as the Cisco Unified Communications Manager H.323 implementation does). Instead, it uses the ISDN Generic Transparency Descriptor (GTD) to transport QSIG messages in the corresponding H.225 message to another Cisco gateway device on the other side of the network.


Question 200
Which number is sent as the caller ID when a user at extension 5001 places a call that matches this translation profile?
Refer to the exhibit.

Which number is sent as the caller ID when a user at extension 5001 places a call that matches this translation profile?
A. 14087775001
B. +4087775001
C. 4087750001
D. +14087775001

Correct Answer: D

Explanation:
When someone dials 5001, it will match rule 2 because it exactly starts with 5(five) using the ^ sign and ends with [0-9] followed by $. In replace pattern you can see +1408777 & means all set of match pattern. Thus, +14087775001.
Which two statements about the show command output are true? (Choose two.)

A. T1 0/2/1 terminates Q.921 signaling to a Cisco Unified Communications Manager server.
B. T1 0/0/0 terminates Q.921 signaling on the gateway.
C. T1 0/0/0 terminates SIP Signaling to a Cisco Unified Communications Manager server.
D. T1 0/0/0 terminates Q.931 signaling to a Cisco Unified Communications Manager server.
E. T1 0/2/1 terminates Q.931 signaling on the gateway.

Correct Answer: BD

Explanation:
As you can see the T1 0/0/0:23 interface is active in layer 1,2,3(multi frame established) & 3, it means Q.931 signaling terminates at gateway and using backhauled technique q931 messages are going to CUCM server.

But in case of T1 0/2/1 port multi frames are not established in layer 2, so it's not configured properly & doesn't backhauling q931 messages to CUCM server.

Question 202

Which two responses from a SIP device, which is the only remote destination on a Cisco Unified Communications Manager SIP trunk with OPTIONS ping enabled, cause the trunk to be marked as “Out of Service”? (Choose two.)

A. 503 Service Unavailable
B. 408 Request Timeout
C. 505 Version Not Supported
D. 504 Server Timeout
E. 484 Address Incomplete
F. 404 Not Found

Correct Answer: AB

Explanation:
The remote peer may be marked as Out of Service if it fails to respond to OPTIONS, if it sends 503 or 408 responses, or if the Transport Control Protocol (TCP) connection cannot be established. If at least one IP address is available, the trunk is In Service; if all IP addresses are unavailable, the trunk is Out of Service.

Reference:

Question 203

Which action can the Cisco Unified Communications Manager systems administrator use to change the response to “200 OK”? (Refer to the exhibit.)

A. Make sure the gateway IP address of the SIP VoIP service provider is defined correctly in Cisco Unified Communications Manager SIP trunk.
B. Enable OPTIONS ping on Cisco Unified Communications Manager SIP trunk.
C. Disable OPTIONS ping on Cisco Unified Communications Manager SIP trunk.
D. Create an SIP response alias to force outgoing 401 messages to “200 OK”.
E. Disable digest authentication on Cisco Unified Communications Manager SIP trunk.

Correct Answer: E

Explanation:
Because Right now CUCM challenges the identity of a SIP user agent and must configure digest credentials for the application user in CUCM or you have to disable it for stop challenging by CUCM.

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/9.0/1/secugd/CUCM_BK_CCB00C40_00_cucm-security-guide-90/CUCM_BK_CCB00C40_00_cucm-security-guide_chapter_011010.html
Question 204
Which two statements about the restrictions for support of H.239 are true?
Which two statements about the restrictions for support of H.239 are true? (Choose two.)
A. SIP to H323 video calls using H.239 are not supported.
B. Redundancy for H.323 calls is not supported.
C. H.239 calls are not supported over intercluster trunks with Cisco Unified Communications Manager.
D. H.239 is not supported with third-party endpoints.
E. Cisco Unified Communications Manager supports a maximum of three video channels when using H.239.
Correct Answer: AB
Explanation/Reference:
Restriction for Support for H.239
The Support for H.239 feature has the following restrictions:
- Interworking SIP-H.323 Video calls using H.239 is not supported.
- Redundancy for H.323 calls is not supported.
- A fast-start request cannot include a request to open an H.239 additional video channel as it is not supported.
- H.239 systems based on H.235 is not supported.
- The SBC does not support call transfer for H.323 calls. When an H.323 endpoint is placed on hold, it closes its media as well as video channels.

Question 205
How many signaling bits are there in each T1 time slot using channel associated signaling with Super Frame?
How many signaling bits are there in each T1 time slot using channel associated signaling with Super Frame?
A. 1
B. 2
C. 4
D. 8
E. 12
Correct Answer: B
Explanation/Reference:
Each T1 CAS has 24 channels that can transmit 8 bits per channel each. This gives us a total of 192 bits. The T1 has one additional bit for framing, bringing the total to 193 bits. Two types of line coding can be used on a T1 CAS. The first type of line coding is called Super Frame (SF). This is an older and less – efficient type of framing. Super Frame bundles 12 of these 193 – bit frames together for transport. It then uses the even numbered frames as signaling bits. The T1 CAS signaling then looks at every sixth frame for signaling information. This comes out to be 2 bits that are referred to as the A and B bits, which reside in frames 6 and 12.

Question 206
Which digital modulation method is used to transmit caller ID information on analog FXS ports on Cisco IOS routers?
Which digital modulation method is used to transmit caller ID information on analog FXS ports on Cisco IOS routers?
A. DTMF
B. PSK
C. FSK
D. MF
E. pulse dialing
Correct Answer: C
Explanation/Reference:

Question 207
Which chipset is the PVDM2-32 DSP hardware based on?
In Cisco IOS routers, which chipset is the PVDM2-32 DSP hardware based on?
A. C5441
B. C549
C. C5510
D. C5421
E. Broadcom 1500
Correct Answer: C
Explanation/Reference:
Table 6.2 DSP Resources on Cisco IOS Hardware Platforms with C5510 Chipset

Table | DSP Configuration | Maximum Number of Voice Terminations (Calls) per DSP and per Module (5 calls per DSP)

<table>
<thead>
<tr>
<th>Hardware Model or Cheassis</th>
<th>Medium Complexity</th>
<th>High Complexity</th>
</tr>
</thead>
<tbody>
<tr>
<td>VG-224</td>
<td>NA</td>
<td>NA</td>
</tr>
<tr>
<td>NM-HD-1V2</td>
<td>4 calls per NM</td>
<td>4 calls per NM</td>
</tr>
<tr>
<td>NM-HD-2V</td>
<td>8 calls per NM</td>
<td>6 calls per NM</td>
</tr>
<tr>
<td>NM-HD-2VE</td>
<td>24 calls per NM</td>
<td>24 calls per NM</td>
</tr>
<tr>
<td>NM-HD2</td>
<td>3 calls per NM</td>
<td>3 calls per NM</td>
</tr>
<tr>
<td>NM-HD2 V2</td>
<td>8 calls per NM</td>
<td>6 calls per NM</td>
</tr>
<tr>
<td>NM-HD2 V2</td>
<td>16 calls per NM</td>
<td>12 calls per NM</td>
</tr>
<tr>
<td>NM-HD2 V2</td>
<td>24 calls per NM</td>
<td>18 calls per NM</td>
</tr>
<tr>
<td>NM-HD2 V2</td>
<td>32 calls per NM</td>
<td>24 calls per NM</td>
</tr>
<tr>
<td>NM-HD2 V4</td>
<td>48 calls per NM</td>
<td>36 calls per NM</td>
</tr>
<tr>
<td>NM-HD2 V4</td>
<td>64 calls per NM</td>
<td>48 calls per NM</td>
</tr>
<tr>
<td>NM-HD2 V4</td>
<td>96 calls per NM</td>
<td>72 calls per NM</td>
</tr>
</tbody>
</table>

http://www.aoowe.com/practice-400-051-3156.html

Question 208
Which ds0-group option should you select to support automated number identification information collection on inbound calls for this digital T1 voice circuit? Refer to the exhibit.

A. e&m-wink-start
B. e&m-delay-dial
C. e&m-delay-dial
D. e&m-lmr
E. e&m-fgd

Correct Answer: E
Explanation/Reference:
Because it can receive ANI information and sends DNIS info. But can't send ANI.

Question 209
How many signaling bits does each T1 timeslot have?
In Channel Associated Signaling on a T1 circuit using Extended Super Frame, how many signaling bits does each T1 timeslot have?
A. 1
B. 2
C. 4
D. 12
E. 24

Correct Answer: C
Explanation/Reference:
Each T1 channel carries a sequence of frames. These frames consist of 193 bits and an additional bit designated as the framing bit, for a total of 193 bits per frame. Super Frame (SF) groups twelve of these 193 bit frames together and designates the framing bits of the even numbered frames as signaling bits. CAS looks specifically at every sixth frame for the timeslot’s or channel’s associated signaling information. These bits are commonly referred to as A- and B-bits. Extended super frame (ESF), due to grouping the frames in sets of twenty-four, has four signaling bits per channel or timeslot. These occur in frames 6, 12, 18, and 24 and are called the A-, B-, C-, and D-bits respectively.

Question 210
Which two types of line codes are configurable for an E1 PRI controller on a Cisco IOS router? (Choose two.)
A. CRC4
B. AMI
C. B8ZS
D. HDB3
E. ESF
F. SF

Correct Answer: BD
Explanation/Reference:
Configuring an NM-xCE1T1-PRI Card for an E1 Interface
Perform this task to select and configure an NM-xCE1T1-PRI network module card as E1.
SUMMARY STEPS
1. enable
2. configure terminal
3. card type e1 slot
4. controller e1 slot / port
5. linecode {ami | hdb3}
6. framing {crc4 | no-crc4}

Question 211
Which method allows administrators to determine the best match impedance on analog voice ports in Cisco IOS router without having to shut and no shut the ports?
A. THL tone sweep
B. original tone sweep
C. ECAN test
D. inject-tone local sweep
E. remote loop

Correct Answer: A

Explanation/Reference:

THL tone sweep allows all available impedances for a single test call to a quiet termination point out to the PSTN. You do not need to manually disable ECAN on the voice port under test. The test feature switches impedances automatically for the tester. The test feature calculates the arithmetic mean ERL and reports the mean for each channel profile at each impedance setting. Then, at the end of the test, the feature specifies the best match impedance setting. This test requires minimal supervision.


Question 212

How many calls are native to this Cisco Unified Border Element?

Refer to the exhibit.

<table>
<thead>
<tr>
<th>CUBE-2#show voice high-availability summary</th>
<th>Voice HA DB INFO</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of calls in HA DB: 28 (MAX: 2048)</td>
<td></td>
</tr>
<tr>
<td>Number of calls in HA sync pending DB: 12</td>
<td></td>
</tr>
<tr>
<td>Number of calls in HA preserved session DB: 9</td>
<td></td>
</tr>
</tbody>
</table>

This output was captured on a Cisco IOS gateway shortly after it became the active Cisco Unified Border Element in a box-to-box redundancy failover. How many calls are native to this Cisco Unified Border Element?

A. 9
B. 12
C. 19
D. 31
E. 40

Correct Answer: D

Explanation/Reference:

Total no of calls = 28 + 12 = 40.
So, native calls are 40 - 9 = 31.


Question 213

Which two Cisco Unified Communications Manager Express hunt group mechanisms keep track of the number of hops in call delivery decisions?

Which two Cisco Unified Communications Manager Express hunt group mechanisms keep track of the number of hops in call delivery decisions? (Choose two.)

A. sequential
B. peer
C. longest idle
D. parallel
E. overlay
F. linear

Correct Answer: BC

Explanation/Reference:

Peer configures hunting in a circular manner among the hunt group member DNs and starts with the DN to the right of the last DN to ring. Longest-idle specifies hunting on the DN which is idle for a longest period of time and the call will go to that DN of the hunt Group.

Reference: http://ccievoice.ksiazek.be/?p=690

Question 214

How much time does a member of the hunt group have to answer a queue call that is ringing on their extension?

Refer to the exhibit.
Assume the B-ACD configuration on a Cisco Unified Communications Manager Express router is operational.

How much time does a member of the hunt group have to answer a queue call that is ringing on their extension?

A. 5 seconds  
B. 10 seconds  
C. 20 seconds  
D. 30 seconds  
E. 40 seconds  

Correct Answer: B

Explanation/Reference:

As you can see the timeout 10 sec in ephone-hunt 1 means hunt group members have to answer the queued call within 10 sec.

Question 215

Which option describes what will happen to an incoming call that entered the call queue but all members of the hunt group are in Do Not Disturb status?

Refer to the exhibit.

A. The call is forwarded to extension 2120.  
B. The call is forwarded to extension 2220.  
C. The call is forwarded to extension 2003.  
D. The call is disconnected with user busy.  
E. The call is forwarded to extension 2100.  

Correct Answer: B
Explanation/Reference:
Explanation:
Because all members of hunt group are unavailable or activate DnD and incoming queued call will forward to voicemail using the param voice-mail 2220 command.

Question 216
Which option describes what will happen to the fourth incoming call?
Refer to the exhibit.

| voice register dn 1
| number 2001
| call-forward b2bua busy 2300
call-forward b2bua noan 2200 timeout 20
| huntstop channel 3
| voice register pool 1
busy-trigger-per-button 3
id mac 1111.1111.1111
| type 7965
| number 1 dn 1
| voice register pool 2
busy-trigger-per-button 2
id mac 2222.2222.2222
| type 7965
| number 1 dn 1

IP phone 1 has MAC address of 1111.1111.1111, and IP phone 2 has MAC address of 2222.2222.2222. The first two incoming calls were answered by IP phone 1, and the third incoming call was answered by IP phone 2.

Which option describes what will happen to the fourth incoming call?
A. Both phones ring, but only IP phone 2 can answer the call.
B. Both phones ring and either phone can answer the call.
C. Both phones ring, but only IP phone 1 can answer the call.
D. Neither phone rings and the call is forwarded to 2100.
E. Neither phone rings and the call is forwarded to 2200.

Correct Answer: D
Explanation/Reference:
Explanation:
IP Phone 1 & 2 both have busy-trigger-per-button configured to 3 & 2 respectively. So, the 4th incoming call will get forwarded to 2100 as busy-triggers are exceeding in IP Phones.

Question 217
Which option describes what will happen to the third incoming call?
Refer to the exhibit.

| voice register dn 1
| number 2001
| call-forward b2bua busy 2100
call-forward b2bua noan 2200 timeout 20
| huntstop channel 3
| voice register pool 1
busy-trigger-per-button 2
id mac 1111.1111.1111
| type 7965
| number 1 dn 1
| voice register pool 2
busy-trigger-per-button 2
id mac 2222.2222.2222
| type 7965
| number 1 dn 1

IP phone 1 has MAC address of 1111.1111.1111, and IP phone 2 has MAC address of 2222.2222.2222. The first two incoming calls rang both phones and were answered by IP phone 2.

Which option describes what will happen to the third incoming call?
A. Both phones ring, but only IP phone 1 can answer the call.
B. Both phones ring and either phone can answer the call.
C. Only IP phone 1 rings and can answer the call.
D. Neither phone rings and the call is forwarded to 2100.
E. Neither phone rings and the call is forwarded to 2200.

Correct Answer: C
Explanation/Reference:
Explanation:
As we can see busy-trigger-per-button set to 2 in voice register pool 1(IP Phone 1). So, IP Phone 1’s channel is free for receiving incoming calls and right now IP Phone 2 is busy answering call.

Question 218
How many additional calls can be placed from ephone 1?
Refer to the exhibit.

http://www.aoowe.com/practice-400-051-3156.html
Three calls are active on ephone 1. Assume ephone 2 will remain idle.
How many additional calls can be placed from ephone 1?
A. 0
B. 1
C. 2
D. 3
E. 5
Correct Answer: C
Explanation/Reference:
Explanation:
As we can see max-calls-per-button set to 5 and 3 calls are active. So, 2 calls remain.

Question 219
How many inbound calls can be handled simultaneously between ephone 1 and ephone 2 before a user busy tone is returned?
Refer to the exhibit.
A. 6
B. 7
C. 8
D. 9
E. 11
Correct Answer: A
Explanation/Reference:
Explanation:
Because hunt stop channel is set to 6 as it enables call hunting to up to six channels of this ephone-dn and remaining 2 channels are available for outgoing call features.

Question 220
Which call state does the Mobility soft key act as a toggle key to enable or disable Single Number Reach for Cisco Unified Communications Manager Express SCCP IP phones?
In which call state does the Mobility soft key act as a toggle key to enable or disable Single Number Reach for Cisco Unified Communications Manager Express SCCP IP phones?
A. idle
B. seized
C. alerting
D. ringing
E. connected
Correct Answer: A
Explanation/Reference:
Explanation:
Pressing the Mobility soft key during the idle call state enables the SNR feature. This key is a toggle; pressing it a second time disables SNR.


Question 221
Which option describes how this Cisco IOS SIP gateway, with an analog phone attached to its FXS port, handles an incoming informational SIP 180 response message without SDP?

Refer to the exhibit.

![SIP gateway show sip-ua status](image)

Which option describes how this Cisco IOS SIP gateway, with an analog phone attached to its FXS port, handles an incoming informational SIP 180 response message without SDP?
A. It will enable early media cut-through.
B. It will generate local ring back.
C. It will do nothing because the message is informational.
D. It will terminate the call because this is an unsupported message format.
E. It will take the FXS port offhook.

Correct Answer: B

Explanation/Reference:
The Session Initiation Protocol (SIP) feature allows you to specify whether 180 messages with Session Description Protocol (SDP) are handled in the same way as 183 responses with SDP. The 180 Ringing message is a provisional or informational response used to indicate that the INVITE message has been received by the user agent and that alerting is taking place. The 183 Session Progress response indicates that information about the call state is present in the message body media information. Both 180 and 183 messages may contain SDP, which allows the early media session to be established prior to the call being answered.

Prior to this feature, Cisco gateways handled a 180 Ringing response with SDP in the same manner as a 183 Session Progress response; that is, the SDP was assumed to be an indication that the far end would send early media. Cisco gateways handled a 180 response without SDP by providing local ringback, rather than early media cut-through. This feature provides the capability to ignore the presence or absence of SDP in 180 messages, and as a result, treat all 180 messages in a uniform manner. The SIP—Enhanced 180 Provisional Response Handling feature allows you to specify which call treatment, early media or local ringback, is provided for 180 responses with SDP.


Question 222
Which codec is supported on the Cisco PVDM2 DSP modules but not on the PVDM3 DSP modules?
A. G.728
B. G.729B
C. G.729AB
D. G.723
E. G.726

Correct Answer: D

Explanation/Reference:
All codecs that are supported on the PVDM2 are supported on the PVDM3, except that the PVDM3 does not support the G.723 (G.723.1 and G.723.1A) codecs. The PVDM2 can be used to provide G.723 codec support or the G.729 codec can be used as an alternative on the PVDM3.


Question 223
Which chipset is the PVDM-12 DSP hardware based on?
In Cisco IOS routers, which chipset is the PVDM-12 DSP hardware based on?
A. C542
B. C549
C. C5510
D. C5421
E. C5409

Correct Answer: B

Explanation/Reference:
NA-HDV has five SIMM sockets (called Banks) that hold the PVDM-12 cards. Each PVDM-12 card contains three TI 549 DSPs.

