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CCIE Collaboration (v2.0)

Version: 23.0

[Total Questions: 705]

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Exam Topic Breakdown

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Topic 1, All New Questions

Question #:1 - (Exam Topic 1)

In Which scenario can you host the most instances of a server?

A. Using Only Virtual Machines

B. Using Micro Services Application

C. Using Virtual machines in containers

D. Using Only containers

E. Using containers in virtual machines

Answer: D

Question #:2 - (Exam Topic 1)

How many RTP streams exist on a network when a Cisco Unified Contact Centre Express agent is engaged in a call that is being silently monitored and recorded? (Choose two.)

A. 8

B. 3

C. 5

D. 6

E. 4

Answer: D

Question #:3 - (Exam Topic 1)

Which statement about what happens to a hunt member who does not answer queuing enabled hunt list call in Cisco Unified Communications Manger 9.1 is true?

A. The hunt member remains logged in if Automatically Logout Hunt Member on No Answer is not selected in the Hunt Pilot configuration page.

B. The hunt member is logged off if Automatically Logout Hunt Member on No Answer is selected on the Line Group Configuration page.

C. The hunt member is logged off automatically and must manually reset the phone to log back in.

D. The hunt member remains logged in if Automatically Logout Hunt Member on No Answer is not selected in Cisco Unified Communications Manger Service Parameters.

E. The hunt member is logged off automatically and guest press HLOG to log back in.

Answer: B

Question #:4 - (Exam Topic 1)

Refer to the exhibit.

```
[ContactServiceEdgeHandlerLogger] [ContactServiceEdgeHandler::edgeIsActive] -
[csf.httpclient] [http::CurlHttpUtils::logOperationTiming] - Network IO timestamps: [name
lookup = 0 ; connect = 0 ; ssl connect = 0 ; pre-transfer = 0 ; start-transfer = 0 ;
total = 0.203 ; redirect = 0]
[csf.httpclient] [http::CurlAnswerEvaluator::curlCodeToResult] - curlCode=[7] error
message=[Failed to connect to 172.16.100.51 port 7080: Connection refused]
result=[HOST_UNREACHABLE ERROR] fips enabled=[false]
[csf.httpclient] [http::executeImpl] - *-----* HTTP response from:
http://voicemailserver:7080/vmevents/cometd/handshake [18] -> 0.
[csf.httpclient] [http::executeImpl] - There was an issue performing the call to
curl_easy_perform: HOST_UNREACHABLE_ERROR
[csf.httpclient] [http::HttpRequestData::returnEasyCURLConnection] - Returning borrowed
EasyCURLConnection from request : 18
[csf.edge.capability.EdgeAccessDirector] [edge::EdgeAccessDirector::getInstance] -
Registering this as a DefaultPoliciesStore observer
[csf.voicemail] [NotificationClient::sendRequest] - [this: 0D06A400] Http operation error
4 , HOST_UNREACHABLE_ERROR
```

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.

Answer: C E

Question #:5 - (Exam Topic 1)

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. Consolidate the exported certificates and reimport into each cluster.

B. Associate a user device profile for the user in the remote cluster.

C. Enable the Allow Proxy service parameter on both clusters.

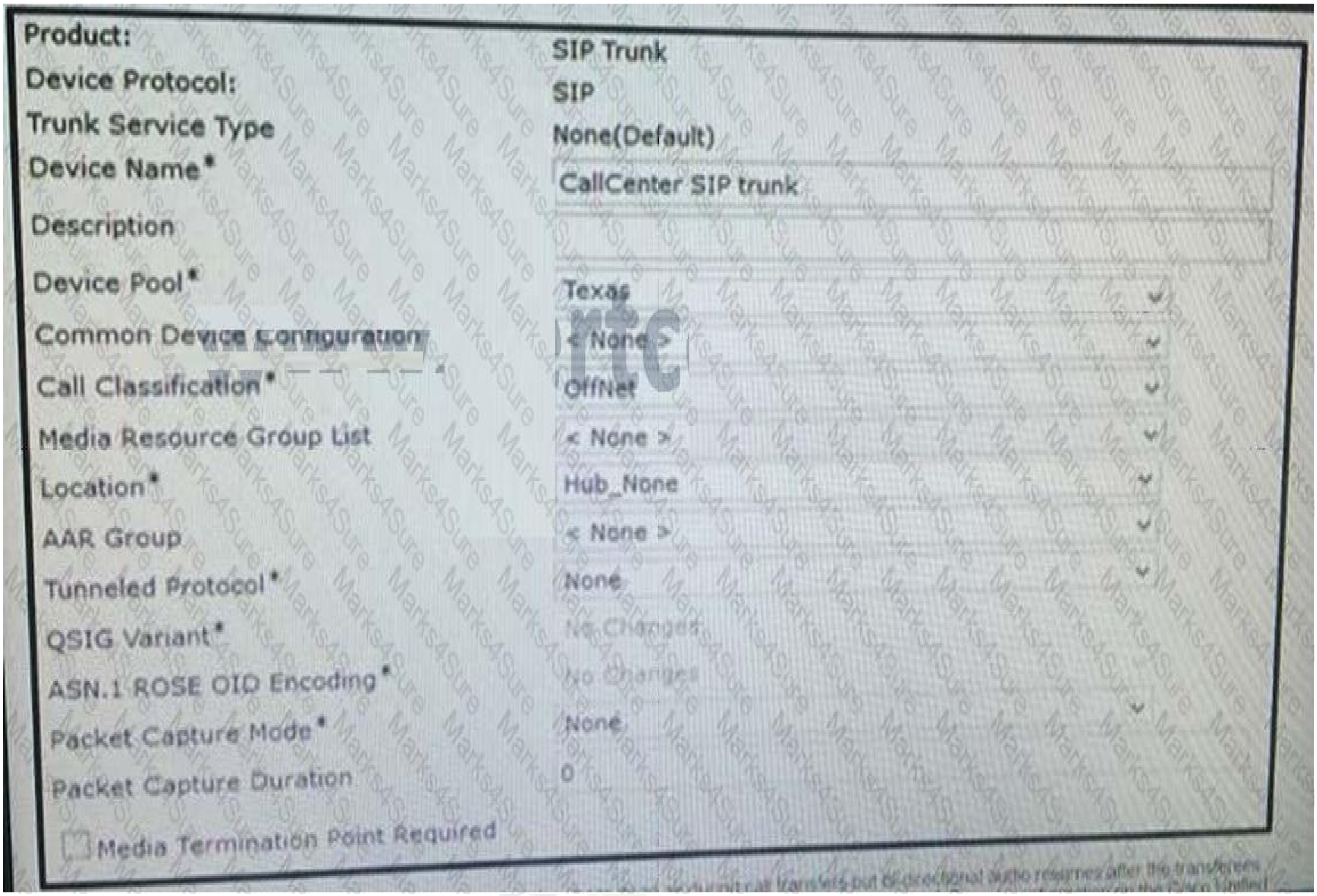
D. Update the Cluster IDs so that they are unique in the EMCC network.

E. Restart the Cisco CallManager and Cisco Tomcat Services

Answer: A E

Question #:6 - (Exam Topic 1)

Refer to the exhibit.



Cisco Unified CM users report that they hear dead air during call transfer but bi- directional audio resumes after the transferees answer the call. The transferees are located across a SIP trunk. A collaboration engineer is checking the SIP trunk configuration on the Cisco Unified CM Which Two configuration changes fix this problem? (Choose two)

A. Assign a Media Resource Group List to the SIP Trunk

B. Place a check mark on Media Termination Point Required

C. Make sure there is an Annunciator Resource available on the MRGL

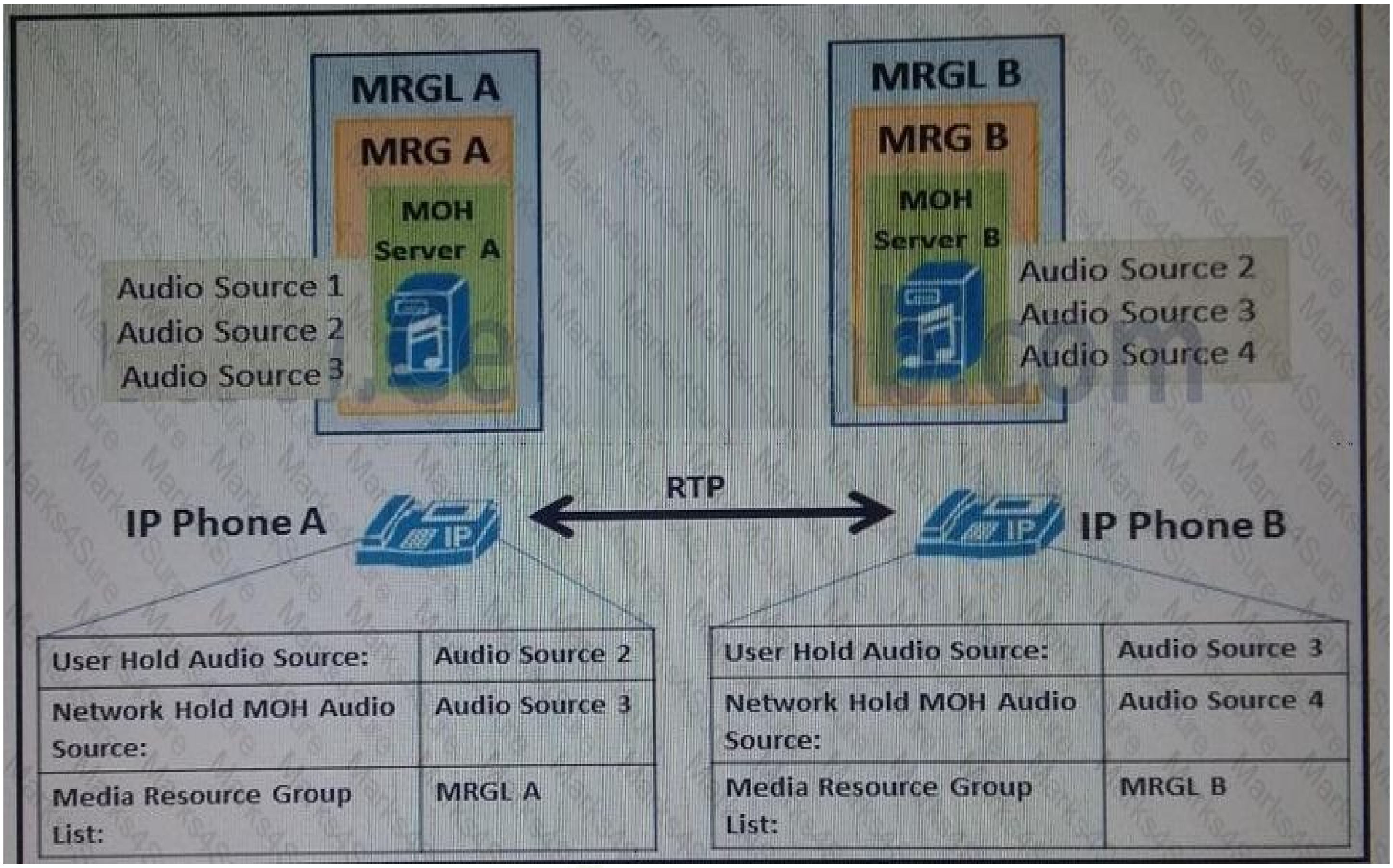
D. Modify the call Classification on the SIP trunk to OnNet

E. Change the "Send H225 User Info" service parameter to "Use ANN for Ringback"

Answer: A B

Question #:7 - (Exam Topic 1)

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 cluster wide, what will happen when the user of IP Phone A presses the Hold softkey?

A. The IP Phone B user hears audio source 2 from MOH Server A.

B. The IP Phone B user hears audio source 2 from MOH Server B.

C. The IP Phone B user hears audio source 3 from MOH Server A.

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.

Answer: D

Question #:8 - (Exam Topic 1)

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

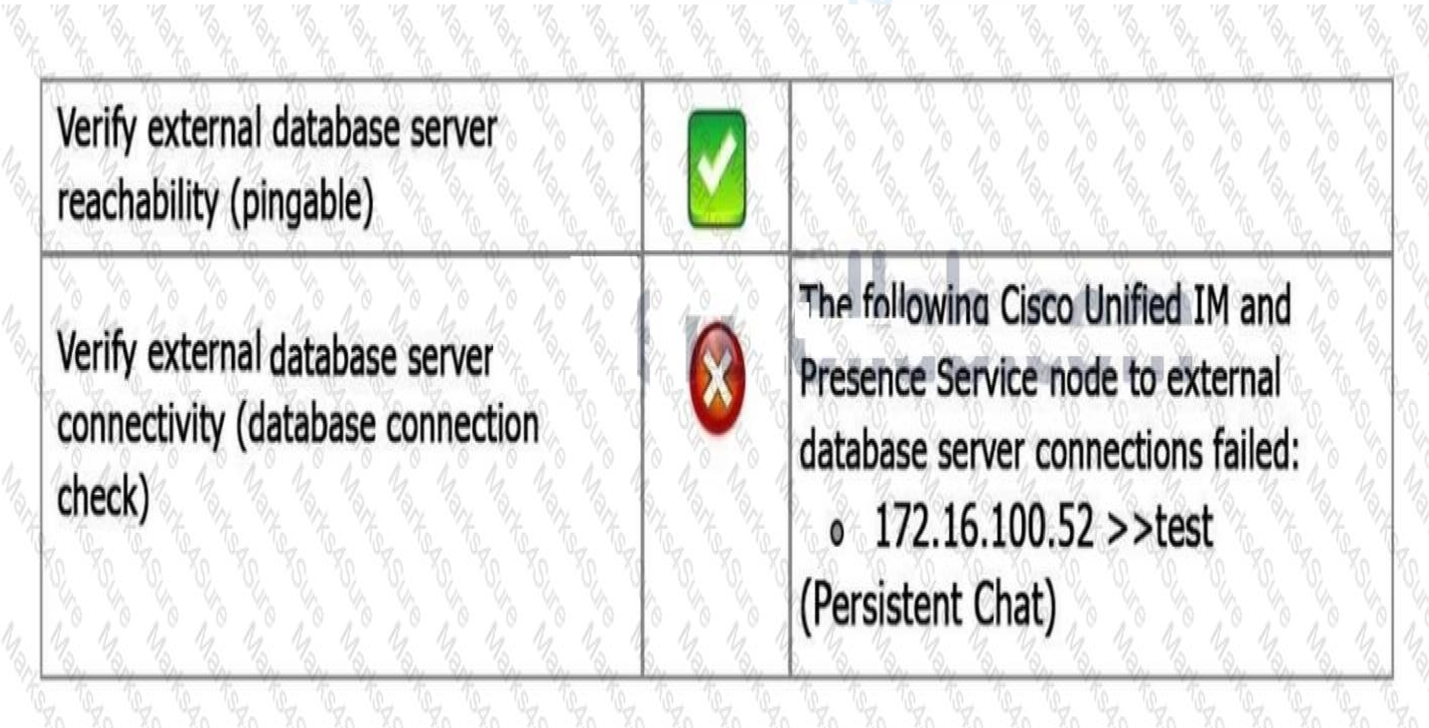
C. police 70000000 13125000 confirm-action transmit exceed-action drop

D. traffic-shape rate 70000000 8750000 8750000

Answer: D

Question #:9 - (Exam Topic 1)

Refer to the exhibit.



Verify external database server reachability (pingable)	✓	
Verify external database server connectivity (database connection check)	✗	The following Cisco Unified IM and Presence Service node to external database server connections failed: o 172.16.100.52 >>test (Persistent Chat)

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 Data store 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.

Answer: A E

Question #:10 - (Exam Topic 1)

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 Postgress 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.

Answer: B C D G

Question #:11 - (Exam Topic 1)

A collaboration engineer has been asked to implement secure real-time protocol between a Cisco

Unified CM and its SIP gateway. Which option is a consideration for this implementation?

A. only T.38 and Cisco fax protocol are supported

B. SIP require the all time be sent in GMT

C. Call hold RE-INVITE is not supported

D. SRTP is supported only in cisco IOS 15.x and higher

Answer: B

Question #:12 - (Exam Topic 1)

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. terminating side of a call back

C. pickup notification

D. hold reversion

E. remotely placed pickup request by a shared line

Answer: D

Question #:13 - (Exam Topic 1)

Which two actions does the cisco Unified IP phone use the initial Trust list to perform? (Choose two)

A. Decrypt secure XML files

B. Encrypt RTP traffic for IP phone that are not register to the same call manager cluster

C. Download background image files

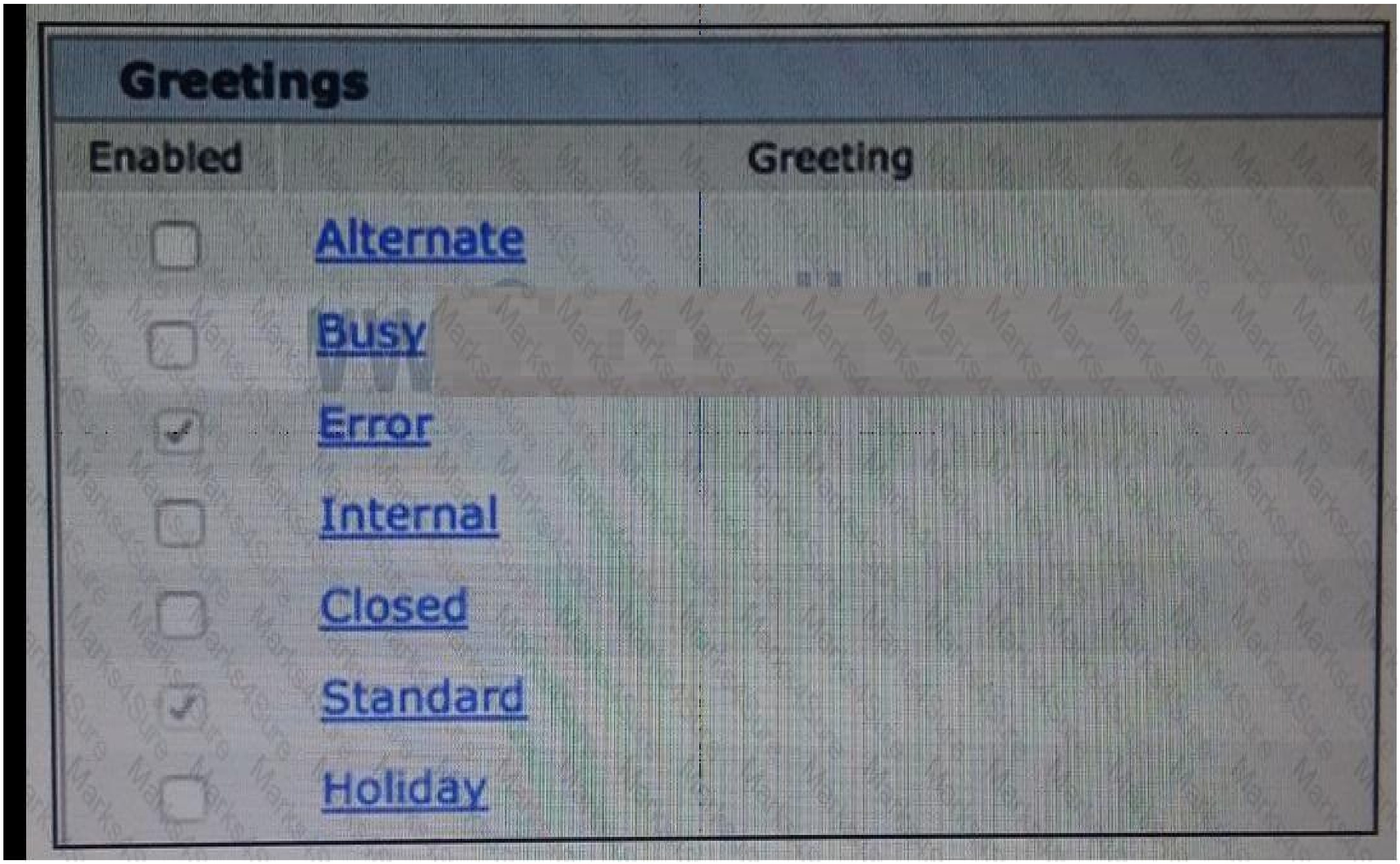
D. Authenticate their configuration file signature

E. Talk securely to CAPF which is a prerequisite to support configuration files encryption

Answer: D E

Question #:14 - (Exam Topic 1)

Exhibit:



What setting is required to play personal recordings during standard, closed and holiday hours?

A. Enable standard, alternate and error greetings

B. Enable holiday, closed and standard greetings

C. Enable holiday, closed, error and standard greetings

D. Enable standard and error greetings

E. Enable holiday, closed, standard, alternate and error greetings

Answer: C

Question #:15 - (Exam Topic 1)

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

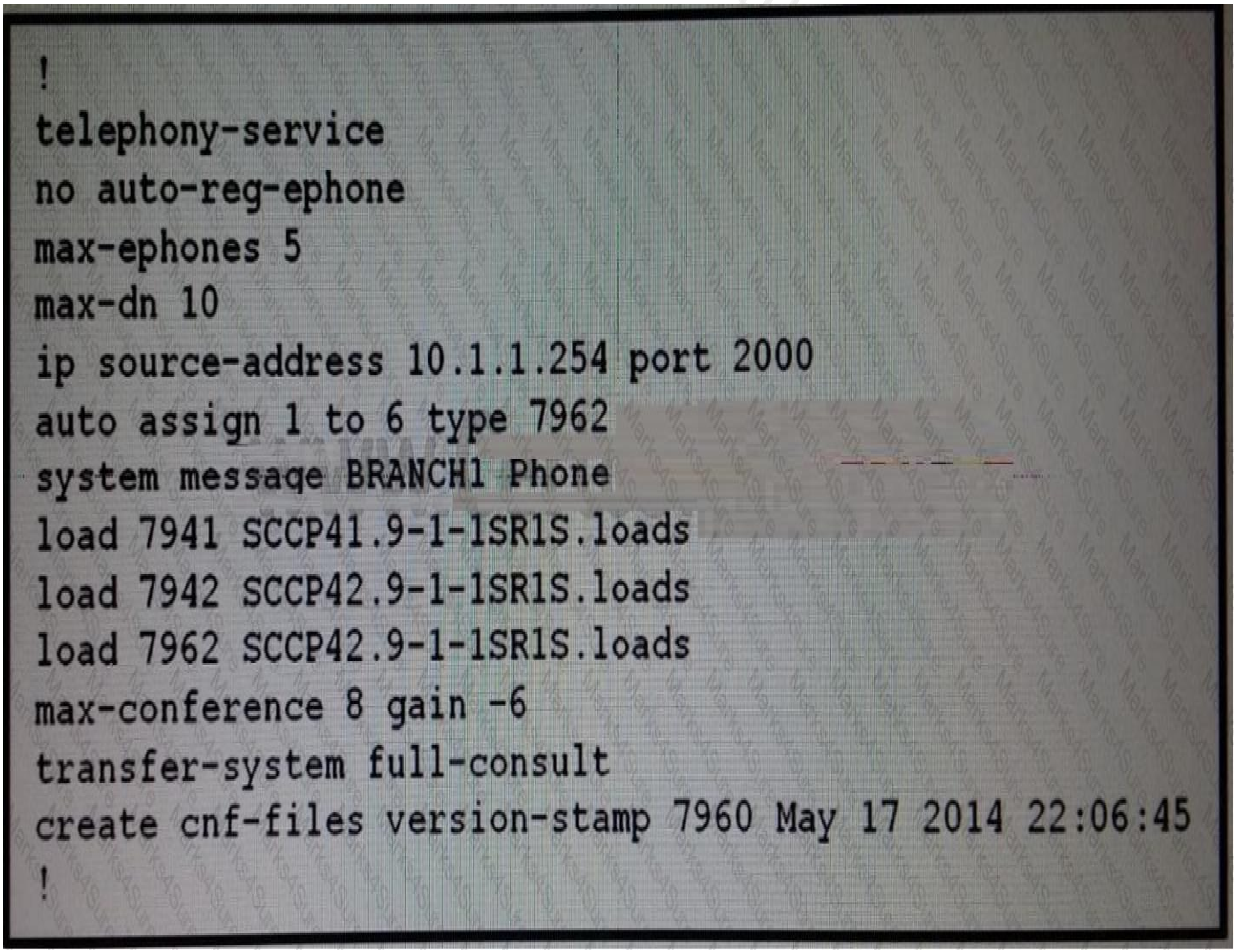
E. Domain Name

F. Certificate Information

Answer: A B

Question #:16 - (Exam Topic 1)

Refer to the exhibit.



```
!
telephony-service
no auto-reg-ephone
max-ephones 5
max-dn 10
ip source-address 10.1.1.254 port 2000
auto assign 1 to 6 type 7962
system message BRANCH1 Phone
load 7941 SCCP41.9-1-1SR1S.loads
load 7942 SCCP42.9-1-1SR1S.loads
load 7962 SCCP42.9-1-1SR1S.loads
max-conference 8 gain -6
transfer-system full-consult
create cnf-files version-stamp 7960 May 17 2014 22:06:45
!
```

Cisco VoIP administrator is configuring CUE VoiceView Express for end users and they are not able to manage their voice message from the Cisco IP phone. Which two configuration commands are required? (Choose two).

A. url

information http://10.10.1.1/information/info.html

B. url

services http://10.10.1.1/voiceview/common/login.do

C. url

authentication http://10.10.1.1/CCMCIP/authenticate.asp

D. url messages http;//10.10.1.1/messages/common/login.do

E. url

services http://10.10.1.1/CMEUser/123456/urlsupport.html

Answer: B C

Question #:17 - (Exam Topic 1)

Switch# show mls qos interface fastEthernet 0/1

FastEthernet 0/1

trust state: not trusted

trust mode: not trusted

COS override: dis

default COS: 0

DSCP Mutation Map: Default DSCP Mutation Map

Trust Device: None

Refer to the exhibit. A cisco VOIP engineer is configuring QoS for the company switches. The configuration must be based on two requirements:

- The switch port must trust all traffic coming from the IP Phone.

- The switch port must remap all traffic coming from the computer to COS 0

Which two sets of commands satisfy these requirements? (choose two)

A. FastEthernet 0/1

mls qos trust cos

B. FastEthernet 0/1

switchport priority extend cos 0

C. FastEthernet 0/1

switchport priority extend trust

D. FastEthernet 0/1

mls qos cos override

E. FastEthernet 0/1

mls qos cos 0

F. FastEthernet 0/1

mls qos trust device cisco-phone

Answer: A B

Question #:18 - (Exam Topic 1)

A SIP carried delivers DIDs to a Cisco Unified Border Element in the form of +155567810XX, 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\(.*\).*\(.*\)/ /\150\2/

B. rule 1 /+ 1555\(…\).\(…\)$/ /\15\2/

C. rule 1 /^\+ 1555\(678\)10\(..\)$/ /\150\2/

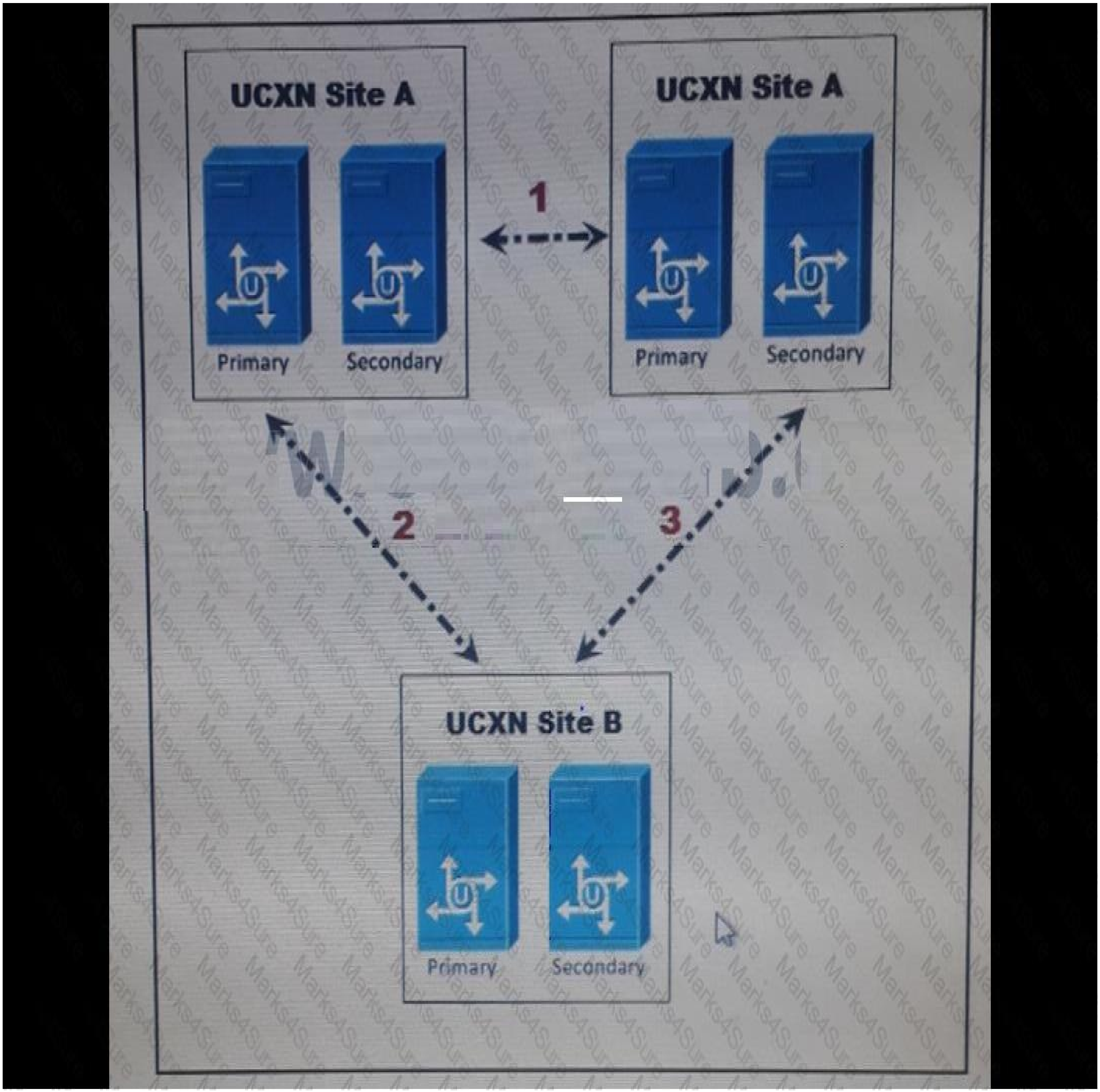
D. rule 1 /^15+678\(… .\)/678\1/

E. rule 1 /.15+678?10?\(..\)/ /67850\1/

Answer: C E

Question #:19 - (Exam Topic 1)

Refer to the exhibit.