Question 224
Which SIP message element is mapped to QSIG FACILITY messages being tunneled across a SIP trunk between two Cisco IOS gateways?
A. SIP UPDATE
B. SIP OPTIONS
C. SIP SUBSCRIBE
D. SIP INFO
E. SIP NOTIFY

Correct Answer: D

Explanation/Reference:
Mapping of QSIG Message Elements to SIP Message Elements
This section lists QSIG message elements and their associated SIP message elements when QSIG messages are tunneled over a SIP trunk.

Reference:

Question 225
Which message is used by a Cisco IOS MGCP gateway to send periodic keepalives to its call agent?
A. CRCX
B. AUCX
C. NTFY
D. RQNT
E. 200 OK

Correct Answer: C

Explanation/Reference:
The gateway maintains this connection by sending empty MGCP Notify (NTFY) keepalive messages to Cisco CallManager at 15-second intervals. If the active Cisco CallManager fails to acknowledge receipt of the keepalive message within 30 seconds, the gateway attempts to switch over to the next highest order Cisco CallManager server that is available. If none of the Cisco CallManager servers respond, the gateway switches into fallback mode and reverts to its default H.323 session application for basic call control support of IP telephony activity in the network.

Question 226
What will the controller T1 1/1 D-channel status be in the output of the show isdn status command?

Assuming the ISDN-enabled T1 PRI configuration on a Cisco IOS router is fully functional, what will the controller T1 1/1 D-channel status be in the output of the show isdn status command?
A. MULTIPLE_FRAME_ESTABLISHED
B. TEI_ASSIGNED
C. AWAITING_ESTABLISHMENT
D. STANDBY
E. INITIALIZED

Correct Answer: B

Explanation/Reference:
TEI_ASSIGNED, which indicates that the PRI does not exchange Layer 2 frames with the switch. Use the show controller t1s command to first check the controller t1 circuit, and verify whether it is clean (that is, it has no errors) before you troubleshoot ISDN Layer 2 problem with the debug isdn q921.

Question 227
How many bearer channels are available to carry voice traffic?

Refer to the exhibit.
From this NFAS-enabled T1 PRI configuration on a Cisco IOS router, how many bearer channels are available to carry voice traffic?

A. 91
B. 92
C. 93
D. 94
E. 95

Correct Answer: E

Explanation:
A T1 circuit typically carries 24 individual timeslots. Each timeslot in turn carries a single telephone call. When a T1 circuit is used to carry Primary Rate ISDN one of the timeslots is used to carry the D channel. A single Primary Rate ISDN circuit is thus sometimes described as 23B + D. There are 23 bearer channels carrying voice or data, and one D channel carrying the Common Channel Signaling. In this case, there are 96 total channels in the group, but only 1 will be needed for use as the D channel, leaving 95 available for bearer channels.

Question 228
Which type of voice port is it?
Refer to the exhibit:

```
voice-port 1/1/0
   caller-id enable
   station-id number 5251234
   station-id name cisco
   ring number 6
```

In an effort to troubleshoot a caller ID delivery problem, a customer emailed you the voice port configuration on a Cisco IOS router. Which type of voice port is it?

A. FXS
B. E&M
C. BRI
D. FXO
E. DID

Correct Answer: D

Explanation:
Configuring FXS and FXO Voice Ports to Support Caller ID

To configure caller-ID on FXS and FXO voice ports, use the following commands beginning in global configuration mode:

```
isdn switch-type primary-dms100
controller T1 1/0
   framing esf
   linecode b8zs
   pri-group timeslots 1-24 nfas_d primary nfas_int 0 nfas_group 1
controller T1 1/1
   framing esf
   linecode b8zs
   pri-group timeslots 1-24 nfas_d backup nfas_int 1 nfas_group 1
controller T1 2/0
   framing esf
   linecode b8zs
   pri-group timeslots 1-24 nfas_d none nfas_int 2 nfas_group 1
controller T1 2/1
   framing esf
   linecode b8zs
   pri-group timeslots 1-24 nfas_d none nfas_int 3 nfas_group 1
```
Question 229
Which two statements describe characteristics of Cisco Unified Border Element high availability, prior to Cisco IOS release 15.2.3T, using a box-to-box redundancy configuration?

Which two statements describe characteristics of Cisco Unified Border Element high availability, prior to Cisco IOS release 15.2.3T, using a box-to-box redundancy configuration? (Choose two.)

A. It leverages HSRP for router redundancy and GLBP for load sharing between a pair of routers.
B. Cisco Unified Border Element session information is checkpointed across the active and standby router pair.
C. It supports media and signal preservation when a switchover occurs.
D. Only media streams are preserved when a switchover occurs.
E. It can leverage either HSRP or VRRP for router redundancy.

Correct Answer: BD

Explanation/Reference:

Configure box-to-box redundancy when you:

- Expect the behavior of the CSSs to be active/standby (only the master CSS processes flows)
- Can configure a dedicated FastEthernet (FE) link between the CSSs for the VRRP heartbeat

Do not configure box-to-box redundancy when you:
- Expect the behavior of the CSSs to be active-active (both CSSs processing flows). Use VIP redundancy instead.
- Cannot configure and dedicated FE link between the CSSs.
- Require the connection of a Layer 2 device between the redundant CSS peers.

Question 230
Which Cisco Unified Communications Manager Express ephone button configuration separator enables overflow lines when the primary line for an overlay button is occupied by an active call?

Which Cisco Unified Communications Manager Express ephone button configuration separator enables overflow lines when the primary line for an overlay button is occupied by an active call?

A. o
B. c
C. w
D. x
E. :

Correct Answer: D

Explanation/Reference:

x expansion/overflow, define additional expansion lines that are used when the primary line for an overlay button is occupied by an active call.

Question 231
Which call hunt mechanism is only supported by the voice hunt group in a Cisco Unified Communications Manager Express router?

Which call hunt mechanism is only supported by the voice hunt group in a Cisco Unified Communications Manager Express router?

A. sequential

http://www.aoowe.com/practice-400-051-3156.html
**Question 232**

Which two statements are requirements regarding hunt group options for B-ACD implementation on Cisco Unified Communications Manager Express routers?

(Choose two.)

A. The ephone hunt group is mandatory.
B. Either the ephone hunt group or the voice hunt group is acceptable.
C. Hunt group members must be SCCP IP phones.
D. Hunt group members can include both SCCP or SIP IP phones.
E. Hunt group members must be SIP IP phones.
F. The member hunting mechanism must be set to sequential.

Correct Answer: AC

**Explanation/Reference:**

The ephone hunt group is mandatory, and while ephone hunt groups only support Cisco Unified SCCP IP phones, a voice hunt group supports either a Cisco Unified SCCP IP phone or a Cisco Unified SIP IP phone.


**Question 233**

What will happen to a new call that enters the call queue when there are already two calls in queue?

Refer to the exhibit.

A. The call will be forwarded to extension 2120.
B. The call will be forwarded to extension 2220.
C. The call will be forwarded to extension 2003.
D. The call will be disconnected with user busy.
E. The call will be forwarded to 2100.

Correct Answer: C

**Explanation/Reference:**

That is because queue over flow is forwarded to 2003 and maximum number of calls in queue is configured as two.

**Question 234**

What will happen to a call in queue that was not answered by any member of the hunt group after the maximum amount of time allowed in the call queue expires?

Refer to the exhibit.
Assume the B-ACD configuration on a Cisco IOS Cisco Unified Communications Manager Express router is operational. What will happen to a call in queue that was not answered by any member of the hunt group after the maximum amount of time allowed in the call queue expires?

A. The call will be forwarded to extension 2120.
B. The call will be forwarded to extension 2220.
C. The call will be forwarded to extension 2003.
D. The call will be disconnected with user busy.
E. The call will be forwarded to 2100.

Correct Answer: B

Explanation:
As we can see in the configuration 2220 is configured as voice mail forwarding extension so the call will forward to voice mail.

Question 235
When multiple greetings are enabled on Cisco Unity Express, which greeting will take the highest precedence?

A. standard
B. meeting
C. busy
D. closed
E. internal

Correct Answer: B

Explanation:
Meeting greeting has the highest priority because it is set by the user when he doesn’t want to take the call and notices the caller he is online.

Question 236
Which type of mailbox on Cisco Unity Express can play a user greeting and disconnect the call, but cannot take or send messages?

A. PIN-less mailbox
B. announcement-only mailbox
C. general delivery mailbox
D. call-handling mailbox
E. personal mailbox

Correct Answer: B

Explanation:
Announcement-only mailbox is set for those users who only want the caller to listen the announcement and leave his message according to the announcement.

Question 237
What is the reason for these disconnected calls?

Refer to the exhibit:

Your customer sent you this debug output, captured on a Cisco IOS router (router A), to troubleshoot a problem where all H.323 calls that originate from another
Cisco IOS router (router B) are being dropped almost immediately after arriving at router A. What is the reason for these disconnected calls?

A. Calls were unsuccessful because of internal, memory-related problems on router A.
B. Calls were rejected because the called number was denied on a configured class of restriction list on router A.
C. Calls were rejected because the VoIP dial peer 1002 was not operational.
D. Calls were unsuccessful because the router B IP address was not found in the trusted source IP address list on router A.
E. Calls were rejected by router A because it received an admission reject from its gatekeeper because of toll fraud suspicion.

Correct Answer: D
Explanation/Reference:
Explanation: Trusted source IP address list on router is a list which secures the connectivity of router if it is enabled then we need to give the trusted entry for any route to reach.

Question 238
Which URL provides Cisco Unity Express end users with a GUI interface to access and manage their messages and mailbox settings?

A. http://10.1.1.1/Web/Common/Login.do
B. http://10.1.1.1/ciscopca
C. http://10.1.1.1/user
D. http://10.1.1.1/inbox
E. http://10.1.1.1/ 

Correct Answer: C
Explanation/Reference:
Explanation: For user access cisco unity has predefined url and it is http://10.1.1.1/user

Question 239
Which statement describes the question mark wildcard character in a SIP trigger that is configured on Cisco Unity Express?

Which statement describes the question mark wildcard character in a SIP trigger that is configured on Cisco Unity Express?

A. It matches any single digit in the range 0 through 9.
B. It matches one or more digits in the range 0 through 9.
C. It matches zero or more occurrences of the preceding digit or wildcard value.
D. It matches one or more occurrences of the preceding digit or wildcard value.
E. It matches any single digit in the range 0 through 9, when used within square brackets.

Correct Answer: C
Explanation/Reference:
Explanation:
Table 5-2 Trigger Pattern Wildcards and Special Characters

<table>
<thead>
<tr>
<th>Character</th>
<th>Description</th>
<th>Examples</th>
</tr>
</thead>
<tbody>
<tr>
<td>X</td>
<td>The X wildcard matches any single digit in the range 0 through 9.</td>
<td>The trigger pattern 900X matches all numbers in the range 900 through 999.</td>
</tr>
<tr>
<td>!</td>
<td>The exclamation point (!) wildcard matches one or more digits in the range 0 through 9.</td>
<td>The trigger pattern 91! matches all numbers in the range 910 through 999.</td>
</tr>
<tr>
<td>?</td>
<td>The question mark (?) wildcard matches zero or more occurrences of the preceding digit or wildcard value.</td>
<td>The trigger pattern 910? matches all numbers in the range 910 through 999.</td>
</tr>
<tr>
<td>+</td>
<td>The plus sign (+) wildcard matches one or more occurrences of the preceding digit or wildcard value.</td>
<td>The trigger pattern 91+ matches all numbers in the range 910 through 999.</td>
</tr>
<tr>
<td>[ ]</td>
<td>The square bracket ([ ]) characters enclose a range of values.</td>
<td>The trigger pattern 8135[10][012345] matches all numbers in the range 813501 through 8135165.</td>
</tr>
<tr>
<td>-</td>
<td>The hyphen (-) character, used with the square brackets, denotes range of values.</td>
<td>The trigger pattern 8135[10][012345] matches all numbers in the range 813501 through 8135165.</td>
</tr>
<tr>
<td>*</td>
<td>The circumflex (*) character, used with the square brackets, negates range of values.</td>
<td>The trigger pattern 8135[10][012345] matches all numbers in the range 81350165.</td>
</tr>
</tbody>
</table>

Question 240
What will happen to the fourth incoming call?
Refer to the exhibit.

http://www.aoowe.com/practice-400-051-3156.html
IP phone 1 has the MAC address 1111.1111.1111, while IP phone 2 has the MAC address 2222.2222.2222. The first two incoming calls were answered by IP phone 1, while the third incoming call was answered by IP phone 2. What will happen to the fourth incoming call?

A. Both phones will ring, but only IP phone 2 can answer the call.
B. Both phones will ring and either phone can answer the call.
C. Only IP phone 2 will ring and can answer the call.
D. Neither phone will ring and the call will be forwarded to 2100.
E. Neither phone will ring and the call will be forwarded to 2200.

Correct Answer: B
Explanation/Reference:
Explanation:
In shared line configuration phone share the same line so it is possible for any phone to answer the call.

Question 241
How many simultaneous inbound calls can be handled by these two IP phones?
Refer to the exhibit.

How many simultaneous inbound calls can be handled by these two IP phones?
A. 2
B. 4
C. 6
D. 9
E. 10

Correct Answer: A
Explanation/Reference:
Explanation:
The line is configured as shared line so it will support maximum two calls at a time.

Question 242
What will happen on ephone 1 when a fourth call arrives for extension 2001?
Refer to the exhibit.

Ephone 1 has three active calls. The first two calls were inbound calls, which the user put on hold to place a third call outbound. What will happen on ephone 1 when a fourth call arrives for extension 2001?
A. The fourth call will be delivered to ephone 1 because it only received two inbound calls, one call less than the busy-trigger-per-button setting.
B. The fourth call will be delivered to ephone 1 because the huntstop channel setting is not yet saturated.

A. The fourth call will be delivered to ephone 1 because it only received two inbound calls, one call less than the busy-trigger-per-button setting.
B. The fourth call will be delivered to ephone 1 because the huntstop channel setting is not yet saturated.
C. The fourth call will be delivered to ephone 1 because it can handle up to five calls on each button.
D. The fourth call will be held temporarily by the IOS Software until ephone 1 disconnects one of the active calls.
E. The fourth call will not be delivered and the caller will hear a user busy tone.

Correct Answer: E
Explanation/Reference:
Because on line maximum 4 calls can be placed when user put the call on hold is consume a channel and reach the maximum number of calls on line.

Question 243
How many simultaneous outbound calls are possible with this Cisco Unified Communications Manager Express configuration on these two phones? Refer to the exhibit.

```
ephone-dn 1 octo-line
number 2001
huntstop channel 6
ephone 1
mac-address 1111.1111.1111
max-calls-per-button 5
busy-trigger-per-button 3
type 7965
button 1:1
ephone 2
mac-address 2222.2222.2222
max-calls-per-button 6
busy-trigger-per-button 4
type 7965
button 1:1
```

How many simultaneous outbound calls are possible with this Cisco Unified Communications Manager Express configuration on these two phones?
A. 6
B. 7
C. 8
D. 9
E. 11

Correct Answer: C
Explanation/Reference:
Explanation:
Ephone is configured as octo line so maximum call number is 8 and it will be divided between lines.

Question 244
When DSP oversubscription occurs on a Cisco IOS router using DSP modules that are based on the C5510 chipset, what will happen when an analog phone connected to a FXS port goes off-hook?

A. A fast busy tone will be played.
B. A slow busy tone will be played.
C. A network busy tone will be played.
D. A dial tone will be played, but digits will not be processed.
E. No tone will be played.

Correct Answer: E
Explanation/Reference:
Explanation:
When DSP oversubscription occurs for both analog ports and digital ports, except PRI and BRI. FXO signaling and application controlled endpoints are not supported. This feature does not apply to insufficient DSP credits due to mid-call codec changes (while a call is already established).

Question 245
Which codec complexity mode, when deployed on Cisco IOS routers with DSPs using the C5510 chipset, supports the most G.711 calls per DSP?

A. Low
B. Medium
C. High
D. Secure
E. Flex

Correct Answer: E
Explanation/Reference:
Explanation:
The flex parameter allows the complexity to automatically adjust to either medium or high complexity depending on the needs of a call. For example, if a call uses the G.711 codec, the C5510 chipset automatically adjusts to the medium-complexity mode. However, if the call uses G.729, the C5510 chipset uses the high complexity mode.

Question 246
Which ds0-group option should you select to send automated number identification information on outbound calls for this digital T1 voice circuit? Refer to the exhibit.
Which ds0-group option should you select to send automated number identification information on outbound calls for this digital T1 voice circuit?

A. e&m-fgd
B. e&m-fgd
C. fgd-eana
D. e&m-delay-dial
E. fgd-os

Correct Answer: C

Explanation/Reference:

E&M signaling is often the preferred option for CAS because it avoids glare, it provides answer/disconnect supervision and it can receive Automatic Number Identification (ANI) with FGD and send ANI with FGD-EANA. In other words, you can have 1 channel-group for incoming calls and 1 channel-group for outgoing calls.

Question 247
Which two statements about Cisco Unified Communications Manager mixed-mode clusters are true?

A. Cluster security mode configures the security capability for your standalone server or a cluster.
B. The device security mode in the phone configuration file is set to nonsecure.
C. The phone makes nonsecure connections with Cisco Unified Communications Manager even if the device security mode specifies authenticated or encrypted.
D. Security-related settings other than device security mode, such as the SRST Allowed check box, get ignored.
E. Auto-registration does not work when you configure mixed mode.

Correct Answer: AE

Explanation/Reference:

Cluster security mode configures the security capability for a standalone server or a cluster.


Question 248
Which option about the primary directory URI for IP phone A is true?

When IP phone A was provisioned in a Cisco Unified Communications Manager, 2001 was configured as the directory number for its first line. Also, [email protected] was defined as the only directory URI on the Directory Number configuration page for this line. A few days later, an end user was created in the same Cisco Unified Communications Manager and was associated with the same phone with the primary extension set to 2001. Also, [email protected] was defined as a directory URI for that end user.

Which option about the primary directory URI for IP phone A is true?

A. [email protected]
B. [email protected]
C. It depends on which radio button was selected next to the Directory URI entries on the Directory Configuration page.
D. Both are primary directory URIs in a manner like a shared line for DNs.
E. Neither are primary directory URIs for IP phone A.

Correct Answer: B

Explanation/Reference:


Question 249
Which description of what happens when the user of IP phone B presses the Transfer soft key is true?

Refer to the exhibit.
All displayed devices are registered to the same Cisco Unified Communications Manager server and the phones are engaged in an active call. Assume that the provided configurations exist at the phone line level and multicast MOH is disabled cluster wide.

Which description of what happens when the user of IP phone B presses the Transfer soft key is true?

A. IP phone A user hears audio source 3 from MOH server A.
B. IP phone A user hears audio source 4 from MOH server B.
C. IP phone A user hears audio source 3 from MOH server B.
D. IP phone A user hears on-hold beep tones.
E. IP phone A user hears no on-hold music or beep tones.

Correct Answer: E

Explanation/Reference:

Question 250

Which two Cisco Unified CM Administration pages can a system administrator define MTU for an SSL VPN tunnel connecting between a Cisco IP phone and a Cisco IOS VPN gateway?

Refer to the exhibit.

On which two Cisco Unified CM Administration pages can a system administrator define MTU for an SSL VPN tunnel connecting between a Cisco IP phone and a Cisco IOS VPN gateway? (Choose two.)

A. VPN Profile
B. VPN Group
C. VPN Gateway
D. VPN Feature Configuration
E. System, followed by Enterprise Parameters
F. System, followed by Enterprise Phone Configuration

Correct Answer: AD

Explanation/Reference:


Question 251

Which three requirements must be met to share Enhanced Location Based Call Admission Control bandwidth usage between clusters?

Which three requirements must be met to share Enhanced Location Based Call Admission Control bandwidth usage between clusters? (Choose three.)

A. The Cisco Unified Communications Manager version must be 8.6 or higher.
B. The location name must be the same on both clusters.
C. SIP ICT must use the Shadow location.
D. The Location Bandwidth Manager Service should be started on only two servers in each cluster.
E. A Location Bandwidth Manager Hub Group must be created for each cluster.
F. Links must be created to the Shadow location.

Correct Answer: BCE

Explanation/Reference:

Explanation:
Recommendations and Design Considerations for Unified CM Session Management Edition Deployments

- All leaf clusters that support E-L CAC should be enabled for intercluster E-L CAC with SME.
- SME can be used as a centralized bootstrap hub for the E-L CAC intercluster hub replication network. See LBM Hub Replication Network, for more information.
- All trunks to leaf clusters supporting E-L CAC should be SIP trunks placed in the shadow location to enable E-L CAC on the trunk between SME and the leaf clusters supporting E-L CAC.

For TelePresence video interoperability, see the section on Call Admission Control Design Recommendations for TelePresence Video Interoperability Architectures.

- Connectivity from SME to any trunk or device other than a Unified CM that supports E-L CAC (some examples are third-party PBXs, gateways. Unified CM clusters prior to release 9.0 that do not support E-L CAC, voice messaging ports or trunks to conference bridges, Cisco Video Communications Server, and so forth) should be configured in a location other than a phantom or shadow location. The reason for this is that both phantom and shadow locations are non-terminating locations; that is, they relay information about locations and are effectively placeholders for user-defined locations on other clusters. Phantom locations are legacy locations that allow for the transmission of location information in versions of Unified CM prior to 9.0, but they are not supported with Unified CM 9.x Enhanced Locations CAC. Shadow locations are special locations that enable trunks between Unified CM clusters that support E-L CAC to accomplish it end-to-end.

- SME can be used as a locations and link management cluster

Question 252
Which method does a Cisco Unified 9971 phone use to send keep-alive messages to Cisco Unified Communications Manager?

Which method does a Cisco Unified 9971 phone use to send keep-alive messages to Cisco Unified Communications Manager?
A. SIP NOTIFY with Event set to keep-alive
B. SIP OPTIONS
C. SIP REGISTER with Expires set to zero
D. SCCP StationRegister
E. SCCP StationServerReq

Correct Answer: C

Explanation/Reference:

Phone registers with primary and establishes keepalive connection with secondary.
Expires = 0 keepalive mechanism allows Cisco SIP Phones to more closely resemble the failover / fallback behavior of SCCP.

Question 253
Which statement about using the Answer File Generator to load a Cisco Unified Communications virtual machine is true?

Which statement about using the Answer File Generator to load a Cisco Unified Communications virtual machine is true?
A. You must copy the output text to a file named platformConfig.txt.
B. Each host should be copied to its own configuration file.
C. The answer file can be used only when performing the new identity process to load the Cisco Unified Communications virtual machine.
D. The configuration file should be placed inside an ISO file and mounted on the virtual machine.

Correct Answer: A

Explanation/Reference:


SIP trunks can be configured with up to 16 destination IP addresses, 16 fully qualified domain names, or a single DNS SRV entry.

Question 254
Which design restriction applies to Cisco Unified Communications Manager Session Management Edition clustering over the WAN deployment with extended round-trip times in Cisco Unified CM 9.1 and later releases?

Which design restriction applies to Cisco Unified Communications Manager Session Management Edition clustering over the WAN deployment with extended round-trip times in Cisco Unified CM 9.1 and later releases?
A. SIP and H.323 intercluster trunks are supported.
B. Only SIP trunk is supported.
C. SIP trunks and H.323 gateways are supported.
D. A minimum of 1.544 Mb/s bandwidth is required for all traffic between any two nodes in the cluster.
E. Only RSVP agents can be configured and registered to the SME cluster as media resources.

Correct Answer: B

Explanation/Reference:

Using only SIP trunks in the SME cluster allows you to deploy a “media transparent” cluster where media resources, when required, are inserted by the end or leaf Unified Communications system and never by SME. Using only SIP trunks also allows you to use extended round trip times (RTTs) between SME nodes when clustering over the WAN.

Question 255
How many destinations can be configured for a SIP trunk on a Cisco Unified Communications Manager 9.1 system when the destination address is an SRV?

How many destinations can be configured for a SIP trunk on a Cisco Unified Communications Manager 9.1 system when the destination address is an SRV?
A. 1
B. 2
C. 3
D. 8
E. 16

Correct Answer: A

Explanation/Reference:

SIP trunks can be configured with up to 16 destination IP addresses, 16 fully qualified domain names, or a single DNS SRV entry.

Question 256
Which two call processing features have a lower priority than the Do Not Disturb settings on a Cisco IP phone?
Which two call processing features have a lower priority than the Do Not Disturb settings on a Cisco IP phone? (Choose two.)
A. park reversion for a locally parked call
B. hold reversion
C. intercom
D. pickup notification
E. terminating side of a call back
F. originating side of a call back

Correct Answer: DE
Explanation/Reference:
Explanation:
For the DND Ringer Off option, only visual notification gets presented to the device.
For the DND Call Reject option, no notification gets presented to the device.
For the terminating side of the call, Do Not Disturb overrides call back:
When the phone that terminates the call uses DND Ringer Off, the Callback Available screen will be displayed on the phone after the terminating side goes off hook and on hook.
When the phone that terminates the call has DND Call Reject enabled but the phone becomes available (goes off hook and on hook), a new screen will be presented to the originating device as “ has become available but is on DND-R”. Callback available notification will be sent only after the terminating side disables DND Call Reject.

Question 257
Which two applications must be connected to a leaf cluster in a Cisco Unified Communications Manager Session Management Edition deployment?

Which two applications must be connected to a leaf cluster in a Cisco Unified Communications Manager Session Management Edition deployment? (Choose two.)
A. Cisco Unified Meeting Place
B. Cisco Unified Contact Center Express
C. H.323-based video conferencing systems
D. Cisco Unity
E. Cisco Unified Communications Manager
F. fax servers

Correct Answer: BE
Explanation/Reference:
Explanation:
The deployment of a Unified CM Session Management Edition enables commonly used applications, such as conferencing or videoconferencing to connect directly to the session management cluster, which reduces the overhead of managing multiple trunks to leaf systems.
Unified CM Session Management Edition supports the following applications:
- Unity, Unity Connection
- Meeting Place, Meeting Place Express
- SIP and H.323-based video conferencing systems
- Third Party voice mail systems
- Fax servers
- Cisco Unified Mobility
The following applications must connect to the leaf cluster:
- Unified Contact Centre, CUCM, Unified Contact Centre Express
- Cisco Unified Presence Server
- Attendant Console
- Manager Assistant
- IP IVR
- Cisco Voice Portal

Question 258
Which two applications can connect directly with a Cisco Unified Communications Manager Session Management Edition cluster?

Which two applications can connect directly with a Cisco Unified Communications Manager Session Management Edition cluster? (Choose two.)
A. Cisco Unity
B. Cisco Unified Meeting Place Express
C. Cisco Unified Contact Center Enterprise
D. Cisco Unified Contact Center Express
E. Cisco Unified Communications Manager Attendant Console
F. Cisco Emergency Responder

Correct Answer: AB
Explanation/Reference:
Explanation:
The deployment of a Unified CM Session Management Edition enables commonly used applications such as conferencing or videoconferencing to connect directly to the Session Management cluster, thus reducing the overhead of managing multiple trunks to leaf systems. Cisco Unity or other voicemail systems can be deployed at all sites and integrated into the Unified CM cluster.

Question 259
How many music on hold servers are required in a trunk-only megacluster of Cisco Unified Communications Manager Session Management Edition?
How many music on hold servers are required in a trunk-only megacluster of Cisco Unified Communications Manager Session Management Edition?
A. 0
B. 1
C. 2
D. 3
E. 4

Correct Answer: A
Explanation/Reference:
Explanation:
When considering a megachuster deployment, the primary areas impacting capacity are as follows:
The megachuster may contain a total of 21 servers consisting of 16 subscribers, 2 TFTP servers, 2 music on hold (MoH) servers (0 required), and 1 publisher.
Server type must be either Cisco MCS 7845-I3/H3 class or Cisco Unified Computing System (UCS) C-Series or B-Series using the 10K Open Virtualization Archive (OVA) template.
Redundancy model must be 1:1.

Question 260
Which response from the SIP remote peer causes the trunk to be marked as “Out of Service”?

http://www.aoowe.com/practice-400-051-3156.html
On a Cisco Unified Communications Manager SIP trunk with a single remote device and OPTIONS ping feature enabled, which response from the SIP remote peer causes the trunk to be marked as "Out of Service"?

A. 401 Unauthorized
B. 505 Version Not Supported
C. 406 Not Acceptable
D. 408 Request Timeout
E. 500 Server Internal Error

Correct Answer: D

Explanation/Reference:
Explaination: 408 Request Timeout

Question 261
When the Cisco Unified Communications Manager service parameter “Auto Call Pickup Enabled” is selected, which two softkeys on an IP phone connect you to an incoming call?

When the Cisco Unified Communications Manager service parameter “Auto Call Pickup Enabled” is selected, which two softkeys on an IP phone connect you to an incoming call? (Choose two.)

A. Pickup
B. Gpickup
C. CallBack
D. Select
E. Join

Correct Answer: AB

Explanation/Reference:
Explaination: Pickup softkey is used to receive a call that is ringing in another phone within the same pickup group and Gpickupsoftkey is used to receive calls that are ringing but that phone is another pickup group.

Question 262
Which two statements about BFCP with Cisco Unified Communications Manager are true?

Which two statements about BFCP with Cisco Unified Communications Manager are true? (Choose two.)

A. BFCP is supported only on full SIP networks.
B. Cisco Unified Communications Manager allows BFCP only over UDP.
C. BFCP is not supported for third-party endpoints.
D. BFCP is not supported through Cisco Unified Border Element.
E. BFCP is supported between Cisco Unified Communications Manager and a TelePresence MCU.

Correct Answer: AB

Explanation/Reference:
Explaination: BFCP configuration tips

To enable BFCP in Cisco Unified Communications Manager, check the Allow Presentation Sharing using BFCP check box in the SIP Profile Configuration window. If the check box is unchecked, all BFCP offers will be rejected. By default, the check box is unchecked.

BFCP is supported only on full SIP networks. For presentation sharing to work, BFCP must be enabled for all SIP endpoints as well as all SIP lines and SIP trunks between the endpoints.

CUCM uses BFCP over user datagram protocol (UDP) in both secure and non-secure BFCP modes.


Question 263
What is the maximum number of option 66 IP addresses that a Cisco IP SCCP phone will accept and use from a DHCP server?

What is the maximum number of option 66 IP addresses that a Cisco IP SCCP phone will accept and use from a DHCP server?

A. 1
B. 2
C. 3
D. 4
E. 5

Correct Answer: A

Explanation/Reference:
Reference: http://www.electronicssolution.com/blog/?p=1201

Question 264
What is the maximum number of option 150 IP addresses that a Cisco IP SCCP phone will accept and use from a DHCP server?

What is the maximum number of option 150 IP addresses that a Cisco IP SCCP phone will accept and use from a DHCP server?

A. 1
B. 2
C. 3
D. 4
E. 5

Correct Answer: B

Explanation/Reference:
Cisco Unified IP Phones use the option 150 value as the TFTP server IP address when Alternate TFTP option is set to No. You can assign only IP addresses as Option 150 values. A maximum of two IP addresses are used, and only the first two IP addresses that the DHCP server provides get accepted.

Question 265
Which description of what will happen when the user of IP phone A presses the Hold soft key is true? Refer to the exhibit.

All displayed devices are registered to the same Cisco Unified Communications Manager server and the phones are engaged in an active call. Assume that the provided configurations exist at the phone line level and multicast MOH is disabled cluster wide.

Which description of what will happen when the user of IP phone A presses the Hold soft key is true?
A. IP phone B receives audio source 2 from MOH server A.
B. IP phone B receives audio source 3 from MOH server A.
C. IP phone B receives audio source 2 from MOH server B.
D. IP phone B receives audio source 3 from MOH server B.
E. IP phone B receives audio source 1 from MOH server A.

Correct Answer: C
Explanation/Reference:
Explanation:
Because audio source 2 is in top of the MRGL List and it will be selected locally first.

Question 266
Which Cisco Unified Communications Manager deployment model for clustering over the IP WAN mandates a primary and a backup subscriber at the same site?

Which Cisco Unified Communications Manager deployment model for clustering over the IP WAN mandates a primary and a backup subscriber at the same site?
A. multisite with centralized call processing
B. multisite with distributed call processing
C. local failover
D. remote failover
E. remote failover with Cisco Unified Communications Manager Express as SRST

Correct Answer: C
Explanation/Reference:
Explanation:
Clustering Over the IP WAN
You may deploy a single Unified CM cluster across multiple sites that are connected by an IP WAN with QoS features enabled. This section provides a brief overview of clustering over the WAN. For further information, refer to the chapter on Call Processing.

Clustering over the WAN can support two types of deployments:
1. Local Failover Deployment Model
   Local failover requires that you place the Unified CM subscriber and backup servers at the same site, with no WAN between them. This type of deployment is ideal for two to four sites with Unified CM.
2. Remote Failover Deployment Model
   Remote failover allows a site to deploy primary and backup call processing servers split across the WAN. Using this type of deployment, you may have up to eight sites with Unified CM subscribers being backed up by Unified CM subscribers at another site.


Question 267
What is the maximum one-way delay, in milliseconds, between any two Cisco Unified Communications Manager servers in a non-Session Management Edition cluster over an IP WAN?

What is the maximum one-way delay, in milliseconds, between any two Cisco Unified Communications Manager servers in a non-Session Management Edition cluster over an IP WAN?
A. 20
B. 40
C. 80
D. 160
E. 250

Correct Answer: B
Explanation/Reference:
Explanation:
The maximum one-way delay between any two Unified CM servers should not exceed 40 msec, or 80 msec round-trip time. Propagation delay between two sites introduces 6 microseconds per kilometer without any other network delays being considered. This equates to a theoretical maximum distance of approximately 3000 km for 20 ms delay or approximately 1860 miles. These distances are provided only as relative guidelines and in reality will be shorter due to other delay incurred within the network.

Question 268
Which SIP header is used by Cisco Unified Communication Manager to support the Redirected Number ID Service?
Which SIP header is used by Cisco Unified Communication Manager to support the Redirected Number ID Service?
A. replaces
B. RPID
C. diversion
D. join
E. P-charging-vector

Correct Answer: C
Explanation/Reference:
CUCM uses sip diversion header in INVITE message to carryout Redirected Number ID service.
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BR_CD2F83FA_00_cucm-system-guide-90/CUCM_BR_CD2F83FA_00_system-guide_chapter_0101000.html#CUCM_TP_R3F173A9_00

Question 269
Which tag in the SIP header is used by Cisco Unified Communications Manager to deliver a blended identity of alpha URI and number?
Which tag in the SIP header is used by Cisco Unified Communications Manager to deliver a blended identity of alpha URI and number?
A. x-cisco-callinfo
B. x-cisco-service-control
C. x-cisco-serviceuri
D. x-cisco-number
E. x-cisco-uri

Correct Answer: D
Explanation/Reference:
Cisco Unified Communications Manager supports blended addressing of directory URIs and directory numbers. When blended addressing is enabled across the network, Cisco Unified Communications Manager inserts both the directory URI and the directory number of the sending party in outgoing SIP Invites, or responses to SIP Invites. The destination endpoint has the option of using either the directory URI or the directory number for its response—both will reach the same destination.
Cisco Unified Communications Manager uses the x-cisco-number tag in the SIP identity headers to communicate a blended address. When both a directory URI and directory number are available for the sending phone and blended addressing is enabled, Cisco Unified Communications Manager uses the directory URI in the From fields of the SIP message and adds the x-cisco-number tag with the accompanying directory number to the SIP Identity headers. The x-cisco-number tag identifies the directory number that is associated with the directory URI.
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmfeat/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100/CUCM_BK_F3AC1C0F_00_cucm-features-services-guide-100_chapter_0110011.html#CUCM_RF_D0008C2B_00

Question 270
What is the trace file name syntax in which detailed SIP messages are logged?
What is the trace file name syntax in which detailed SIP messages are logged?
A. SDL
B. SDI
C. CCM
D. Call logs
E. Traces

Correct Answer: A
Explanation/Reference:
SDL files log SIP messages from CCM.

Question 271
Which SIP request is used by Cisco Unified Communications Manager to signal DND status changes to a Cisco 9971 IP Phone?
Which SIP request is used by Cisco Unified Communications Manager to signal DND status changes to a Cisco 9971 IP Phone?
A. OPTIONS
B. NOTIFY
C. INFO
D. REFER
E. UPDATE

Correct Answer: D
Explanation/Reference:
Cisco Unified Communications Manager supports Do Not Disturb that a SIP device initiates or that a Cisco Unified Communications Manager device initiates. A DND status change gets signaled from a SIP device to Cisco Unified Communications Manager by using the SIP PUBLISH method (RFC3999). A DND status change gets signaled from a Cisco Unified Communications Manager to a SIP device by using a dndupdate Remote-cc REFER request. Cisco Unified Communications Manager can also publish the Do Not Disturb status for a device, along with the busy and idle status for the device.

Question 272
Which SIP request is used by a Cisco 9971 IP Phone to signal DND status changes to Cisco Unified Communications Manager?
Which SIP request is used by a Cisco 9971 IP Phone to signal DND status changes to Cisco Unified Communications Manager?
A. REGISTER
B. NOTIFY
C. INFO
D. PUBLISH
E. UPDATE
Correct Answer: D
Explanation/Reference:
Explaination:
Cisco Unified Communications Manager supports Do Not Disturb that a SIP device initiates or that a Cisco Unified Communications Manager device initiates. A DND status change gets signaled from a SIP device to Cisco Unified Communications Manager by using the SIP PUBLISH method (RFC3909). A DND status change gets signaled from a Cisco Unified Communications Manager to a SIP device by using a msgupdate Remote-cm REFER request. Cisco Unified Communications Manager can also publish the Do Not Disturb status for a device, along with the busy and idle status for the device.