Cisco unity connection site A has two locations and Cisco Unity connection Site B has one Location. Which protocol connect the location and servers together for messaging and replication?

A. 1 SMTP

2 - HTTP/HTTPS, SMTP

3 None

B. 1 HTTP/HTTPS, SMTP

2 SMTP

3 None

C. 1 - HTTP/HTTPS, SMTP

2 - HTTP/HTTPS, SMTP

3 - HTTP/HTTPS, SMTP

D. 1 SMTP

1 SMTP

1 SMTP

Answer: A

Question #:20 - (Exam Topic 1)

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 Enabled Flag and CDR Log Calls with Zero Duration Flag to True.

C. Set CDR Enabled Flag to True and set Call Diagnostics Enabled to Enable Only When CDR Enabled Flag is 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.

Answer: A

Question #:21 - (Exam Topic 1)

An IT team has decided to deploy Jabber for instant messaging and presence to utilize the existing Cisco Unified Communications infrastructure for on-premises Jabber deployment. Which feature can be used by

Jabber deployed in this model?

A. Instant message and chat for Jabber using WebEx messenger service

B. Softphone mode for Jabber using Cisco Unified Communications Manager

C. Visual voicemail for Jabber on Cisco unity

D. Audio and video conference for Jabber using WebEx meeting center

Answer: B

Question #:22 - (Exam Topic 1)

Refer to the exhibit.

```
Jun  1 17:59:16.839: ISDN Se0/0/0:15 Q931: RX <- SETUP pd = 8  callref = 0x17FF
        Bearer Capability i = 0x8090A3
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98392
                Exclusive, Channel 18
        Progress Ind i = 0x8283 - Origination address is non-ISDN
        Calling Party Number i = 0x2181, '2014582589'
                Plan:ISDN, Type:National
        Called Party Number i = 0xC1, '6727498'
                Plan:ISDN, Type:Subscriber(local)
        Sending Complete
Jun  1 17:59:16.847: ISDN Se0/0/0:15 Q931: TX -> CALL_PROC pd = 8  callref = 0x97
FF
        Channel ID i = 0xA98392
                Exclusive, Channel 18
PSTN_VG01#
```

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

Answer: B E

Question #:23 - (Exam Topic 1)

Which two characteristics should a collaboration engineer be aware of before enabling LATM on a Cisco Unified Border Element router? (Choose two.)

A. Box-To-Box High Availability Support feature is not supported.

B. Configure LATryl under a voice class or dial peer is not supported.

C. SIP UPDATE message outlined in RFC3311 is not supported.

D. Codec transcoding between LATM and other codecs is not supported

E. Dual tone multi-frequency interworking with LATM codec is not supported.

F. Basic calls using flow-around or flow-through is not supported.

Answer: D E

Question #:24 - (Exam Topic 1)

A user has reported that when trying to access Visual Voicemail the following error is received

“Unable to open application. Please try again later. If it continues to fail contact your administrator”. The collaboration engineer is working on the problem found on the following phone logs:

CVMInstallerModule STATUS_install_cancelled

STATUS_INSTALL)_ERROR [thread=installer MQThread][class=cip midp midletsuite installerModule][function=update status] Midlet install Canceled/ERROR...visual

Voicemail

How can this issue be resolved?

A. Replace the sever name with the server IP on service URL field

B. Eliminate the space in the service Name field

C. Configure DNS on phone configuration so it can resolve server name

D. Check the Enable checkbox on IP phone service configuration

Answer: B

Question #:25 - (Exam Topic 1)

A service provider wants to use a controller to automate the provisioning of service function chaining. Which two overlay technologies can be used with EVPN MP- BGP to create the service chains in the data centre?

(Choose two.)

A. VXLAN

B. MPLSoGRE

C. Provider Backbone Bridging EVPN

D. 802.1Q

E. MPLS L2VPN

Answer: A C

Question #:26 - (Exam Topic 1)

Refer to the exhibit.

Cluster Detailed View from PUB (3 Servers):

SERVER-NAME	IP ADDRESS	PING (msec)	RPC?	REPLICATION STATUS	REPL. QUEUE	DBver& TABLES	RPL LOOP?	REPL. REPLICATION SETUP (RTMT) & details
CUCMPUB	172.16.100.50	0.033	Connected	0	Match	Yes	(3)	PUB Setup Completed
CUCMSub1	172.16.100.51	0.855	Connected	0	Match	Yes	(4)	Setup Failed
CUCMSub2	172.16.100.52	1.025	Connected	0	Match	Yes	(4)	Setup Failed
CUCMSub3	172.16.100.53	3.250	Connected	0	Match	Yes	(4)	Setup Failed

Users on a four-node CUCM cluster are reporting call problems when attempting to call out to internal extension and PSTN. An engineer troubleshooting issue found a replication of the cluster is in status 4.

Which three steps will resolve the replication problem? (Choose three.)

A. run the command utils dbreplication dropadmindb on all subscribers

B. run the command utils dbreplication repairable all from the publisher

C. run the command utils dbreplication stop on the publisher

D. run the command utils dbreplication reset all from the publisher

E. run the command utils dbreplication repair all from the publisher

F. run the command utils dbreplication stop on all subscribers

Answer: C D F

Question #:27 - (Exam Topic 1)

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

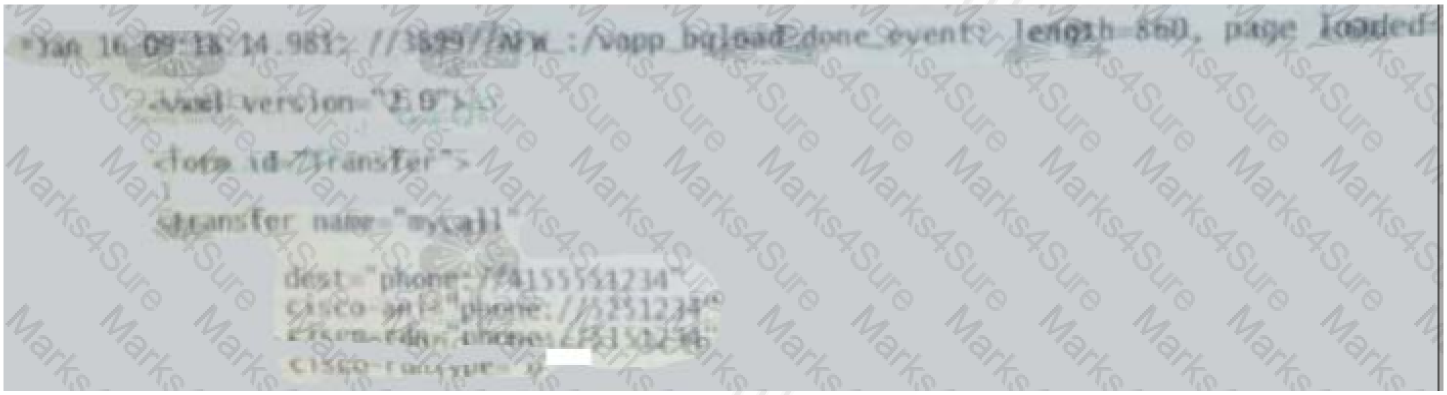
D. encrypted call signaling and media

E. Not enough information provided to answer this question.

Answer: D

Question #:28 - (Exam Topic 1)

Refer to the exhibit.



This VXML message was captured on a Cisco IOS H.323 gateway communication wit aCisco Unified Communication Manager server after a Cisco Unified Mobile Voice Access user authenticatedwith his pin number and entered the number be wanted to call. Which Cisco Unified Mobile Access Voice directory number is configured on the Cisco UCM?

A. The Cisco Unified Mobility Access Voice directory number is not shown in the exhibit

B. 4155551234

C. 5151234

D. 5251234

Answer: B

Question #:29 - (Exam Topic 1)

In the OpenStack, which two statements about the NOVA component are true? (choose two)

A. it launches virtual machine instances

B. it is considered the cloud computing fabric controller

C. it provides persistent block storage to running instances of virtual machines

D. it provides the authentication and authorization services

E. it tracks cloud usage statics for billing purposes

Answer: A B

Question #:30 - (Exam Topic 1)

Which statement describes the key security service that is provided by the TLS proxy function on a Cisco ASA appliance?

A. It enables internal phones to communicate with the external phones without encryption.

B. It only applies to encrypted voice calls where both parties utilize encryption,

C. It only provides internetworking to ensure that external IP phone traffic is encrypted, even if the rest of the system is unencrypted.

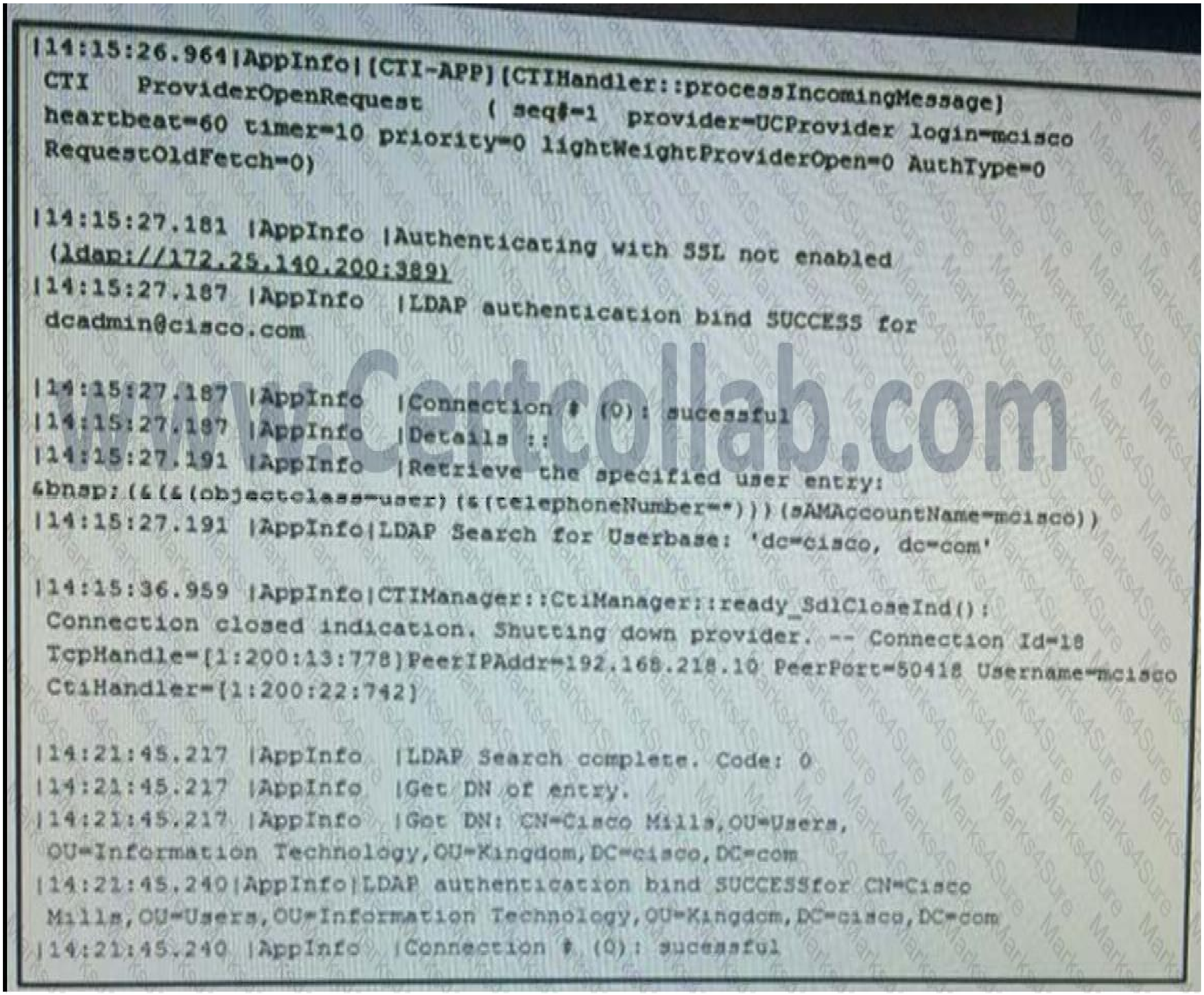
D. It protects Cisco Unified Communications Manager from rogue soft clients and attackers on the data VLAN

E. It manipulates the call signalling to ensure that all media is routed via the adaptive security appliance.

Answer: E

Question #:31 - (Exam Topic 1)

Refer to the exhibit.



```
|14:15:26.964|AppInfo|[CTI-APP][CTIHandler::processIncomingMessage]
 CTI   ProviderOpenRequest     ( seq#=1  provider=UCProvider login=mcisco
 heartbeat=60 timer=10 priority=0 lightWeightProviderOpen=0 AuthType=0
 RequestOldFetch=0)

|14:15:27.181 |AppInfo |Authenticating with SSL not enabled
 (ldap://172.25.140.200:389)
|14:15:27.187 |AppInfo  |LDAP authentication bind SUCCESS for
 dcadmin@cisco.com

|14:15:27.187 |AppInfo  |Connection # (0): sucessful
|14:15:27.187 |AppInfo  |Details ::
|14:15:27.191 |AppInfo  |Retrieve the specified user entry:
&bnsp:(&(&(objectclass=user)(&(telephoneNumber=*)))(sAMAccountName=mcisco))
|14:15:27.191 |AppInfo|LDAP Search for Userbase: 'dc=cisco, dc=com'

|14:15:36.959 |AppInfo|CTIManager::CtiManager::ready_SdlCloseInd():
 Connection closed indication. Shutting down provider. -- Connection Id=18
 TcpHandle=[1:200:13:778]PeerIPAddr=192.168.218.10 PeerPort=50418 Username=mcisco
 CtiHandler=[1:200:22:742]

|14:21:45.217 |AppInfo  |LDAP Search complete. Code: 0
|14:21:45.217 |AppInfo  |Get DN of entry.
|14:21:45.217 |AppInfo  |Got DN: CN=Cisco Mills,OU=Users,
 OU=Information Technology,OU=Kingdom,DC=cisco,DC=com
|14:21:45.240|AppInfo|LDAP authentication bind SUCCESSfor CN=Cisco
 Mills,OU=Users,OU=Information Technology,OU=Kingdom,DC=cisco,DC=com
|14:21:45.240 |AppInfo  |Connection # (0): sucessful
```

A jabber user reports that desktop mode does not work. Connection status shows the following message

"Connection error. Ensure the server information in the phone services tab on the option window is correct. contact your system administrator for assistance."

Which two actions will resolve the problem? (Choose two)

A. Change LDAP port from 389 to 636 in LDAP directory configuration

B. Restart DirSync service under unified serviceability

C. Change LDAP port from 389 to 443 in LDAP director configuration

D. Change LDAP port from 389 to 3268 in LDAP director configuration

E. Restart CTI manager service under unified serviceability

F. Restart TFTP service under Unified serviceability

Answer: D E

Question #:32 - (Exam Topic 1)

Which description of route list digit manipulation behavior in Cisco Unified Communication manager is true?

A. Called party transformation at route list level is not shown on the display of the calling phone

B. Called party transformations at route list level is reflected on the display of the calling phone

C. Digit manipulation occur once per route list

D. Only called transformation is available at route list level

E. Called party transformations at route list level is replaced by called party transformations at route pattern level

Answer: E

Question #:33 - (Exam Topic 1)

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.

Answer: A B

Explanation

SCCP: Configuring a Virtual SNR DN

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, 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.

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmesnr.html#

Question #:34 - (Exam Topic 1)

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.

Answer: E

Question #:35 - (Exam Topic 1)

A collaboration engineer is configuring Cisco Unified Mobility and found that when mobile phone hangs up, the calling party gets reorder tone and resume softkey does not show on deskphone. If calling party waits for 45 seconds, the resume softkey appears and call can be connected to the deskphone. Which two configuration modifications resolve this issue? (Choose two)

A. Configure voice call send-alert on voice gateway

B. Configure voice call convert-discpi-to-prog command on the gateway

C. Set the Retain media on disconnect with PI for active call UCM service parameter to False

D. Configure voice call disc-pi-off command on the gateway

E. Set the call screening timer UCM service parameter to a higher value

F. Set the Send call to Mobile Menu Timer UCM service parameter to a lower value

Answer: C D

Question #:36 - (Exam Topic 1)

An Engineer is configuring a CUBE interoperability with a SIP Service Provider. What are the three different ways Mid Call Re - Invites function to ensure smooth interoperability of supplementary services? (Choose three.)

A. provides early offer to delay offer codec change in 200 OK message

B. provides support for media flow around in early offer forced call flows

C. converts a delayed offer to an early offer

D. allows interoperability for video related features

E. allows pass through of mid- call signalling on media change

F. blocks all mid- call signalling for specific SIP trunk

Answer: D E F

Question #:37 - (Exam Topic 1)

Exhibit:

```
INVITE sip:5124182222@172.16.100.90:5060 SIP/2.0
Via: SIP/2.0/TCP 172.16.100.50:5060;branch=z9hG4bK4e561fd2b0b
From: <sip:4051@172.16.100.50>;tag=1251~9dd03ed8-9382-4ac1-b8a1-b7247e0f115a-31897011
To: <sip:5124182222@172.16.100.90>
Call-ID: 25bb600-57a14b05-4ce-326410ac@172.16.100.50
Min-SE:  1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 15
```

A network engineer is testing a new SIP deployment and sees this message. Which two situations cause a cancel message?

A. The called device did not have call forward setting

B. SIP session timer header is too small

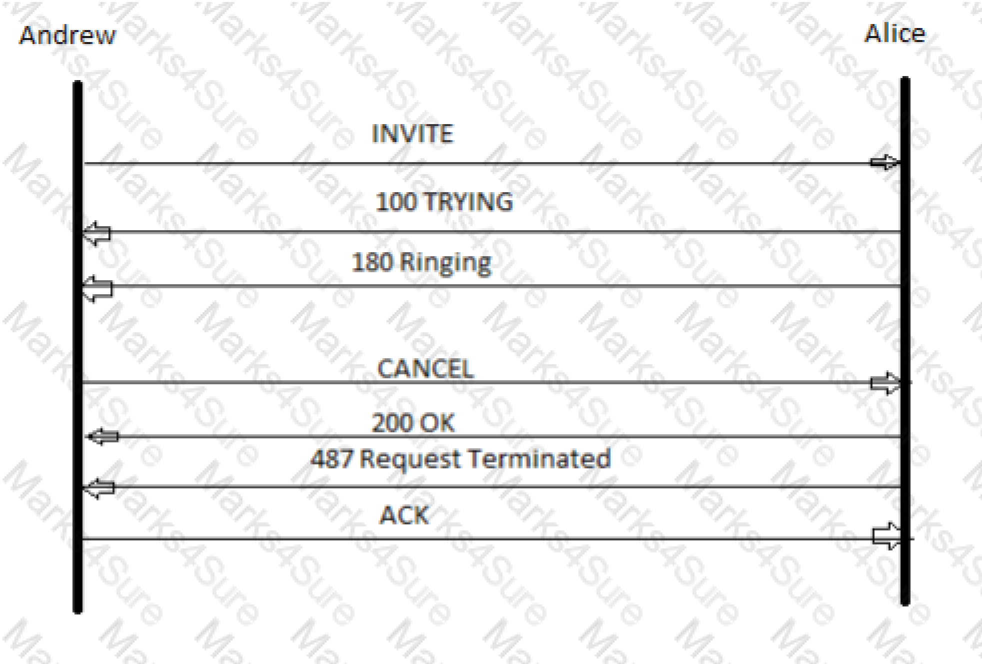
C. SIP Min-SE timer header is set to small

D. SIP Refresher header is set to UAS instead of UAC

E. Calling user terminated the call

Answer: A E

Question #:38 - (Exam Topic 1)



Refer to the exhibit. How many SIP transactions are involved in this call flow?

A. 1

B. 2

C. 3

D. 4

E. 5

Answer: B

Question #:39 - (Exam Topic 1)

O: 104

Which two types of patterns are always nonurgent in Cisco Unified Communications Manager version 11.1? (choose two)

A. Voice Mail Directory Number

B. Route Pattern

C. Translation Pattern

D. Remote Destination Directory Number

E. Hunt Pilot

F. Voice Mail Pilot

Answer: A F

Question #:40 - (Exam Topic 1)

Which three features work over MRA in a environment Cisco Unified Communications Manager IM & P 10.5 Cisco Jabber 11.6 and Expressway X.8.7? (choose three)

A. peer-to-peer file transfer

B. Deskphone Control (CTI/QBE)

C. Directory Integration LDAP lookups

D. Cisco Unified Communication Manager User Data Service directory lookup

E. Instant Messaging

F. Managed File Transfer

G. Enhanced Directory Integration LDAP lookups

Answer: D E F

Question #:41 - (Exam Topic 1)

Refer to the exhibit.

The SIP service Provider does not support re-invites unless media changes. Which two commands are needed on the CUBE to standby the SIP service provider requirements? (choose two)

A. media flow-through

B. midcall signaling preserve-codec

C. media flow-around

D. midcall-signaling passthru media-change

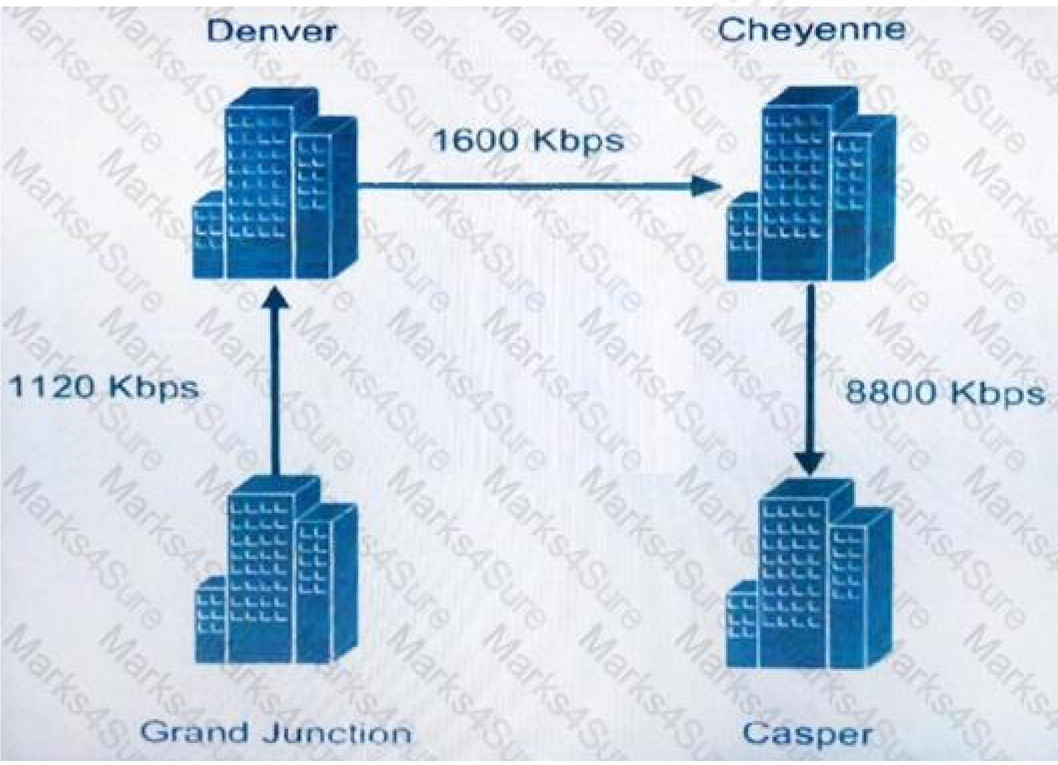
E. media anti-trombone

F. midcall-signaling block

Answer: D F

Question #:42 - (Exam Topic 1)

Refer to the exhibit.



A collaboration engineer configures Cisco Unified CM location using G.711 and iLBC for each site. The bandwidth for each link is shown. Which two options represent the maximum concurrent number of calls supported by grand junction to Casper for each Codec? (Choose two.)

A. 20 G.711 calls

B. 18 G.711 calls

C. 36 iLBC calls

D. 42 iLBC calls

E. 11 G.711 calls

F. 51 iLBC calls

Answer: C E

Question #:43 - (Exam Topic 1)

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

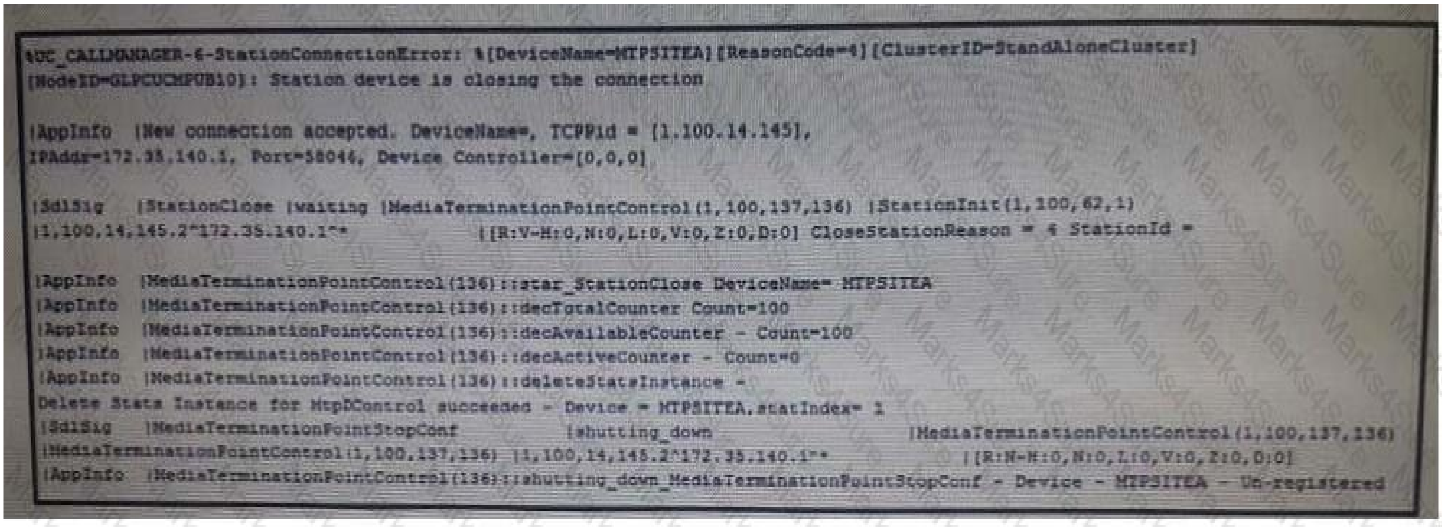
D. RSIP

E. AUCX

Answer: C

Question #:44 - (Exam Topic 1)

Refer to the exhibit.



```
%UC_CALLMANAGER-6-StationConnectionError: %[DeviceName=MTPSITEA][ReasonCode=4][ClusterID=StandAloneCluster]
[NodeID=GLPCUCMPUB10]: Station device is closing the connection

|AppInfo  |New connection accepted. DeviceName=, TCPPid = [1.100.14.145],
IPAddr=172.35.140.1, Port=58046, Device Controller=[0,0,0]

|SdlSig   |StationClose |waiting |MediaTerminationPointControl(1,100,137,136) |StationInit(1,100,62,1)
|1,100,14,145.2^172.35.140.1^*          |[R:V-H:0,N:0,L:0,V:0,Z:0,D:0] CloseStationReason = 4 StationId =

|AppInfo  |MediaTerminationPointControl(136)::star_StationClose DeviceName= MTPSITEA
|AppInfo  |MediaTerminationPointControl(136)::decTotalCounter Count=100
|AppInfo  |MediaTerminationPointControl(136)::decAvailableCounter - Count=100
|AppInfo  |MediaTerminationPointControl(136)::decActiveCounter - Count=0
|AppInfo  |MediaTerminationPointControl(136)::deleteStateInstance -
Delete State Instance for MtpDControl succeeded - Device = MTPSITEA.statIndex= 1
|SdlSig   |MediaTerminationPointStopConf        |shutting_down        |MediaTerminationPointControl(1,100,137,136)
|MediaTerminationPointControl(1,100,137,136) |1,100,14,145.2^172.35.140.1^*      |[R:N-H:0,N:0,L:0,V:0,Z:0,D:0]
|AppInfo  |MediaTerminationPointControl(136)::shutting_down_MediaTerminationPointStopConf - Device - MTPSITEA - Un-registered
```

A cisco collaboration engineer discovers that an instance of IOS media termination point (MTP) could not maintain stable registration with CUCM. Call manager traces is showing in the exhibit. What is the reason for the flapping registration?

A. The CCM version on IOS configuration does not match the CUCM version.

B. The IOS MTP is experiencing high CPU and is missing its keep-alive.

C. A Firewall is blocking port 2000 intermittently between IOS Device and CUCM.

D. Another IOS Media device is attempting to register with the same name.

Answer: D

Question #:45 - (Exam Topic 1)

Which set of information is replicated when Global Dial Plan Replication is configured?

A. local and learned directory URIs, enterprise alternate numbers, +E.164 numbers, and number patterns throughout ILS network

B. local and learned directory URIs, route partitions, directory numbers, and calling search space

C. local and learned directory URIs, translation pattern, route patterns, and calling/called number

transformation patterns throughout the ILS network

D. local and learned directory URIs, translation patterns, +E.164 numbers and route patterns throughout ILS network

E. local and learned directory URIs, enterprise alternate numbers, +E.164 numbers, and SIP route patterns throughout ILS network

Answer: E

Question #:46 - (Exam Topic 1)

An engineer wants to configure a Cisco IOS router to allow for up to four simultaneous conferences in CME mode. Which configuration meets the requirement?