Question 273
Which statement about what happens to a hunt member who does not answer queuing-enabled hunt-list call in Cisco Unified Communications Manager 9.1 is true?
Which statement about what happens to a hunt member who does not answer queuing-enabled hunt-list call in Cisco Unified Communications Manager 9.1 is true?
A. The hunt member is logged off automatically and must press HLOG to log back in.
B. The hunt member remains logged in if Automatically Logout Hunt Member on No Answer is not selected in Cisco Unified Communications Manager Service Parameters.
C. The hunt member is logged off automatically and must manually reset the phone to log back in.
D. The hunt member is logged off if Automatically Logout Hunt Member on No Answer is selected on the Line Group configuration page.
E. The hunt member remains logged in if Automatically Logout Hunt Member on No Answer is not selected in Hunt Pilot configuration page.

Correct Answer: D
Explanation/Reference:
Explaination:
If a line member does not answer a queue-enabled call, that line member is logged off the hunt group only if the setting “Automatically Logout Hunt Member on No Answer” is selected on the line group page.
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmfg/cUCM_BK_A34970C5_00_admin-guide-91/CUCM_BK_A34970C5_00_admin-guide-91_chapter_0100011.html

Question 274
Which statement about what happens to incoming calls to a Cisco Unified Communications Manager native call queue when no hunt members are logged in or registered is true?
Which statement about what happens to incoming calls to a Cisco Unified Communications Manager native call queue when no hunt members are logged in or registered is true?
A. Calls are handled according to the Forward Hunt No Answer settings on the Hunt Pilot configuration page.
B. Calls are handled according to the Not Available Hunt Option settings on the Line Group Configuration page.
C. Calls are handled according to the Forward Hunt Busy settings on the Hunt Pilot configuration page.
D. Calls are handled according to the Forward Busy Calls To destination if configured; otherwise the calls are disconnected.
E. Calls are handled according to the correspondent parameters under the Queuing section on the Hunt Pilot Configuration page.

Correct Answer: E
Explanation/Reference:
Explaination:
There are three main scenarios where alternate numbers are used:
- When queue is full
- When maximum wait time is met
- When no hunt members are logged in or registered
When queue is full
Call Queuing allows up to 100 callers to be queued per hunt pilot (the maximum number of callers allowed in queue on a hunt pilot page). Once this limit for new callers has been reached on a particular hunt pilot, subsequent calls can be routed to an alternate number. This alternate number can be configured through the Hunt Pilot configuration page (through the “When queue is full” settings).
When maximum wait time is met
Each caller can be queued for up to 3600 seconds per hunt pilot (the maximum wait time in queue). Once this limit is reached, that caller is routed to an alternate number. This alternate number can be configured through the Hunt Pilot configuration page (through the “Maximum wait time in queue” settings).
When no hunt members are logged in or registered
In a scenario where none of the members of the hunt pilot are available or registered at the time of the call, hunt pilot configuration provides an alternate number field (through the “When no hunt members are logged in or registered” setting) where calls can be routed. For Call Queuing, a hunt pilot member is considered available if that member has both deactivated do not disturb (DND) and logged into the hunt group. In all other cases, the line member is considered unavailable or logged off.
Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmfeat/CUCM_BK_CEF0C471_00_cucm-features-services-guide-90/CUCM_BK_CEF0C471_00_cucm-features-and-services-guide_chapter_0111.html

Question 275
Which statement about what happens to this queued call is true?
A queued call has reached the maximum wait time configured for a Cisco Unified Communications Manager native call queue.
Which statement about what happens to this queued call is true?
A. Calls are handled according to the Forward Hunt No Answer settings on the Hunt Pilot configuration page.
B. Calls are handled according to the When Maximum Wait Time Is Met settings on the Hunt Pilot Configuration page.
C. Calls are handled according to the When Maximum Wait Time Is Met settings in Cisco Unified Communications Manager Service Parameters.
D. Calls are handled according to the Not Available Hunt Option settings on the Line Group Configuration page.
E. Calls are handled according to the When Queue Is Full settings on the Hunt Pilot Configuration page.

Correct Answer: B
Explanation/Reference:
Explaination:
There are three main scenarios where alternate numbers are used:
- When queue is full
- When maximum wait time is met
- When no hunt members are logged in or registered
When queue is full
Call Queuing allows up to 100 callers to be queued per hunt pilot (the maximum number of callers allowed in queue on a hunt pilot page). Once this limit for new callers has been reached on a particular hunt pilot, subsequent calls can be routed to an alternate number. This alternate number can be configured through the Hunt Pilot configuration page (through the “Destination When Queue is Full” settings).
When maximum wait time is met
Each caller can be queued for up to 3600 seconds per hunt pilot (the maximum wait time in queue). Once this limit is reached, that caller is routed to an alternate number. This alternate number can be configured through the Hunt Pilot configuration page (through the “Maximum wait time in queue” settings).
When no hunt members are logged in or registered

http://www.aoowe.com/practice-400-051-3156.html
In a scenario where none of the members of the hunt pilot are available or registered at the time of the call, hunt pilot configuration provides an alternate number field (through the "When no hunt members are logged in or registered" setting) where calls can be routed. For Call Queuing, a hunt pilot member is considered available if that member has both deactivated do not disturb (DND) and logged into the hunt group. In all other cases, the line member is considered unavailable or logged off.


**Question 276**

Which statement about what happens to additional incoming calls is true?

The number of calls waiting in a Cisco Unified Communications Manager native call queue has reached its maximum limit.

**Correct Answer:** E

**Explanation/Reference:**

There are three main scenarios where alternate numbers are used:

A. Calls are handled according to the Forward Hunt Busy settings on the Hunt Pilot configuration page.
B. Calls are handled according to the Forward Hunt No Answer settings on the Hunt Pilot configuration page.
C. Calls are handled according to the Hunt Options settings on the Line Group configuration page.
D. Calls are handled according to the When Queue Is Full settings on the Hunt Pilot configuration page.


**Question 277**

Which configuration parameter defines whether or not the user portion of a directory URI is case sensitive on Cisco Unified Communications Manager 9.1 or later?

A. URI Dialing Display Preference in Cisco CallManager Service Parameter
B. URI Lookup Policy in Cisco CallManager Service Parameter
C. URI Dialing Display Preference in Enterprise Parameters
D. URI Lookup Policy in Enterprise Parameters
E. The user portion of a directory URI is always case sensitive and cannot be changed.

**Correct Answer:** D

**Explanation/Reference:**

Cisco Unified Communications Manager supports the following formats in the user portion of a directory URI (the portion before the @ symbol):

- Accepted characters are a-z, A-Z, 0-9, !, $, %, &, *.
- The user portion has a maximum length of 47 characters.
- The user portion accepts percent encoding from %2[0-9A-F] through %7[0-9A-F]. For some accepted characters, Unified CM automatically applies percent encoding. See below for more information on percent encoding.
- The user portion is case-sensitive or case-insensitive depending on the value of the URI Lookup Policy enterprise parameter. The default value is case-sensitive.


**Question 278**

Which two user portion format conditions are true for directory URI on Cisco Unified Communications Manager 9.1 or later?

Which two user portion format conditions are true for directory URI on Cisco Unified Communications Manager 9.1 or later? (Choose two.)

A. It supports the $ character.
B. It supports space between characters.
C. It has a maximum length of 50 characters.
D. It has a maximum length of 254 characters.
E. It is always case sensitive.
F. It cannot be a directory number.

**Correct Answer:** AB

**Explanation/Reference:**

Cisco Unified Communications Manager supports the following formats in the user portion of a directory URI (the portion before the @ symbol):

- Accepted characters are a-z, A-Z, 0-9, !, $, %, &.
- The user portion has a maximum length of 47 characters.
- The user portion accepts percent encoding from %2[0-9A-F] through %7[0-9A-F]. For some accepted characters, Unified CM automatically applies percent encoding. See below for more information on percent encoding.
- The user portion is case-sensitive or case-insensitive depending on the value of the URI Lookup Policy enterprise parameter. The default value is case-sensitive.


**Question 279**

Which two host portion format conditions are true for directory URI on Cisco Unified Communications Manager?

Which two host portion format conditions are true for directory URI on Cisco Unified Communications Manager? (Choose two.)

A. It cannot be a directory number.
B. It has a maximum length of 254 characters.
C. It has a maximum length of 50 characters.
D. It supports space between characters.
E. It supports the $ character.
F. It cannot be a directory number.

**Correct Answer:** 

**Explanation/Reference:**

Cisco Unified Communications Manager supports the following formats in the host portion of a directory URI (the portion after the @ symbol):

- The host portion has a maximum length of 254 characters.
- The host portion accepts percent encoding from %2[0-9A-F] through %7[0-9A-F]. For some accepted characters, Unified CM automatically applies percent encoding. See below for more information on percent encoding.
- The host portion is always case sensitive.

A. It is case sensitive.
B. It cannot start with a hyphen.
C. It must have at least one character.
D. It supports IPv4 or IPv6 addresses, or fully qualified domain names.
E. It cannot end with a hyphen.
F. It supports the & character.

Correct Answer: BE

Explanation/Reference:
Cisco Unified Communications Manager supports the following formats in the host portion of a directory URI (the portion after the @ symbol):
- Supports IPv4 addresses or fully qualified domain names.
- Accepted characters are a-z, A-Z, 0-9, hyphens, and dots.
- The host portion cannot start or end with a hyphen.
- The host portion cannot have two dots in a row.
- Minimum of two characters.
- The host portion is not case sensitive.


Question 280
Which option is a characteristic of the Enhanced Location Call Admission Control mechanism on Cisco Unified Communications Manager?
A. It accounts for network protocol rerouting.
B. It accounts for network downtime and failures.
C. It supports dynamic bandwidth adjustments based on WAN topology changes.
D. It supports asymmetric media flows such that different bit rates in each direction are deducted accordingly.
E. Unidirectional media flows are deducted as if they were bidirectional.

Correct Answer: E

Explanation/Reference:
Network Modeling with Locations, Links, and Weights
Enhanced Location CAC is a model-based static CAC mechanism. Enhanced Location CAC involves using the administration interface in Unified CM to configure Locations and Links to model the “Routed WAN Network” in an attempt to represent how the WAN network topology routes media between groups of endpoints for end-to-end audio, video, and immersive calls. Although Unified CM provides configuration and serviceability interfaces in order to model the network, it is still a “static” CAC mechanism that does not take into account network failures and network protocol rerouting. Therefore, the model needs to be updated when the WAN network topology changes. Enhanced Location CAC is also call oriented, and bandwidth deductions are per call not per stream, so asymmetric media flows where the bit-rate is higher in one direction than in the other will always deduct for the highest bit rate. In addition, unidirectional media flows will be deducted as if they were bidirectional media flows.


Question 281
Which statement about how digits are forwarded to the Cisco Unified Communications Manager for further call processing is true?
Refer to the exhibit.

A. As each digit is pressed on the SIP IP phone, it is sent to the Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event.
B. The SIP IP phone waits for the inter-digit timer expiry and then sends each digit to the Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.
C. As soon as the user selects the Dial soft key, the SIP IP phone forwards all digits to the Cisco Unified Communications Manager in a SIP INVITE message.
D. As soon as the Dial soft key is selected, the SIP IP phone forwards the first digit in a SIP INVITE and the subsequent digits in SIP INFORMATION messages.
E. The SIP IP phone waits for the inter-digit timer expiry, and then sends all digits to the Cisco Unified Communications Manager in a SIP INVITE message.

Correct Answer: C

Explanation/Reference:

Question 282
Which call routing practice is critical to prevent unnecessary toll charges caused by internal calls routed through the PSTN?
In a Cisco Unified Communications Manager design where +E.164 destinations are populated in directory entries, which call routing practice is critical to prevent unnecessary toll charges caused by internal calls routed through the PSTN?
A. forced on-net routing
B. automated alternate routing
C. forced authorization codes
D. client matter codes
E. tail-end hop-off

Correct Answer: A

Explanation/Reference:
Forced On-Net Routing
It is not uncommon for the dialing habits for on-net/inter-site and off-net destinations to use the same addressing structure. In this case the call control decides...
whether the addressed endpoint, user, or application is on-net or off-net based on the dialed address, and will treat the call as on-net or off-net, respectively.

Figure 14-4 shows an example of this forced on-net routing. All four calls in this example are dialed as 91 plus 10 digits. But while the calls to +1 408 555 2345 and +1 212 555 7000 are really routed as off-net calls through the PSTN gateway, the other two calls are routed as on-net calls because the call control identifies the ultimate destinations as on-net destinations. Forced on-net routing clearly shows that the dialing habit used does not necessarily also determine how a call is routed. In this example, some calls are routed as on-net calls even though the used PSTN dialing habit seems to indicate that an off-net destination is called.

Figure 14-4 Forced On-Net Routing

Forced on-net routing is especially important if dialing of +E.164 destinations from directories is implemented. In a normalized directory, all destinations are defined as +E.164 numbers, regardless of whether the person that the number is associated with is internal or external. In this case forced on-net routing is a mandatory requirement to avoid charges caused by internal calls routed through the PSTN.


Question 283
Which statement describes the Maximum Serving Count service parameter of the Cisco TFTP service on Cisco Unified Communications Manager?

A. It specifies the maximum number of files in the TFTP server disk storage.
B. It specifies the maximum number of TFTP client requests to accept and to serve files at a given time.
C. It specifies the maximum file support by the Cisco TFTP service.
D. It specifies the maximum file counts, in cache as well as in disk, that are supported by the Cisco TFTP service.
E. It specifies the maximum number of TFTP client requests to accept and to serve files in a 120-minute window.

Correct Answer: B

Explanation/Reference:

This parameter specifies the maximum number of client requests to accept and to serve files at a time. Specify a low value if you are serving files over a low bandwidth connection. You can set it to a higher number if you are serving small files over a larger bandwidth connection and when CPU resources are available, such as when no other services run on the TFTP server. Use the default value if the TFTP service is run along with other Cisco CallManager services on the same server. Use the following suggested values for a dedicated TFTP server: 1500 for a single-processor system and 3000 for a dual-processor system. If the dual-processor system is running Windows 2000 Advanced Server, the serving count can be up to 5000. This is a required field.

.Maximum: 5000.

Question 284
Which call processing feature overrides the Do Not Disturb settings on a Cisco IP phone?

A. park reversion for remotely parked calls by a shared line
B. hold reversion
C. remotely placed pickup request by a shared line
D. pickup notification
E. terminating side of a call back

Correct Answer: B

Explanation/Reference:

Hold Reversion and Intercom

Hold reversion and intercom override DND (both options), and the call gets presented normally.


Question 285
Which statement about how digits are forwarded to the Cisco Unified Communications Manager for further call processing is true?

Refer to the exhibit.

A user is going through a series of dialing steps on a SIP Type A IP phone to call a SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assume that the calling SIP phone is not associated with any SIP dial rules. Which statement about how digits are forwarded to the Cisco Unified Communications Manager for further call processing is true?

A. As each digit is pressed on the SIP IP phone, it is sent to the Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event.
B. The SIP IP phone waits for the inter-digit timer expiry and then sends each digit to the Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.
C. The SIP IP phone waits for the inter-digit timer expiry or for the Dial soft key to be selected before it sends all digits to the Cisco Unified Communications Manager in a SIP INVITE message.

http://www.aoowe.com/practice-400-051-3156.htm
D. The SIP IP phone waits for the inter-digit timer expiry, or for the Dial soft key to be selected before it sends the first digit in a SIP INVITE and the subsequent digits in SIP NOTIFY messages.
E. The SIP IP phone sends all digits to the Cisco Unified Communications Manager in a SIP INVITE message as soon as the fourth digit is pressed.

Correct Answer: C
Explanation/Reference:
Because Type A SIP phone with no SIP dial rules sends digit in Enbloc style. All digits are sent to CUCM after the user completes the dialing and press the Dial softkey.

Question 286
Which procedure eliminates the wait time?
You have implemented 5-digit forced authorization codes to all international route patterns on Cisco Unified Communications Manager. Your users report that after entering the FAC codes, they must wait for more than 10 seconds before the call is routed.
Which procedure eliminates the wait time?
A. Check and eliminate any existing route patterns that overlap with the international route pattern.
B. Go to the Cisco Unified Communications Manager Service Parameters and reduce the T-304 number to 5000 milliseconds.
C. Request your long distance telephone service provider to reduce the call setup time to 5 seconds.
D. Configure a # (hash) sign to the end of the forced authorization codes to signal the end of dialing.
E. Educate the users to press # (hash) after entering the forced authorization codes.

Correct Answer: E
Explanation/Reference:
Because it immediately stops taking digits and route the digits to CUCM, otherwise the call occurs after the interdigit timer expire which is 15 seconds by default.

Question 287
Which user inputs are sent from the calling IP phone to the Cisco Unified Communications Manager, in the form of SCCP messages, after the user takes the phone off-hook?
Refer to the exhibit.

A user is performing a series of dialing steps on a SCCP IP phone (extension 1001) to call another SCCP IP phone (extension 2003). Both phones are registered to the same Cisco Unified Communications Manager cluster.
Which user inputs are sent from the calling IP phone to the Cisco Unified Communications Manager, in the form of SCCP messages, after the user takes the phone off-hook?
(The commas in the options are logical separators, not part of the actual user input or SCCP messages.)
A. A separate SCCP message is sent to the Cisco Unified Communications Manager for each of these user inputs: 2, 0, 0, 3.
B. A separate SCCP message is sent to the Cisco Unified Communications Manager for each of these user inputs: 2, 0, 1, <<, 0, 3.
C. The IP phone collects all keypad and soft key events until user inputs stops, then it sends a single SCCP message to report that 2003 has been dialed.
D. The IP phone collects all keypad and soft key events until user inputs stops, then it sends a single SCCP message to report that 201<<03 has been dialed.
E. A separate SCCP message is sent to the Cisco Unified Communications Manager for each of these user inputs: 2, 0, 1, <<, 2, 0, 0, 3.

Correct Answer: B
Explanation/Reference:
Because sccp phones send digits DIGIT-by-DIGIT i.e. it sends each digit in real time.

Question 288
What is the maximum number of Cisco Unified Communications Manager subscriber pairs in a megacluster deployment?

A. 4
B. 8
C. 12
D. 16
E. 32

Correct Answer: B
Explanation/Reference:
Because there can be up to 8 pairs of subscribers, 16 subscribers total and must be in a 1:1 redundancy mode (8 active, 8 standby).

Question 289
Which message is used by a Cisco Unified Communications Manager respond to periodic keepalives from a Cisco IOS MGCP gateway?
Which message is used by a Cisco Unified Communications Manager respond to periodic keepalives from a Cisco IOS MGCP gateway?
A. AUEP
B. RQNT
C. NTFY
D. 200
E. AUCX
Correct Answer: D
Explanation/Reference:
Explanation:
(2xx) Successful completion: The requested transaction was executed normally.

Question 290
Which Call Control Discovery function allows the local Cisco Unified Communications Manager to listen for advertisements from remote call-control entities that use the SAF network?
A. CCD advertising service
B. CCD requesting service
C. SAF forwarder
D. SAF enabled trunks
E. CCD registration service

Correct Answer: B
Explanation/Reference:
Explanation:
SAF and CCD will allow large distributed multi-cluster deployments to have the directory number (DN) ranges of each call routing element advertised dynamically over SAF. Cisco routers act as SAF Forwarders (SAFF), while the call routing elements (e.g. CUCM) act as clients that register with the routers to advertise their DN ranges and listen to the advertisements of other routers.