A. dspfram profile 1 conferenceCodec G711ulawMaximum sessions 4Maximum conferenceparticipants 8Telephony-serviceSdspfarm units 4Sdspfarm tag 4 hwcfbConference hardware

B. dspfram profile 1 conferenceCodec G711ulawMaximum conference-participants 8TelephonyserviceSdspfarm units 4Sdspfarm tag 4 hwcfbConference hardwareMax-conference-participants 8

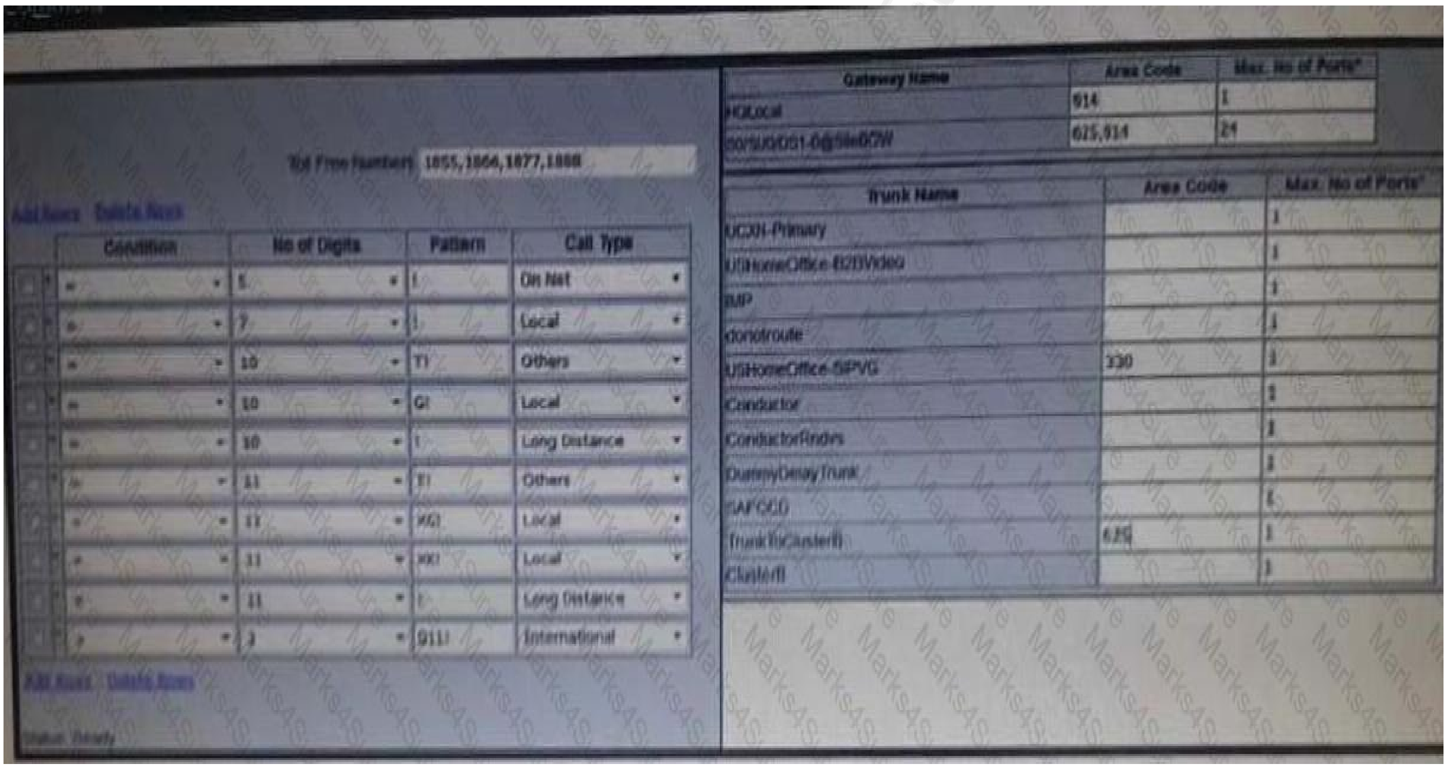
C. dspfram profile 1 conferenceCodec G711ulawMaximum sessions 4Maximum conferenceparticipants 8Telephony-serviceSdspfarm units 1Sdspfarm tag 1 hwcfbConference hardwareMaxconferences 2

D. dspfram profile 1 conferenceCodec G711ulawMaximum sessions 4Maximum conferenceparticipants 8Telephony-serviceSdspfarm units 1Sdspfarm tag 1 hwcfbConference hardwareMaxconferences 4

Answer: D

Question #:47 - (Exam Topic 1)

Refer to the exhibit.



A customer is configuring CAR costing for call. When the customer runs the costing reports calls are not being

tagged correctly. Which two changes allow proper costing to be determined for these calls? (Choose two)

A. The toll free area code field must be updated to include all toll free area codes

B. A new local pattern must be added with the pattern "k!"

C. A new pattern must be added for the 914 and 625 area codecs

D. The items are out of order and must be sorted with the most specific at the top

E. Overlapping area codec on the trunks must be removed

F. All external patterns must be change to include the outside access code

Answer: A B

Explanation

Choose System > System Parameters > Dial Plan Configuration.

The Dial Plan Configuration window displays.

In the Toll Free Numbers field, enter the numbers in your dial plan that can be placed without a charge.

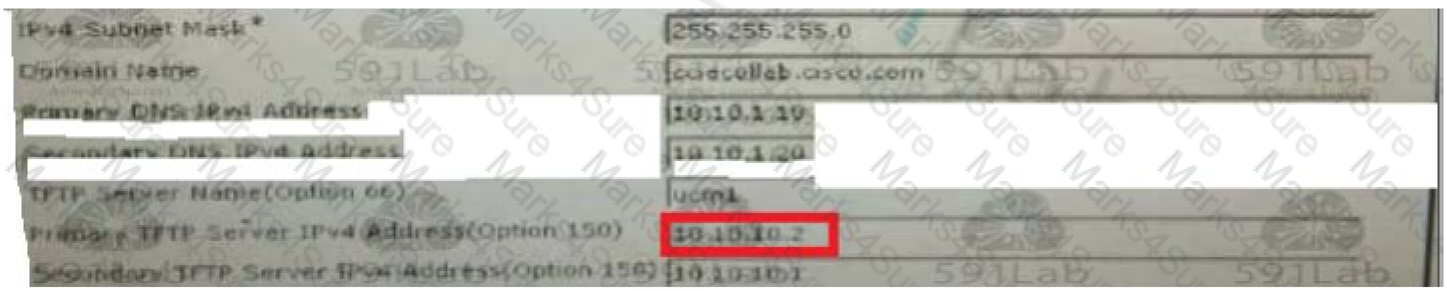
If the number of digits dialed equals 10 and the pattern is K! (more than one digit, in this case a 10-digit number that starts with a trunk code),

the call gets classified as Local.

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/service/9_0/car/CUCM_BK_CB39F074_00_cdr-ar

Question #:48 - (Exam Topic 1)

Refer to the exhibit.



Assume that an IP phone is using this DHCP server. Which TFTP server does the IP

phone select to request its configuration file?

A. ucm1

B. 10.10.1.10

C. 10.10.10.2

D. ucm1.cciecollab.cisco.com

Answer: C

Question #:49 - (Exam Topic 1)

What is the minimum number of TCP sessions needed to complete a H323 call between two H323 gateways using slow start?

A. 0

B. 3

C. 2

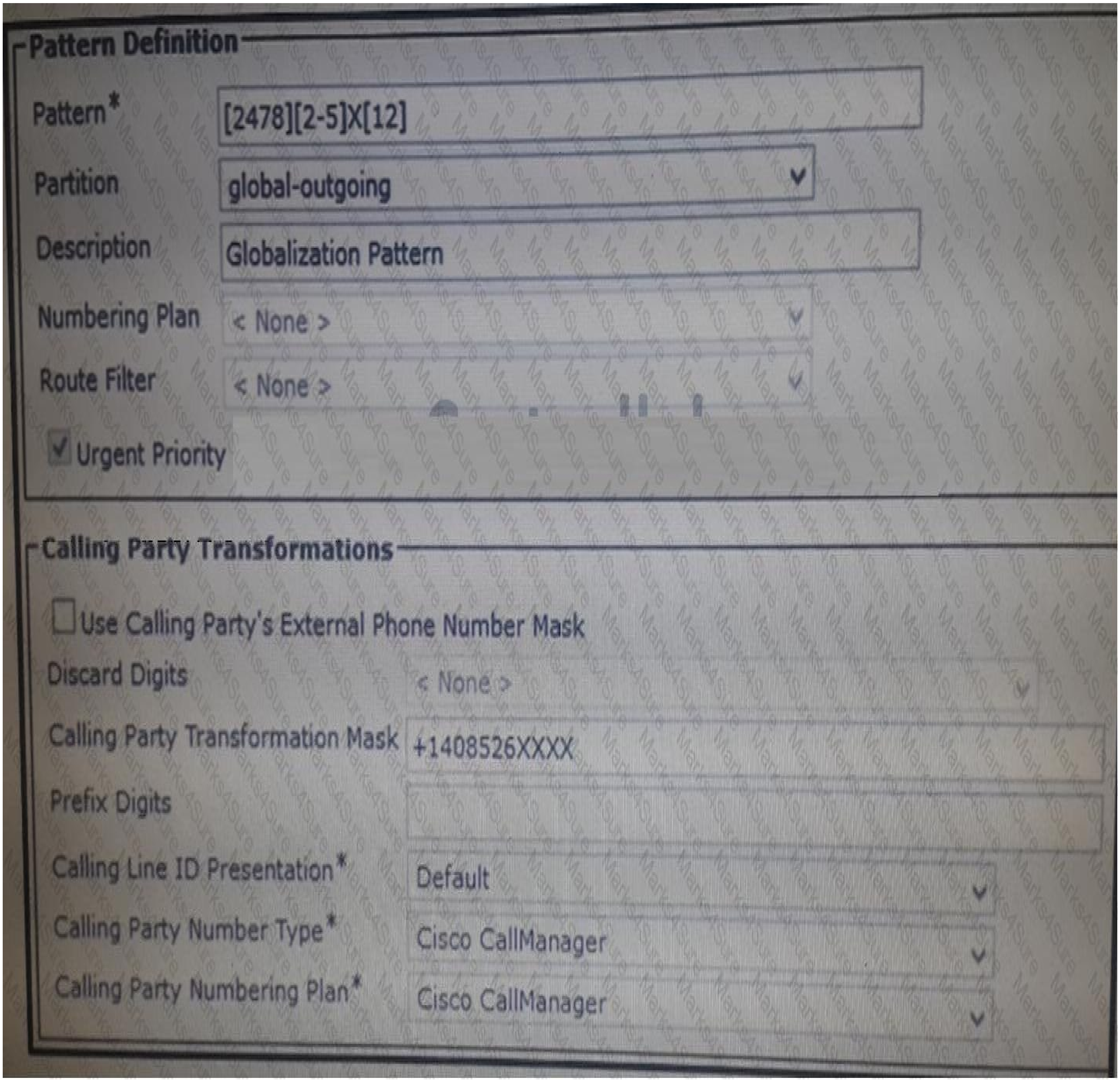
D. 4

E. 1

Answer: C

Question #:50 - (Exam Topic 1)

Refer to the Exhibit,



Which three cisco Unified CM internal extension match the globalization pattern shown and provide a globalized calling party number? (Choose three)

A. 2401

B. 2671

C. 3392

D. 4202

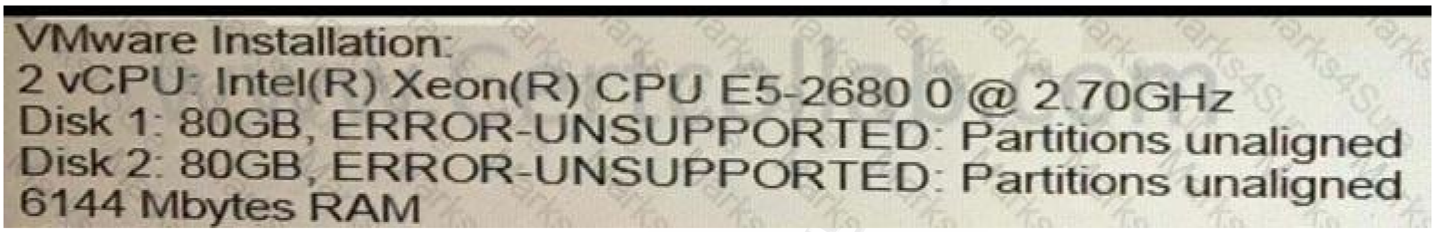
E. 7352

F. 8253

Answer: A D E

Question #:51 - (Exam Topic 1)

Refer to the exhibit.



VMware Installation:
2 vCPU: Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz
Disk 1: 80GB, ERROR-UNSUPPORTED: Partitions unaligned
Disk 2: 80GB, ERROR-UNSUPPORTED: Partitions unaligned
6144 Mbytes RAM

An IT engineer upgraded Cisco Unified Communications Manager to version 9.1.2. When accessing CLI of the server, this output is displayed.

Which three actions must be taken to correct this issue? (Choose Three)

A. From the recovery disk menu options, select option [F] to check and correct disk file system

B. Login to DRS and perform Cisco Unified CM restore from the backup

C. From the recovery disk menu option, select option [Q] to quit recovery program and reboot the virtual machine

D. Download Cisco unified CM recovery iso, boot the virtual machine from it and verify disk partitioning layout

E. Create a new virtual machine from Cisco ova template and create a fresh install with the Cisco Unified CM bootable iso

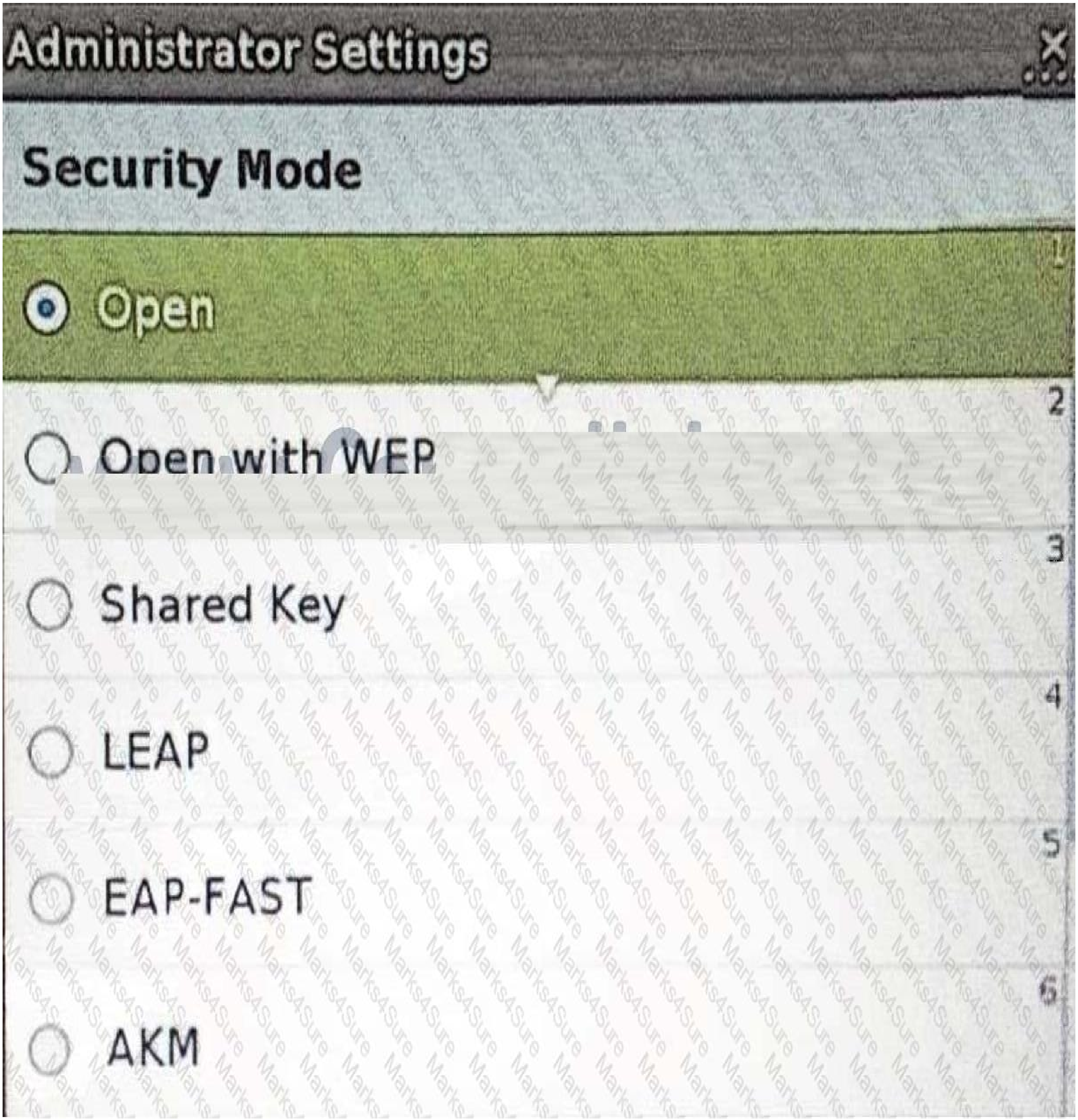
F. Take the backup of the system with disaster recovery system

G. From the recovery disk menu options, select option [A] to align the partitions of the virtual machines

Answer: B E F

Question #:52 - (Exam Topic 1)

Refer to the exhibit.



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

Answer: B E F

Question #:53 - (Exam Topic 1)

Which computing model does Fog use? (Choose two.)

A. Cluster

B. Distributed

C. Centralized

D. Grid

Answer: B

Question #:54 - (Exam Topic 1)

Refer to the exhibit.

```
Router#show dial-peer voice sum
dial-peer hunt 0
              AD                                          PRE PASS                     OUT
TAG     TYPE  MIN   OPER PREFIX    DEST-PATTERN           FER THRU SESS-TARGET         STAT PORT      KEE

4300    voip  up    up             4...                     0 syst ipv4:10.1.1.4                      act
2300    voip  up    up             [2-3]...                 0 syst ipv4:10.1.1.3                      act
1111    voip  up    down           1111                     0 syst ipv4:10.1.1.1                      bus
20001   pots  up    up             2001$                    0                             50/0/1
20002   pots  up    up             2002$                    0                             50/0/2
```

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

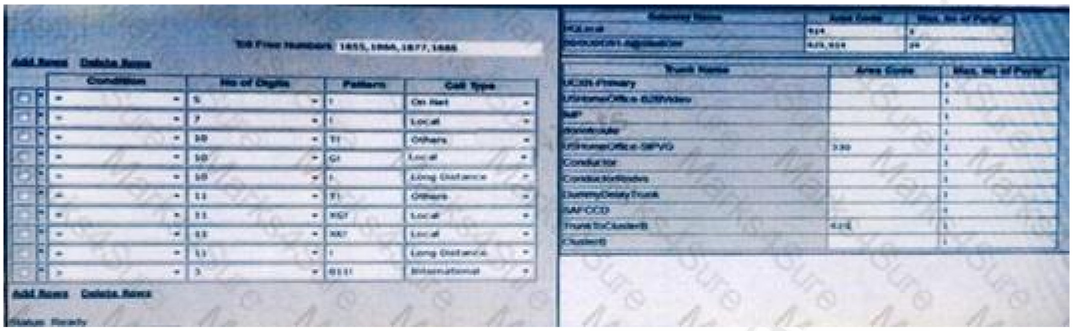
D. 482 Loop Detected

E. 500 Server Internal Error

Answer: B

Question #:55 - (Exam Topic 1)

Refer to the exhibit.



A customer is configuring CAR costing for call. When the customer runs the costing reports calls are not being tagged correctly. Which two changes allow proper costing to be determined for these calls? (Choose two)

A. The toll free area code field must be updated to include all toll free area codes

B. A new local pattern must be added with the pattern "k!"

C. A new pattern must be added for the 914 and 625 area codecs

D. The items are out of order and must be sorted with the most specific at the top

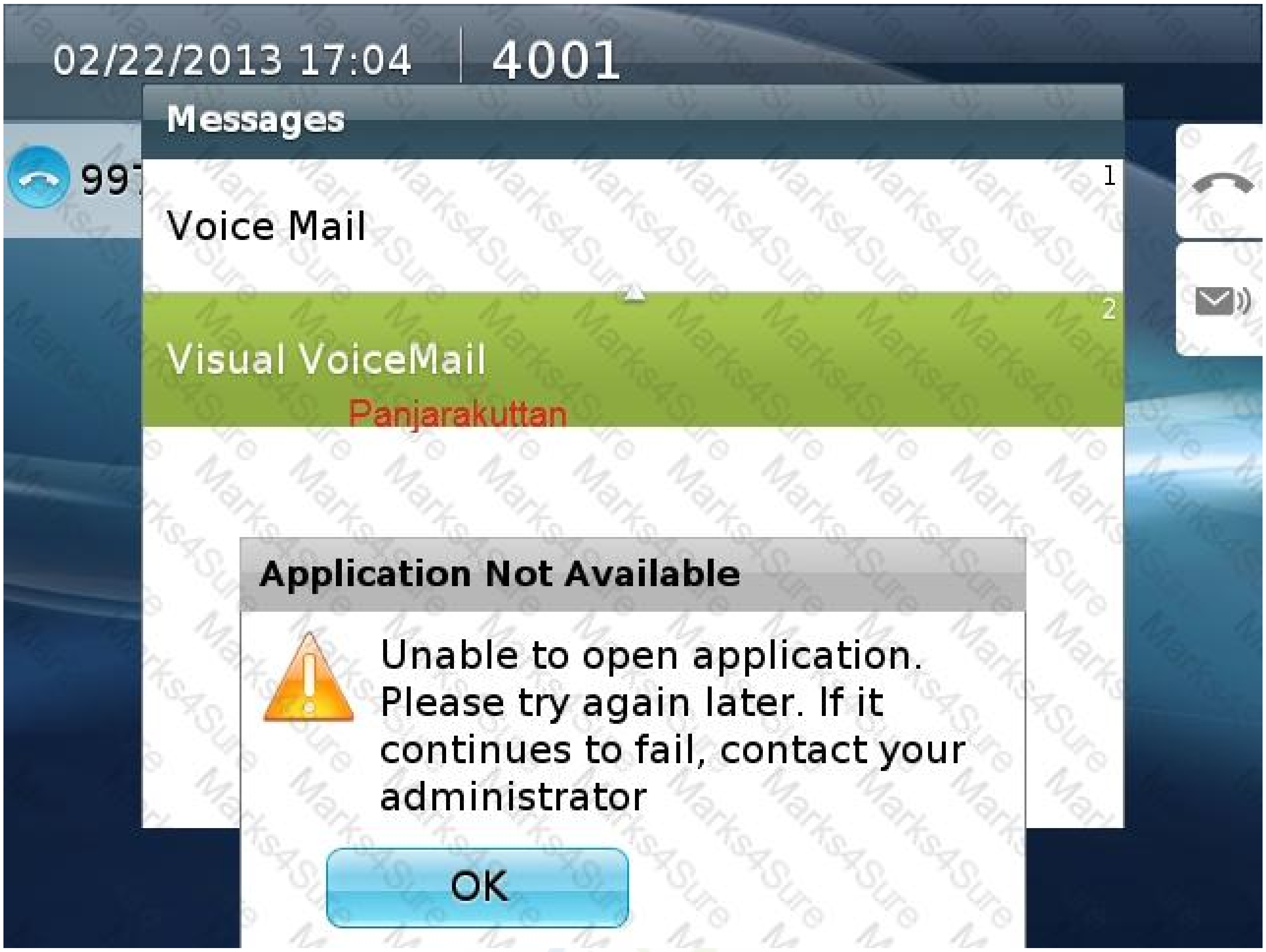
E. Overlapping area codec on the trunks must be removed

F. All external patterns must be change to include the outside access code

Answer: A B

Question #:56 - (Exam Topic 1)

Exhibit:



A user has reported that when trying to access Visual Voicemail the following error is received "Unable to open application. Please try again later. If it continues to fail contact your administrator". The collaboration engineer is working on the problem found on the following phone logs:

6532 NOT 13:49:35.357489 CVM-InstallerModule.STATUS_INSTALL_CANCELLED &

STATUS_INSTALL_ERROR: [thread=installer MQThread][class=cip.midp.midletsuite.InstallerModule]

[function=updateStatus] Midlet Install Canceled/Error...Visual VoiceMail

How can this issue be resolved?

A. Replace the sever name with the server IP on service URL field

B. Eliminate the space in the service Name field

C. Configure DNS on phone configuration so it can resolve server name

D. Check the Enable checkbox on IP phone service configuration

Answer: B

Explanation

Looks like a simple error in phone service's display name: Visual VoiceMail

It needs to be exactly VisualVoiceMail without spaces (delete the space in the service Name field).

Question #:57 - (Exam Topic 1)

A company is closing two offices and are transferring employees to a new location over a 30 day period. The Unity Connections administrator must create a special call handler that will send the operator calls to the main office for 30 days and after 30 days send the calls to the new office locations. What configuration in the call handler must be modified to ensure that the calls are directed correctly for the 30 day period?

A. Modify the "Caller Input"

B. Modify the "Enabled Until" in the Standard Transfer Rule.

C. Modify the "Active Schedule" in the Call Handler Basics Page

D. Modify the "Enabled Until" in the Standard Greeting.

Answer: C

Question #:58 - (Exam Topic 1)

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

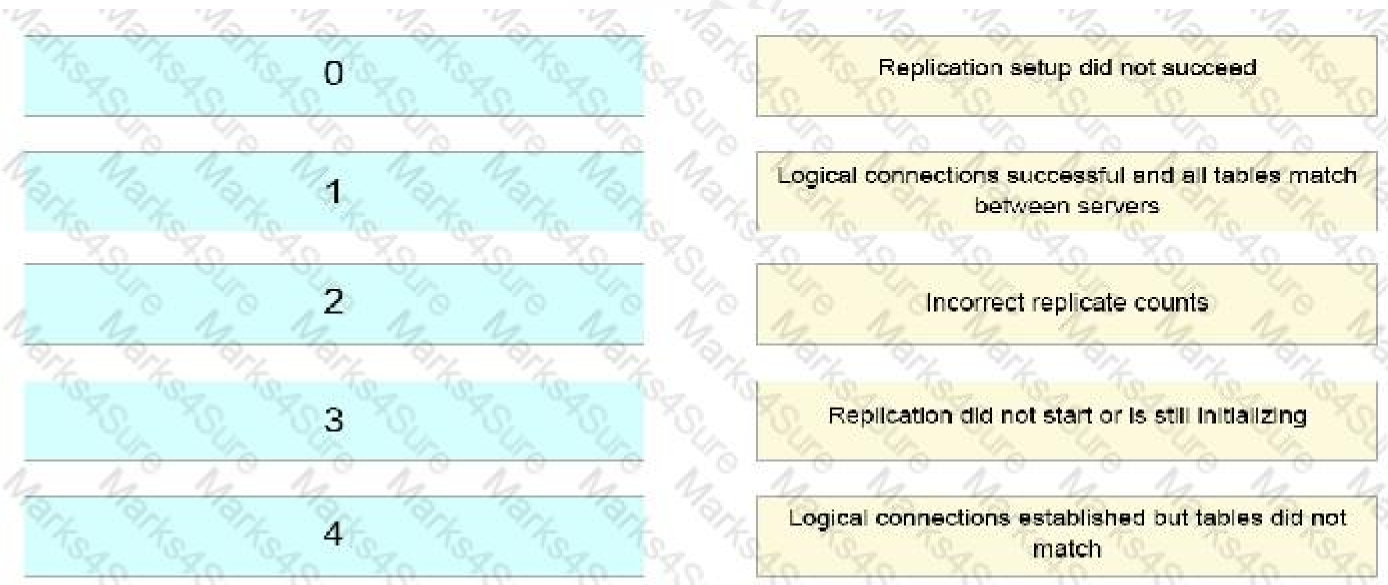
D. 9

E. 11

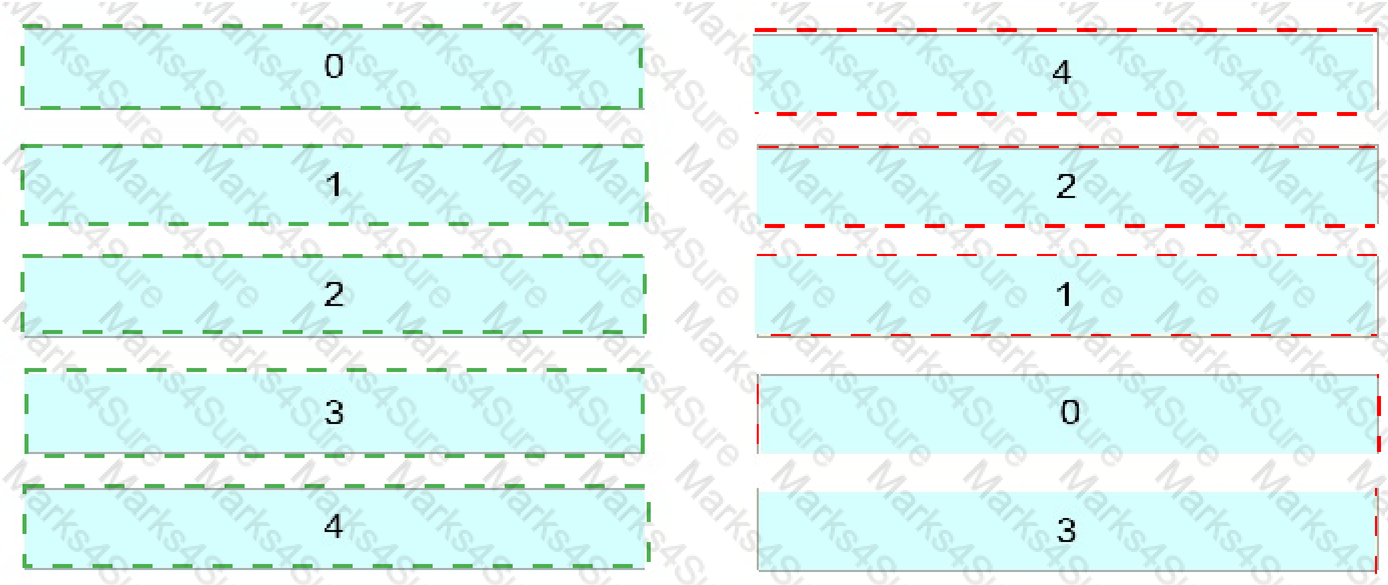
Answer: C

Question #:59 - (Exam Topic 1)

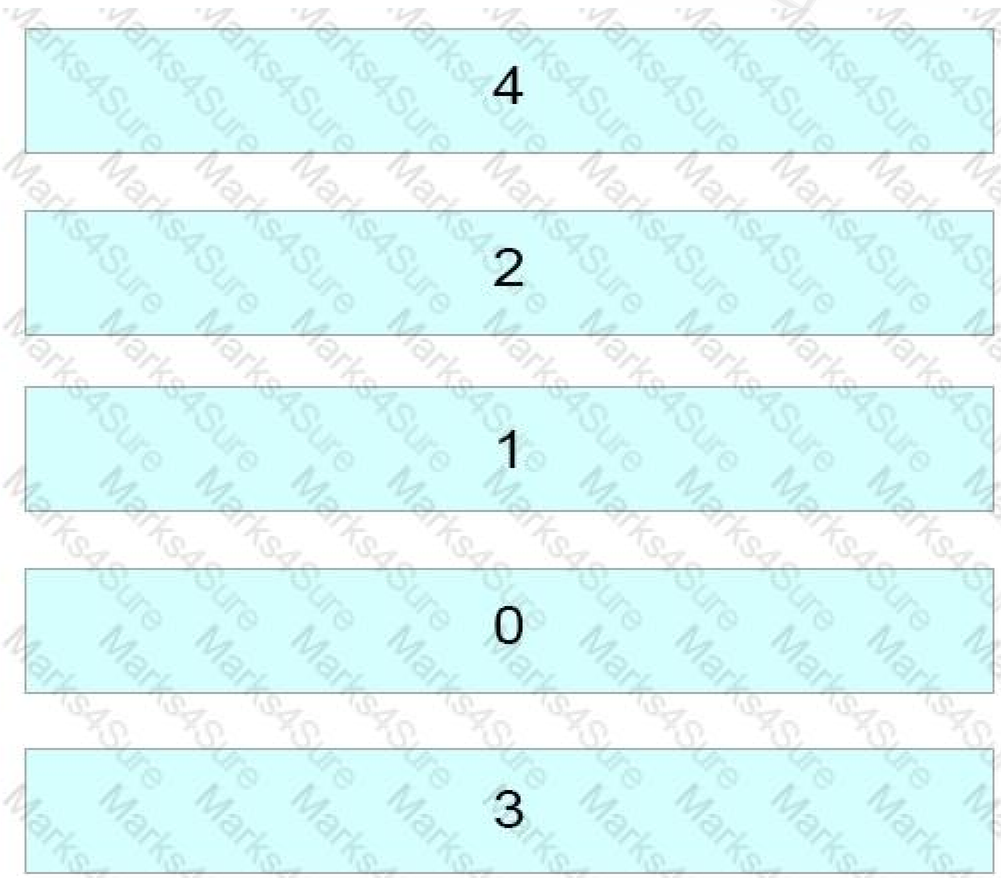
Drag and drop the Cisco Unified CM database replication status values on the left to the correct replication status definition on the right.