Question 291
Which Cisco Unified Communications Manager partition will be associated with a directory URI that is configured for an end user with a primary extension?
A. null
B. none
C. directory URI
D. default
E. any partition that the Cisco Unified Communications Manager administrator desires

Correct Answer: C
Explanation/Reference:
Explanation:
Cisco Unified Communications Manager supports dialing using directory URIs for call addressing. Directory URIs look like email addresses and follow the [email protected] format where the host portion is an IPv4 address or a fully qualified domain name. A directory URI is a uniform resource identifier, a string of characters that can be used to identify a directory number. If that directory number is assigned to a phone, Cisco Unified Communications Manager can route calls to that phone using the directory URI. URI dialing is available for SIP and SCCP endpoints that support directory URIs.

Question 292
What is the number of directory URIs with which a Cisco Unified Communications Manager directory number can be associated?
A. 1
B. up to 2
C. up to 3
D. up to 4
E. up to 5

Correct Answer: E
Explanation/Reference:
Explanation:
Cisco Unified Communications Manager supports dialing using directory URIs for call addressing. Directory URIs look like email addresses and follow the [email protected] format where the host portion is an IPv4 address or a fully qualified domain name. A directory URI is a uniform resource identifier, a string of characters that can be used to identify a directory number. If that directory number is assigned to a phone, Cisco Unified Communications Manager can route calls to that phone using the directory URI. URI dialing is available for SIP and SCCP endpoints that support directory URIs.

Question 293
Which two Cisco Unified Communications Manager SIP profile configuration parameters for a SIP intercluster trunk are mandatory to enable end-to-end RSVP SIP Preconditions between clusters? (Choose two.)
A. Set the RSVP over SIP parameter to Local RSVP.
B. Set the RSVP over SIP parameter to E2E.
C. Set the SIP Rel1XX Options parameter to Disabled.
D. Set the SIP Rel1XX Options parameter to Send PRACK If 1xx Contains SDP.
E. Set the SIP Rel1XX Options parameter to Send PRACK For All 1xx Messages.
F. Check the Fall Back to Local RSVP check box.

Correct Answer: BD
Explanation/Reference:
Explanation:
Each Unified Communications Manager cluster and Unified CME should have the same configuration information. For example, Application ID should be the same on each Unified Communications Manager cluster and Unified CME. RSVP Service parameters should be the same on each Unified Communications Manager cluster.

Question 294
Which Call Admission Control mechanism is supported for the Cisco Extension Mobility Cross Cluster solution?
A. Location CAC
B. RSVP CAC
C. H.323 gatekeeper
D. intercluster Enhanced Location CAC

Correct Answer: A
Explanation/Reference:
Explanation:
Each Unified Communications Manager cluster and Unified CME should have the same configuration information. For example, Application ID should be the same on each Unified Communications Manager cluster and Unified CME. RSVP Service parameters should be the same on each Unified Communications Manager cluster.
E. visiting cluster’s LBM hub

Correct Answer: B
Explanation/Reference:
Explanation:
Configuring extension mobility cross cluster (EMCC) is nothing you should take lightly. EMCC requires a lot of configuration parameters including the exporting and importing of each neighbor cluster’s X.509v3 digital certificates. EMCC is supported over SIP trunks only. Presence is another feature that’s only supported over SIP trunks. If you want to be able to perform scalable Call Admission Control (CAC) in a distributed multi-cluster call processing model, you will need to point an H.225 or Gatekeeper controlled trunk to an H.323 Gatekeeper for CAC... but if you want to support presence and EMCC between clusters and maintain CAC.

Question 295
Which two Device Pool configuration settings will override the device-level settings when a device is roaming within or outside a device mobility group?

Which two Device Pool configuration settings will override the device-level settings when a device is roaming within or outside a device mobility group? (Choose two.)

A. Adjunct CSS
B. Device Mobility CSS
C. Network Locale
D. Called Party Transformation CSS
E. AAR CSS
F. Device Mobility Group

Correct Answer: CF
Explanation/Reference:
Explanation:
The parameters under these settings will override the device-level settings when the device is roaming within or outside a Device Mobility Group. The parameters included in these settings are:
- Date/time Group
- Region
- Media Resource Group List
- Location
- Network Locale
- SRST Reference
- Physical Location
- Device Mobility Group

The roaming sensitive settings primarily help in achieving proper call admission control and voice codec selection because the location and region configurations are used based on the device’s roaming device pool.

Question 296
Which Device Pool configuration setting will override the device-level settings only when a device is roaming within a device mobility group?

Which Device Pool configuration setting will override the device-level settings only when a device is roaming within a device mobility group?

A. Region
B. Location
C. SRST Reference
D. Calling Party Transformation CSS
E. Media Resource Group List

Correct Answer: D
Explanation/Reference:
Explanation:
Device Mobility Related Settings:
The parameters under these settings will override the device-level settings only when the device is roaming within a Device Mobility Group. The parameters included in these settings are:
Device Mobility Calling Search Space
AAR Calling Search Space
AAR Group
Calling Party Transformation CSS
Called Party Transformation CSS

The device mobility related settings affect the dial plan because the calling search space dictates the patterns that can be dialed or the devices that can be reached.

Question 297
Which Media Resource Group and Media Resource Group List configuration should be implemented if an administrator wants to make sure that all provisioned DSPs on router A are consumed before router B’s DSP is used?

Router A and router B are Cisco IOS routers with hardware CFB resources that are registered to the same Cisco Unified Communications Manager server. Which Media Resource Group and Media Resource Group List configuration should be implemented if an administrator wants to make sure that all provisioned DSPs on router A are consumed before router B’s DSP is used?

A. Router A’s CFB and router B’s CFB should each be configured in its own MRG. Both MRGs should then be grouped into the same MRGL, but the MRG that contains router A’s CFB should be listed in higher order than the MRG that contains router B’s CFB. Finally, associate the MRG with all conference resource consumers.
B. Router A’s CFB and router B’s CFB should each be configured in its own MRG. Both MRGs should then be further separated into different MRGLs. Finally, associate the MRG that contains router A’s CFB in higher order than router B’s CFB to all conference resource consumers.
C. Router A’s CFB and router B’s CFB should both be configured in the same MRG with router A’s CFB listed higher than that of router B. Then associate the MRG with an MRGL and apply it to all conference resource consumers.
D. Router A’s CFB and router B’s CFB should both be configured in the same MRG. Make sure router A’s CFB is listed in a higher alphabetical order than router B’s CFB. Then associate the MRG with an MRGL and apply it to all conference resource consumers.
E. Router A’s CFB and router B’s CFB should both be configured in the same MRG. Use Cisco Unified Communications Manager service parameters to assign a higher priority to router A’s CFB. Then associate the MRG with an MRGL and apply it to all conference resource consumers.

Correct Answer: A
Explanation/Reference:

Question 298
What will happen when the user of IP Phone A presses the Transfer softkey?

Refer to the exhibit.
All displayed devices are registered to the same Cisco Unified Communications Manager server and the phones are engaged in an active call. Assuming the provided configurations exist at the phone line level and multicast MOH is disabled clusterwide, what will happen when the user of IP Phone A presses the Transfer softkey?

A. The IP Phone B user hears audio source 3 from MOH Server A.
B. The IP Phone B user hears audio source 4 from MOH Server B.
C. The IP Phone B user hears audio source 3 from MOH Server B.
D. The IP Phone B user hears audio source 2 from MOH Server B.
E. The IP Phone A user hears no on-hold music.

Correct Answer: C

Explanation/Reference:
Held parties determine the media resource group list that a Cisco Unified Communications Manager uses to allocate a music on hold resource.

Question 299
What will happen when the user of IP Phone B presses the Hold softkey?
Refer to the exhibit.

A. IP Phone A receives audio source 2 from MOH Server A.
B. IP Phone A receives audio source 3 from MOH Server A.
C. IP Phone A receives audio source 2 from MOH Server B.
D. IP Phone A receives audio source 3 from MOH Server B.
E. IP Phone A receives audio source 1 from MOH Server A.

Correct Answer: B

Explanation/Reference:
Held parties determine the media resource group list that a Cisco Unified Communications Manager uses to allocate a music on hold resource.

Question 300
Which system location is used for intercluster Enhanced Location CAC on Cisco Unified Communications Manager?

A. Hub_None
B. Default
C. Intercluster
D. Phantom
E. Shadow
The shadow location is used to enable a SIP trunk to pass Enhanced Location CAC information such as location name and Video-Traffic-Class (discussed below), among other things, required for Enhanced Location CAC to function between clusters. In order to pass this location information across clusters, the SIP intercluster trunk (ICT) must be assigned to the “shadow” location. The shadow location cannot have a link to other locations, and therefore no bandwidth can be reserved between the shadow location and other locations. Any device other than a SIP ICT that is assigned to the shadow location will be treated as if it was associated to Hub_None. That is important to know because if a device other than a SIP ICT ends up in the shadow location, bandwidth deductions will be made from that device as if it were in Hub_None, and that could have varying effects depending on the location and links configuration.

**Question 301**
When neither the active or standby Location Bandwidth Manager in the configured LBM group is available, what will the Cisco CallManager service on a subscriber Cisco Unified Communications Manager server do to make location CAC decisions?
A. It will attempt to communicate with the first configured member in the Location Bandwidth Manager hub group.
B. It will use the Call Treatment When No LBM Available service parameter with the default action to allow calls.
C. It will use the Call Treatment When No LBM Available service parameter with the default action to reject calls.
D. It will attempt to communicate with the local LBM service for location CAC decisions.
E. It will allow all calls until communication is reestablished with any configured servers in the LBM group.

**Correct Answer:** D

**Explanation/Reference:**
By default the Cisco CallManager service communicates with the local LBM service; however, LBM groups can be used to manage this communication. LBM groups provide an active and standby LBM in order to create redundancy for Unified CM call control.

**Question 302**
What is the amount of audio bandwidth, in kilobits per second, that is used in the Cisco Unified Communications Manager location bandwidth calculation for a G.728 call?
What is the amount of audio bandwidth, in kilobits per second, that is used in the Cisco Unified Communications Manager location bandwidth calculation for a G.728 call?
A. 8
B. 16
C. 24
D. 29
E. 80

**Correct Answer:** B

**Explanation/Reference:**
G.728 — Low-bit-rate codec that video endpoints support. It supports kilobits per second.

**Question 303**
Which configuration component in Cisco Unified Communications Manager Enhanced Location Call Admission Control is designated to participate directly in intercluster replication of location, links, and bandwidth allocation data?
Which configuration component in Cisco Unified Communications Manager Enhanced Location Call Admission Control is designated to participate directly in intercluster replication of location, links, and bandwidth allocation data?
A. an active member of a Location Bandwidth Manager Group
B. a member of a Location Bandwidth Manager Hub Group
C. a standby member of a Location Bandwidth Manager Group
D. all members of a Location Bandwidth Manager Group
E. a shadow member of a Location Bandwidth Manager Hub Group

**Correct Answer:** B

**Explanation/Reference:**
A Location Bandwidth Manager (LBM) service that has been designated to participate directly in intercluster replication of fixed locations, links data, and dynamic bandwidth allocation data. LBMs assigned to an LBM hub group discover each other through their common connections and form a fully meshed intercluster replication network. Other LBM services in a cluster with an LBM hub participate indirectly in intercluster replication through the LBM hubs in their cluster.

**Question 304**
Which statement about the effective path in the Enhanced Location Call Admission Control mechanism on Cisco Unified Communications Manager is true?
Which statement about the effective path in the Enhanced Location Call Admission Control mechanism on Cisco Unified Communications Manager is true?
A. It is the sequence of links and intermediate locations that connect a pair of locations.
B. It is used to define the bandwidth that is available between locations.
C. Only one effective path is used between two locations.
D. There could be multiple effective paths between a pair of locations.
E. It logically represents the WAN link.

**Correct Answer:** C

**Explanation/Reference:**
The effective path is the path used by Unified CM for the bandwidth calculations, and it has the least cumulative weight of all possible paths. Weights are used on links to provide a “cost” for the “effective path” and are pertinent only when there is more than one path between any two locations.

**Question 305**
What does a weight represent in the Enhanced Location Call Admission Control mechanism on Cisco Unified Communications Manager?
What does a weight represent in the Enhanced Location Call Admission Control mechanism on Cisco Unified Communications Manager?
A. It defines the bandwidth that is available between locations.
B. It defines the bandwidth that is available on a link.
C. It is the amount of bandwidth allocation for different types of traffic.
D. It is used to provide the relative priority of a link between locations.

**Correct Answer:** B

**Explanation/Reference:**
The weight represents the bandwidth available on a link.
A weight provides the relative priority of a link in forming the effective path between any pair of locations. The effective path is the path used by Unified CM for the bandwidth calculations, and it has the least cumulative weight of all possible paths. Weights are used on links to provide a "cost" for the "effective path" and are pertinent only when there is more than one path between any two locations.

Question 306
What does a period accomplish when it is used in a SIP Dial Rule pattern that is associated with a Cisco 9971 IP Phone that is registered to Cisco Unified Communications Manager?

A. It matches any single digit from 0 to 9.
B. It inserts a 500-millisecond pause between digits.
C. It is a delimiter and has no significant dialing impact.
D. It indicates a timeout value of 5000 milliseconds.
E. It is an obsolete parameter and will be ignored.

Correct Answer: B
Explanation/Reference:
Explanation:
Comma is accepted in speed dial as delimiter and pause. -Comma used to delineate dial string, FAC, CMC, and post connect digits For post connect digits, commas get matched by the wildcard characters * and period (.).

Question 307
What does a comma accomplish when it is used in a SIP Dial Rule pattern that is associated with a Cisco 9971 IP Phone that is registered to Cisco Unified Communications Manager?

A. It matches any single digit from 0 to 9.
B. It inserts a 500-millisecond pause between digits.
C. It is a delimiter and has no significant dialing impact.
D. It matches any single digit from 0 to 9, or the asterisk (*) or pound (#) symbols.
E. It matches one or more characters. The * gets processed as a wildcard character. You can override this by preceding the * with a backward slash (/) escape sequence, which results in the sequence *. The phone automatically strips the , so it does not appear in the outgoing dial string. When # is received as a dial digit, it gets matched by the wildcard characters * and period (.).

Correct Answer: D
Explanation/Reference:
Explanation:
Asterisk (*) matches one or more characters. The * gets processed as a wildcard character. You can override this by preceding the * with a backward slash (/) escape sequence, which results in the sequence *. The phone automatically strips the , so it does not appear in the outgoing dial string. When # is received as a dial digit, it gets matched by the wildcard characters * and period (.).

Question 308
Which statement about how digits are forwarded to Cisco Unified Communications Manager for further call processing is true?

Refer to the exhibit.

A user is going through a series of dialing steps on a SIP Type B IP phone (for example, a Cisco 7975) to call an SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assuming the calling SIP phone is associated with a SIP Dial Rule with a pattern value of 2001, which statement about the call setup process of this call is true?

A. Each digit will arrive at Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event, and Cisco Unified Communications Manager will extend the call as soon as the collected digits match the extension of the SCCP IP phone, bypassing class of service configuration on both IP phones.
B. Each digit will arrive at Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event. When the collected digits match the extension of the SCCP IP phone, Cisco Unified Communications Manager will extend the call only if the class of service configuration on both phones permits this action.
C. As soon as the user selects the Dial softkey, the SIP IP phone will forward all digits to Cisco Unified Communications Manager in a SIP INVITE message. Cisco Unified Communications Manager will extend the call as soon as the collected digits match the extension of the SCCP IP phone, bypassing class of service configuration on both IP phones.
D. As soon as the user selects the Dial softkey, the SIP IP phone will forward all digits to Cisco Unified Communications Manager in a SIP INVITE message. Cisco Unified Communications Manager will extend the call only if class of service configuration on both phones permits this action.
E. The SIP IP phone will wait for the interdigit timer to expire, and then send all digits to Cisco Unified Communications Manager in a SIP INVITE message. Cisco Unified Communications Manager will extend the call as soon as the collected digits match the extension of the SCCP IP phone, bypassing class of service configuration on both IP phones.

Correct Answer: D
Explanation/Reference:
Explanation:
Cisco Type B SIP Phones offer functionality based SIP INVITE Message. Every key the end user presses triggers an individual SIP message. The first event is communicated with a SIP INVITE, but subsequent messages use SIP NOTIFY messages. The SIP NOTIFY messages send KPML events corresponding to any buttons or soft keys pressed by the user. Cisco Type B SIP IP Phones with SIP dial rules operate in the same manner as Cisco Type A phones with dial rules.

Question 309
Which statement about how digits are forwarded to Cisco Unified Communications Manager for further call processing is true?

Refer to the exhibit.
A user is going through a series of dialing steps on a SIP Type B IP phone (for example, a Cisco 7975) to call an SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assuming the calling SIP phone is associated with a SIP dial rule with a pattern value of 2001, which statement about how digits are forwarded to Cisco Unified Communications Manager for further call processing is true?

A. As each digit is pressed on the SIP IP phone, it is sent to Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event.
B. The SIP IP phone will wait for the interdigit timer to expire, and then send each digit to Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.
C. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending all digits to Cisco Unified Communications Manager in a SIP INVITE message.
D. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending the first digit in a SIP INVITE and the subsequent digits in SIP INFORMATION messages.
E. The SIP IP phone will wait for the interdigit timer to expire, and then send all digits to Cisco Unified Communications Manager in a SIP INVITE message.

Correct Answer: E

Explanation/Reference:
Cisco Type B SIP Phones offer functionality based SIP INVITE Message. Every key the end user presses triggers an individual SIP message. The first event is communicated with a SIP INVITE, but subsequent messages use SIP NOTIFY messages. The SIP NOTIFY messages send KPML events corresponding to any buttons or soft keys pressed by the user. Cisco Type B SIP IP Phones with SIP dial rules operate in the same manner as Cisco Type A phones without dial rules.

Question 310
Which statement about how digits are forwarded to Cisco Unified Communications Manager for further call processing is true?

Refer to the exhibit.

A user is going through a series of dialing steps on a SIP Type B IP phone (for example, a Cisco 7975) to call an SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assuming that the calling SIP phone is not associated with any SIP dial rules, which statement about how digits are forwarded to Cisco Unified Communications Manager for further call processing is true?

A. Each digit is sent to Cisco Unified Communications Manager in a SIP NOTIFY message KPML event, at the time that the user enters the digit on the keypad.
B. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending each digit to Cisco Unified Communications Manager as a separate KPML event in a SIP NOTIFY message.
C. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending all digits to Cisco Unified Communications Manager in a SIP INVITE message.
D. The SIP IP phone will wait for the interdigit timer to expire, or for the Dial softkey to be selected before sending the first digit in a SIP INVITE and the subsequent digits in SIP INFORMATION messages.
E. The SIP IP phone will send all digits to Cisco Unified Communications Manager in a SIP INVITE message as soon as the fourth digit is pressed.

Correct Answer: A

Explanation/Reference:
KPML procedures use a SIP SUBSCRIBE message to register for DTMF digits. The digits themselves are delivered in NOTIFY messages containing an XML encoded body. And it is Out of Band DTMF.

Question 311
Which two fields in the National Significant Number code may be further subdivided?

According to ITU-T E.164 recommendations, which two fields in the National Significant Number code may be further subdivided? (Choose two.)