Answer:



Explanation



Question #:60 - (Exam Topic 1)

An engineer is converting all gateways to SIP and wants to ensure that the device is protected from traffic with malicious intent? Which two parts must the engineer enable to protect the Cisco Unified Communications Manager from SIP attacks on any newly created SIP trunks? (Choose two.)

A. SIP Station TCP Port Throttle Threshold

B. Denial - of - Service Protection Flag

C. SIP TCP Unused Connection Timer

D. SIP Max incoming Message Headers

E. SIP Max incoming Message Size

F. SIP trunk TCP Port Throttle Threshold

Answer: C E

Question #:61 - (Exam Topic 1)

What is the maximum delay requirement, in milliseconds, for deploying Cisco Unity Connection servers in active/active pairs over different sites?

A. 150

B. 200

C. 100

D. 250

Answer: C

Question #:62 - (Exam Topic 1)

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

Answer: A D

Question #:63 - (Exam Topic 1)

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

Answer: D

Question #:64 - (Exam Topic 1)

Which description of how dialed numbers are recorded in the Call detail record in Cisco Unified

Communication Manager is true?

A. Client Matter codes are not included in CDR for security reasons

B. Neither forced authorization codes or names are recorded in CDR

C. CDR records the dialed numbers after route list level digit manipulation

D. CDR records the dialed numbers prior to transition pattern level digit manipulation

E. CDR records the dialed numbers after route pattern level digit manipulation

Answer: C

Question #:65 - (Exam Topic 1)

Which two file transfer protocols are used to send Call Detail Record flat files to third party billing servers?

(choose two)

A. SFTP

B. SOAP/HTTPS

C. FTP

D. FTP over SSL

E. SCP

F. TFTP

Answer: A C

Question #:66 - (Exam Topic 1)

Refer to the exhibit.

```
|14:15:26.964|AppInfo|[CTI-APP][CTIHandler::processIncomingMessage]
 CTX ProviderOpenRequest ( seq0=1 provider=UCProvider login=mcisco
 heartbeat=60 timer=10 priority=0 lightWeightProviderOpen=0 AuthType=0
 RequestOldFetch=0)

|14:15:27.181 |AppInfo |Authenticating with SSL not enabled
 (ldap://172.25.140.200.389)
|14:15:27:187 |AppInfo |LDAP authentication bind SUCCESS for
 dcadmin@cisco.com

|14:15:27:187 |AppInfo |Connection # (0): successful
|14:15:27:187 |AppInfo |Details ::
|14:15:27:191 |AppInfo |Retrieve the specified user entry:
&bnsp:(&(&(objectclass-user)(&(telephoneNumber"*)))(sAMAccountName-mcisco))
|14:15:27:191 |AppInfo |LDAP search tor Userbase: 'dc=cisco, dc=com'

|14:15:36:959 |AppInfo |CTIManager::CtiManager::ready_SdlCloseInd():
 Connection closed indication. Shutting down provider. -- Connection Id=10
 TcpHandler=[1:200:13:778]PeerIPAddr=192.168.218.10 PeerPort=50418 Username=mcisco
 CtiHandler=[1:200:22:742]

|14:21:45:217 |AppInfo |LDAP Search complete. Code: 0
|14:21:45:217 |AppInfo |Get DN of entry.
|14:21:45:217 |AppInfo |Get DN: CN=Cisco Mills.OU=Users.
 OU=Information Technology.OU=Kingdom.DC=cisco.DC=com
|14:21:45:240|AppInfo|LDAP authentication bind SUCCESSfor CN=Cisco
 Mills.OU=Users.OU=Information Technology.OU=Kingdom.DC=cisco.DC=com
|14:21:45:240 |AppInfo |Connection # (O): successful
```

A jabber user reports that desktop mode does not work. Connection status shows the following message: “Connection error. Ensure the server information in the phone services tab on the option window is correct. Contact your system administrator for assistance.”

Which two actions will resolve the problem? (Choose two)

A. Change LDAP port from 389 to 636 in LDAP directory configuration.

B. Restart DirSync service under unified serviceability.

C. Change LDAP port from 389 to 443 in LDAP director configuration.

D. Change LDAP port from 389 to 3268 in LDAP director configuration.

E. Restart CTI manager service under unified serviceability.

F. Restart TFTP service under Unified serviceability

Answer: D E

Question #:67 - (Exam Topic 1)

Which of the 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. 484 Address Incomplete

B. 408 Request Timeout

C. 503 Service Unavailable

D. 504 Server Timeout

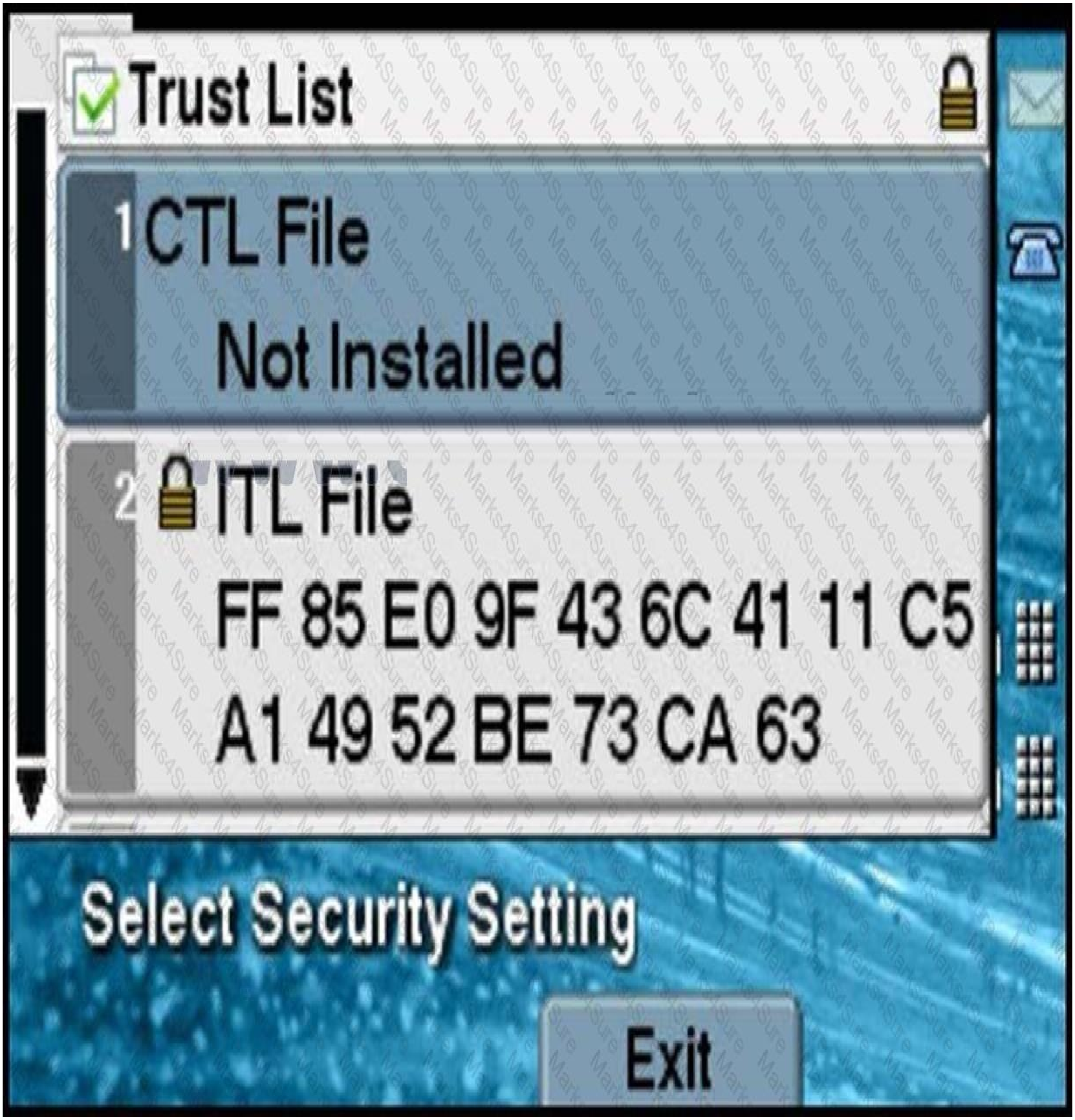
E. 404 Not Found

F. 505 Version Not Supported

Answer: B C

Question #:68 - (Exam Topic 1)

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 signalling but unencrypted call media

D. Encrypted call media but unencrypted call signalling

E. Encrypted call signalling and media

F. Local trust verification on the

Answer: A B

Question #:69 - (Exam Topic 1)

Refer to the exhibit.

```
Jan 10 02:31:25.598: h323chan_gw_conn: Created socket fd=1
Jan 10 02:31:25.598: h323chan_gw_conn: Created socket fd=2h323chan_dgram_send:Sent UDP msg.
                     Bytes sent: 50 to 224.0.1.41:1718 fd=2

Jan 10 02:31:25.598: RASLib::GW_RASSendGRQ: GRQ (seq# 47) sent to 224.0.1.41
Jan 10 02:31:25.598: h323chan_chn_process_read_socket
Jan 10 02:31:25.598: h323chan_chn_process_read_socket: fd=2 of type CONNECTED has data
Jan 10 02:31:25.598:  h323chan_chn_process_read_socket: h323chan accepted/connected fd=2

Jan 10 02:31:25.598: h323chan_dgram_recvdata:rcvd from [10.1.1.2:1718] on fd=2

Jan 10 02:31:25.598: GCF (seq# 47) rcvd from h323chan_dgram_send:Sent UDP msg.
```

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

Answer: E

Question #:70 - (Exam Topic 1)

A collaboration engineer is designing 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

Answer: A D

Explanation

The following IM and Presence Service features require an external database:

Persistent Group Chat

Message Archiver (IM Compliance)

External database:

PostgreSQL database, versions 8.3.x through 9.4.x are supported, and in IM and Presence Service Release, 11.0(1) versions: 9.1.9, 9.2.6, 9.3.6, 9.4.1 have been tested.

Note: You can also use Version 8.1.x of the PostgreSQL database, but the configuration of these versions may be different to the PostgreSQL database configuration described in this section. See the PostgreSQL documentation for details on how to configure these PostgreSQL database versions. If you use Version 8.1.x of the PostgreSQL database, the database configuration on IM and Presence Service is the same as described in this section.

Oracle database, versions 9g, 10g, 11g, and 12c are supported, and in IM and Presence Service Release, 11.0(1) versions: 11.2.0.1.0 and 12.1.0.1.0 have been tested.

You can install the database on either a Linux or a Windows operating system. See the relevant database documentation for details on the supported operating systems and platform requirements.

IPv4 and IPv6 are supported.

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/database_setup/10_0_1/CUP0_BK_D

Question #:71 - (Exam Topic 1)

In OpenStack, while project stores and retreives arbitrary unestructured data objects?

A. Nova

B. Swift

C. Cinder

D. Keystone

Answer: D

Question #:72 - (Exam Topic 1)

Which two SCCP call states support the Meet Me soft key? (Choose two.)

A. On Hook

B. Connected

C. On Hold

D. Off Hook

E. Ring Out

F. Connected Conference

Answer: A D

Question #:73 - (Exam Topic 1)

A Cisco Unified CM cluster is being set up for call control discover using the service advertising framework. An engineer discovers that patterns are not being learned by the cluster. Which two items must be checked in an attempt to resolve the issue? (Choose two)

A. The CCD block patterns are not preventing remote patterns from being entered into the local cache.

B. The hostedDN group on the cluster matches the patterns that should be learned.

C. The CCD advertising service is activated in Cisco unified CM serviceability.

D. A CCD route partition has been assigned for learned patterns.

E. The CCD requesting service is activated in Cisco unified CM serviceability.

F. The Sip trunk is enabled for call control discover.

Answer: A F

Explanation

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_0_2/ccmfeat/fsgd-802-cm/fscallcontroldis

After you configure call control discovery, you may block learned patterns that remote call-control entities send to the local Cisco Unified Communications Manager. (Call Routing > Call Control Discovery > Blocked Learned Patterns

Ensure that the CCD block patterns are not preventing remote patterns from being entered into the local cache.

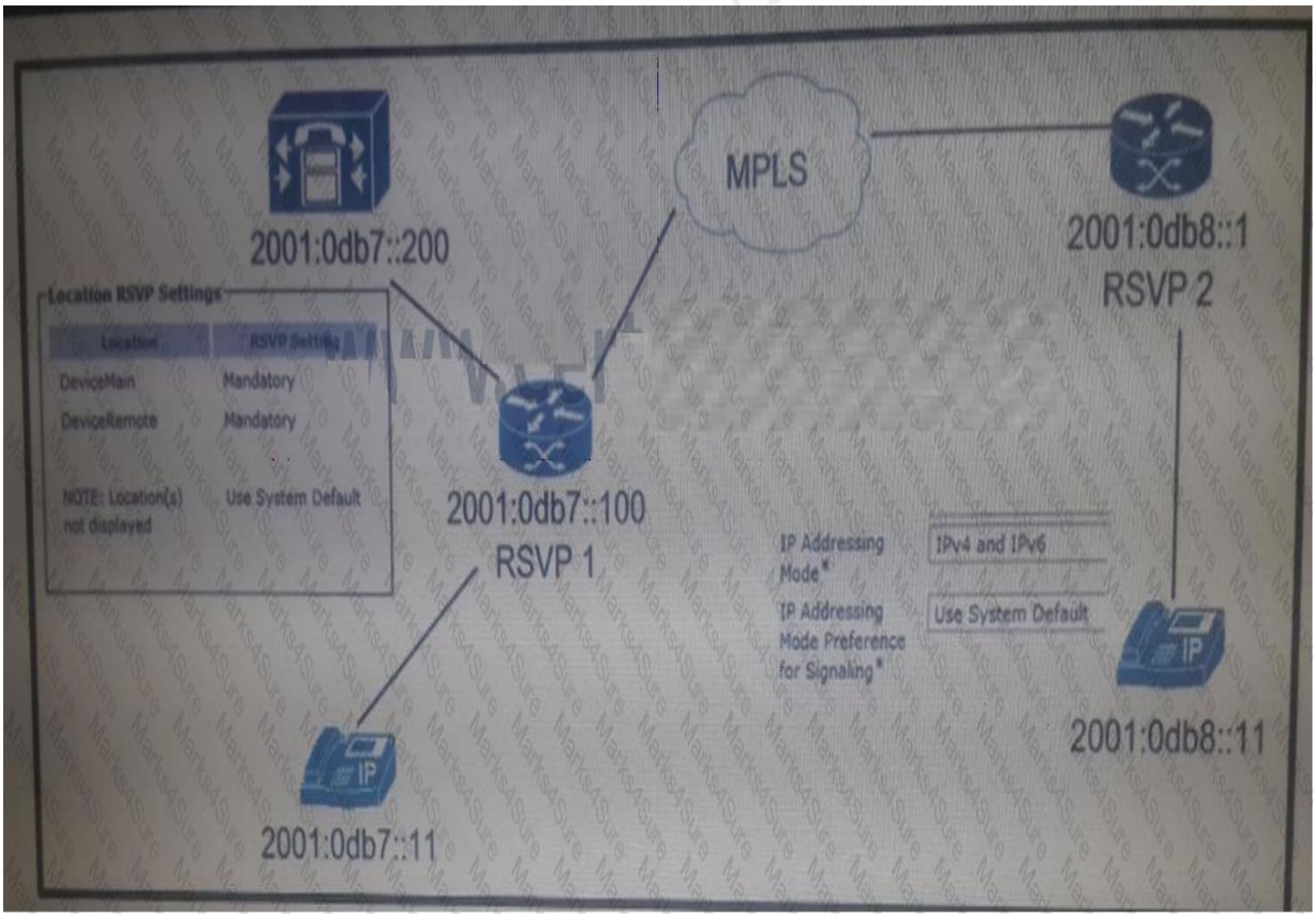
The local Cisco Unified Communications Manager cluster uses SAF-enabled trunks that are assigned to the CCD requesting service to route outbound calls to remote call-control entities that use the SAF network.

The Cisco Unified Communications Manager cluster advertises the SAF-enabled trunks that are assigned to the CCD advertising service along with the range of hosted DNs; therefore, when a user from a remote call-control entity makes an inbound call to a learned pattern on this Cisco Unified Communications Manager, this Cisco Unified Communications Manager receives the inbound call from this SAF-enabled trunk and routes the call to the correct DN.

Ensure that the Sip trunk is enabled for call control discover.

Question #:74 - (Exam Topic 1)

Refer to the exhibit.



A collaboration engineer is troubleshooting a cluster that has been configured to use RSVP. The calls are being rejected and the caller receives a busy tone. What is the root cause of this problem?

A. The RSVP Agents are only using an Ipv6 address.

B. The IP Addressing mode preference for signalling is set to use System Default.

C. The RSVP relationship between Main and Remote is set to Mandatory

D. The IP Addressing Mode is set to use Ipv4 and Ipv6

Answer: A

Question #:75 - (Exam Topic 1)

A Cisco Unified Cm user is set up with one remote destination profile that has two remote destination numbers: the first destination number is the user's mobile phone in

country A and the second is a mobile phone located in country . All outbound calls are centralized from the gateway at country A. The user reports that inbound calls are properly routed to the mobile phone as long as the user is in country A. But inbound calls are not successfully routed to country B. Which two options could be the cause of this? (Choose two)

A. The enable mobile connect option must be selected under the user's second remote destination number.

B. The value of remote destination limits should be chaned to 2 instead of the default value of 4 under the end user page.

C. The enable mobile voice access option must be selected under the end user page.

D. The value of maximum wait time for desk pickup should be changed to 20000 instead of the default of D. 10000, under the end user page.

E. The rerouting calling search space assigned to the user's remote destination profile must have access to international calls.

Answer: A E

Question #:76 - (Exam Topic 1)

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**

Answer: D

Question #:77 - (Exam Topic 1)

A Cisco collaboration architect is evaluating a list of codecs to use in a voice infrastructure. Which three facts are associated with iSAC and should be considered in the decision? (Choose three)

A. The codec has better quality with less bandwidth for sideband applications.

B. The codec will not be supported in TDM voice gateways.

C. The codec will adjust its bandwidth consumption to the network conditions.

D. The codec will not be available for H.323 and MGCP devices.

E. The codec will not support low complexity.

F. The codec will not be supported by SCCP configured on DSPFARMS.

Answer: A C E

Question #:78 - (Exam Topic 1)

Which three issues prevent a customer from seeing the presence status of a new contact in their Jabber contact list? (Choose three.)

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

Answer: B C D

Explanation

No Presence Information After Login

Problem

You receive no Presence information after login.

Solution

Complete these steps in order to resolve this issue:

- Make sure that the DNS server the PC is pointed to can resolve the fully qualified name of the CUPS server.

The host entry will not suffice, you must resolve via DNS.

- Check the SUBSCRIBE CSS on the SIP trunk to CUP.

This CSS must include the partitions of the devices you are trying to receive status on.

- The CUP SIP proxy incoming access control list (ACL) is not allowing incoming SIP presence messages to reach the presence engine. As a test, set the incoming ACL to ALL and reset the SIP proxy and presence engine. Log in again to the CUPC and try to reconfigure the incoming ACL properly.

http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-presence/97443-cups-cupc-ts

Question #:79 - (Exam Topic 1)

A company has migrated email service from Microsoft exchange to Microsoft Office365. After the migration, end users cannot receive voicemail using Office365 email account. What action is needed to receive voicemail via email?

A. Change the service type field

B. Select the user corporate email address option

C. Uncheck synchronize connection and exchange mailboxes

D. Update the Cisco unified Messaging service

Answer: D

Question #:80 - (Exam Topic 1)

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

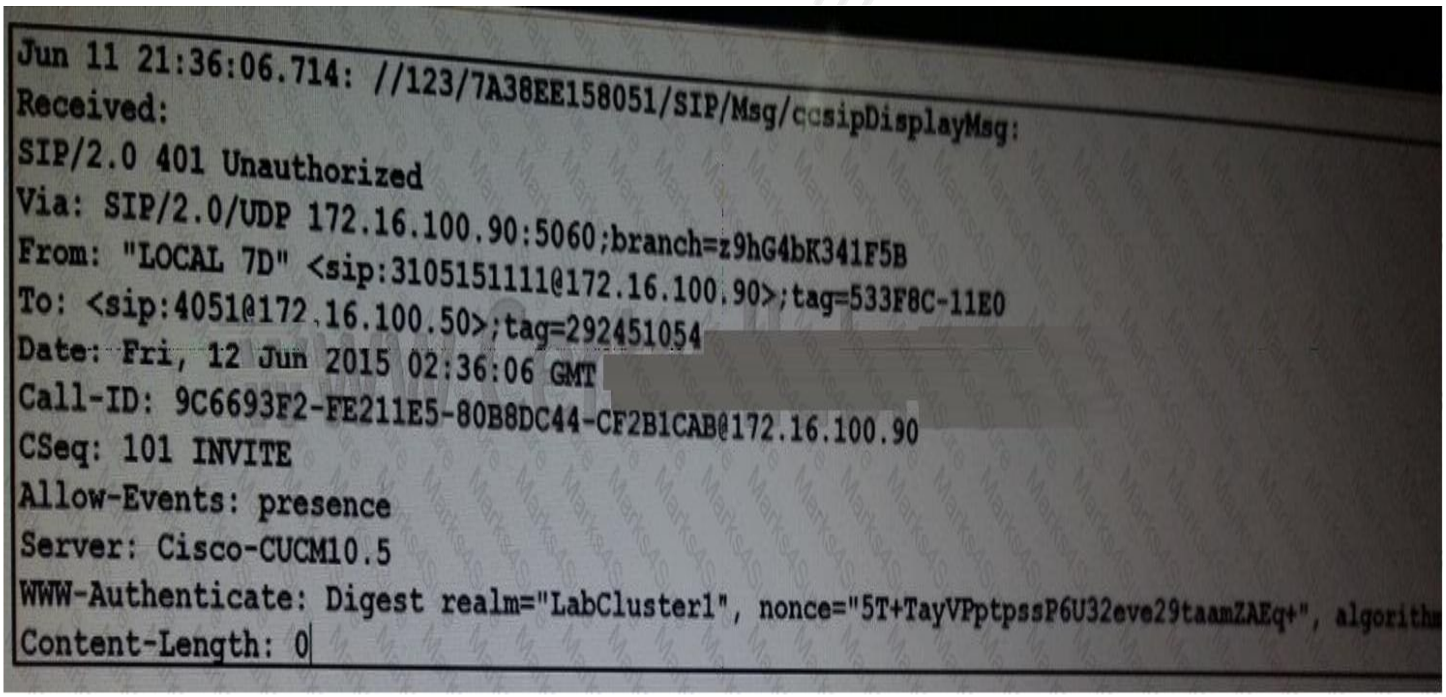
F. cBarge

G. Join

Answer: D E F

Question #:81 - (Exam Topic 1)

Refer to the exhibit.



Network administrator is implementing SIP trunk digest authentication. After making outbound tests, the calls are failing with 503 service unavailable error from the router after being challenged.

Which set of commands on the router fix this problem?

A. sip-ua

Registrar ipv4:172.16.100.90 expires 3600

Authentication cisco realm MD5 LabCluster1

B. sip-ua

Authentication username cisco password 7 cisco realm LabCluster1

C. Voice service voip

Sip

Authenticate digest MD5 LabCluster1

D. Voice service voip

Sip

Authenticate digest LabCluster1 MD5

Answer: B

Question #:82 - (Exam Topic 1)

An engineer received this requirement from a service provider:

Diversion header should match the network DID "123456@company.com" 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:(.*>)" "123456@company.com>"

request REINVITE sip-header Diversion modify "sip:(.*>)" "123456@company.com>"

B. voice class sip-profiles 200

request INVITE sdp-header Diversion modify "sip:(.*>)" "123456@company.com>"

request REINVITE sdp-header Diversion modify "sip:(.*>)" "123456@company.com>"

C. voice class sip-profiles 200

response 200 sdp-header Diversion modify "sip:(.*>)" "123456@company.com>"

D. voice class sip-profiles 200

response 200 sip-header Diversion modify "sip:(.*>)" "123456@company.com>"

Answer: A

Question #:83 - (Exam Topic 1)

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

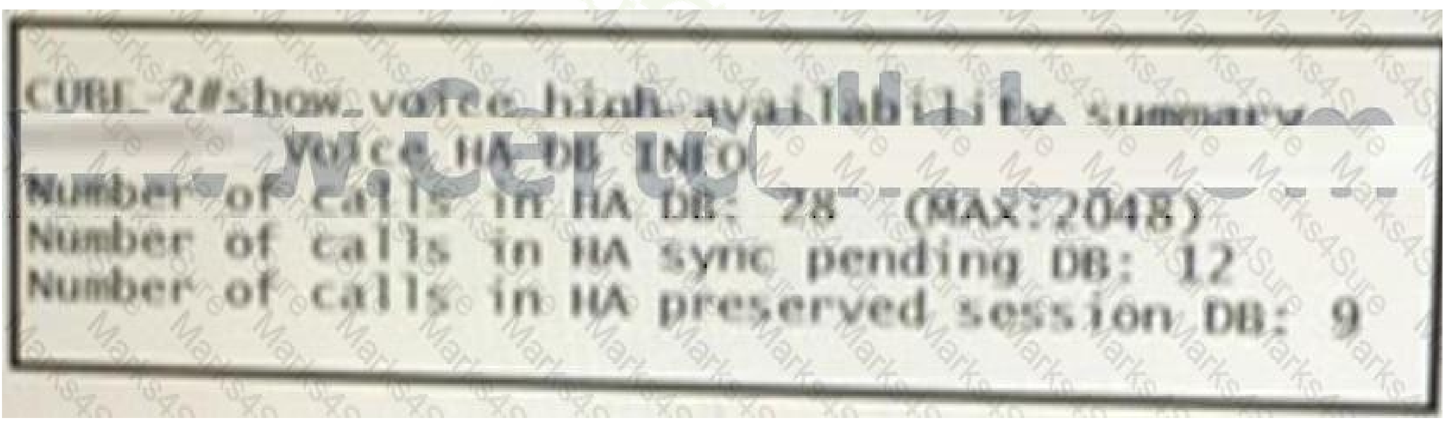
F. a minimum of two USB e-token

G. a minimum of one soft e-token

Answer: A B D F

Question #:84 - (Exam Topic 1)

Refer to the exhibit.



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

Answer: D

Explanation

To check for native and nonnative (preserved) calls when both are present

The numbers of calls on the system are shown as follows:

Total number of calls = "Number of calls in HA DB" + "Number of calls in HA sync pending DB".

Total number of preserved (nonnative) calls = "Number of calls in HA preserved session DB".

Total number of native calls (calls set up since the failover and therefore not preserved over the failover) is the difference in the previous two numbers. In this example, it is (28+12) - 9 = 31.

Question #:85 - (Exam Topic 1)

A collaboration engineer has been asked to implement secure real-time protocol between a Cisco Unified CM and its SIP gateways. Which option is a consideration for this implementation? (Choose two.)

A. SRTP is supported only in Cisco IOS 15.x and higher

B. Only T.38 and Cisco Fax protocols are supported.

C. Call hold RE-INVITE is not supported.

D. SIP requires that all times be sent in GMT

Answer: D

Question #:86 - (Exam Topic 1)

Refer to the exhibit.

Which of the following domain must be configured on the Expressway C in a Multidomain

MRA setup?A. Domain1 and Domain2

B. Domain 1 only

C. Domain 4 only

D. Domain 2 only

E. Domain1 and Domain4

E

Question #:87 - (Exam Topic 1)

Refer to the exhibit.

```
v=0
o=BroadWorks 12628880 1 IN IP4 172.31.200.196
s=-
c=IN IP4 172.31.200.196
t=0 0
a=sqn: 0
a=cdsc: 1 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
m=audio 10086 RTP/AVP 18 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=fmtp:18 annexb=no
```

Which three facts can be determined about the audio parameters of this call from this session description protocol? (Choose Three)

A. The DTMF relay will be RFC2833

B. The Codec will be G711

C. The codec will be G729

D. VAD will be disabled for this call

E. VAD will be enabled for this call

F. The call will be a T38 fax call

Answer: A C D

Question #:88 - (Exam Topic 1)

A Cisco Jabber and Cisco Unified Communication Manager IM&P on premise customer wants to eliminate certificate warning messages when Jabber client launch. The customer environment uses Jabber services Discovery. After some investigations, you find that the CUCM IM&P server is running with self-signed certificates.

Which two certificates on the CUCM IM&P servers must be signed by CA trusted by the Cisco Jabber client to eliminate certificate warning message when the Jabber clients start? (choose two)

A. cup

B. cup-xmpp

C. cup-xmpp-s2s

D. tomcat

E. ipsec

Answer: B D

Question #:89 - (Exam Topic 1)

A Cisco collaboration engineer is troubleshooting unexpected SIP call disconnects. Which three responses correspond to the 5xx range (Chose three).