A. Country Code
B. National Destination Code
C. Subscribers Number
D. Regional Significant Number
E. Local User Code
F. National Numbering Plan

Correct Answer: BC

Explanation/Reference:
A telephone number can have a maximum of 15 digits. The first part of the telephone number is the country code (one to three digits). The second part is the national destination code (NDC). The last part is the subscriber number (SN). The NDC and SN together are collectively called the national (significant) number.

Question 312
What is the maximum length of any numeric geographic area address in ITU recommendation E.164?

What is the maximum length of any numeric geographic area address in ITU recommendation E.164?

A. 15
B. 18
C. 21
D. 22
E. 25

Correct Answer: A
Explanation/Reference:
Explanation:
E. 164 defines a general format for international telephone numbers. Plan-conforming numbers are limited to a maximum of 15 digits. The presentation of numbers is usually prefixed with the character + (plus sign), indicating that the number includes the international country calling code (country code), and must typically be prefixed when dialing with the appropriate international call prefix, which is a trunk code to reach an international circuit from within the country of call origination.

Question 313
Which device is the initiator of a StationD message in a Cisco Unified Communications Manager SDI trace?
A. SCCP IP phone
B. SIP IP phone
C. Cisco Unified Communications Manager
D. MGCP analog gateway
E. digital voice gateway

Correct Answer: C
Explanation/Reference:
Explanation:
All messages to and from a skinny device are preceded by either the words StationId or StationInit. StationD messages are sent from call manager to IP phone. Skinny message transmission such as this between the IP phone and CallManager occurs for every action undertaken by the IP phone, including initialization, registration, on-hook, off-hook, dialing of the digits, key press on the phone, and so much more.

Question 314
What is the purpose of this message?
Refer to the exhibit.

You received this debug output to troubleshoot a Cisco IOS MGCP gateway media-related problem at a customer site. What is the purpose of this message?
A. The MGCP gateway is responding to an RQNT message from Cisco Unified Communications Manager to poll the media capabilities on its endpoints.
B. The MGCP gateway is responding to an AUEP message from Cisco Unified Communications Manager to poll the media capabilities on its endpoints.
C. The MGCP gateway is responding to an AUCX message from Cisco Unified Communications Manager to poll the active calls on its endpoints.
D. The MGCP gateway is responding to an MDCX message from Cisco Unified Communications Manager during a call setup.
E. The MGCP gateway is responding to a CRCX message from Cisco Unified Communications Manager during a call setup.

Correct Answer: E
Explanation/Reference:

Question 315
Which user agent has the recipient role in this SIP REFER call transfer?
Refer to the exhibit.

Which user agent has the recipient role in this SIP REFER call transfer?
A. user agent A
B. user agent B
C. user agent C
D. user agent B and C
E. user agent A and B

Correct Answer: B
Explanation/Reference:
The Refer method has three main roles:
Originator — User agent that initiates the transfer or Refer request.
Recipient — User agent that receives the Refer request and is transferred to the final-recipient.
Final-Recipient — User agent introduced into a call with the recipient.

Question 316
Which SIP message is being used by the recipient to notify the originator that the final recipient was successfully contacted?
In a SIP REFER-based call transfer, which SIP message is being used by the recipient to notify the originator that the final recipient was successfully contacted?
A. 200 OK
B. NOTIFY with a message body of 200 OK
C. 202 Accepted
D. 100 Trying
E. 200 BYE

Correct Answer: B
Explanation/Reference:
The Refer method always begins within the context of an existing call and starts with the originator. The originator sends a Refer request to the recipient (user agent receiving the Refer request) to initiate a triggered Invite request. The triggered Invite request uses the SIP URL contained in the Refer-To header as the destination of the Invite request. The recipient then contacts the resource in the Refer-To header (final-recipient), and returns a SIP 202 (Accepted) response to the originator. The recipient also must notify the originator of the outcome of the Refer transaction—whether the final-recipient was successfully or unsuccessfully contacted. The notification is accomplished using the Notify Method, SIP's event notification mechanism. A Notify message with a message body of SIP 200 OK indicates a successful transfer, while a body of SIP 503 Service Unavailable indicates an unsuccessful transfer. If the call was successful, a call between the recipient and the final-recipient results.

Question 317
Which two types of SIP messages can the OGW use to tunnel a waiting QSIG message?
Refer to the exhibit.

During a QSIG tunneling over SIP call establishment, which two types of SIP messages can the OGW use to tunnel a waiting QSIG message? (Choose two.)
A. SIP re-INVITE
B. SIP NOTIFY
C. SIP INFO
D. SIP OPTIONS
E. SIP UPDATE
F. SIP REFER

Correct Answer: AE
Explanation/Reference:
The TGW sends and the OGW receives a 200 OK response—the OGW sends an ACK message to the TGW and all successive messages during the session are encapsulated into the body of SIP INFO request messages. There are two exceptions:
When a SIP connection requires an extended handshake process, renegotiation, or an update, the gateway may encapsulate a waiting QSIG message into a SIP re-INVITE or SIP UPDATE message during QSIG call establishment.
When the session is terminated, gateways send a SIP BYE message. If the session is terminated by notice of a QSIG RELEASE COMPLETE message, that message can be encapsulated into the SIP BYE message.

Question 318
Which SIP response message should the TGW send if it cannot process the tunneled QSIG messages from the OGW?
Refer to the exhibit.

Which SIP response message should the TGW send if it cannot process the tunneled QSIG messages from the OGW?
A. 405 Method Not Allowed
B. 406 Not Acceptable
C. 412 Conditional Request Failed
D. 415 Unsupported Media Type
E. 485 Ambiguous
Correct Answer: D
Explanation/Reference:
Explanation:
Fallback from QSIG Tunneling
In some situations, QSIG tunneling will fail or need to fall back:
Remote party does not support multipart MIME body: In this case, the remote side sends a “415 Media Not Supported” response. Upon receiving this response, OGW will fall back to normal mode and send an INVITE request without any tunneled content. This procedure helps ensure that at least the basic call will work normally.
Remote party does not understand tunneled content: If the remote side does not support the tunneled content, it should drop the tunneled content and continue as a normal call. Because all essential parameters are present in the original INVITE, the call can go through without the need for fallback.

Question 319
Which SIP message header is used to tunnel QSIG messages across the SIP network when the OGW receives a call bound for the TGW? Refer to the exhibit.

Correct Answer: B
Explanation/Reference:
Explanation:
Tunneling over SIP
The Cisco gateway receives QSIG messages from the PBX side and then identifies the destination of the message (or call). The QSIG messages received from the PBX are encapsulated within SIP messages as Multipurpose Internet Mail Extensions (MIME) bodies and are sent (tunneled) across the IP network to the recipient gateway.
When encapsulating a QSIG message (for switch type primary-qsig) inside a SIP message, Cisco gateways include the QSIG message in a MIME body of the SIP request or response using media type application/QSIG:
Content-Type: application/QSIG
Reference:

Question 320
Which SIP reason phrase maps to SIP response reason code 181?

Correct Answer: B
Explanation/Reference:
Explanation:
1xx — Provisional Responses
100 Trying
Extended search being performed may take a significant time so a forking proxy must send a 100 Trying response.
180 Ringing
Destination user agent received INVITE, and is alerting user of call.
181 Call Is Being Forwarded
Servers can optionally send this response to indicate a call is being forwarded.[1]: §21.1.3
182 Queued
Indicates the destination was temporarily unavailable, so the server has queued the call until the destination is available. A server may send multiple 182 responses to update progress of the queue.
183 Session In Progress
This response may be used to send extra information for a call which is still being set up.
199 Early Dialog Terminated
Can be used by User Agent Server to indicate to upstream SIP entities (including the User Agent Client (UAC)) that an early dialog has been terminated.

Question 321
Which SIP trunk deployment model is shown in this enterprise VoIP topology? Refer to the exhibit.
Which SIP trunk deployment model is shown in this enterprise VoIP topology?
A. mixed TDM and VoIP
B. centralized
C. hybrid
D. traditional TDM
E. distributed

Correct Answer: C

Explanation/Reference:
Hybrid SIP Trunk Model In a hybrid SIP trunk deployment, some of the businesses' sites conform to a distributed SIP trunk deployment model. In this model each site has direct SIP session connectivity to the IP PSTN, and other sites conform to a centralized SIP trunk deployment, accessing the IP PSTN through a central hub, which has SIP session connectivity to the IP PSTN (Figure 3). The hybrid SIP trunk deployment model may have multiple 'central' hubs in different geographic regions, or for specific business functions, such as call centers.

Figure 3 Hybrid SIP Trunk Deployment Mode


Question 322
Which SIP request is used to deliver the actual DTMF digits?
In Key Press Markup Language, which SIP request is used to deliver the actual DTMF digits?
A. SUBSCRIBE
B. INFO
C. NOTIFY
D. INVITE
E. ACK

Correct Answer: C

Explanation/Reference:
KPML procedures use a SIP SUBSCRIBE message to register for DTMF digits. The digits themselves are delivered in NOTIFY messages containing an XML encoded body.

Question 323
What is the minimum number of H.225 messages required to establish an H.323 call with bidirectional media?

What is the minimum number of H.225 messages required to establish an H.323 call with bidirectional media?
A. 1
B. 2
C. 3
D. 4
E. 5

Correct Answer B

Explanation/Reference:
A typical H.245 exchange looks similar to below figure.

For this reason, H.323 version 2 (published in 1998) introduced a concept called Fast Connect, which enables a device to establish bi-directional media flows as part of the H.225.0 call establishment procedures. With Fast Connect, it is possible to establish a call with bi-directional media flowing with no more than two messages, like in figure 3.

Fast Connect is widely supported in the industry. Even so, most devices still implement the complete H.245 exchange as shown above and perform that message exchange in parallel to other activities, so there is no noticeable delay to the calling or called party.
Question 324
Which statement about G.722.1 codec support on Cisco Unified Communications Manager is true?
A. It is always preferred by Cisco Unified Communications Manager over G.711.
B. It is a high-complexity wideband codec.
C. It operates at bit rates of 15.2 and 13.3 kb/s.
D. It is supported for SIP and SCCP devices.
E. It is supported for SIP and H.323 devices.
Correct Answer: E
Explanation/Reference:
Explanation:
G.722.1 is a low-complexity wideband codec operating at 24 and 32 kb/s. The audio quality approaches that of G.722 while using at most half the bit rate. As it is optimized for both speech and music, G.722.1 has slightly lower speech quality than the speech-optimized iSAC codec. G.722.1 is supported for SIP and H.323 devices.

Question 325
Which two types of devices on Cisco Unified Communications Manager support iSAC?
A. MGCP
B. SIP
C. SCCP
D. Music on Hold server
E. H.323
Correct Answer: BC
Explanation/Reference:
Explanation:
iSAC-Internet Speech Audio Codec (iSAC) is an adaptive wideband audio codec, specially designed to deliver wideband sound quality with low delay in both low and medium-bit rate applications. Using an adaptive bit rate of between 10 and 32 kb/s, iSAC provides audio quality approaching that of G.722 while using less than half the bandwidth. In deployments with significant packet loss, delay, or jitter, such as over a WAN, iSAC audio quality is superior to that of G.722 due to its robustness. iSAC is supported for SIP and SCCP devices. The Cisco Unified Communications Manager IP Voice Media Streaming App (IPVMSApp), which includes Media Termination Point, Conference Bridge, Music on Hold Server, and Annunciation does not support iSAC. MGCP devices are not supported.

Question 326
Which two data frame lengths are supported by iLBC?
A. 10 milliseconds
B. 20 milliseconds
C. 30 milliseconds
D. 40 milliseconds
E. 50 milliseconds
F. 60 milliseconds
Correct Answer: BC
Explanation/Reference:
Explanation:
iLBC-Internet Low Bit Rate Codec (iLBC) provides audio quality between that of G.711 and G.729 at bit rates of 15.2 and 13.3 kb/s, while allowing for graceful speech quality degradation in a lossy network due to the speech frames being encoded independently. By comparison, G.729 does not handle packet loss, delay, and jitter well, due to the dependence between speech frames. iLBC is supported for SIP, SCCP, H.323, and MGCP devices.

Question 327
Which statement about the iSAC on Cisco Unified Border Element is true?
A. It is a narrow-band codec.
B. It has a fixed frame of 30 milliseconds.
C. It has an adaptive frame of up to 60 milliseconds.
D. It is designed to deliver wideband sound quality in high-bit-rate applications only.
E. It is not yet supported on the Cisco Unified Border Element (CUBE)
F. It is not yet supported on Cisco Unified Border Element.
Correct Answer: C
Explanation/Reference:
Explanation:
iSAC-Internet Speech Audio Codec (iSAC) is an adaptive wideband audio codec, specially designed to deliver wideband sound quality with low delay in both low and medium-bit rate applications. Using an adaptive bit rate of between 10 and 32 kb/s, iSAC provides audio quality approaching that of G.722 while using less than half the bandwidth. In deployments with significant packet loss, delay, or jitter, such as over a WAN, iSAC audio quality is superior to that of G.722 due to its robustness. iSAC is supported for SIP and SCCP devices. The Cisco Unified Communications Manager IP Voice Media Streaming App (IPVMSApp), which includes Media Termination Point, Conference Bridge, Music on Hold Server, and Annunciation does not support iSAC. MGCP devices are not supported.

Question 328
Which ITU-T recommendation defines the procedures for using more than one video channel in H.320-based systems?
Which ITU-T recommendation defines the procedures for using more than one video channel in H.320-based systems?
A. H.324
B. H.230
C. H.239
D. H.264
E. H.329
Correct Answer: C
Explanation/Reference:

H.239 is the ITU standard for a second video channel; this is supported by all the major vendors, and is the only content channel standard supported by Cisco acquired Codian products or Cisco TelePresence Serial Gateway Series products on H.323 video calls. Cisco acquired Codian products need to be configured to enable H.239. Any H.323 endpoint that also supports the H.239 protocol can source this channel, as can a VNC connection, though some endpoints need to be configured to use H.239 instead of their proprietary solution.

Question 329
When a Cisco IOS gatekeeper receives an LRQ, what is the first step it will take in an attempt to resolve the destination address?
A. Check to see if the LRQ reject-unknown-prefix flag is set.
B. Check to see if the destination address matches the technology prefix.
C. Check to see if the destination address matches the hop-off technology prefix.
D. Check to see if the destination address matches the remote zone prefix.
E. Check to see if the LRQ forward-queries flag is set.
Correct Answer: B
Explanation/Reference:

LRQ — These messages are exchanged between gatekeepers and are used for inter-zone (remote zone) calls. For example, gatekeeper A receives an ARQ from a local zone gateway requesting call admission for a remote zone device. Gatekeeper A then sends an LRQ message to gatekeeper B. Gatekeeper B replies to the LRQ message with either a Location Confirm (LCF) or Location Reject (LRJ) message, which depends on whether it is configured to admit or reject the inter-zone call request and whether the requested resource is registered.

Question 330
When a Cisco IOS gatekeeper receives an ARQ from a registered endpoint, what is the first step it will take in an attempt to resolve the destination address?
A. Check to see if the destination address is locally registered.
B. Check to see if the destination address matches the technology prefix.
C. Check to see if the destination address matches the local zone prefix.
D. Check to see if the destination address matches the remote zone prefix.
E. Check to see if the destination address matches the default technology prefix.
Correct Answer: B
Explanation/Reference:

Admission Request (ARQ) and Location Request (LRQ) are the two H.225 Registration, Admission, Status (RAS) messages that trigger a gatekeeper to initiate the call routing decision process.

ARQ — Local zone messages that are sent by H.323 endpoints (usually gateways) to the Cisco gatekeeper. Gatekeepers receive ARQs from an endpoint if:
- A local zone endpoint initiates a call. OR
- A local zone endpoint request permission to admit an incoming call.

Question 331
Which RAS message is used between two gatekeepers to resolve an alias address?
A. GRQ
B. ARQ
C. IRQ
D. LRQ
E. RRQ
Correct Answer: D
Explanation/Reference:

LRQ — These messages are exchanged between gatekeepers and are used for inter-zone (remote zone) calls. For example, gatekeeper A receives an ARQ from a local zone gateway requesting call admission for a remote zone device. Gatekeeper A then sends an LRQ message to gatekeeper B. Gatekeeper B replies to the LRQ message with either a Location Confirm (LCF) or Location Reject (LRJ) message, which depends on whether it is configured to admit or reject the inter-zone call request and whether the requested resource is registered.

Question 332
Which two VoIP protocols use SDP to describe streaming media sessions?
Which two VoIP protocols use SDP to describe streaming media sessions? (Choose two.)
A. SCCP
B. H.323
C. SIP
D. MGCP
E. RAS
F. cRTP
Correct Answer: CD
Explanation/Reference:

The Session Description Protocol (SDP), defined in RFC 2327, describes the content of sessions, including telephony, Internet radio, and multimedia applications. SDP includes information about [9]:
- Media streams: A session can include multiple streams of differing content. SDP currently defines audio, video, data, control, and application as stream types, similar to the MIME types used for Internet mail.
Addresses: SDP indicates the destination addresses, which may be a multicast address, for a media stream.
Ports: For each stream, the UDP port numbers for sending and receiving are specified.
Payload types: For each media stream type in use (for example, telephony), the payload type indicates the media formats that can be used during the session.
Start and stop times: These apply to broadcast sessions, for example, a television or radio program. The start, stop, and repeat times of the session are indicated.
Originator: For broadcast sessions, the originator is specified, with contact information. This may be useful if a receiver encounters technical difficulties.

Question 333
Which procedure uses H.225 messages to exchange H.245 Master-Salve Determination information?
Which procedure uses H.225 messages to exchange H.245 Master-Salve Determination information?
A. H.323 Fast Connect
B. H.245 tunneling
C. H.225 tunneling
D. H.323 early media
E. H.245 terminal capability set
Correct Answer: B
Explanation/Reference:
Explanation:
The H.245 protocol is a media control protocol that is a part of the H.323 protocol suite. The H.245 protocol is used primarily to negotiate master-slave relationship between communicating endpoints. These endpoints exchange terminal capabilities and logical channel manipulations (open, close, modify). The H.245 messages can be encapsulated and carried between H.225 controlled endpoints within H.225 messages. This way of "piggy-backing" an H.245 message to a H.225 message is referred to as H245 Tunneling. The H.245 Tunneling method is optional and negotiable between communicating H.323 endpoints. If both endpoints support this option, usually the H.245 Media Controlled messages are exchanged via the Tunneling method.

Question 334
What is the name of the logical channel proposal that is transmitted from the called entity to the calling entity in H.323 Fast Connect?
What is the name of the logical channel proposal that is transmitted from the called entity to the calling entity in H.323 Fast Connect?
A. Forward Logical Channel
B. Backward Logical Channel
C. Reverse Logical Channel
D. Originator Logical Channel
E. Destination Logical Channel
Correct Answer: C
Explanation/Reference:
Explanation:
Unlike the OpenLogicalChannel request used by H.323 for video uni-directional logical channels, the request used by H.324 for opening video bi-directional logical channels specifies the temporalSpatialTradeOff Capability in both the forward and reverse directions in the forwardLogicalChannelParameters.dataType and reverseLogicalChannelParameters.dataType components, respectively. The semantics of temporalSpatialTradeOff Capability used in forward LogicalChannelParameters.dataType is described in the previous section. The semantics for its presence in the reverse direction is described in this section.