A. Server timeout

B. Bad request

C. Temporarily unavailable

D. Service unavailable

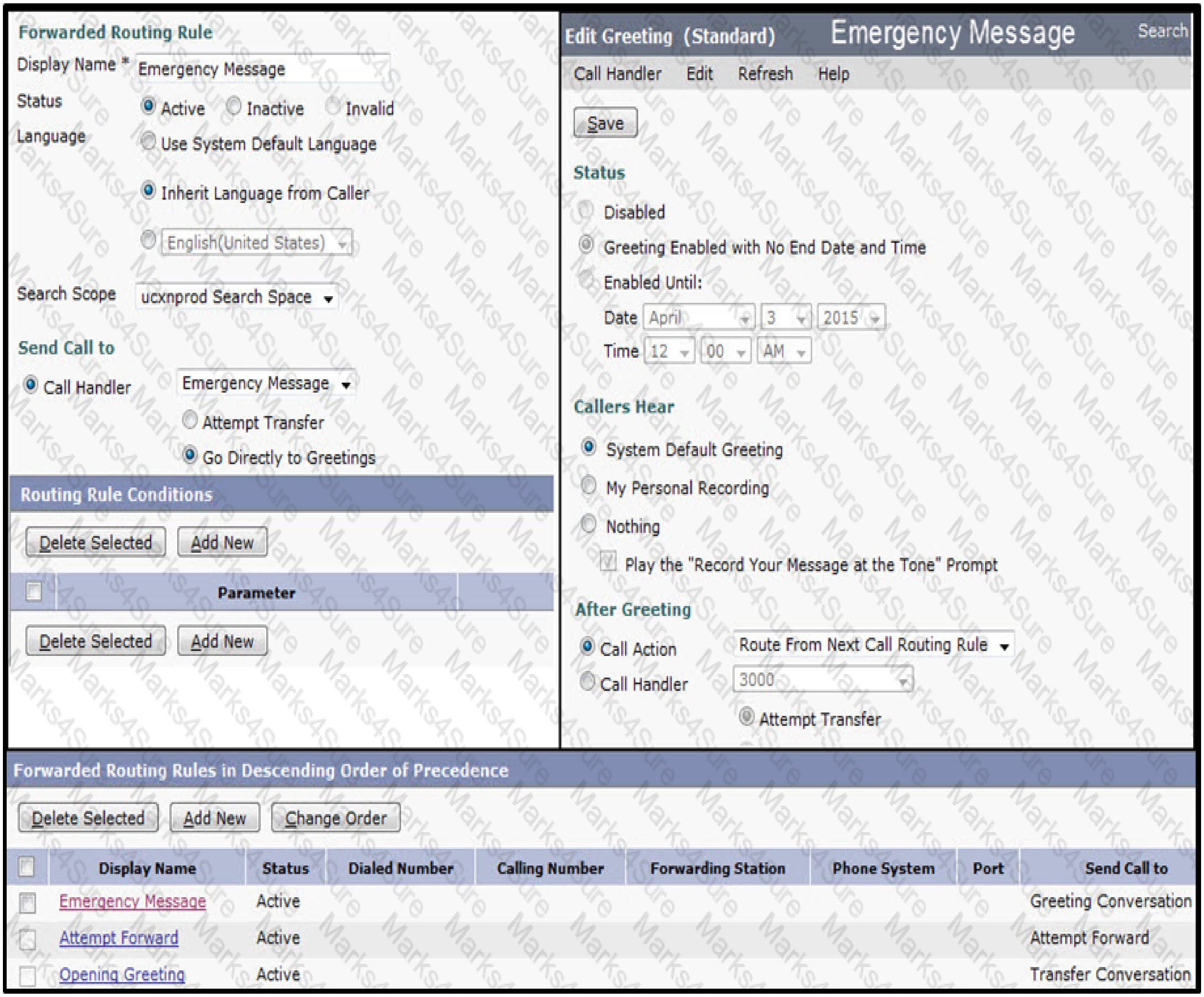
E. Forbidden

F. Version not supported

Answer: A D F

Question #:90 - (Exam Topic 1)

Refer to the exhibit.



What does an outside caller hear when calling a user and forwarding to Cisco Unity Connection?

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.

Answer: B

Question #:91 - (Exam Topic 1)

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

Answer: C D

Question #:92 - (Exam Topic 1)

During a Cisco Unity 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 Enterprise Parameters.

B. Reduce Caller Input timeout in Cisco Unity Connection Service Parameters.

C. Lower the timer Wait for Additional Digits on the caller input page.

D. Enable Ignore Additional Input on the Edit Caller Input page for the selected key.

E. Enable Prepend Digits to Dialled Extensions and configure complete extension number on the Edit Caller Input page for the selected key.

Answer: D

Question #:93 - (Exam Topic 1)

Refer to the exhibit.

```
voice service voip
 clid substitute name
 clid network-provided
 address-hiding
 mode border-element
 allow-connections sip to sip
 allow-connections sip to h323
 allow-connections h323 to sip
 allow-connections h323 to h323
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
 no fax-relay sg3-to-g3
 modem passthrough nse codec g711ulaw
 sip
  session transport tcp
  min-se 360 session-expires 360
  ds0-num
  header-passing
  media flow-around
  pass-thru content sdp
  error-passthru
  registrar server expires max 600 min 60
  options-ping 90
  early-offer forced
  midcall-signaling passthru
  no call service stop
```

A user is on an outbound call through a Cisco Unified border Element gateway. When the user places the call on hold, the remote party hears silence. The Cisco Unified Communication Manager Cluster is using multicast on hold. The Cisco Unified Border Element Gateway is on the same subnet as the Cisco Unified Communication Manager Cluster.

Which two options will resolve this issue? (Choose two)

A. Media flow-through must be configured.

B. CCM-manager music-on-hold should be removed from the configuration.

C. The session transport UDP command must be configured.

D. The Cisco unified border Element router must be set up for gateway-based MOH.

E. The pass-thru content sdp command should be removed.

Answer: A E

Explanation

Configuring the media flow-around command is required for Session Description Protocol (SDP) pass-through. When flow-around is not configured, the flow-through mode of SDP pass-through will be functional.

• When the dial-peer media flow mode is asymmetrically configured, the default behavior is to fallback to SDP pass-through with flow-through.

SDP pass-through is addressed in two modes:

• Flow-through—Cisco UBE plays no role in the media negotiation, it blindly terminates and re-originates the RTP packets irrespective of the content type negotiated by both the ends. This supports address hiding and NAT traversal.

• Flow-around—Cisco UBE neither plays a part in media negotiation, nor does it terminate and re-originate media. Media negotiation and media exchange is completely end-to-end.

When SDP pass-through is enabled, some of interworking that the Cisco Unified Border Element currently performs cannot be activated. These features include:

• Delayed Offer to Early Offer Interworking

• Supplementary Services with triggered Invites

• DTMF Interworking scenarios

• Fax Interworking/QoS Negotiation

• Transcoding

Question #:94 - (Exam Topic 1)

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

Answer: A B C

Question #:95 - (Exam Topic 1)

Refer to the exhibit.

```
Jan 10 05:55:35.130: MGCP Packet sent to 10.1.1.2:2427--->
NTFY 217738192 *@MGCP-gateway.cisco.com MGCP 0.1
X: 0
O:
<---

Jan 10 05:55:35.130: MGCP Packet received from 10.1.1.2:2427 --->
200 217738192
<---
```

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 it will 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

Answer: C

Question #:96 - (Exam Topic 1)

Refer to the exhibit.

```
RAS INCOMING PDU ::=

value RasMessage ::= registrationReject :
   {
    requestSeqNum 24
    protocolIdentifier {0 0 8 2250 0 3 }
    rejectReason duplicateAlias:
    {
    }
    gatekeeperIdentifier {"gk"}
 }
```

A CISCO collaboration engineer is troubleshooting a gateway and gatekeeper problem and sees this output from a debug command. Which two configurations can cause this problem? (Choose two)

A. The same zone prefix is configured in two different gatekeepers.

B. The same H323-ID is configured in two different gateways.

C. The same gw-type-prefix is configured in two different zone subnets IDs.

D. The same zone subnet ID is configured in two different gatekeepers.

E. The same E164-ID is configured in two different gateways.

Answer: B E

Question #:97 - (Exam Topic 1)

XMPP Protocol

FEATURES(stream) []

STARTTLS [xmlns="urn:ietf:params:xnl:ns:xmpp-tls*]

xmlns: urn:ietf:params:xml:ns:xmpp-tls

REQUIRED

Refer to the exhibit. Which message is used to negotiate the TLS requirement while federating with an external domain?

A. xmpp-server message

B. FEATURES message

C. server hello

D. STARTTLS message

E. client hell

Answer: D

Question #:98 - (Exam Topic 1)

A Cisco collaboration engineer is troubleshooting unexpected SIP call disconnects. Which three responses correspond to the 5xx range? (Choose three.)

A. Temporarily Unavailable

B. Version not supported

C. Bad request

D. Forbidden

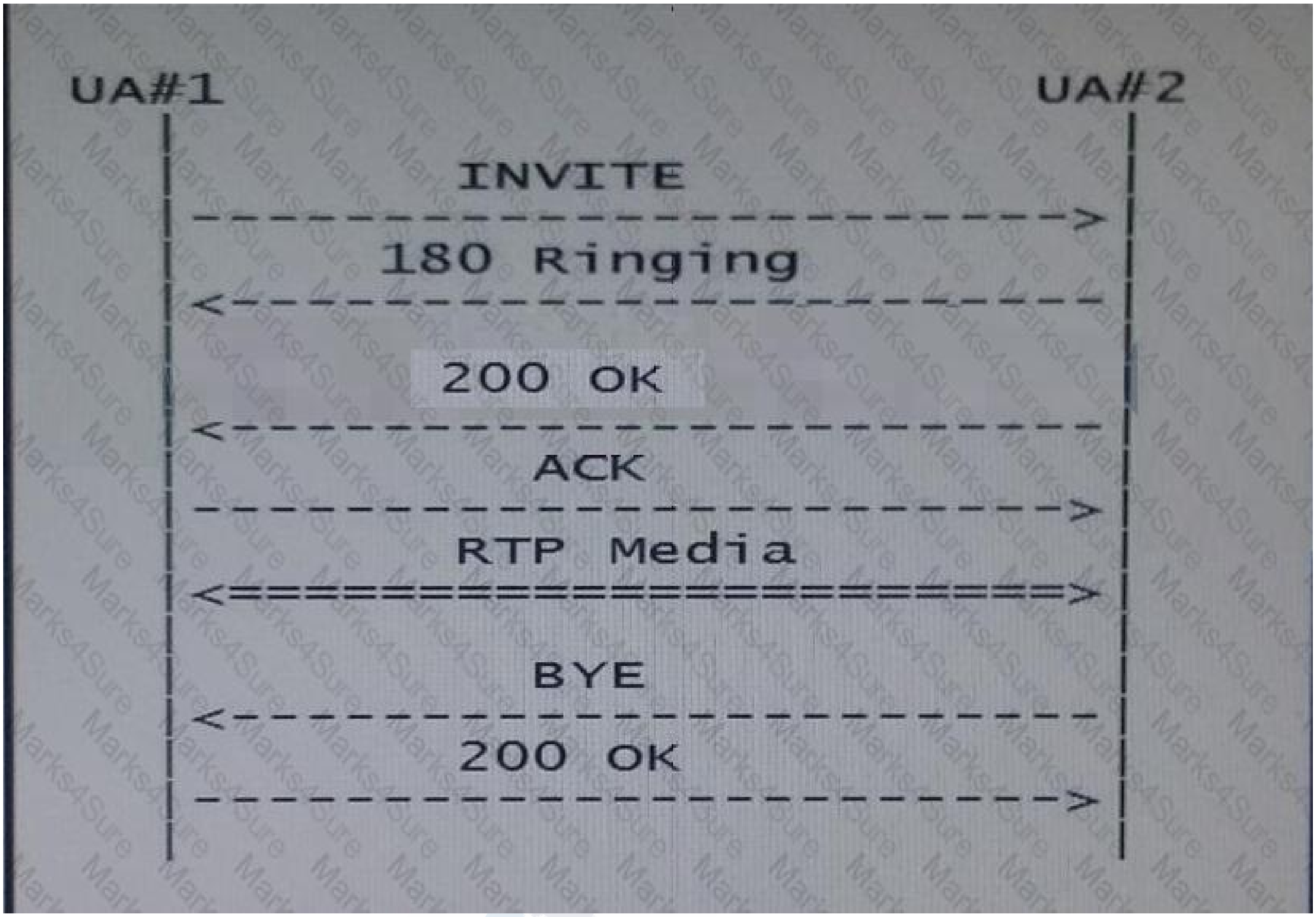
E. Server Timeout

F. Service Unavailable

Answer: B E F

Question #:99 - (Exam Topic 1)

Refer to Exhibit:



How many SIP signalling 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

Answer: C

Question #:100 - (Exam Topic 1)

Which two options are examples of data at rest? (Choose two.)

A. An email received from a colleague.

B. An email accessed via a web browser.

C. An email saved on a USB drive.

D. An email archived locally on a laptop hard drive.

E. An email sent to a colleague.

Answer: C D

Question #:101 - (Exam Topic 1)

Which statement describes virtual SNR DN configuration and behaviour on a Cisco Communication Manager Express IOS router?

A. A virtual SNR DN is a DN that must be associated with multiple registered IP phones

B. Mid-calls on virtual SNR DN can be pulled back as soon as a phone becomes associated with the DN

C. SNR feature can only be invoked if the virtual SNR DN is associated with at least one registered IP phone

D. A call that is ringing a virtual SNR DN prior to its association with a registered phone, cannot be answered by the phone even after the association is made

E. Virtual SNR DN supports either SCCP or SIP IP phone DNs

Answer: D

Question #:102 - (Exam Topic 1)

A company has one CUCM located in North America and another cluster located in Europe. IT department has piloted deployment to enable SIP URI dialling and call from video endpoints between clusters. The engineer performs following three steps,

- Define Cluster ID on both CUCM clusters

- Exchange tomcat certificates with other nodes

- Setup Role option for primary/hub cluster

Which additional three steps are required to make SIP URI dialing work between the clusters? (Choose three)

A. Configure the route list

B. Define SIP route pattern

C. Setup role option for secondary cluster/spoke cluster

D. Configure the hunt list

E. Configure the SIP trunk

F. Configure the route pattern

G. Configure the hunt pilot

Answer: B C E

Question #:103 - (Exam Topic 1)

Which two requests use the same Cseq number of an earlier INVITE request? (choose two)

A. NOTIFY

B. UPDATE

C. REFER

D. BYE

E. ACK

F. CANCEL

Answer: E F

Question #:104 - (Exam Topic 1)

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.

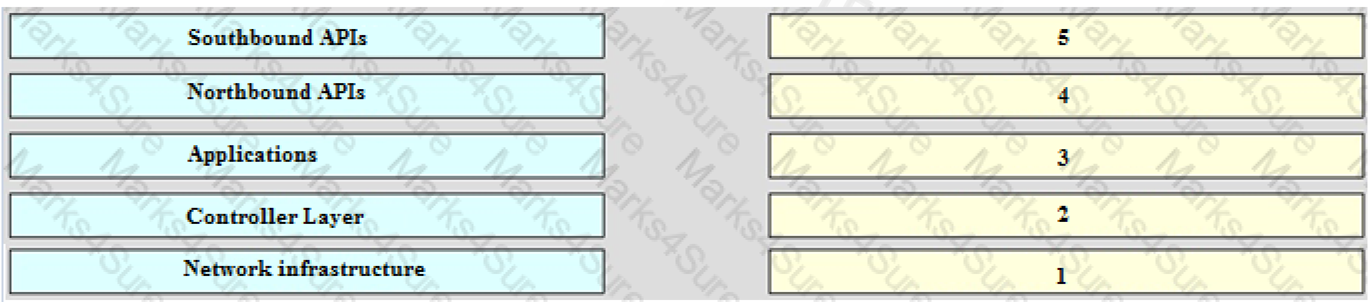
D. The administrator informs the user that this feature is a built-in option to the user Cisco PCA page under caller input.

E. The administrator informs the user that this feature is a built-in option for the user in the TUI under personal settings.

Answer: A

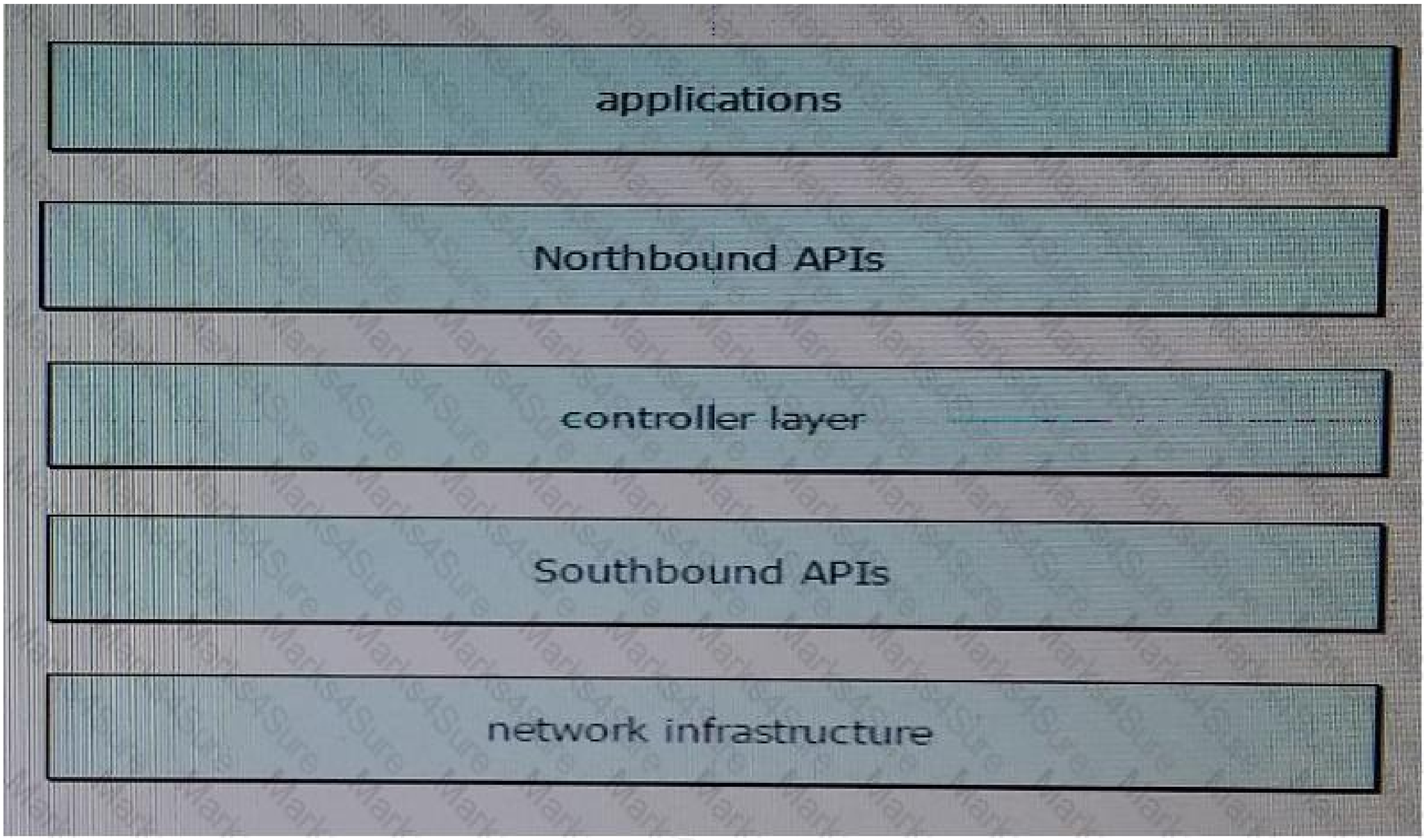
Question #:105 - (Exam Topic 1)

Drag and drop the logical components of a traditional SDN infrastructure into the correct order. Similar to open system interconnection model (OSI model), working your way up from the network infrastructure layer.



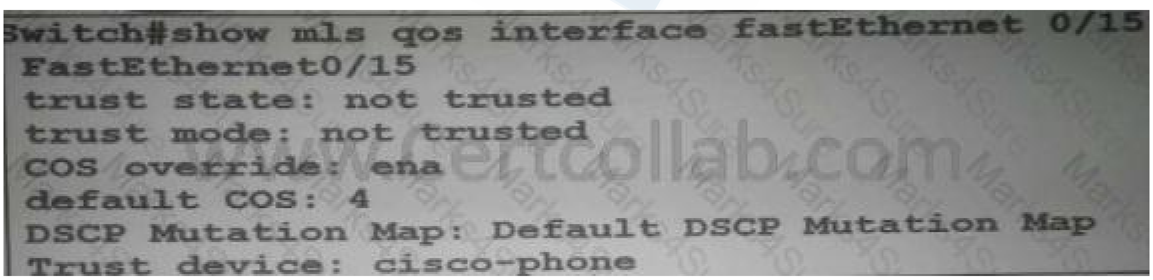
Answer:

Explanation



Question #:106 - (Exam Topic 1)

Refer to the exhibit.



```
Switch#show mls qos interface fastEthernet 0/15
FastEthernet0/15
trust state: not trusted
trust mode: not trusted
COS override: ena
default COS: 4
DSCP Mutation Map: Default DSCP Mutation Map
Trust device: cisco-phone
```

A VOIP engineer must configure QoS for the switch ports to remap PC traffic to COS 2

while allowing Cisco phone COS values to be used on the network. Which three commands satisfy these requirements? (Choose three)

A. FastEthernet 0/15

mls qos cos 2

B. FastEthernet 0/15

No mls qos cos 4

C. FastEthernet 0/15

Switchport priority extend cos

D. FastEthernet 0/15

mls qos cos override

E. FastEthernet 0/15

no mls qos cos override

F. FastEthernet 0/15

no mls qos trust device cisco-phone

G. FastEthernet 0/15

no switchport priority extend trust

Answer: A E G

Question #:107 - (Exam Topic 1)

You consider using RPL in a new loT environment. Which definition of a DIO is true?

A. A DIO is an ICMPv6 RPL control message whose main function is to perform DODAG secure message Counters

B. A DIO is an ICMPv4 RPL control message whose main function is to perform DODAG discovery, formation, and maintenance

C. A DIO is an ICMPv6 RPL control message whose main function is to perform DODAG discovery, formation, and maintenance

D. A DIO is an ICMPv4 RPL control message whose main function is to propagate destination information in a RPL network

Answer: C

Question #:108 - (Exam Topic 1)

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.

Answer: C

Question #:109 - (Exam Topic 1)

A voice engineer is preparing to migrate from PRI to SIP. Which three SIP request methods are available? (choose three)

A. NOTIFY

B. TRYING

C. PRACK

D. USE PROXY

E. UNAUTHORIZED

F. OPTIONS

Answer: A C F

Question #:110 - (Exam Topic 1)

Which description of the expected behavior of a SIP User Agent Client that received a reliable provisional response that contains an offer is true?

A. The UAC must include an answer in a PRACK

B. The UAC must include an answer in a PRACKACK

C. The UAC may include an answer in PRACKACK

D. The UAC may include an answer in PRACK

E. The UAC may include an answer in ACK

Answer: B

Question #:111 - (Exam Topic 1)

Refer to the exhibit.

```
INVITE sip:95124182222@172.16.100.90:5060 SIP/2.0
Via: SIP/2.0/TCP 172.16.100.50:5060
From: "Agent A" <sip:51241830010172.16.100.50>
To: <sip:95124182222@172.16.100.90>
Date: Tue, 10 Mar 2015 14:25:03 GMT
Call-ID: 3c748f00-4fe1febr-35ff9-2bef12ac0172.16.100.50
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE
Cisco-Guid: 1014271744-0000065536-0000221138-0737068172
Content Type: application/sdp
Content-Length: 202

v=0
o=CiscoSystemsCCM-SIP 2371 1 IN IP4 172.16.100.50
s=SIP Call
c=IN IP4 172.16.100.50
t=0 0
m=audio 24582 RTP/AVP 0 4 8 9 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Which SIP message will trigger the calling device to open channels for early media reception?

A. 180 Ringing

B. ACK

C. INVITE

D. 183 session-progress

E. 200 ok

Answer: D

Question #:112 - (Exam Topic 1)

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. PostgreeSQL database

E. Informix SQL database

Answer: A D

Question #:113 - (Exam Topic 1)

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.

Answer: B

Question #:114 - (Exam Topic 1)

A collaboration engineer is troubleshooting an MOH problem on a Cisco IOS SIP gateway in searching through a debug CCSIP massage output. Which three parameters in the SIP massages can be used to determine if the call was placed on hold? (Choose three)

A. OPTIONS WITH 301 CALLHOLD

B. INVITE WITH a=INACTIVE

C. INVITE WITH a=SENDONLY

D. OPTION WITH c=INACTIVE

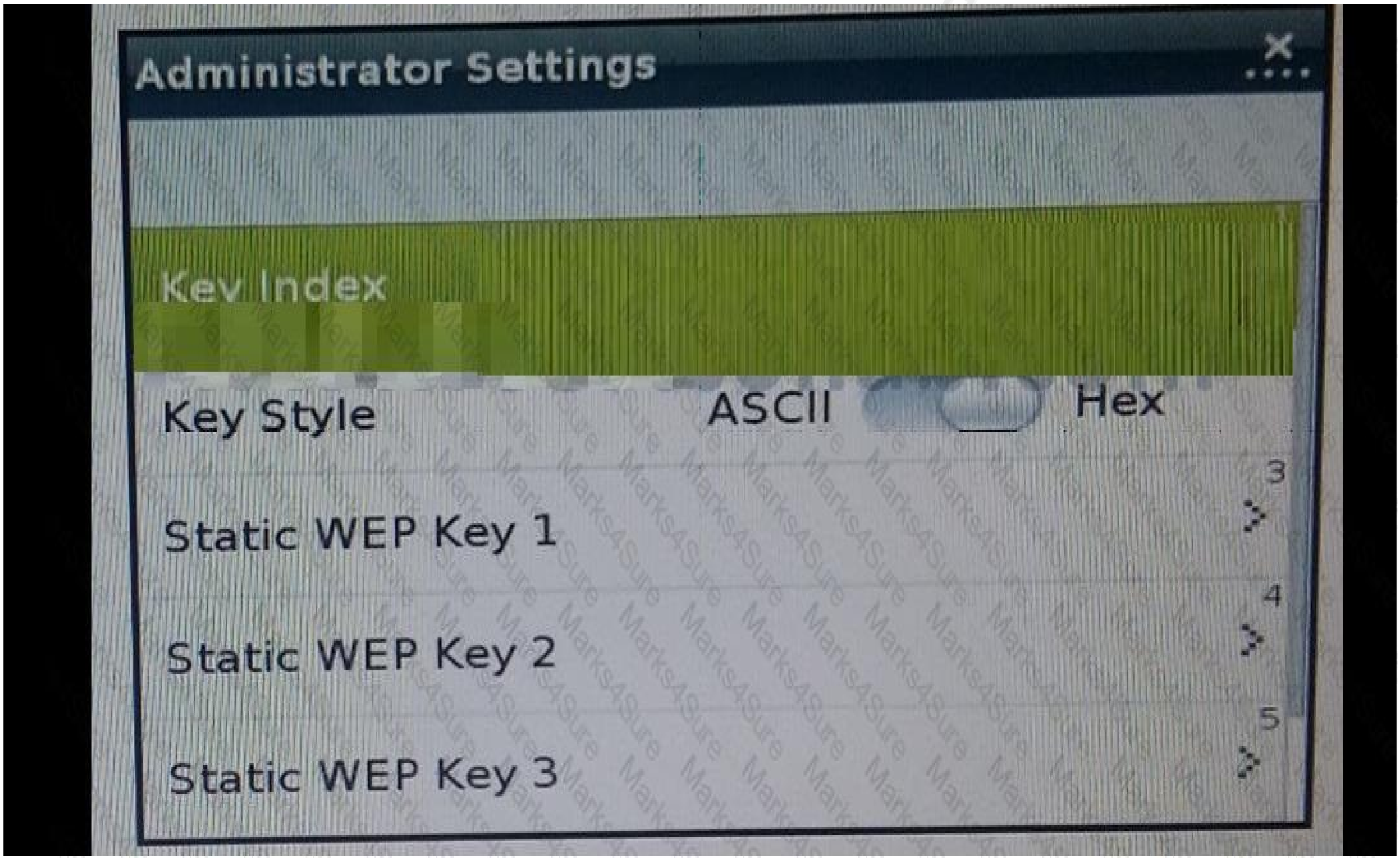
E. c=IN IP4 0.0.0.0

F. BYE WITH A = CALLHOLD

Answer: B C E

Question #:115 - (Exam Topic 1)

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

Answer: A F

Question #:116 - (Exam Topic 1)

Refer to the Cisco Unified Communication Manager configuration descriptions below. When a call is made from phone A line 1 to 30001, using line 1, which route pattern is chosen by Cisco Unified Communication Manager?

Phone A device calling search space is CSS_Dev_A

Phone A line 1 is assigned calling search space CSS_Line_A

Route Pattern 30XXX is placed in Partition Part_1

Route Pattern 3XXXX is placed in Partition Part_2

Route Pattern 300XX is placed in Partition Part_3

CSS_Dev_A contains partition(s) Part_1

CSS_Line_A contains partition(s) Part_2

A. 300XX in partition Part_3

B. 3XXXX in partition Part_2

C. 30XXX in partition Part_1

D. No match exists and the user receives a reorder tone

Answer: C

Question #:117 - (Exam Topic 1)

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

A. 60 seconds

B. 1 seconds

C. 1440 seconds

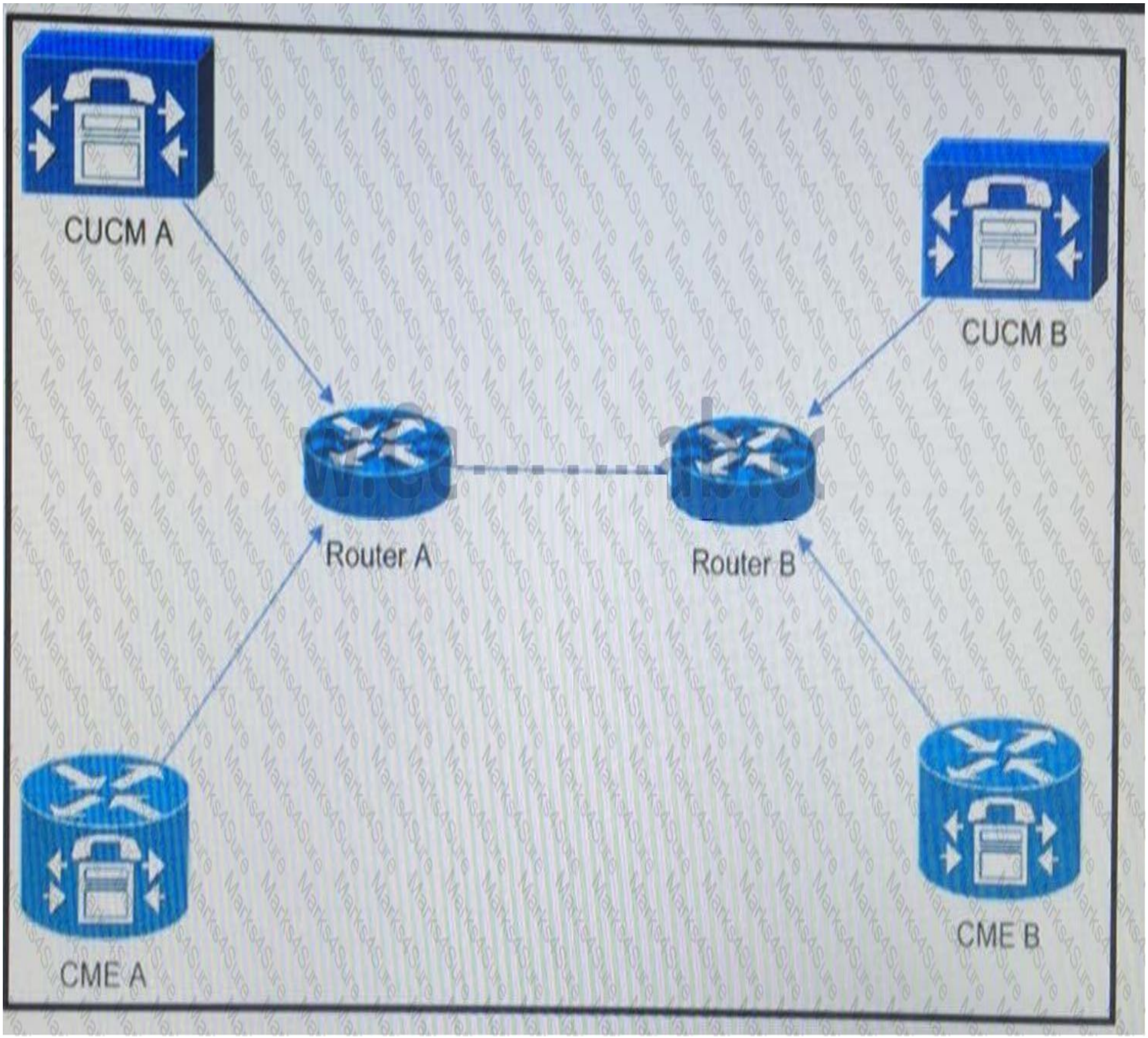
D. 600 seconds

E. 3600 seconds

Answer: A

Question #:118 - (Exam Topic 1)

Refer to the exhibit.



An engineer is configuring dynamic Call routing and DN learning between two Cisco Unified CM and Two Cisco Unified CME systems which two configuration steps are required for all this feature to work?

(Choose two)

A. Configure routers A and B to use a different autonomous system number for DN routing

B. Configure routers A and B to use EIGRP for IP Routing

C. Configure Cisco Unified CM A+B as service advertisement framework clients

D. Configure router A and B to use OSPF for IP Routing

E. Configure Cisco Unified CME A+B as service advertisements forwarders

F. Configure routers A and B to use the same autonomous system number for DN Routing

Answer: C F

Question #:119 - (Exam Topic 1)

Refer to the exhibit.

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

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/+\0\1/ to /+1\0\1\2/

B. edit rule 1 and change \([27]..\) to (7+.\)

C. edit rule 3 and change /+1\1\2/ to /+\1\2/

D. edit rule 2 and change /+\0\2/ to /+1\0\1\2/

E. edit rule 2 and change \([567][456].\) to \([5-7][4-6].\)

F. edit rule 4 and change \(.*\) to \(776\)\(.*\)

G. edit rule 1 and change /+\1\2/ to /+1\1\2/

H. edit rule 3 and change \(7+ \) to ([27]..\)

Answer: D F G H

Question #:120 - (Exam Topic 1)

The Cisco Cal Manager service is activated and running in the publisher node in a Cisco Unified

Communication Manager cluster. Which service is responsible for transferring the Call Detail Record flat file to the cdr_repository structure on the Publisher?

A. Cisco CAR Scheduler

B. Cisco CAR DB

C. Cisco CDR Repository Manager

D. Cisco CDR Agent

E. Cisco CallManager

Answer: D

Question #:121 - (Exam Topic 1)

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.

Answer: B

Question #:122 - (Exam Topic 1)

Which three statements about configuring partitioned intradomain federation to Lync are true? (Choose three.)

A. Microsoft RCC must be enabled

B. The enable use of Email Address when federating option can be turned on if SIP URI's are different between IM&P and Lync.

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. Intradomain Federation to Lync is only possible using SIP

E. A static route must be added to point the local presence domain to the Lync server.

F. IM&P and Lync should federate to any required remote domains.

Answer: C D E

Question #:123 - (Exam Topic 1)

A cisco Unified CM user is set up with one remote destination profile that has two remote destination numbers.

First destination number is the user's mobile phone in country A and the Second is a mobile phone located in country B. All outbound calls are centralized from the gateway at country A. The user reports that inbound calls are properly routed to the mobile phone as long as the user is in country A. but inbound calls are not successfully routed to country B?

What could resolve this issue?

(Choose two)

A. The enable mobile connect option must be selected under the user's second remote destination number

B. The value of remote destination limits should be change to 2 instead of the default value of 4 under the end user page

C. The enable mobile voice access option must be selected under the end user page

D. The value of maximum wait time for desk pickup should be change 20000 instead of the default of 10000, under the end user page

E. The rerouting calling search space assigned to the user's remote destination profile must have access to international calls

Answer: A E

Question #:124 - (Exam Topic 1)

Which two services must be enabled on the routing servers when configuring Partitioned Intradomain Federation? (Choose two.)

A. Cisco XCP Directory Service

B. Cisco XCP Router

C. Cisco Presence Engine

D. Cisco XCP SIP Federation Connection Manager

E. Cisco XCP Connection Manager

F. Cisco SIP Proxy

Answer: B C

Question #:125 - (Exam Topic 1)

Exhibit:

```
Gatekeeper#show gatekeeper endpoint

                    GATEKEEPER ENDPOINT REGISTRATION
                    ================================
CallSignalAddr  Port  RASSignalAddr   Port  Zone Name        Type    Flags
--------------- ----- --------------- ----- ---------        ----    -----
10.1.1.1        1720  10.1.1.1        49960  GK              VOIP-GW
    H323-ID: HQGK_1
    Voice Capacity Max.=  Avail.=  Current.=
10.1.1.2        1720  10.1.1.2        49309  GK              VOIP-GW
    H323-ID: HQGK_2
    Voice Capacity Max.=  Avail.=  Current.= 0
20.1.1.1        1720  20.1.1.1        49262  GK              H323-GW
    H323-ID: RemoteGK-1
    Voice Capacity Max.=  Avail.=  Current.= 0
Total number of active registrations = 3
```

10.1.1.1 and 10.1.1.2 are node IP addresses of a Cisco Unified CM cluster. Which statement describes the correct Cisco Unified CM configurations that produced the output shown in 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 NO:.

Answer: B

Question #:126 - (Exam Topic 1)

A collaboration engineer has been asked to implement secure real-time protocol between a Cisco Unified CM and SIP gateway. Which option is a consideration for this implementation?

A. Only T.38 and Cisco fax protocol are supported

B. SIP require the all the time be sent in GMT

C. Call hold RE-INVITE is not supported

D. SRTP is supported only in cisco IOS 15.x and higher

Answer: B

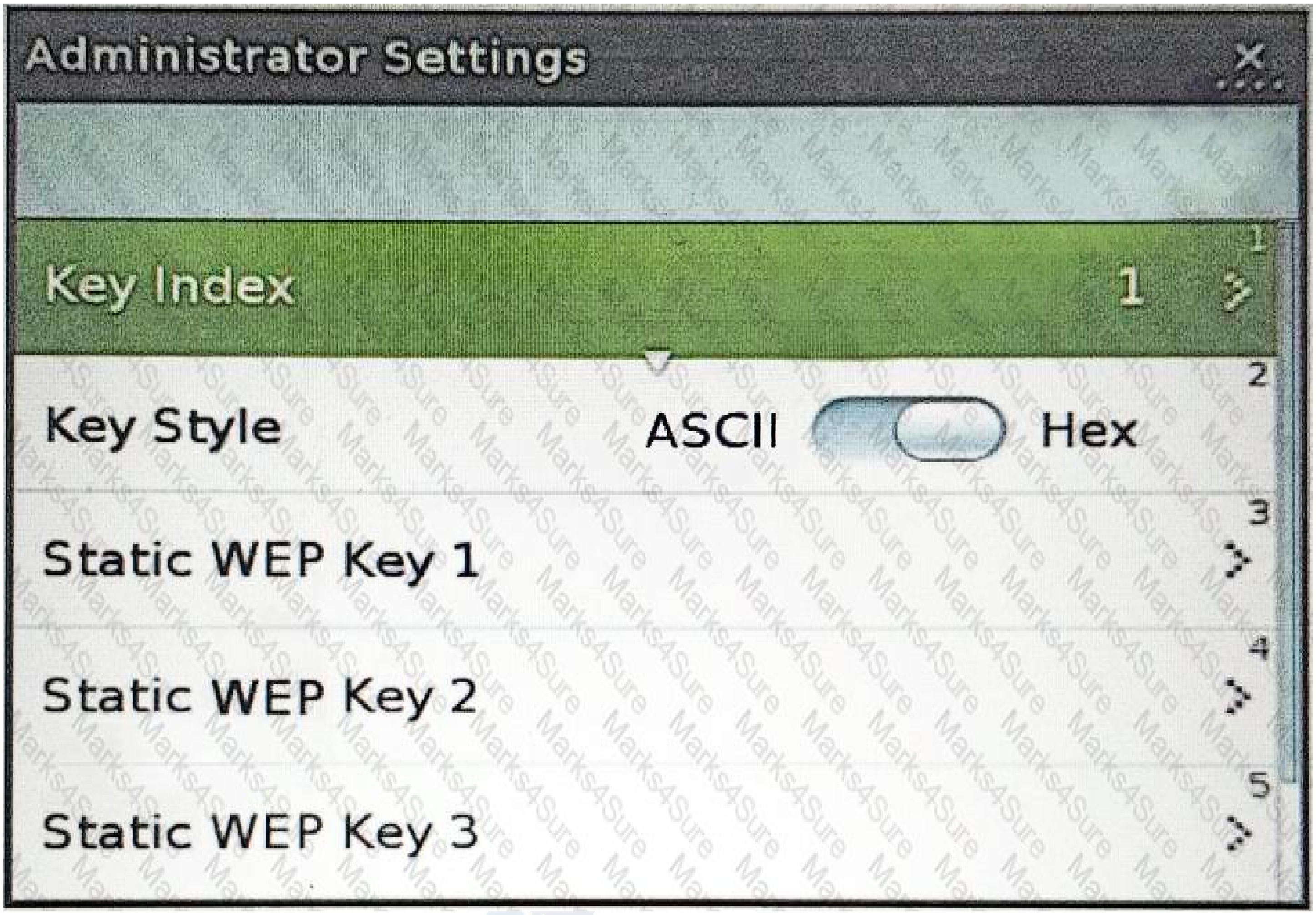
Explanation

As necessary, configure the router to use Greenwich Mean Time (GMT). SIP requires that all times be sent in GMT. SIP INVITE messages are sent in GMT. However, the default for routers is to use Coordinated Universal Time (UTC). To configure the router to use GMT, issue the clock timezone command in global configuration mode and specify GMT.

http://www.cisco.com/c/en/us/td/docs/ios/voice/cube/configuration/guide/vb_book/vb_book/vb_8240.html

Question #:127 - (Exam Topic 1)

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

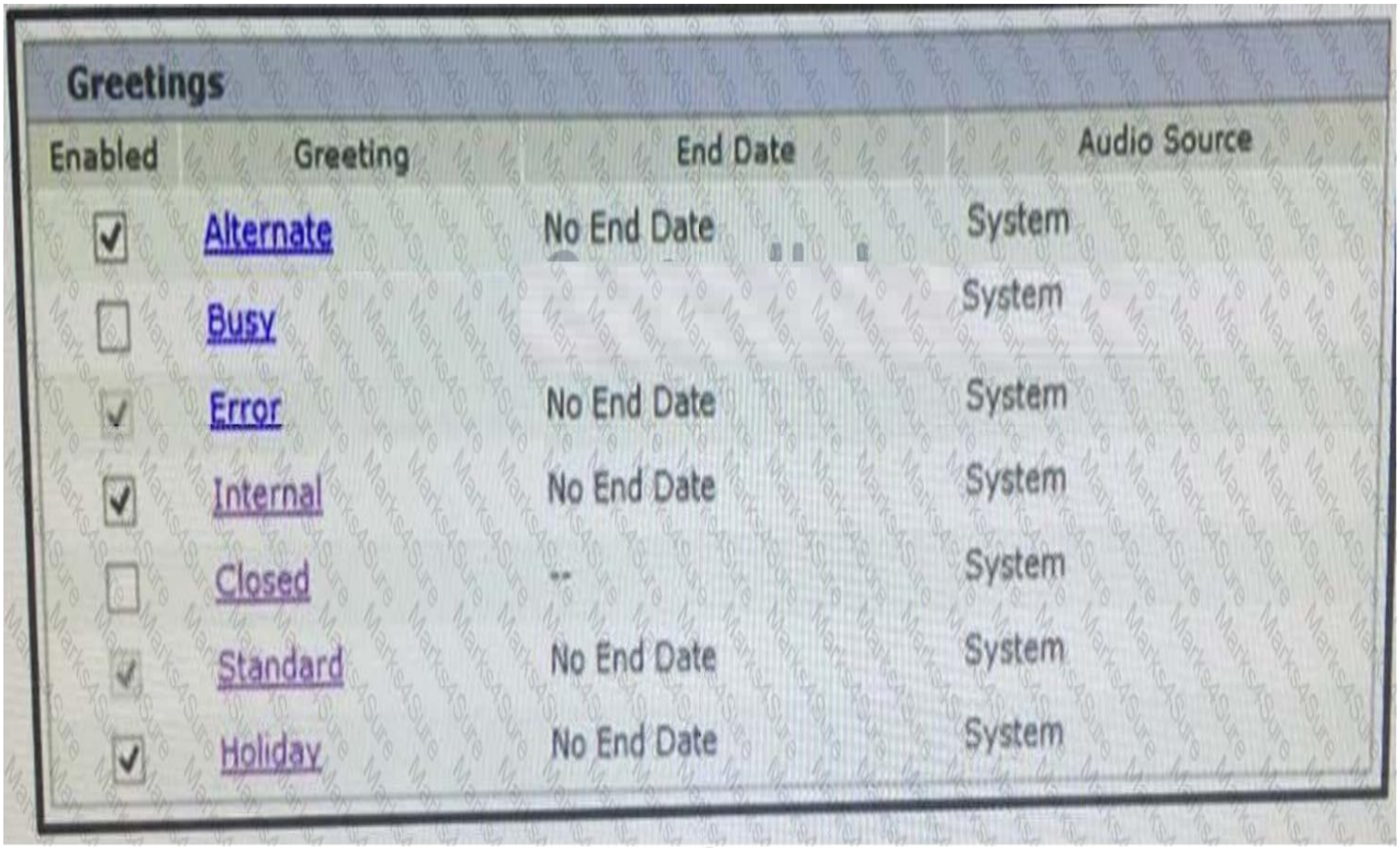
E. LEAP

F. Open with WEP

Answer: A F

Question #:128 - (Exam Topic 1)

Refer to the exhibit.



Greetings

Enabled	Greeting	End Date	Audio Source
☑	Alternate	No End Date	System
☐	Busy		System
☑	Error	No End Date	System
☑	Internal	No End Date	System
☐	Closed	--	System
☑	Standard	No End Date	System
☑	Holiday	No End Date	System

A voicemail administrator was asked to create a call handler for the sales department with

the following requirements,

- No end date for any of the configured greetings

- Play a specific greeting on business-approved days off

- Play a specific greeting when a sales agent calls the call handler number

- After creating the call handler and making some test calls only the default system greeting is heard

Which four configuration changes are needed to company with this business request? (Choose Four)

A. Disable the Alternate Greeting under Call Handler Greetings

B. Create a new closed schedule and assign it to the sales Call Handler

C. Record a new Greeting and assign it to the Alternate Greeting

D. Record a new Greeting and assign it to the Holiday Greeting

E. Record a new Greeting and assign it to the Internal Greeting

F. Create a new holiday schedule to be used by the Holiday Greeting

G. Disable the Internal Greeting under Call Handler Greeting

H. Enable the Closed Greeting under Call Handler Greetings

Answer: A D F G

Question #:129 - (Exam Topic 1)

"Parse ITL FILE"

"This etoken was used to sign the ITL file"

"The ITL file was verified successfully"

Refer to the exhibit. Which description of the Cisco Unified Communications Manager Cluster that produced the identity trust List records is true?

A. IP phones registered to this cluster have ITL and CTL files installed

B. it is a mixed-mode cluster that uses tokenless solution

C. it is a mixed-mode cluster that uses hardware USB token

D. IP phones registered to this cluster have only ITL files installed

E. IP phones registered to this cluster have neither ITL nor CTL files installed

Answer: D

Question #:130 - (Exam Topic 1)

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

Answer: A D

Question #:131 - (Exam Topic 1)

Refer to the exhibit.

```
Gatekeeper#show gatekeeper gw

GATEWAY TYPE PREFIX TABLE
=========================
Prefix: 1*
  Zone GK master gateway list:
    10.1.1.2:49392 HQGK_2
    10.1.1.1:50972 HQGK_1
```

10.1.1.1 and 10.1.1.2 are node IP addresses of a Cisco Unified CM cluster. 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? (Choose two.)

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.

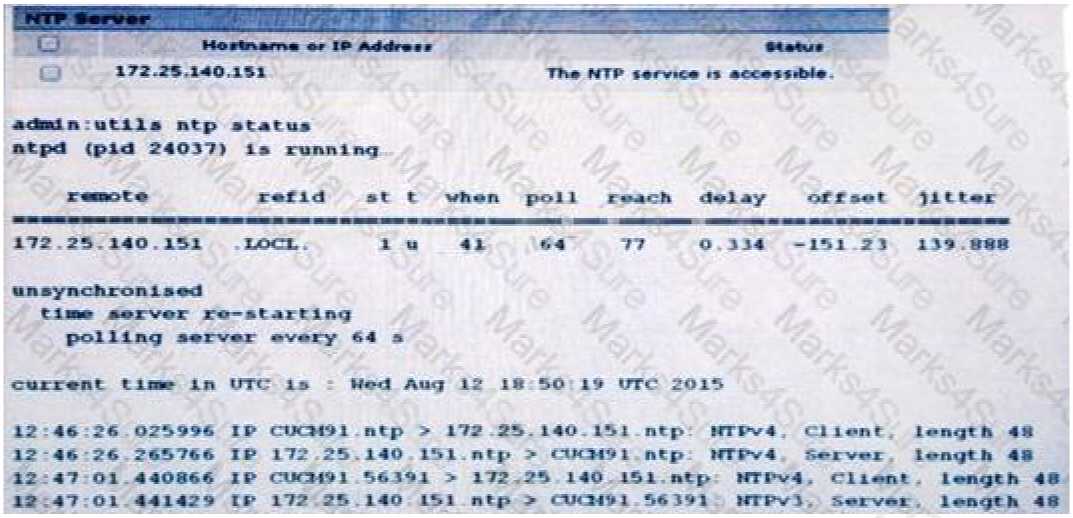
E. Technology Prefix is 1*.

F. Zone name is HQGK.

Answer: B E

Question #:132 - (Exam Topic 1)

Refer to the exhibit.



A network engineer is troubleshooting a NTP synchronized issue in CUCM. Why is NTP unsynchronized?

A. The NTP server used is a Windows based NTP server.

B. The IOS Command NTP server 172.25.140.151 version 3 is advertising NTPv3.

C. The NTP server stratum is higher than four.

D. A firewall is blocking NTP port 123.

Answer: B

Question #:133 - (Exam Topic 1)

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

Answer: C F

Question #:134 - (Exam Topic 1)

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-codec** 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.

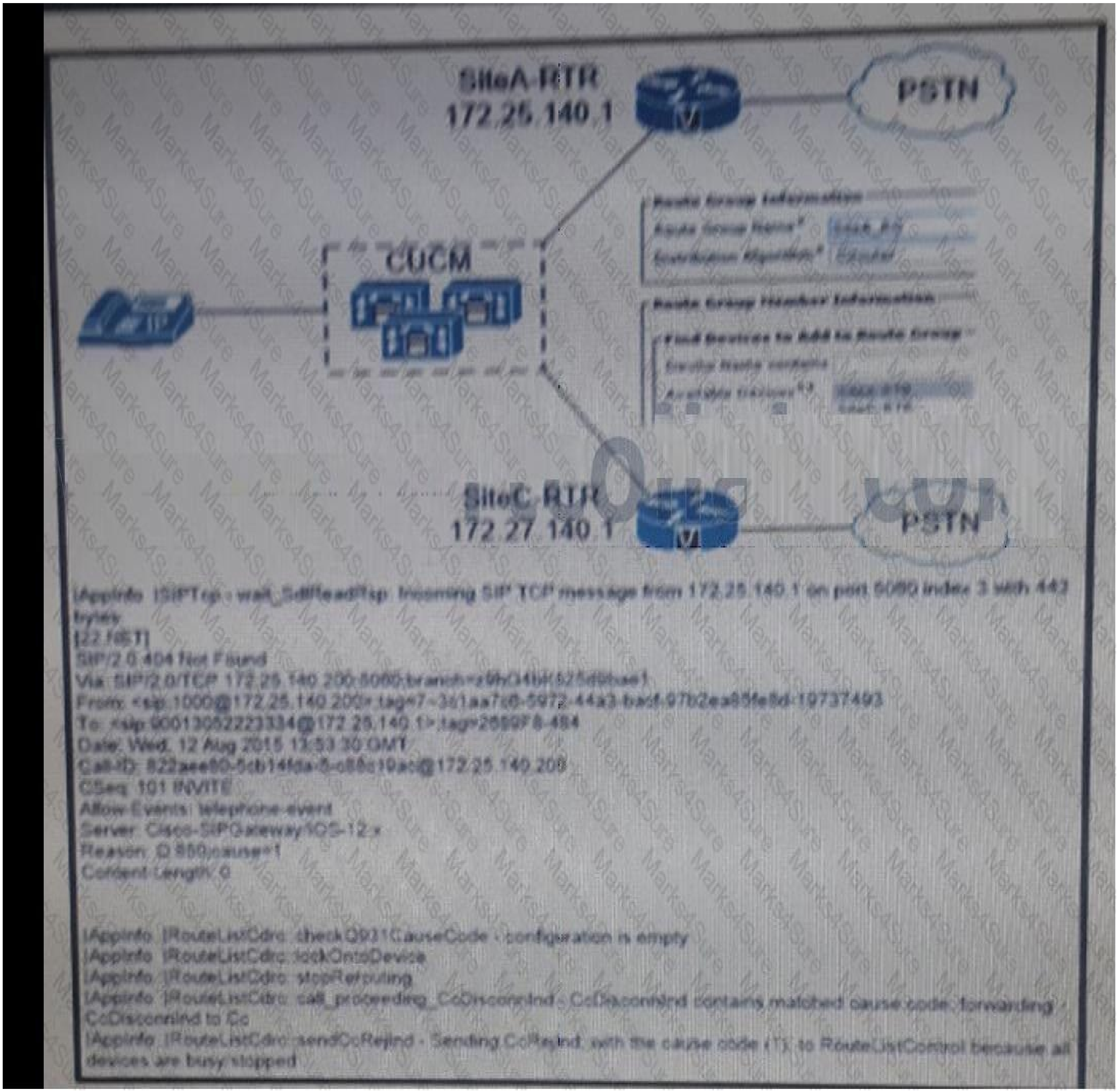
E. Configure the **midcall-signaling preserve-codec** command under each outbound ITSP dial peer.

F. Configure the **midcall-signaling passthru media-change** command under **voice service voip.**

Answer: A D

Question #:135 - (Exam Topic 1)

Refer to the exhibit.



A network engineer is troubleshooting a call routing issue where failed calls on the primary path (SiteA-RTR) was not sent to the secondary path (SiteC-RTR). Why is CUCM unable to extend the call setup through SiteC-RTR?

A. Stop routing on Q.931 Disconnect cause code is set to 27

B. Stop routing on Unallocated Number Flag is set to true

C. Stop Routing on User Busy Flag is set to true

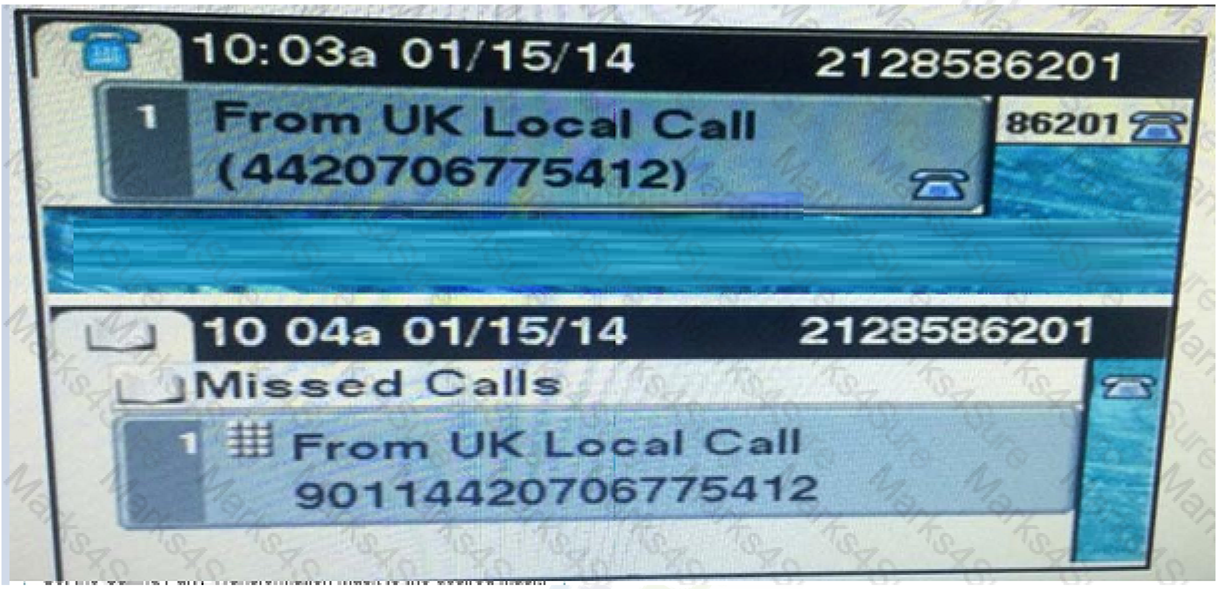
D. Retry count for SIP Invite is set to 1

E. Retry count for SIP response is set to 1

Answer: B

Question #:136 - (Exam Topic 1)

Refer to the exhibit.



A CUCM engineer is working with Globalization and localization on H323 gateway. Which four configuration changes are needed to achieve the result on the exhibit? (Choose four)

A. Create as CSS and PT for calling party transformation pattern

B. Create a transformation profile and add 9011 in the international number prefix field

C. Assign a transformation profile in the incoming transformation profile setting in the E 164 transformation number prefix field

D. Assign the calling party transformation CSS to the device pools in the cluster

E. Uncheck the use device pool calling party transformation CSS on all the phones

Answer: A B C D

Question #:137 - (Exam Topic 1)

A CUCM engineer has deployed Type SIP Phones on a remote site and no SIP dial rules were deployed for these phones. How will CUCM receive the DTMF after the phone goes off- hook and the buttons are pressed?

A. Each digit will be received by CUCM in a SIP NOTIFY message as soon as they are pressed.

B. The first digit will be received in a SIP INVITE and subsequent digits will be received using notify MASSAGE as soon as they are pressed.

C. Each digit will be received by CUCM in a SIP INVITE as soon as the dial softkey has been pressed.

D. All digits will be received by CUCM in a SIP INVITE as soon as the dial softkey has been pressed

Answer: B

Question #:138 - (Exam Topic 1)

A Call is made between two desk phones enabled with single number reach that are registered to a cisco unified CM cluster. The device pool for each device has a local route group defined. When the call is placed to exit the system, which device pool control the destination gateway?

A. Destination RDP

B. Source Phone

C. Source RDP

D. Destination phone

Answer: B

Question #:139 - (Exam Topic 1)

Under which three conditions will a Cisco 9971 IP Phone request the "xmldefault.cnf.xml" file from TFTP server in a Cisco Unified CM cluster? (Choose three)

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

Answer: B D F

Question #:140 - (Exam Topic 1)

Which three basic elements does the architecture for voice and video over WLAN include? (Choose three.)

A. Wireless LAN controllers.

B. Wireless access points.

C. Wireless bridge

D. Wireless VLAN

E. Wired call elements

F. DHCP server

Answer: A B D

Question #:141 - (Exam Topic 1)

Which two descriptions of +E.164 and enterprise alternate number for directory numbers in Cisco Unified communications Manager 10.6 are true(choose two)

A. They cannot be advertised as PSTN fail over number

B. They can be added into local partition

C. If the number mask is not configured, the alternate number is invalid

D. They cannot be added into local partition

E. They are not eligible to be advertised using Global Dial Plan Replication

F. If the number mask it is not configured, use DN as alternate number

Answer: B F

Question #:142 - (Exam Topic 1)

Which option is a mandatory LDAP attribute for a user to be synchronized to Cisco Unified Communications Manager?

A. uid

B. telephone Number

C. employee Number

D. sn

E. mail

Answer: D

Question #:143 - (Exam Topic 1)

"Login failed due a configuration error with your phone and JTAPI or Unified CM Contact your Administrator"

Refer to the exhibit. Which two of the following are possible causes of the error message when an agent attempted to log into the Cisco Agent Desktop

A. The Resource is not Available under Cisco Desktop Administrator

B. The RMCM is stuck in the initializing state

C. The MAC of the agent phone is not associated with RMCM application user on the Cisco Unified Communication Manager

D. THE IPCC extension is not associated with the end user

E. An incorrect extension was entered by the agent while logging onto Cisco Agent Desktop

Answer: B C

Question #:144 - (Exam Topic 1)

Which two statements about the Peer Firmware Sharing option for IP phone firmware distribution are true? (Choose two.)

A. The option must be enabled on Cisco Unified Communications Manager service parameters for Cisco TFTP.

B. This option allows falling back to the TFTP server in the Cisco Unified Communications Manager cluster.

C. This option mandates that the parent phone and child phones be identical selected phone models

D. This option uses a parent-child hierarchy that must be manually defined by the Cisco Unified Communications Manager administrator.

E. This option allows firmware transfers between phones in different subnets as long as the round-trip delay is less than 5 milliseconds.

Answer: B C

Question #:145 - (Exam Topic 1)

Refer to the exhibit.

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.10.10.1:5060;branch=z9hG4bK246da4c33fa3e
From: <sip:1234@10.10.10.1>;tag=169850-fb41edd8-7bc7-4ced-b8b0-9b10a31db5c4
To: <sip:1001@10.10.10.251>;tag=42C10134-1BC8
Date: Sat, 24 Dec 2016 05:50:29 GMT
Call-ID: c071a100-35d10ff1-24695-ab4b420a@10.10.10.1
Server: Cisco-SIPGateway/IOS-15.2.4.M2.8
CSeq: 102 PRACK
Content-Length: 0
```

Which description of a User Agent Server that sends this message is true?