Question 335
Which element was added to H.225 messages to enable Fast Connect in H.323 version 2?
Which element was added to H.225 messages to enable Fast Connect in H.323 version 2?
A. fastStart
B. fastConnect
C. H.245 PDU
D. User-User Information
E. Connection Information
Correct Answer: A
Explanation/Reference:
Explanation:
Fast start allows for H323 media connections to be started at the beginning of a call. This is helpful for ringback scenarios, and also reduces the amount of time calls take to establish media. H245 is still negotiated later, but the actual media can be done earlier through H225 messages.

Question 336
What is the minimum number of TCP sessions needed to complete a H.323 call between two H.323 gateways using slow start?
What is the minimum number of TCP sessions needed to complete a H.323 call between two H.323 gateways using slow start?
A. 0
B. 1
C. 2
D. 3
E. 4
Correct Answer: C
Explanation/Reference:
Explanation:
H.323 has two modes of operation: slow start and fast start. The initiation of a call may proceed in a slow start or fast start in H.323. In a slow start, H.323 signaling consists of Setup, Call Proceeding, Alerting, and Connect steps. After these steps, the H.245 media negotiation is performed. When a call is initiated in H.323 fast start, the H.245 media negotiation is performed within the initial Setup message. With slow start, multiple TCP connections are needed for an H.323 call, such as one H.225 signaling channel and one H.245 signaling channel if required (minimum of these two).

Question 337
Which two statements describe characteristics of Binary Floor Control Protocol?
Which two statements describe characteristics of Binary Floor Control Protocol? (Choose two.)
A. Its binary encoding is designed to work in high-bandwidth environments.
B. It is designed for audio or video conference sessions of three or more participants.
C. It enables management of shared content resources independent of video streams.
D. It supports TLS-based authentication.
E. It supports SIP as well as H.323.
Correct Answer: CD
Explanation/Reference:
Explanation:
BFCP is a deliverable developed as part of the Internet Engineering Task Force (IETF) XCON Centralized Conferencing working group. The IETF XCON working group was formed to focus on delivering a standards-based approach to managing IP conferencing while promoting broad interoperability between software and equipment vendors.

Question 338
How many SIP signaling dialog(s) took place in this SIP message exchange between two SIP user agents?
Refer to the exhibit.

How many SIP signaling dialog(s) took place in this SIP message exchange between two SIP user agents?
A. 1
B. 2
C. 3
D. 4
E. 5
F. 6

Correct Answer: A
Explanation/Reference:
Explanation:
During the establishment, maintenance and termination of a SIP session, signaling messages are exchanged between the two SIP endpoints. There are two different kinds of signaling "conversations" that those messages take part in: transactions and dialogs.
A transaction is a SIP message exchange between two user-agents that starts with a request and ends with its final response (it can also contain zero or more provisional responses in between). For example, during the termination of a SIP session, one user releases the call by sending a BYE request and the other party replies back with a 200 OK response. This message exchange is called a transaction.
But what happens in the case of the INVITE request? The establishment of a SIP session starts basically with an INVITE request and is considered as completed upon the receipt of the ACK. In this case, the transaction starts with the INVITE request and ends with the 200 OK, so the ACK is not part of the transaction. The ACK can be considered as a transaction on its own. However, when the final response to an INVITE is not a 2xx response, then the ACK is considered as part of the transaction.
A dialog is a complete exchange of SIP messages between two user-agents. That means that transactions are actually parts of a dialog. For example, in the case of a SIP session establishment, a dialog starts with the INVITE-200 OK transaction, continues with the ACK and ends with the BYE-200 OK transaction.
The picture below depicts the dialog and transactions that take place during the establishment of a SIP session:

Note: There can also be subsequent requests that belong to the same dialog, such as a BYE or a re-INVITE message. As out-of-dialog requests are considered such as an initial INVITE request for a new session or an OPTIONS message for checking capabilities.
There are different SIP headers/parameters that identify the dialogs and transactions, and they will be analyzed in later posts.
Reference: https://telconotes.wordpress.com/2013/03/13/sip-transactions-vs-dialogs/
How many SIP signaling transaction(s) took place in this SIP message exchange between two SIP user agents?

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

Correct Answer: C

Explanation/Reference:

During the establishment, maintenance and termination of a SIP session, signaling messages are exchanged between the two SIP endpoints. There are two different kinds of signaling "conversations" that those messages take part in: transactions and dialogs.

A transaction is a SIP message exchange between two user-agents that starts with a request and ends with its final response (it can also contain zero or more provisional responses in between). For example, during the termination of a SIP session one user releases the call by sending a BYE request and the other party replies back with a 200 OK response. This message exchange is called a transaction.

But what happens in the case of the INVITE request? The establishment of a SIP session starts basically with an INVITE request and is considered as completed upon the receipt of the ACK. In this case, the transaction starts with the INVITE request and ends with the 200 OK, so the ACK is not part of the transaction. The ACK can be considered as a transaction on its own. However, when the final response to an INVITE is not a 2xx response, then the ACK is considered as part of the transaction.

A dialog is a complete exchange of SIP messages between two user-agents. That means that transactions are actually parts of a dialog. For example, in the case of a SIP session establishment, a dialog starts with the INVITE-200 OK transaction, continues with the ACK and ends with the BYE-200 OK transaction.

The picture below depicts the dialog and transactions that take place during the establishment of a SIP session:

Note: There can also be subsequent requests that belong to the same dialog, such as a BYE or a re-INVITE message. As out-of-dialog requests are considered messages such as an initial INVITE request for a new session or an OPTIONS message for checking capabilities. There are different SIP headers/parameters that identify the dialogs and transactions, and they will be analyzed in later posts.

Reference: https://telconotes.wordpress.com/2013/03/13/sip-transactions-vs-dialogs/
D. Client Failure
E. Server Failure
Correct Answer: C
Explanation/Reference:
Explanation:
The 301 response from the Web server should always include an alternative URL to which redirection should occur. If it does, a Web browser will immediately retry the alternative URL. So you never actually see a 301 error in a Web browser, unless perhaps you have a corrupt redirection chain e.g. URL A redirects to URL B which in turn redirects back to URL A. If your client is not a Web browser, it should behave in the same way as a Web browser i.e. immediately retry the alternative URL.

Question 341
Which SIP message will UA#2 use to communicate its media capability to UA#1?
Refer to the exhibit.

If this SIP call is initiated using delayed offer, which SIP message will UA#2 use to communicate its media capability to UA#1?
A. INVITE
B. 180 Ringing
C. 200 OK
D. ACK
E. RTP Media
Correct Answer: C
Explanation/Reference:
Explanation:
200 OK Indicates the request was successful.

Question 342
Which statement about this message is true?
Refer to the exhibit.

You received this debug output to troubleshoot a Cisco IOS MGCP gateway call quality issue at a customer site. Which statement about this message is true?
A. The MGCP gateway is responding to an RQNT message from Cisco Unified Communications Manager to poll the call statistics of an active call.
B. The MGCP gateway is responding to an AUEP message from Cisco Unified Communications Manager to poll the call statistics of a terminating call.
C. The MGCP gateway is responding to an MDCX message from Cisco Unified Communications Manager during a call setup.
D. The MGCP gateway is responding to an AUCX message from Cisco Unified Communications Manager about an active call.
E. The MGCP gateway is responding to a DLCX message from Cisco Unified Communications Manager about a terminating call.
Correct Answer: E
Explanation/Reference:
DeleteConnection — used by a call agent to instruct a gateway to delete an existing connection. DeleteConnection can also be used by a gateway to release a connection that can no longer be sustained.

Question 343
What is the purpose of this message?
Refer to the exhibit.
You received this debug output to troubleshoot a Cisco IOS MGCP gateway problem at a customer site. What is the purpose of this message?

A. The MGCP gateway uses this message to respond to an RQNT message from Cisco Unified Communications Manager.
B. The MGCP gateway uses this message to respond to an AUCX message from Cisco Unified Communications Manager.
C. The MGCP gateway uses this message to respond to an AUEP message from Cisco Unified Communications Manager.
D. The MGCP gateway uses this message to respond to a DLCX message from Cisco Unified Communications Manager.
E. The MGCP gateway uses this message to respond to an NTFY message from Cisco Unified Communications Manager.

Correct Answer: C

Explanation/Reference:
This message requests the status of an endpoint. Information that can be audited with this includes Requested Events, DigitMap, SignalRequests, RequestIdentifier, QuarantineHandling, Notified Entity, Connection Identifiers, Detect Events, Observed Events, Event States, Bearer Information, Restart Method, Restart Delay, ReasonCode, PackageList, Max MGCP Datagram, and Capabilities. The response will include information about each of the items for which auditing info was requested.

Question 344
Which DTMF relay method is advertised when the originating SIP gateway sends an INVITE message with a Call-Info header shown? Refer to the exhibit.

A. RFC 2833
B. SIP INFO
C. SIP NOTIFY
D. SIP KPML
E. In-band audio

Correct Answer: C

Explanation/Reference:
You can develop user-specific applications that reside on your network entity and have the ability to subscribe for event services supported by the IMG. If the network entity wants the ability to detect an entered DTMF digit (only telephone event of "###" are currently supported) from the TDM-side of a call to the IP side of a call, the entity can subscribe to the IMG for these events and receive SIP NOTIFY events containing the digit event.

Question 345
How are DTMF digits transported in RFC 2833?

A. In the RTP stream with the named telephone events payload format.
B. In the RTP stream with the regular audio payload format.
C. In SIP NOTIFY messages.
D. In SIP INFO messages.
E. In SIP SUBSCRIBE messages.

Correct Answer: A

Explanation/Reference:
DTMF digits and named telephone events are carried as part of the audio stream, and MUST use the same sequence number and time-stamp base as the regular audio channel to simplify the generation of audio waveforms at a gateway. The default clock frequency is 8,000 Hz, but the clock frequency can be redefined when assigning the dynamic payload type.

Question 346
Which user inputs are sent from the calling IP phone to the Cisco Unified Communications Manager, in forms of SCCP messages, after the user pressed the Dial softkey? Note that the commas in answer choices below are logical separators, not part of the actual user input or SCCP messages. Refer to the exhibit.

A. SCCP IP Phone Extension 1001
B. SCCP IP Phone Extension 1002
C. SCCP IP Phone Extension 2003

Correct Answer: C

Explanation/Reference:
SCCP messages are used to transfer user inputs from the calling IP phone to the Cisco Unified Communications Manager. The diagram shows the steps involved:
1. Phone is on-hook
2. User presses 201, digit by digit
3. User presses the "<" soft key to delete the last digit entered
4. User enters the remaining digits, 03, digit by digit
5. User selects the "Dial" soft key
A user is going through a series of dialing steps on an SCCP IP phone (extension 1001) to call another SCCP IP phone (extension 2003). Both phones are registered to the same Cisco Unified Communications Manager cluster. Which user inputs are sent from the calling IP phone to the Cisco Unified Communications Manager, in forms of SCCP messages, after the user pressed the Dial softkey? Note that the commas in answer choices below are logical separators, not part of the actual user input or SCCP messages.

A. A separate SCCP message is sent to Cisco Unified Communications Manager for each of the following user inputs: 2, 0, 0, 3.
B. A separate SCCP message is sent to Cisco Unified Communications Manager for each of the following user inputs: 2, 0, 1, <<, 0, 3.
C. A single SCCP message is sent to Cisco Unified Communications Manager to report that digits 2003 have been dialed.
D. A single SCCP message is sent to Cisco Unified Communications Manager to report that digits 201<<03 have been dialed.
E. A separate SCCP message is sent to Cisco Unified Communications Manager for each of the following user inputs: 2, 0, 1, <<, 2, 0, 0, 3.

Correct Answer: C

Explanation/Reference:

After the user delete phone stop the digit by digit dialing and send it as a whole setup.

Question 347

Which two compression formats for high-definition video have technical content that is identical to H.264?
Which two compression formats for high-definition video have technical content that is identical to H.264? (Choose two.)
A. MPEG-4 Part 10
B. MPEG-4 Part 14
C. MPEG-2 Part 7
D. AVC
E. VC3
F. VP8

Correct Answer: AD

Explanation/Reference:

Explanation:

MPEG-4 Part 10, also known as MPEG-4 AVC (Advanced Video Coding), is actually defined in an identical pair of standards maintained by different organizations, together known as the Joint Video Team (JVT). While MPEG-4 Part 10 is a ISO/IEC standard, it was developed in cooperation with the ITU, an organization heavily involved in broadcast television standards. Since the ITU designation for the standard is H.264, you may see MPEG-4 Part 10 video referred to as either AVC or H.264. Both are valid, and refer to the same standard.

Question 348

Which H.245 information is exchanged within H.225 messages in H.323 Fast Connect?

A. Terminal Capability Set
B. Open Logical Channel
C. Master-Slave Determination
D. Call Setup
E. Call Progress

Correct Answer: B

Explanation/Reference:

Explanation:

With the standard H.245 negotiation, the two endpoints need three round-trips before they agree on the parameters of the audio/video channels (1. master/slave voting, 2. terminal capability set exchange, and finally, 3. opening the logical channels). In certain situations and especially with high-latency network links, this can last too long and users will notice the delay.

Question 349

Which two responses are examples of client error responses in SIP protocol?
Which two responses are examples of client error responses in SIP protocol? (Choose two.)
A. 302 Moved Temporarily
B. 404 Not Found
C. 503 Service Unavailable
D. 502 Bad Gateway
E. 604 Does Not Exist Anywhere
F. 408 Request Timeout

Correct Answer: BF

Explanation/Reference:

Explanation:

Client Error (400 to 499) — Request contains bad syntax or cannot be fulfilled at this server. This class of 400 to 499 contains only error messages.

Question 350

Which SIP message will UA#1 use to communicate its media capability to UA#2?

Refer to the exhibit.

If this SIP call is initiated using delayed offer, which SIP message will UA#1 use to communicate its media capability to UA#2?
A. INVITE
B. 180 Ringing
C. 200 OK
D. ACK
E. RTP Media

Correct Answer: D
Explanation/Reference:

In the Delayed Offer process, the calling does not send its offer in the SIP INVITE Message. The callee sends the offer within the SDP fields of its answer (SIP 200 OK). The calling answers within the ACK message.

**Question 351**
Which SIP message will UA#2 use to communicate its media capability to UA#1?
Refer to the exhibit.

![Exhibit](http://www.aoowe.com/practice-400-051-3156.html)

If this SIP call is initiated using early offer, which SIP message will UA#2 use to communicate its media capability to UA#1?
A. INVITE
B. 180 Ringing
C. 200 OK
D. ACK
E. RTP Media

Correct Answer: C
Explanation/Reference:

In Early offer, SIP Send SDP in the invite, the other node will send the SDP in the 200 message.

**Question 352**
Which two SDP content headers can be found in a SIP INVITE message?
(Choose two.)
A. Expires
B. Contact
C. Connection Info
D. Media Attributes
E. Allow
F. CSeq

Correct Answer: CD
Explanation/Reference:

Connection info is optional field in SDP whether Media attributes decide the codec and media type for that call.

**Question 353**
Which SIP response is considered a final response?
A. 183 Session in Progress
B. 199 Early Dialog Terminated
C. 200 OK
D. 180 Ringing
E. 100 Trying

Correct Answer: C
Explanation/Reference:

200 OK Indicates the request was successful. Whether other options state the request is still in progress or request is initiated.

**Question 354**
Which SIP request method enables reliability of SIP 1xx response types?
A. ACK
B. PRACK
C. OPTIONS
D. CANCEL
E. REGISTER

Correct Answer: B
Explanation/Reference:

Explanation:
In order to achieve reliability for provisional responses, we do nearly the same thing. Reliable provisional responses are retransmitted by the TU with an exponential backoff. Those retransmissions cease when a PRACK message is received. The PRACK request plays the same role as ACK, but for provisional responses. There is an important difference, however. PRACK is a normal SIP message, like BYE. As such, its own provisional responses. There is an important difference, however. PRACK is a normal SIP message, like BYE. As such, its own

**Question 355**
Which SIP response class do the SIP response codes 300 to 399 belong to?
A. Provisional
B. Client Failure
C. Server Failure
D. Successful
E. Redirection

Correct Answer: A
Explanation/Reference:

Redirection — further action needs to be taken in order to complete the request. That is what this class implies.

**Question 356**
What is the purpose of this message?
Refer to the exhibit.

You received this debug output to troubleshoot a Cisco IOS MGCP gateway media-related problem at a customer site. What is the purpose of this message?
A. The MGCP gateway is responding to an RQNT message from Cisco Unified Communications Manager to poll the media capabilities on its endpoints.
B. The MGCP gateway is responding to an AUEP message from Cisco Unified Communications Manager to poll the media capabilities on its endpoints.
C. The MGCP gateway is responding to an AUCX message from Cisco Unified Communications Manager to poll the active calls on its endpoints.
D. The MGCP gateway is responding to an MDCX message from Cisco Unified Communications Manager during a call setup.
E. The MGCP gateway is responding to a CRCX message from Cisco Unified Communications Manager during a call setup.

Correct Answer: D
Explanation/Reference:

See MGCP packet debugging examples and their meanings at the Reference link below.
Reference: Sample of Debug MGCP Packets

**Question 357**
Which statement about this endpoint on the Cisco MGCP gateway is true?
Refer to the exhibit.

You received this debug output to troubleshoot a Cisco IOS MGCP gateway problem at a customer site. Which statement about this endpoint on the Cisco MGCP gateway is true?
A. This endpoint is on a T1 Controller 0/1/0.
B. This endpoint is on an E1 Controller 0/1/0.
C. This endpoint is on a T1 Controller 0/1/1.
D. This endpoint is on an E1 Controller 0/1/2.
E. This endpoint is on a T1 Controller 0/1/2.

Correct Answer: A
Explanation/Reference:

The s0/Su1/DS1-0 refers to the slot and port information (0/1/0). It is also a DS1 as shown by this output, which means it is a T1 not an E1.

**Question 358**
Which device is the initiator of a StationInit message in a Cisco Unified Communications Manager SDI trace?
Which device is the initiator of a StationInit message in a Cisco Unified Communications Manager SDI trace?
A. Cisco Unified Communications Manager
B. MGCP gateway
C. Cisco Music on Hold server
D. SCCP IP phone
E. SIP Proxy Server

Correct Answer: D
Explanation/Reference:

Station Init means that an inbound Transmission Control Protocol (TCP) message from a Skinny station reached CallManager. A Skinny station is any endpoint that uses the Skinny protocol to communicate with CallManager.
Which two SCCP call signaling messages are sent by an IP phone to Cisco Unified Communications Manager? (Choose two.)

A. SoftKeyEvent
B. OpenReceiveChannelAck
C. StartMediaTransmission
D. SelectSoftKeys
E. CloseReceiveChannel
F. StopTone

Correct Answer: AB

Explanation:
This message indicates which soft key was pressed. Upon receipt of this message, CallManager invokes the action associated with the pressed soft key. For example, if Hold was the pressed soft key, CallManager places the active call on user hold. In some trace files you might see a soft key number without the corresponding description. The following list defines each soft key number.