A. The UAS sends this message in response to an earlier PRACK received

B. The UAS sends this message in response to an earlier INVITE that contains PRACK

C. The UAS sends this message in response to an earlier ACK received

D. It is not possible for the UAS to send this message

E. The UAS sends this message in response to an earlier INVITE received

Answer: A

Question #:146 - (Exam Topic 1)

What is the maximum number of subclusters supported on a Cisco IM & Presence

A. 4

B. 3

C. 2

D. 5

E. 1

Answer: B

Question #:147 - (Exam Topic 1)

Refer to the Exhibit.

```
INVITE sip:95124182222@172.16.100.90:5060 SIP/2.0
Via: SIP/2.0/TCP 172.16.100.50:5060;branch=z9hG4bK12af194b3e2624
From: "Agent A" <sip:5124183001@172.16.100.50>;tag=1001843~825a8e37-305d-403f-9a76-ee5343b3b431-61539586
To: <sip:95124182222@172.16.100.90>
Date: Tue, 10 Mar 2015 14:25:03 GMT
Call-ID: 3c748f00-4fe1febf-35ff9-2bef12ac@172.16.100.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM9.1
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Cisco-Guid: 1014271744-0000065536-0000221138-0737088172
Session-Expires:  1800
P-Asserted-Identity: "Agent A" <sip:5124183001@172.16.100.50>
Remote-Party-ID: "Agent A" <sip:5124183001@172.16.100.50>;party=calling;screen=yes;privacy=off
Contact: "Agent A" <sip:5124183001@172.16.100.50:5060;transport=tcp>;isFocus
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 202

v=0
o=CiscoSystemsCCM-SIP 2371 1 IN IP4 172.16.100.50
s=SIP Call
c=IN IP4 172.16.100.50
t=0 0
m=audio 24582 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

An agent initiated a video call but was establish as audio only. The support engineer

collected and analysed the Cisco Unified CM traces. Which two options caused this problem? (Choose two)

A. A hardware MTP was assigned to the call

B. SIP Notify DTMF was requested and negotiated

C. MTP required was checked on the SIP Trunks

D. Use Trusted Relay Point is set on one of the phone

E. MRGL assigned to phones with Trusted Relay Point

Answer: D E

Question #:148 - (Exam Topic 1)

Refer to the exhibit.

```
Jul 18 05:00:21.834: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
NOTIFY sip:1001@10.10.10.31:50800;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.10.10.254:5060;branch=z9hG4bKE698F
From: <sip:1001@10.10.10.254>;tag=194EE3C-596
To: <sip:1001@10.10.10.31>
Call-ID: D62DE77B-6AAC11E7-8120BED4-64A83368@10.10.10.254
CSeq: 101 NOTIFY
Max-Forwards: 70
Date: Tue, 18 Jul 2016 05:00:21 GMT
User-Agent: Cisco-SIPGateway/IOS 15.2.4.M5
Event: message-summary
Subscription-State: active
Contact: <sip:1001@10.10.10.254:5060;transport=tcp>
Content-Type: application/simple-message-summary
Content-Length: 23

Messages-Waiting: yes
```

Which description of the event captured in the SIP message on a Cisco Unified

Communication Manager Express router with Cisco Unified Express and registered IP phones (SIP and SCCP)is true?

A. The Cisco UCM Express router notifies a SIP phone to turn on its MWI

B. Cisco Unified Express notifies the Cisco UCM Express router to turn on MWI for a sip IP phone

C. The Cisco UCM Express router hairpins SIP message to itself to notify an SCCP IP phone to turn on MWI

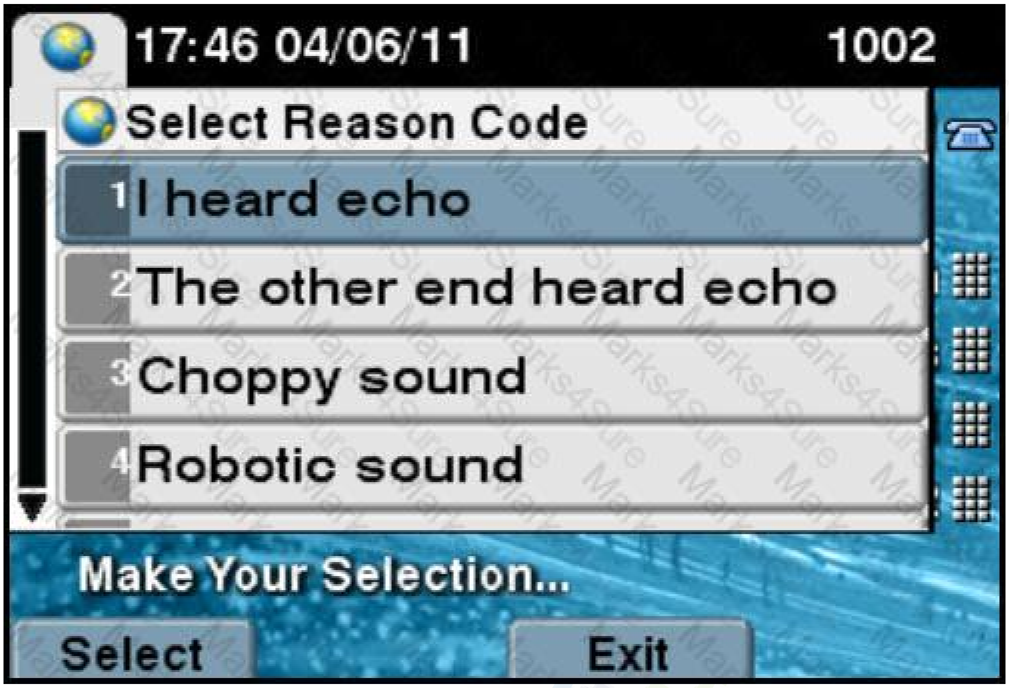
D. The Cisco UCM Express router notifies Cisco Unified Express that MWI has been turned on for a SIP phone

E. Cisco Unified Express notifies a SIP IP phone to turn on its MWI

Answer: A

Question #:149 - (Exam Topic 1)

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

Answer: D

Question #:150 - (Exam Topic 1)

The UCM 9.0 publisher server in a five node cluster failed to boot after a power outage. Which 4 configuration modifications could still be made on the remaining notes?

A. Device security profile

B. Device mobility

C. Busy Trigger

D. Add a new DN

E. PIN Reset

F. Extension mobility login

G. Message waiting indication

H. Call forward all

Answer: B F G H

Question #:151 - (Exam Topic 1)

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

Answer: A

Question #:152 - (Exam Topic 1)

In the Cisco Unified Mobile Voice Access feature on Cisco Unified Communications Manager, which two calling search space configuration options are significant in routing the second stage call after the user is authenticated and dialed digits are collected? (choose two)

A. Concatenation of a remote destination profile CSS and line CSS

B. calling number transformation CSS of the remote destination

C. remote destination profile rerouting CSS

D. phone line level CSS of the mobility user

E. trunk or gateway inbound CSS

F. mobility user's phone device level CSS

Answer: A D

Question #:153 - (Exam Topic 1)

Auto-registartion is enabled on a mixed-mode Cisco Unified Communications Manager with software version 11.0 A new Cisco 7963 SCCP IP Phone just downloaded the default phone configuration file from the TFTP server in the cluster. Which result true?

A. The IP Phone does not attempt to auto-register

B. The IP phone attempts to auto-register with the highest priority Cisco UCM node on the default phone configuration file, and the registration is successful

C. The IP phone attempts to auto-register with the highest priority Cisco UCM node on the default phone configuration file, but the registration is rejected

D. The IP phone attempts to auto-register with a Cisco UCM node UCM1, and the registration is rejected

E. The IP phone attempts to auto-register with the TFTP server, and the registartion is rejected

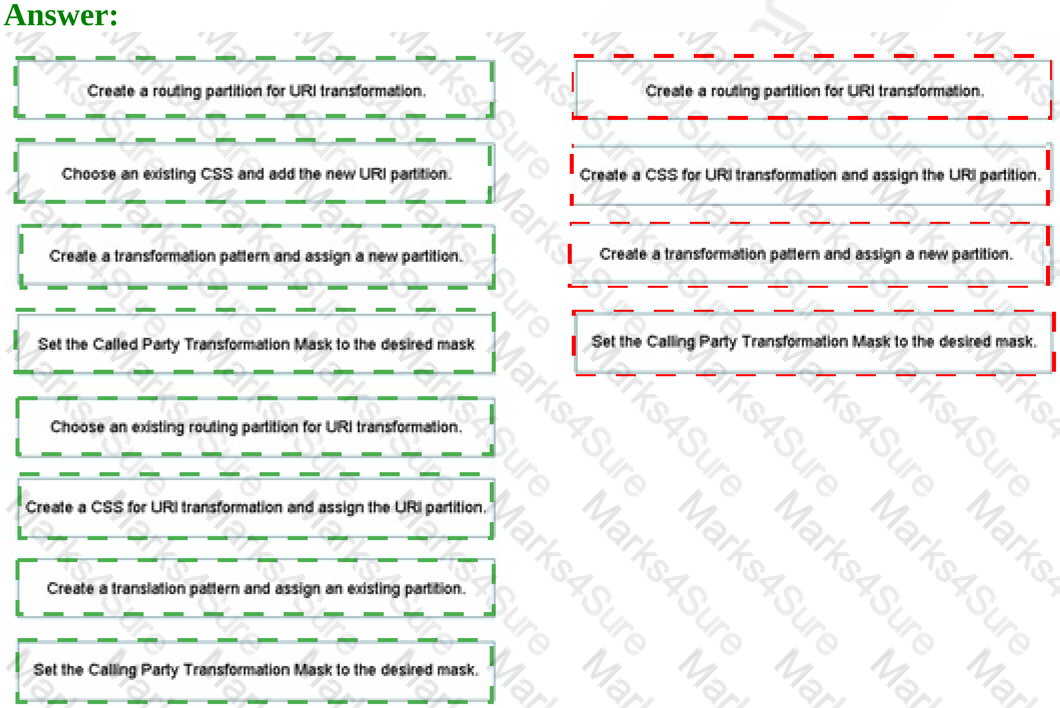
Answer: C

Question #:154 - (Exam Topic 1)

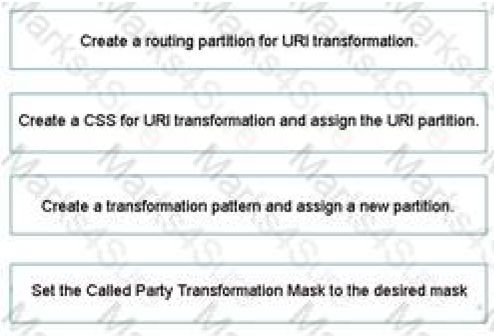
Drag the configuration steps on the left to the correct order for configuring digits transformation for

URI dialing according to CISCO best practices on the right. Not all options will be used.

Answer:

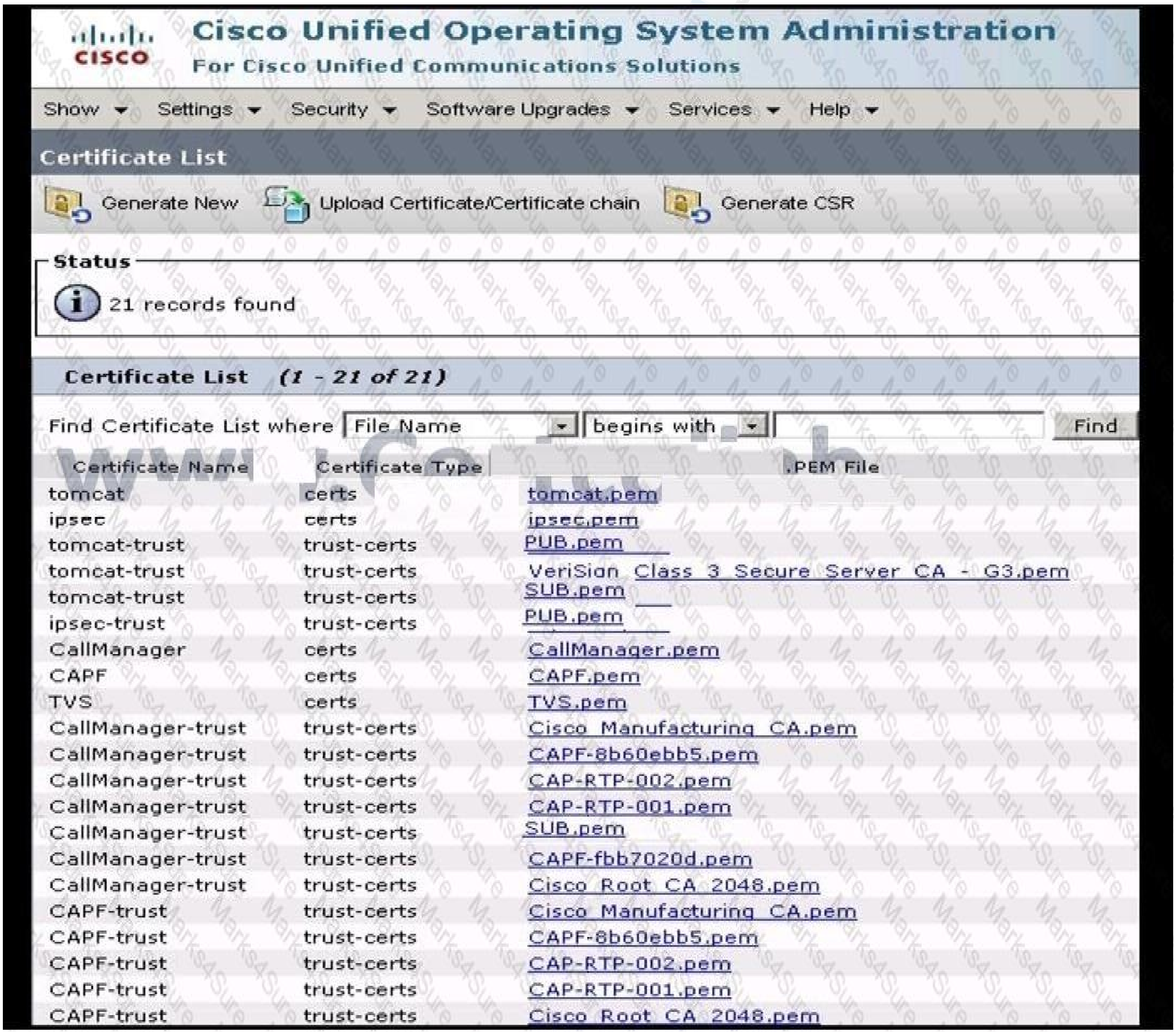


Explanation



Question #:155 - (Exam Topic 1)

Refer to the exhibit.



Which certificate file contains the private key used to sign the TFTP configuration file for download authentication with Initial Trust List enabled IP phones?

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

Answer: E

Question #:156 - (Exam Topic 1)

A voice engineer is trying to configure CUCM to satisfy the following customer codec requirements.

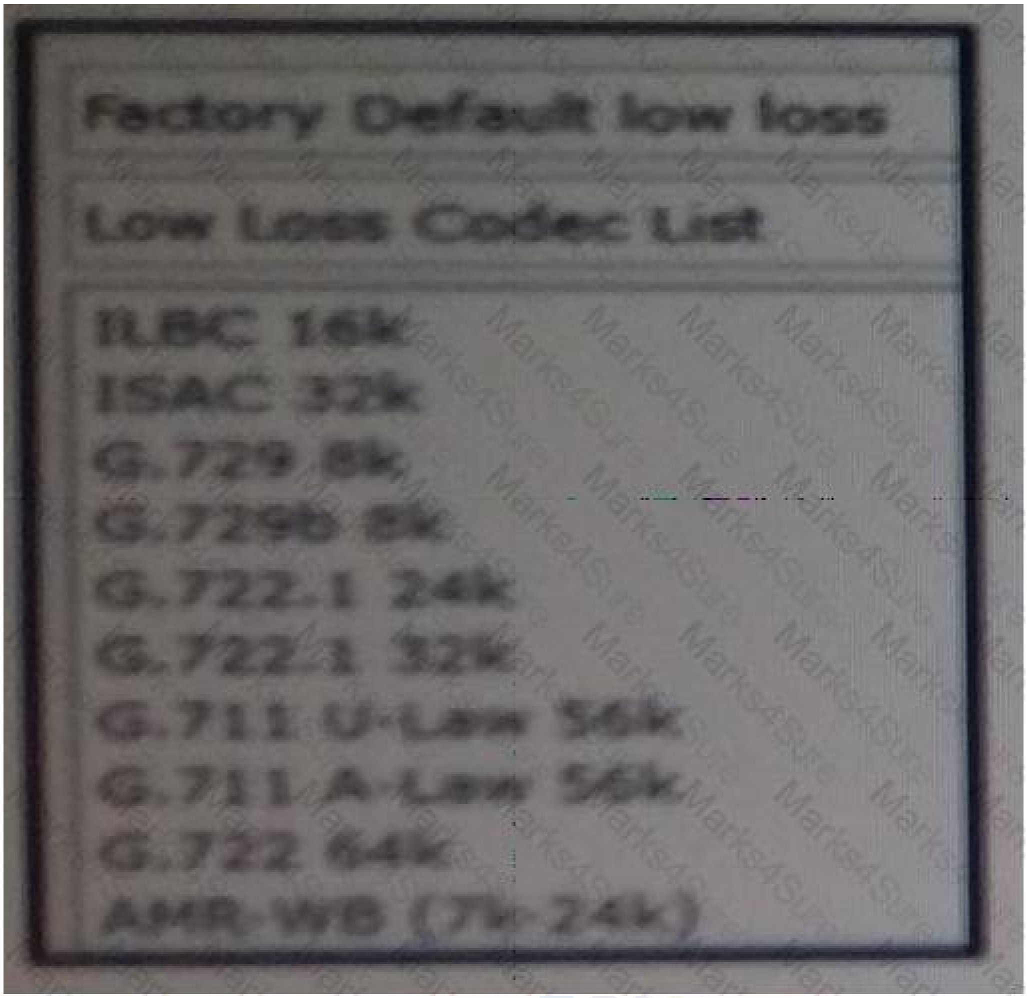
- All IP phones should use audio codec at less than 10K bandwidth and maintain voice quality

- The IP phone audio codec selected should be optimized for speech and music

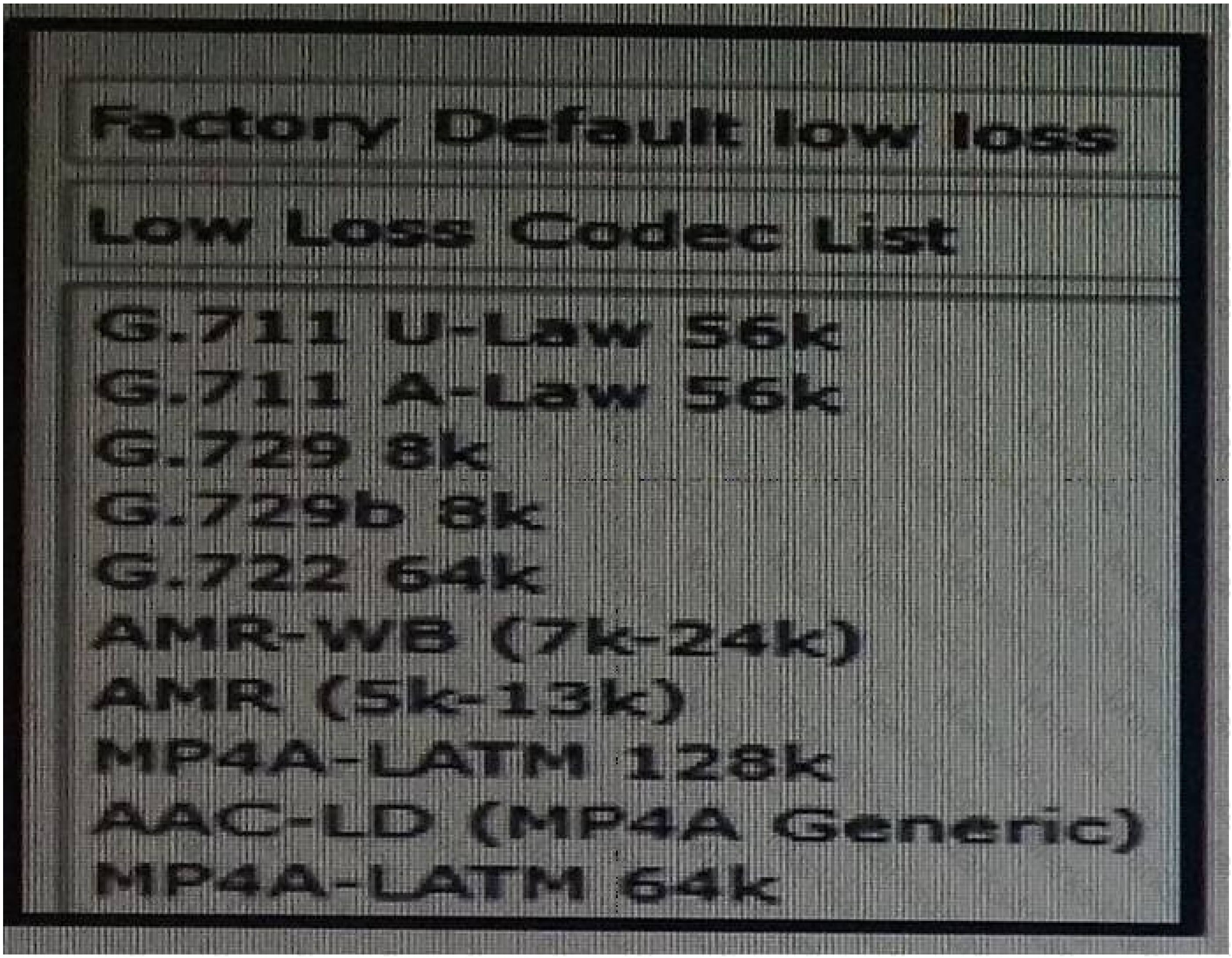
- The audio codec should be used in outbound SIP early offer signaling

Which configuration best satisfy the requirement?

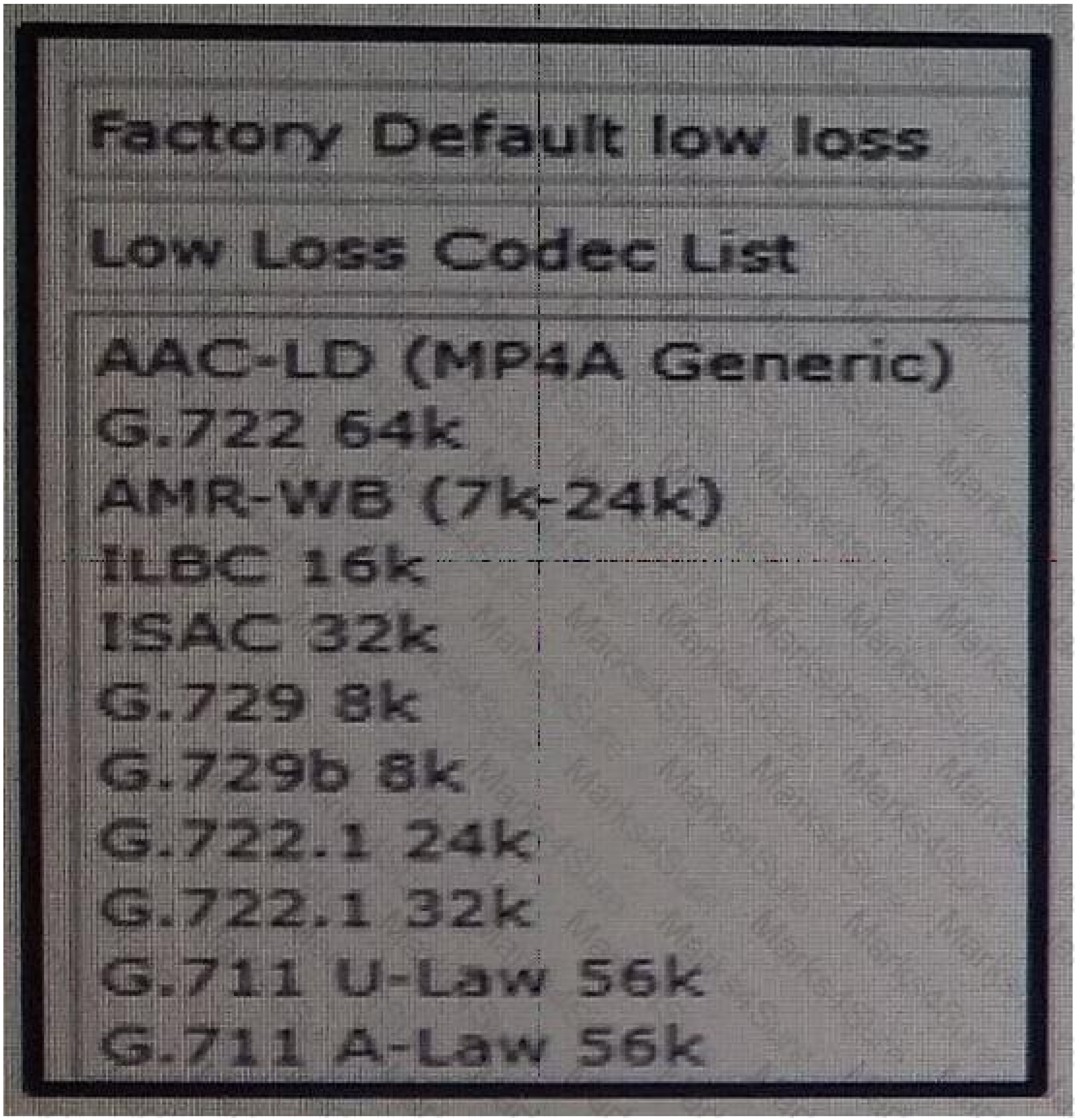
A)


Factory Default low loss
Low Loss Codec List
iLBC 16k
ISAC 32k
G.729 8k
G.729b 8k
G.722.1 24k
G.722.1 32k
G.711 U-Law 56k
G.711 A-Law 56k
G.722 64k
AMR-WB (7k-24k)

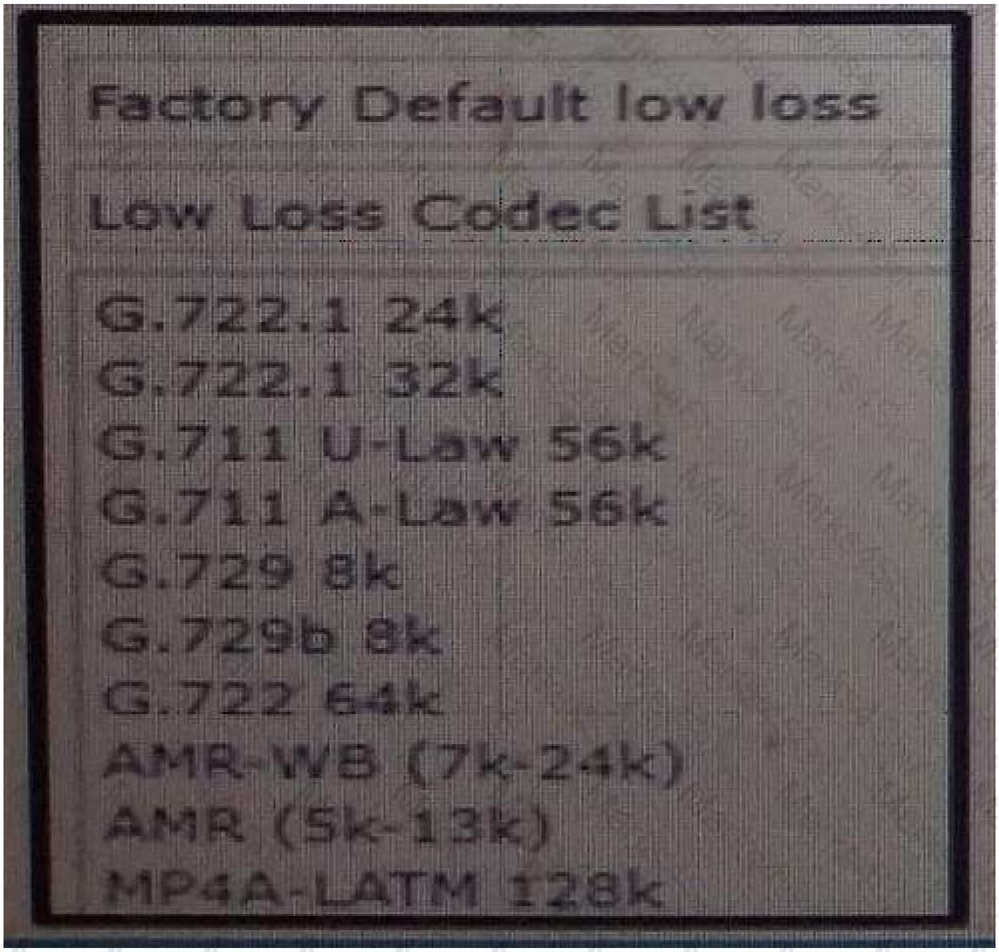
B)


Factory Default low loss
Low Loss Codec List
G.711 U-Law 56k
G.711 A-Law 56k
G.729 8k
G.729b 8k
G.722 64k
AMR-WB (7k-24k)
AMR (5k-13k)
MP4A-LATM 128k
AAC-LD (MP4A Generic)
MP4A-LATM 64k

C)


Factory Default low loss
Low Loss Codec List
AAC-LD (MP4A Generic)
G.722 64k
AMR-WB (7k-24k)
ILBC 16k
ISAC 32k
G.729 8k
G.729b 8k
G.722.1 24k
G.722.1 32k
G.711 U-Law 56k
G.711 A-Law 56k

D)



A. Exhibit A

B. Exhibit B

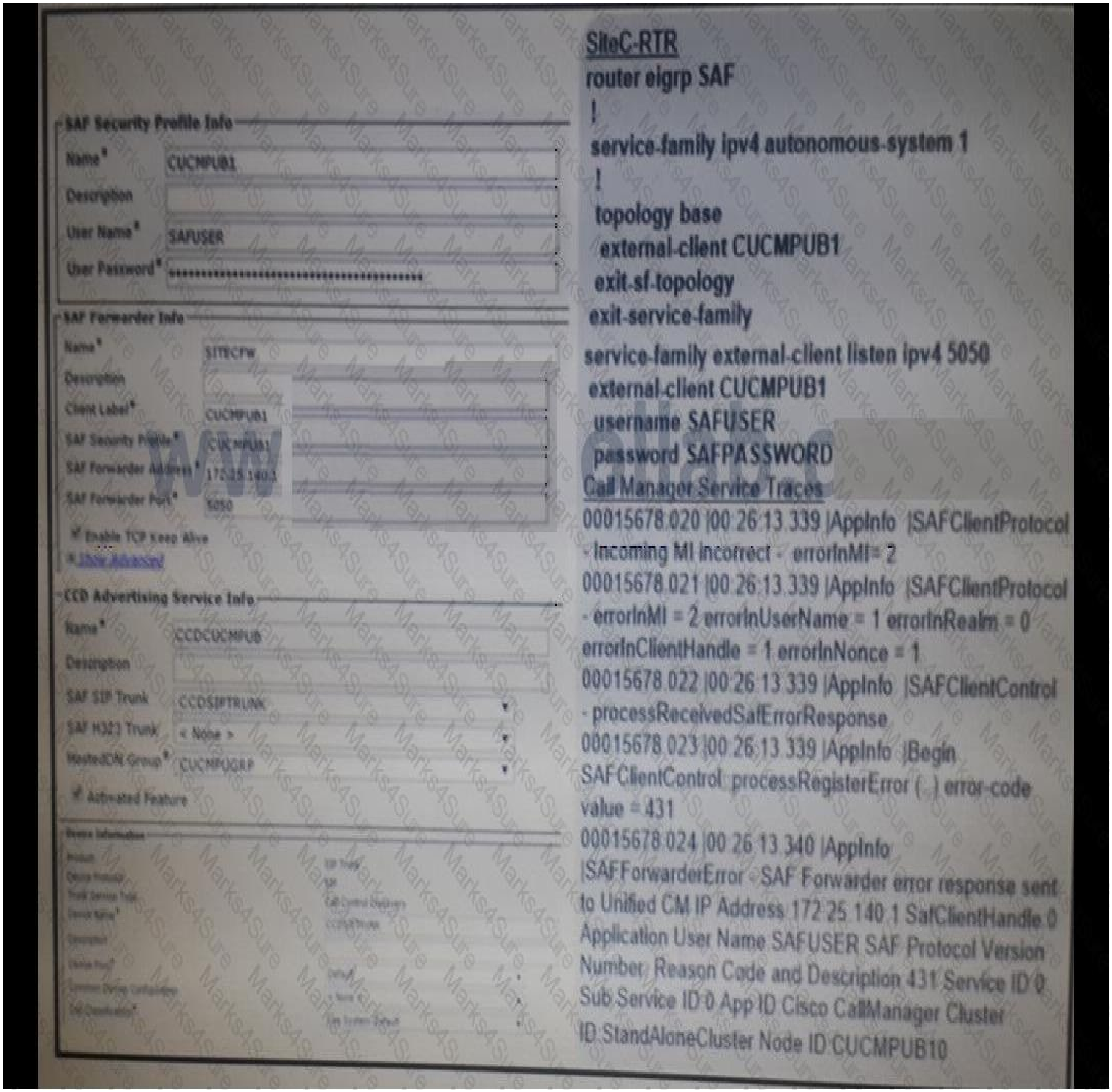
C. Exhibit C

D. Exhibit D

Answer: C

Question #:157 - (Exam Topic 1)

Refer to the exhibit.



A collaboration engineer using the Show eigrp service-family External-client IOS command, noticed that a CUCM failed to register as an external SAF Client on Cisco IOS router named Site CRTR. The engineer has collected snippets of the IOS configuration screenshots and CUCM trace shown in the exhibit.

What is the reason for the registration failure?

A. Password mismatch between CUCM and Router SAF configuration

B. Sf-interface loopback0 command missing under service-family ipv4 autonomous-system 1

C. SIP trunk IP address pointing to a different address than SAF Forwarder address

D. Name mismatch between SAF Forwarder name info field and external-client name on router

E. IP multicast-routing command missing on router configuration

Answer: B

Question #:158 - (Exam Topic 1)

Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 call-forward b2bua busy 2100
 call-forward b2bua noan 2200 timeout 20
 shared-line max-calls 5
 huntstop channel 4
!
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 3
 id mac 2222.2222.2222
 type 7965
 number 1 dn 1
!
```

IP phone 1 has the MAC address 11111.1111.1111, 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 phones 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.