---

Which two SCCP call signaling messages are initiated by Cisco Unified Communications Manager to an IP phone? (Choose two.)

A. SoftKeyEvent
B. CloseReceiveChannelAck
C. CallState
D. KeypadButton
E. OpenReceiveChannel
F. Offhook

Correct Answer: CE

Explanation:
Upon receiving an OpenReceiveChannel message, the IP phone selects the UDP port number it wants to use to receive RTP packets and reports this information to call manager.

With the SCCP protocol architecture, the majority of the H.323 processing power resides in an H.323 proxy — the Cisco CallManager. The end stations (IP phones) run the Skinny client, which consumes less processing overhead. The client communicates with CallManager using connection-oriented (TCP/IP-based) communication to establish a call with another H.323-compliant end station. Once Cisco CallManager has established the call, the two H.323 end stations use connectionless (UDP/IP-based) communication for audio transmissions.

---

What is the maximum number of call-processing subscribers in a standard deployment of a Cisco Unified Communications Manager Session Management Edition cluster?

A. 3
B. 4
C. 5
D. 8
E. 16

Correct Answer: D

Explanation:
There is no deployment difference between CUCM & CUCM session management Edition cluster. The only difference is that CUCM SME is designed to support a large number of trunk to trunk connections. Thus, 8 subscribers.

---

Which four attributes are needed to determine the time to complete a TFTP file transfer process? (Choose four.)

A. file size
B. file type
C. network interface type
D. round-trip time
E. packet loss percentage
F. response timeout
G. network throughput

Correct Answer: ADEF

Explanation:
Four attributes that are needed to determine the time to complete TFTP file transfer process is:
1. File Size
2. Round-trip time
3. Packet loss percentage
4. Response timeout


---

Which two file types can be found when you choose Software Upgrades, followed by TFTP File Management on the Cisco Unified Operating System Administration web page for Cisco Unified Communications Manager?

Refer to the exhibit.
Assuming that the administrator has never performed any manual custom uploads, which two file types can be found when you choose Software Upgrades, followed by TFTP File Management on the Cisco Unified Operating System Administration web page for Cisco Unified Communications Manager? (Choose two.)

A. IP phone configuration files
B. announcement audio files
C. ringer files
D. IP phone license files
E. sample music-on-hold audio files
F. softkey template files

Correct Answer: BC

Explanation/Reference:
The two file types that we get are Announcement Audio Files and Ringer Files.

Question 364
Which statement about Cisco EnergyWise domain member neighbor formation is true?

A. Cisco EnergyWise supports static neighbors, but the neighbor relationship is only possible if a noncontiguous domain member and a contiguous domain member have a static neighbor entry pointing to each other.
B. Cisco EnergyWise static neighbors can be formed even if domain members are not physically contiguous.
C. Static neighbors can be manually defined on Cisco EnergyWise domain members, but TCP protocols must be used.
D. Static neighbors can be manually defined on Cisco EnergyWise domain members, but they have a lower priority compared to the autodiscovered members.
E. Static neighbors can be manually defined on Cisco EnergyWise domain members and the TCP or UDP protocol can be used.

Correct Answer: B

Explanation/Reference:

Question 365
Which two terms describe a Cisco IP phone?

In a Cisco EnergyWise domain, which two terms describe a Cisco IP phone? (Choose two.)

A. endpoint
B. domain member
C. child domain member
D. EnergyWise agent
E. Cisco power distribution unit

Correct Answer: AC

Explanation/Reference:

Question 366
Which capability is support by LLDP-MED but not by Cisco Discovery Protocol?

Which capability is support by LLDP-MED but not by Cisco Discovery Protocol?

A. LAN speed discovery
B. network policy discovery
C. location identification discovery
D. power discovery
E. trust extension

Correct Answer: A

Explanation/Reference:
LLDP-MED supports both LAN speed and duplex discovery. Cisco Discovery Protocol supports duplex discovery only, but this limited support is not seen as a problem because if there is a speed mismatch, LLDP-MED and Cisco Discovery Protocol cannot be exchanged and thus cannot be used to detect the mismatch.

Question 367
Which definition is included in a Cisco UC on UCS TRC?

Which definition is included in a Cisco UC on UCS TRC?

A. required RAID configuration, when the TRC uses direct-attached storage
B. configuration of virtual-to-physical network interface mapping
C. step-by-step procedures for hardware BIOS, firmware, drivers, and RAID setup
D. configuration settings and patch recommendations for VMware software
E. server model and local components (CPU, RAM, adapters, local storage) by name only; part numbers are not included because they change over time

Correct Answer: A

Explanation/Reference:
Definition of server model and local components (CPU, RAM, adapters, local storage) at the orderable part number level.
Required RAID configuration (e.g. RAID5, RAID10, etc.) – including battery backup cache or SuperCap – when the TRC uses DAS storage

Guidance on hardware installation and basic setup.

– Configuration of Virtual-to-physical network interface mapping is design-dependent and not included in TRC definition.
– Configuration of adapters (such as Cisco VIC, 3rd-party CNICNIC / HBA) is design-dependent and not included in TRC definition.
– Network routing and switching (e.g. routers, gateways, MCUs, ethernet/FC/PCoE switches, Cisco Catalyst/Nexus/MDS, etc.)
– QoS configuration of route/switch network devices
– Cisco UCS B-Series chassis and switching components (e.g. Cisco UCS 6100/6200, Cisco UCS B200/2200, Cisco UCS 5100)
– Storage arrays (such as those from EMC, NetApp or other vendors)

Configuration settings, patch recommendations or step by step procedures for VMware software are not included in TRC definition. Infrastructure solutions such as Vblock from Virtual Computing Environment may also be leveraged for configuration details not included in the TRC definition.

Question 368
Which two statements about the Cisco UC on UCS specs-based virtualization support model are true?

Which two statements about the Cisco UC on UCS specs-based virtualization support model are true? (Choose two.)

A. It has a configuration-based approach.
B. It has a rule-based approach.
C. It has less hardware flexibility compared to the third-party server specs-based support model.
D. It has less hardware flexibility compared to the UC on UCS TRC support model.
E. VMware vCenter is optional with this support model.

Correct Answer: BC

Explanation/Reference:
Reference: http://docwiki.cisco.com/wiki/UC_Virtualization_Supported_Hardware#UC_on_ucs Tested_Reference_Configurations

Question 369
Which two statements about the Peer Firmware Sharing option for IP phone firmware distribution are true?

Which two statements about the Peer Firmware Sharing option for IP phone firmware distribution are true? (Choose two.)

A. This option uses a parent-child hierarchy in which a firmware image is downloaded by a parent phone to up to three directly associated child phones.
B. This option must be enabled on Cisco Unified Communications Manager service parameters for Cisco TFTP.
C. This option mandates that the parent phone and child phones be identical, selected phone models.
D. This option allows firmware transfers between phones in different subnets, as long as the round-trip delay is less than 5 milliseconds.
E. This option uses a parent-child hierarchy that must be manually defined by the Cisco Unified Communications Manager administrator.
F. This option allows falling back to the TFTP server in the Cisco Unified Communications Manager cluster.

Correct Answer: CF

Explanation/Reference:
Peer Firmware Sharing works by setting up a parent-child hierarchy of the phones in which a firmware image is downloaded by the parent phone to a child phone. The advantage of using Peer Firmware Sharing is that instead of all phones individually retrieving a firmware image, they pass the image along from one phone to another phone on the same subnet.

Advantage of PFS:
Hierarchy is automatic
One download per phone model on a subnet
Uses TCP
Fails back to TFTP
Speeds up LAN upgrades
Reduces TFTP CPU load during upgrade

Question 370
Which two statements about using the Load Server option for IP phone firmware distribution are true?

Which two statements about using the Load Server option for IP phone firmware distribution are true? (Choose two.)

A. This option must be enabled on at least two servers in a Cisco Unified Communications Manager cluster.
B. This option must be enabled on Cisco Unified Communications Manager service parameters for Cisco TFTP.
C. Phone firmware must be manually copied to any applicable load servers.
D. The load server will not function if its IP address is not in the same subnet as the IP phones.
E. This option is only available for newer IP phone models.
F. This option does not accommodate falling back to Cisco TFTP on error.

Correct Answer: CF

Explanation/Reference:
Choosing the Right Distribution Method

Which of the three different image-distribution methods discussed so far is the best for a customer deployment? Each method has advantages and disadvantages, and they are summarized in Table 1.

<table>
<thead>
<tr>
<th>Distribution Method</th>
<th>Peer Firmware Sharing</th>
<th>Load Server</th>
<th>Traditional TFTP</th>
</tr>
</thead>
</table>
| **Advantages** | • Hierarchy is automatic
• One download per phone model on a subnet
• Uses TCP
• Fails back to TFTP
• Speeds up LAN upgrades
• Reduces TFTP CPU load during upgrade | • Has same download time as LAN image distribution
• Distributes TFTP load over multiple TFTP servers | • Faster distribution
• Default behavior |
| **Disadvantages** | • Must be enabled on each phone
• Hierarchy is formed for each phone model
• Hierarchy is limited to subnet | • IP must be set on each phone
• Administrator must manually request
• No failback to TFTP on error
• More prone to user error | • High-bandwidth requirements
• Multiple requests for the same file (hour) |


Question 371

http://www.aoowe.com/practice-400-051-3156.html
Which statement describes a disadvantage of using the Cisco TFTP service to serve IP phone load files?
A. The Cisco TFTP services can run on only one Cisco Unified Communications Manager server in a cluster.
B. Because TFTP operates on top of UDP, there is a high risk of corrupted load file delivery at the completion of the TFTP process due to undetected data loss in the network.
C. If a response is not received in the timeout period, the TFTP server will not resend the data packet.
D. Packet loss can significantly increase the TFTP session completion time.
E. Because TFTP operates with an adaptive timeout period, the time to complete the file transfer is unpredictable.

Correct Answer: D
Explanation/Reference:
Voice traffic cannot recapture lost packets. Rather than retransmitting a lost network connection, the phone resets and attempts to reconnect its network connection.

Question 372
Which two file types can be found when you choose Software Upgrades, followed by TFTP File Management on the Cisco Unified Operating System Administration web page?
Refer to the exhibit.
Assuming that the administrator has never performed any manual custom uploads, which two file types can be found when you choose Software Upgrades, followed by TFTP File Management on the Cisco Unified Operating System Administration web page? (Choose two.)
A. IP phone configuration files
B. sample music-on-hold audio files
C. Identity Trust List files
D. IP phone license files
E. Mobile Voice Access audio files
F. softkey template files

Correct Answer: CE
Explanation/Reference:
We get option for Identity Trust list Files and Mobile Voice Access audio files.

Question 373
Which protocol does the Cisco Prime LAN Management Solution application use to communicate with Cisco EnergyWise domain members?
Which protocol does the Cisco Prime LAN Management Solution application use to communicate with Cisco EnergyWise domain members?
A. UDP broadcast
B. Cisco Discovery Protocol
C. UDP unicast
D. TCP
E. multicast

Correct Answer: D
Explanation/Reference:
Cisco Prime LMS 4.1 uses TCP port 45440.

Question 374
Which two mechanisms does Cisco EnergyWise use for neighbor discovery?
Which two mechanisms does Cisco EnergyWise use for neighbor discovery? (Choose two.)
A. multicast
B. LLDP-MED
C. UDP broadcast
D. Cisco Discovery Protocol
E. TCP

Correct Answer: CD
Explanation/Reference:
The Cisco EnergyWise Neighbor Discovery Process
Cisco EnergyWise CDP packets
UDP broadcast packets are automatically sent out switch ports which support Cisco EnergyWise, regardless of whether the interfaces are configured with the no energywise interface-level command. CDP packets are sent when CDP is configured for the switch ports.
Question 375
Which capability is supported by Cisco Discovery Protocol but not by LLDP-MED?
A. LAN speed and duplex discovery
B. Network policy discovery
C. Location identification discovery
D. Power discovery
E. Trust extension

Correct Answer: E

Explanation/Reference:
Cisco Discovery Protocol provides an additional capability not found in LLDP-MED that allows the switch to extend trust to the phone. In this case, the phone is now trusted to mark the packets received on the PC port accordingly. This feature can be used to off-load the switch because now it does not need to police the information being received from the phone.

Question 376
Which definition is included in a Cisco UC on UCS TRC?
A. storage arrays such as those from EMC or NetApp, if applicable
B. configuration of virtual-to-physical network interface mapping
C. step-by-step procedures for hardware BIOS, firmware, drivers, and RAID setup
D. server model and local components (CPU, RAM, adapters, local storage) at the part number level
E. configuration settings and patch recommendations for VMware software

Correct Answer: D

Explanation/Reference:
What does a TRC definition include?

Definition of server model and local components (CPU, RAM, adapters, local storage) at the orderable part number level.
Required RAID configuration (e.g. RAID5, RAID10, etc.) – including battery backup cache or SuperCap – when the TRC uses DAS storage
Guidance on hardware installation and basic setup (e.g. click here).
– Click here for detailed Cisco UCS server documentation regarding hardware configuration procedures.
– Configuration of Virtual-to-physical network interface mapping is design-dependent and not included in TRC definition.
– Configuration of adapters (such as Cisco VIC, 3rd-party CNA / NIC / HBA) is design-dependent and not included in TRC definition.
– Configuration settings or step by step procedures for hardware BIOS, firmware, drivers, RAID setup are not included. Click here for detailed Cisco UCS server documentation.

Design, installation and configuration of external hardware is not included in TRC definition, such as:
– Network routing and switching (e.g. routers, gateways, MCUs, ethernet/PC/FC/CNd switches, Cisco Catalyst Nexus/MDM, etc.)
– QoS configuration of route/switch network devices
– Cisco UCS B-Series chassis and switching components (e.g. Cisco UCS 6100/6200, Cisco UCS 7400/7400, Cisco UCS 5100)
– Storage arrays (such as those from EMC, NetApp or other vendors)

Configuration settings, patch recommendations or step by step procedures for VMware software are not included in TRC definition.

Infrastructure solutions such as Vblock from Virtual Computing Environment may also be leveraged for configuration details not included in the TRC definition.

Question 377
Which statement about the Cisco UC on UCS TRC and the third-party server specs-based virtualization support model is true?

Which statement about the Cisco UC on UCS TRC and the third-party server specs-based virtualization support model is true?
A. Both the UC on UCS TRC and the third-party servers specs-based support models have rule-based approaches.
B. The UC on UCS TRC support model has a rule-based approach and the third-party servers specs-based support model has a configuration-based approach.
C. The UC on UCS TRC support model requires a high level of virtualization experience while the third-party server specs-based support model requires a low to medium level virtualization experience.
D. VMware vCenter is mandatory for the UC on UCS TRC support model but it is optional for the third-party server spec-based support model.
E. VMware vCenter is optional for the UC on UCS TRC support model but it is mandatory for the third-party server spec-based support model.

Correct Answer: E

Explanation/Reference:
VMware vCenter is optional when deploying on UCS TRC. Reference: Configuration hardware mandatory when deploying on UCS TRC Specs-based and Third-party Server Specs-based hardware. The UC on UCS TRC support model has a rule-based approach and the third-party servers specs-based support model has a configuration-based approach. The UC on UCS TRC support model requires a high level of virtualization experience while the third-party server specs-based support model requires a low to medium level virtualization experience.
D. VMware vCenter is mandatory for the UC on UCS TRC support model but it is optional for the third-party server spec-based support model.
E. VMware vCenter is optional for the UC on UCS TRC support model but it is mandatory for the third-party server spec-based support model.

Correct Answer: E

Explanation/Reference:
VMware vCenter is optional when deploying on UCS TRC. Reference: Configuration hardware mandatory when deploying on UCS TRC Specs-based and Third-party Server Specs-based hardware. vCenter Statistics Level 4 logging is mandatory so that Cisco TAC is able to provide effective support.
Click here for how to configure VMware vCenter to capture these logs. If not configured by default, Cisco TAC may request enabling these settings in order to provide effective support. Also note that configuration of specific VMware vSphere management features may require vCenter and/or a higher feature Edition of vSphere ESXi. Cisco Collaboration does not require its own dedicated vCenter.
Note that when VMware vCenter is not required and is not used, then VMware vSphere ESXi’s default management interface is its free/included VMware vSphere Client (formerly branded VI Client).

Question 378
Which statement about installation media support is true for this migration?

Company ABC is planning to migrate from MCS-hosted Cisco Unified Communications Manager applications to Cisco UC on UCS B-Series servers. Which statement about installation media support is true for this migration?
A. The install log can be written to a USB flash drive that is attached to the UCS server.
B. The answer file that is generated by the Answer File Generator (platformConfig.xml) can be read from a USB flash drive to perform an unattended installation on the UCS server.
C. The Cisco Music on Hold USB audio sound card can be mapped to a virtual USB port on a VMware virtual machine on the UCS server.
D. The answer file that is generated by the Answer File Generator (platformConfig.xml) can be read from an FLP image that is mounted in a virtual floppy drive.
E. The Cisco Music on Hold USB audio sound card can be mapped to a virtual serial port on a VMware virtual machine on the UCS server.

Correct Answer: D

Explanation/Reference:
Installation media support:
A. Both the UC on UCS TRC and the third-party servers specs-based support models have rule-based approaches.
B. The UC on UCS TRC support model has a rule-based approach and the third-party servers specs-based support model has a configuration-based approach.
C. The UC on UCS TRC support model requires a high level of virtualization experience while the third-party server specs-based support model requires a low to medium level virtualization experience.
D. VMware vCenter is mandatory for the UC on UCS TRC support model but it is optional for the third-party server spec-based support model.
E. VMware vCenter is optional for the UC on UCS TRC support model but it is mandatory for the third-party server spec-based support model.

Correct Answer: D

Explanation/Reference:
VMware vCenter is optional when deploying on UCS TRC. Reference: Configuration hardware mandatory when deploying on UCS TRC Specs-based and Third-party Server Specs-based hardware. vCenter Statistics Level 4 logging is mandatory so that Cisco TAC is able to provide effective support.
Click here for how to configure VMware vCenter to capture these logs. If not configured by default, Cisco TAC may request enabling these settings in order to provide effective support. Also note that configuration of specific VMware vSphere management features may require vCenter and/or a higher feature Edition of vSphere ESXi. Cisco Collaboration does not require its own dedicated vCenter.
Note that when VMware vCenter is not required and is not used, then VMware vSphere ESXi’s default management interface is its free/included VMware vSphere Client (formerly branded VI Client).
Using the AFG will allow you to get this license mac before even touching the server. It is provided after filling in the main form of the AFG but it can also be found by looking at the last few lines of your platformconfig.xml file.

Once you have the xml files, you will need to map those to the floppy drive of the VM (no usb support on the VM OVA). There are many ways to do this. I simply use a freeware virtual floppy app that I drop the platformconfig.xml file on and then copy the *.flp image out to the datastore. I'll end up with a directory on my datastore called AFG that has the host named *.flp images that I will use during install. It also serves as archival of these files in the event the server needs to be re-imaged. This is important because the license mac will change if every parameter is not entered exactly as it was prior. If the license mac changes, you will have to go through the process of requesting new license files to be generated.