Answer: B

Explanation

In shared line configuration phone share the same line so it is possible for any phone to answer the call.

Question #:159 - (Exam Topic 1)

Which two analog telephony signalling methods are most vulnerable to glare conditions? (Choose two.)

A. E & M Immediate - start

B. FXO Ground - start

C. E & M Feature Group D

D. E & M Delay - dial

E. E & M Wink start

F. FXS Loop-start

Answer: A F

Question #:160 - (Exam Topic 1)

SIP Provisional Response Acknowledgement is used in all of the following SIP 1XX responses except

A. 100 Trying

B. 183 Session In Progress

C. 182 Queued

D. 199 Early Dialog Terminated

E. 180 Ringing

Answer: A

Question #:161 - (Exam Topic 1)

An engineer received this requirement from a service provider. Diversion header should match the network DID "123456@company.com" 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:(.*>)" "123456@company.com>" request REINVITE sipheader

Diversion modify "sip:(.*>)" "123456@company.com>"

B. voice class sip-profiles 200

request INVITE sdp-header Diversion modify "sip:(.*>)" 123456@company.com> request REINVITE sdpheader

Diversion modify "sip:(.*>)" "123456@company.com>"

C. voice class sip-profiles 200

response 200 sdp-header Diversion modify "sip:(.*>)" "123456@company.com>"

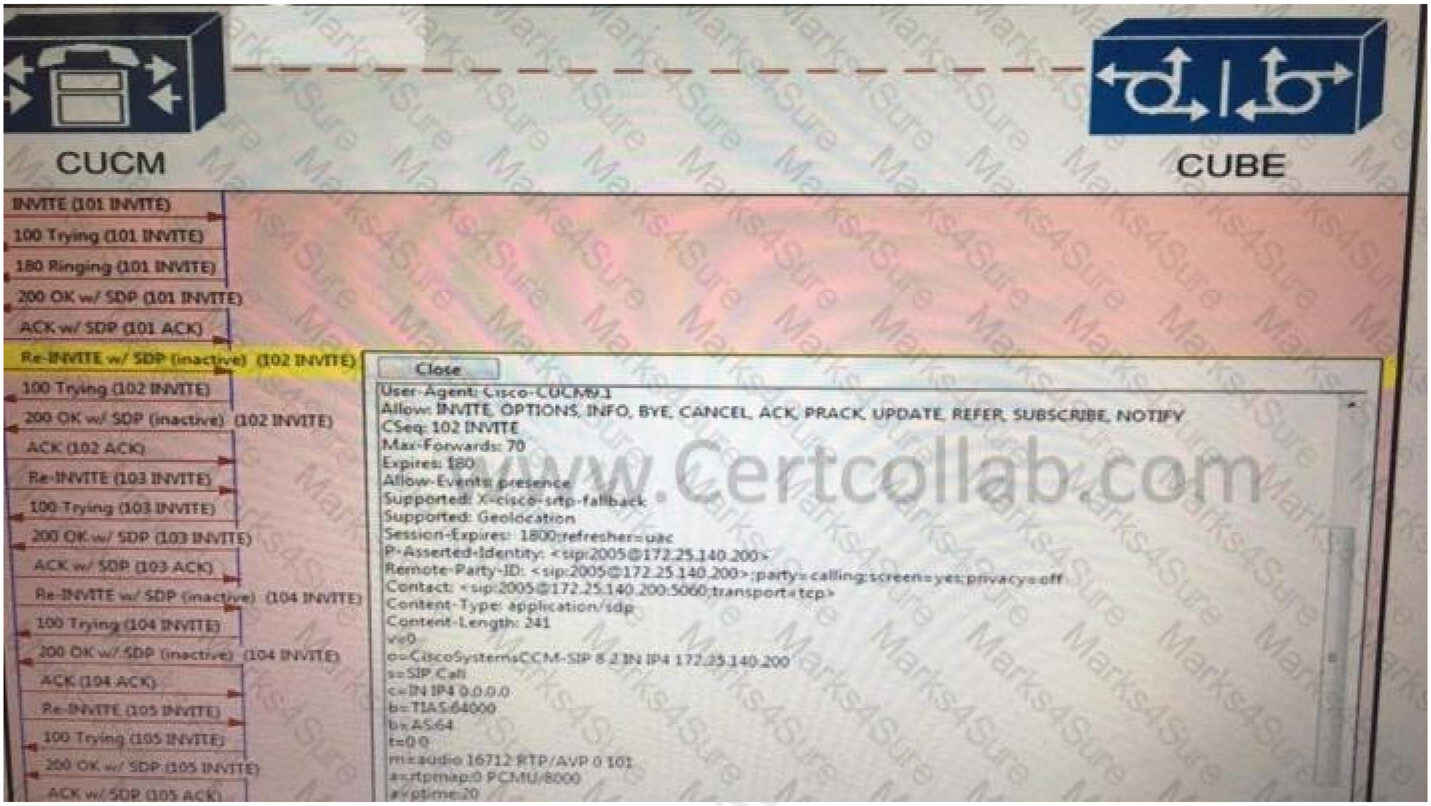
D. voice class sip-profiles 200

response 200 sip-header Diversion modify "sip:(.*>)" 123456@company.com>"

Answer: A

Question #:162 - (Exam Topic 1)

Refer to the exhibit.



An engineer is analysing a SIP call between CUCM on the left and CUBE on the right.

Based on the information displayed, which behaviour can be determined from the exhibit?

A. The mid-call re-invite negotiated for call hold/resume supplementary service

B. The call has interworking issues where user is not experiencing audio

C. The midcall-signaling block is configured on Cisco Unified border element so re-invites are not consumed

D. The call is terminated because 200 Ok messages are not arriving to Cisco Unified Communications Manager

Answer: A

Explanation

Explanation/Reference:

A Re-INVITE, it comes after a session has been established. This means that it will apply to an existing INVITE

after a final response has been received and an ACK has been sent.

You send an UPDATE message prior to session establishment, but that’s an article for another day.

A re-INVITE will have the same Call-ID and From tag as the INVITE it is modifying. It can change every other

header as well as the message body, but those two things tell the SIP stack that this is not a new INVITE.

The most common use for re-INVITE is call hold. The party putting the call on hold sends a re-INVITE with

SDP indicating that media will no longer be sent. That same party will take the call off hold by sending another

re-INVITE with SDP indicating that media transmission will resume.

To demonstrate this, I placed a call to my desk telephone, answered it, started up the Avaya traceSM utility,

put the call on hold, stopped traceSM, and then took a few screen shots of the resultant call flow.

https://andrewjprokop.wordpress.com/2015/02/10/understanding-sip-re-invite/

Question #:163 - (Exam Topic 1)

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

Answer: A C

Question #:164 - (Exam Topic 1)

After configuring EM in the CUCM cluster, users are receiving `Host not found' error message after pressing the Services button. What should be done to fix this problem?

A. Start the EM service and reset the phone

B. Reset CCM service on each node starting with Publisher

C. Set IP address instead of hostnames on the URLs and reset the phones

D. Associate the EM service to the phone and reset the phones

Answer: C

Question #:165 - (Exam Topic 1)

Refer to the exhibit.

```
RAS INCOMING PDU ::=

value RasMessage ::= registrationReject :
   {
     requestSeqNum 24
     protocolIdentifier { 0 0 8 2250 0 3 }
     rejectReason duplicateAlias:
     {
     }
     gatekeeperIdentifier {"gk"}
 }
```

A cisco collaboration engineer is troubleshooting a gateway and gatekeeper problem and sees this output from a debug command. Which two configuration can cause this problem? (Choose two)

A. The same zone prefix is configured in two different gatekeepers

B. The same H323-ID is configured in two different gateways

C. The same gw-type-prefix is configured in two different zone subnets IDs

D. The same zone subnet ID is configured in two different gatekeepers

E. The same E164-ID is configured in two different gateways

Answer: B E

Explanation

This output from the debug h225 asn1 command shows a registration reject reason of duplicateAlias.

RAS INCOMING PDU ::=

value RasMessage ::= registrationReject :

{

requestSeqNum 24

protocolIdentifier { 0 0 8 2250 0 3 }

rejectReason duplicateAlias:

{

}

gatekeeperIdentifier {"gk"}

}

This is usually the result of the gateway registering a duplicate of an E164-ID or H323-ID: Another gateway has already been registered to the gatekeeper. If it is a duplicated E164-ID, change the destination pattern configured under a POTS dial-peer associated with an FXS port. If it is a duplicated H323-ID, change the gateway's H.323 ID under the H.323 VoIP interface.

http://www.cisco.com/c/en/us/support/docs/voice/h323/22378-gk-reg-issues.html#rr1

Question #:166 - (Exam Topic 1)

ABC Company has Cisco unified CM version 9.1 cluster with seven nodes. The publisher server suffered a catastrophic hard disk failure without Cisco Disaster Recovery System Backups. Which method restore the Publisher node is valid?

A. Take a DRS backup from a subscriber and reinstall the Publisher from that backup

B. Reinstall the publisher node and restore the publisher database from a subscriber database

C. Take a full DRS backup from all subscribers and reinstall the publisher from that backup

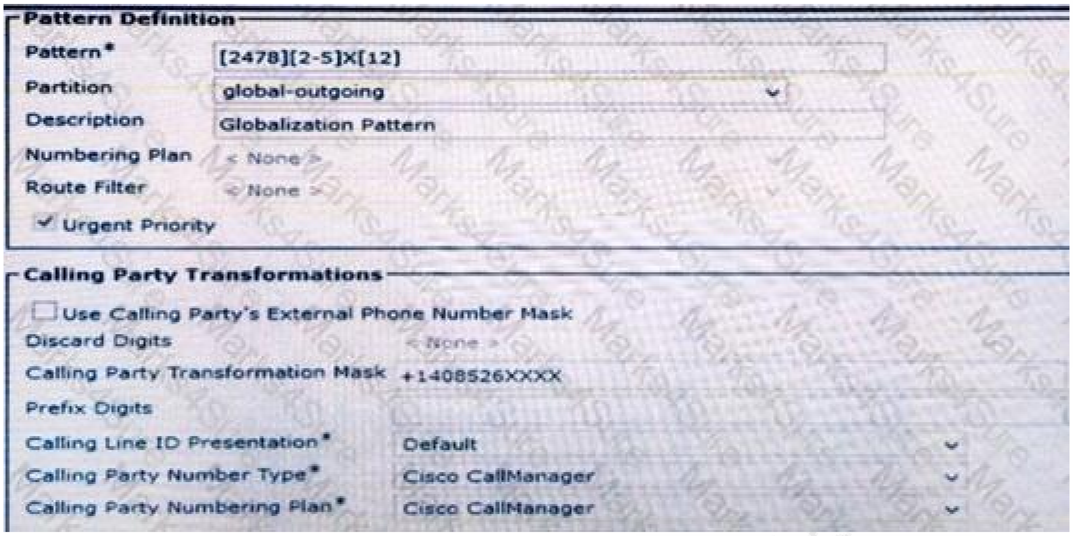
D. Promote one of the remaining subscriber then install a new subscriber

E. The publisher node cannot be restored but the remaining subscribers should be sufficient to support the collaboration devices and Service

Answer: A

Question #:167 - (Exam Topic 1)

Refer to the Exhibit.



Which three Cisco Unified CM internal extension match the globalization pattern shown and provide a globalized calling party number? (Choose three.)

A. 2041

B. 2671

C. 3392

D. 4202

E. 7352

F. 8253

Answer: A D E

Question #:168 - (Exam Topic 1)

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 Signalling Disconnect

B. Hookflash Duration Signalling Disconnect

C. CP-TONE Dual Supervisory Disconnect

D. Power Denial-based Supervisory Disconnect

E. Ground-start Signalling Disconnect

F. Tone-based Supervisory Disconnect

Answer: D E F

Question #:169 - (Exam Topic 1)

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

Answer: C

Question #:170 - (Exam Topic 1)

Which TFTP server address selection option has the highest precedence on Cisco SCCP IP phones using firmware release 8.0(4) 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

Answer: A

Question #:171 - (Exam Topic 1)

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 passwrod 0 password12345
domain 1 default

voice service saf
profile trunk-toute 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.

Answer: B

Question #:172 - (Exam Topic 1)

Which Cisco IOS multipoint video conferencing profile reserves DSPs when it is created in the configuration?

A. flex mode video

B. guaranteed-audio119

C. rendezvous

D. heterogeneous

E. guaranteed-video

Answer: D

Question #:173 - (Exam Topic 1)

Refer to the exhibit.

What happens to the USB e-token after the administrator fails to enter the correct password at the next attempt?

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.

Answer: E

Question #:174 - (Exam Topic 1)

Assume that your customer domain is customer.com and the Cisco Unified Communication Manager IM &

Presences environment is 8.X

Which internal DNS SRV record(s) is needed on the DNS server to facilitate service discovery so the Cisco Jabber Client can automatically find the appropriate servers to connect?

A. _cisco-uds_tcp customer.com

B. _collab-edge_tls customer.comC. _cuplogin_tcp customer.com

C. _cisco-uds_tls customer.com

D. _collab-edge_tcp customer.com

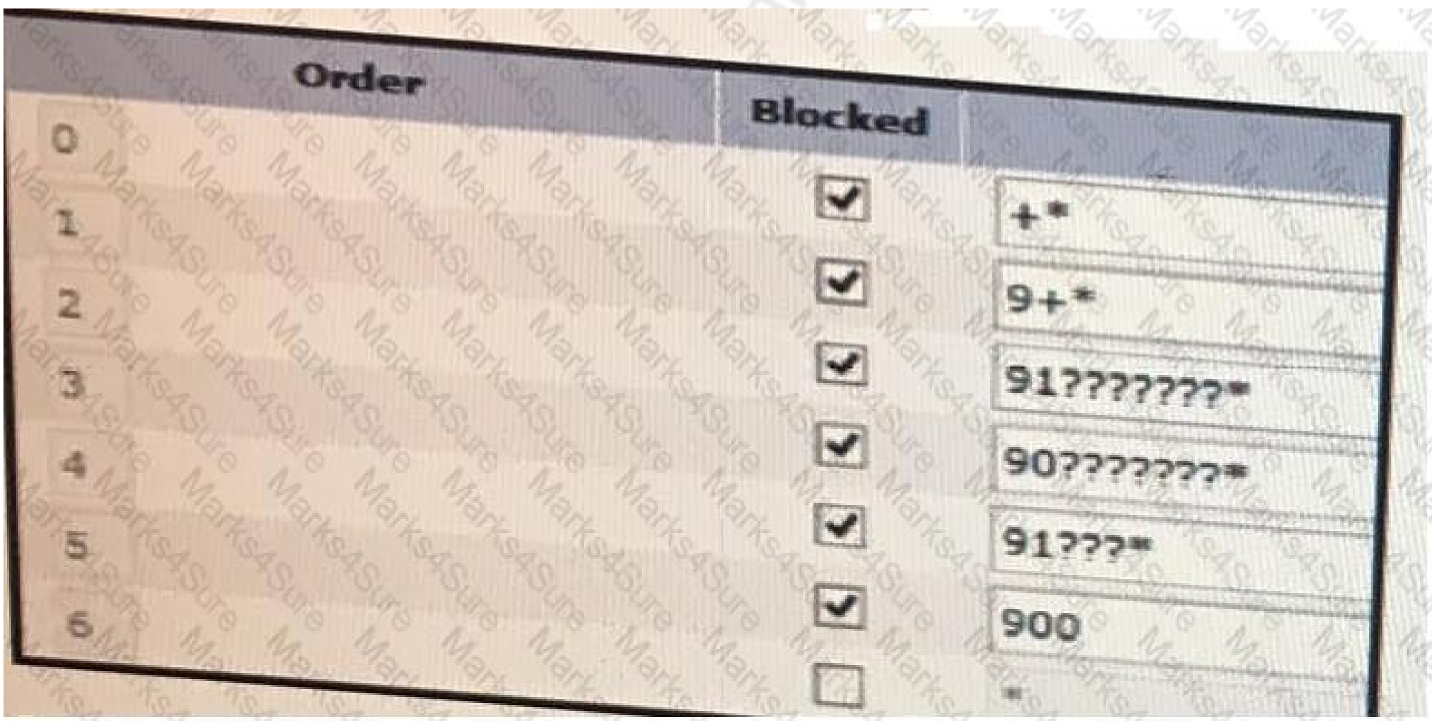
Answer: C

Question #:175 - (Exam Topic 1)

A collaboration engineer is configuring toll fraud prevention in the dial plan. Which two sets of patterns allow Cisco Unity Connection to transfer calls to local and long distance numbers while blocking all other patterns?

(Choose two).

A)



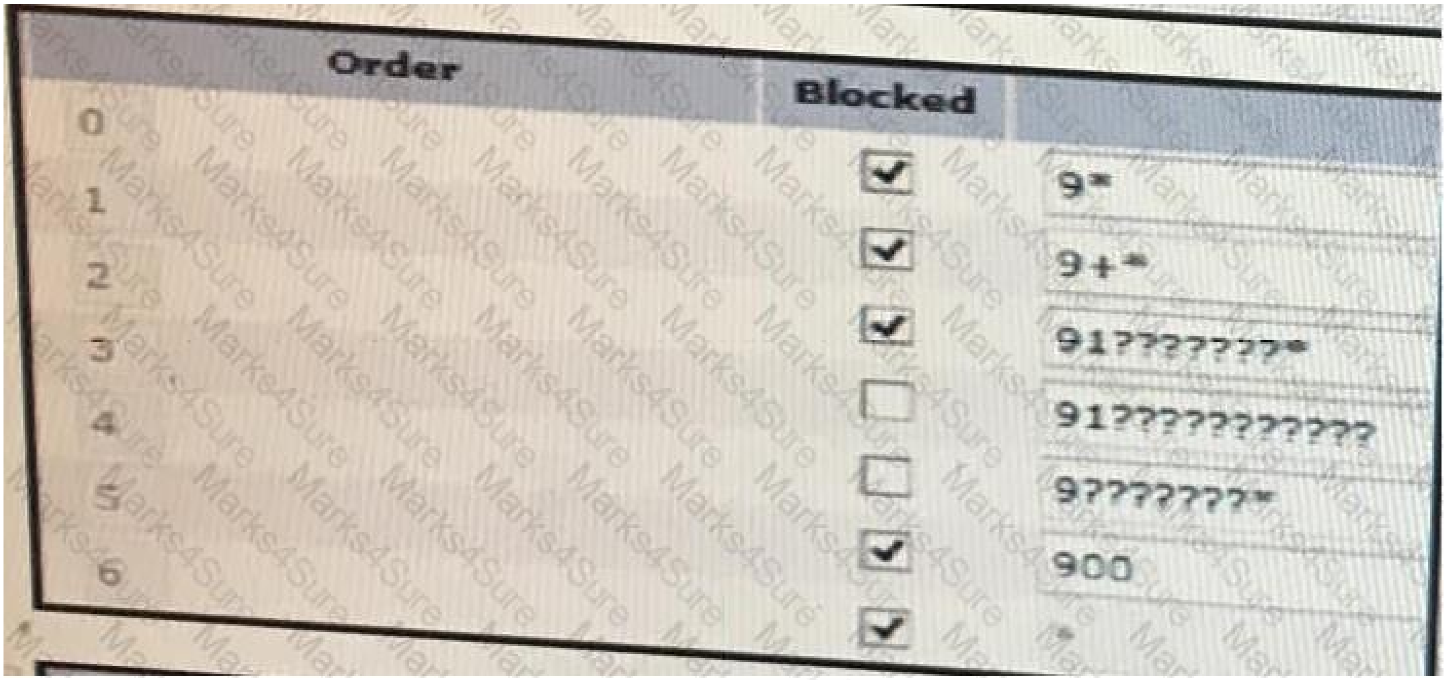
B)

Order	Blocked	
0	☑	+*
1	☑	9+*
2	☐	91???????*
3	☑	9011???????*
4	☐	9???????????*
5	☐	9???????*
6	☑	*

C)

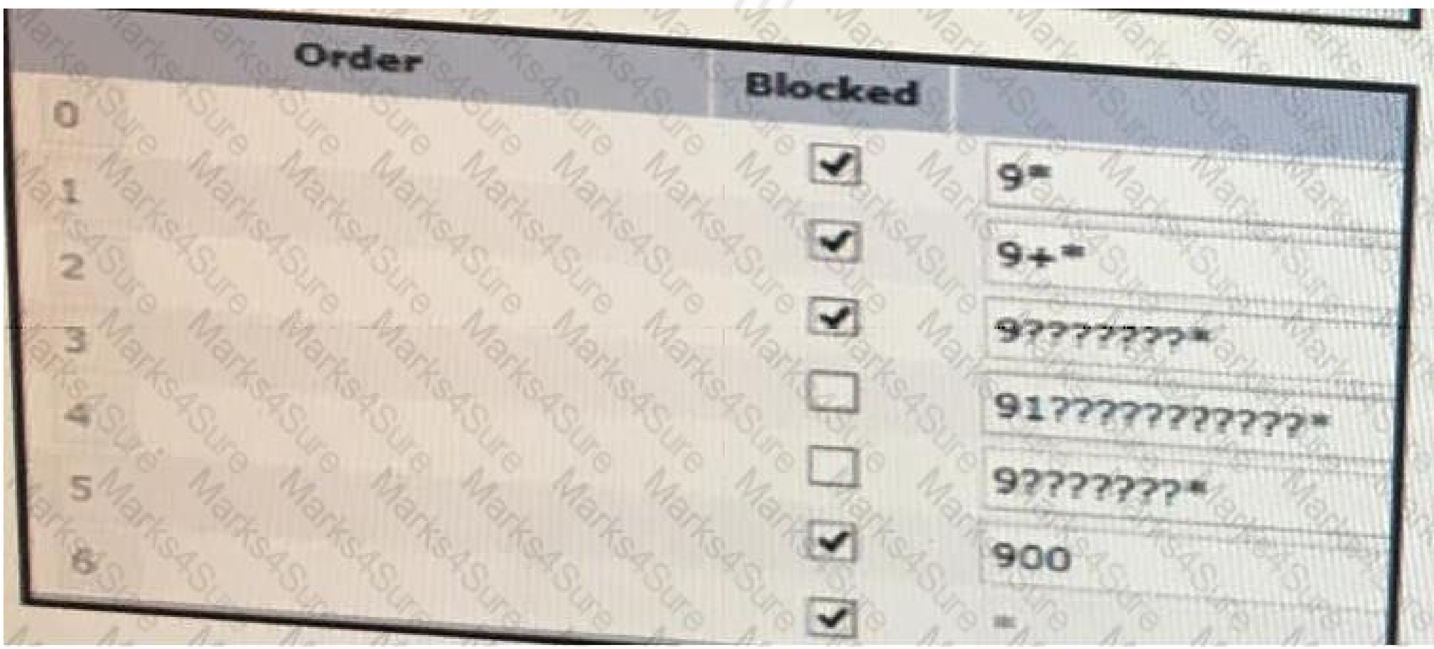
Order	Blocked	
0	☑	9011*
1	☑	+*
2	☑	9+*
3	☐	91???????*
4	☐	9???????*
5	☐	9???????????*
6	☑	*

D)



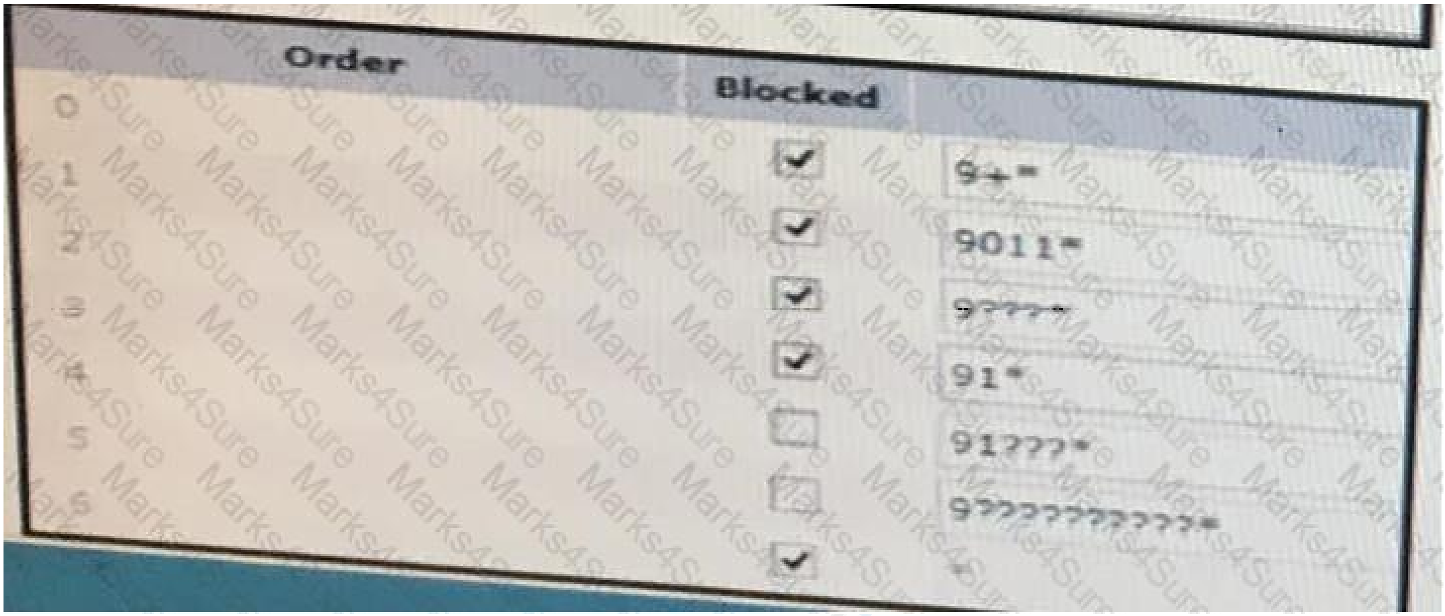
Order	Blocked	
0	✓	9*
1	✓	9+*
2	✓	91???????*
3	☐	91????????????
4	☐	9???????*
5	✓	900
6	✓	*

E)



Order	Blocked	
0	✓	9*
1	✓	9+*
2	✓	9???????*
3	☐	91???????????*
4	☐	9???????*
5	✓	900
6	✓	*

F)



A. Exhibit A

B. Exhibit B

C. Exhibit C

D. Exhibit D

E. Exhibit E

F. Exhibit F

Answer: B C

Question #:176 - (Exam Topic 1)

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

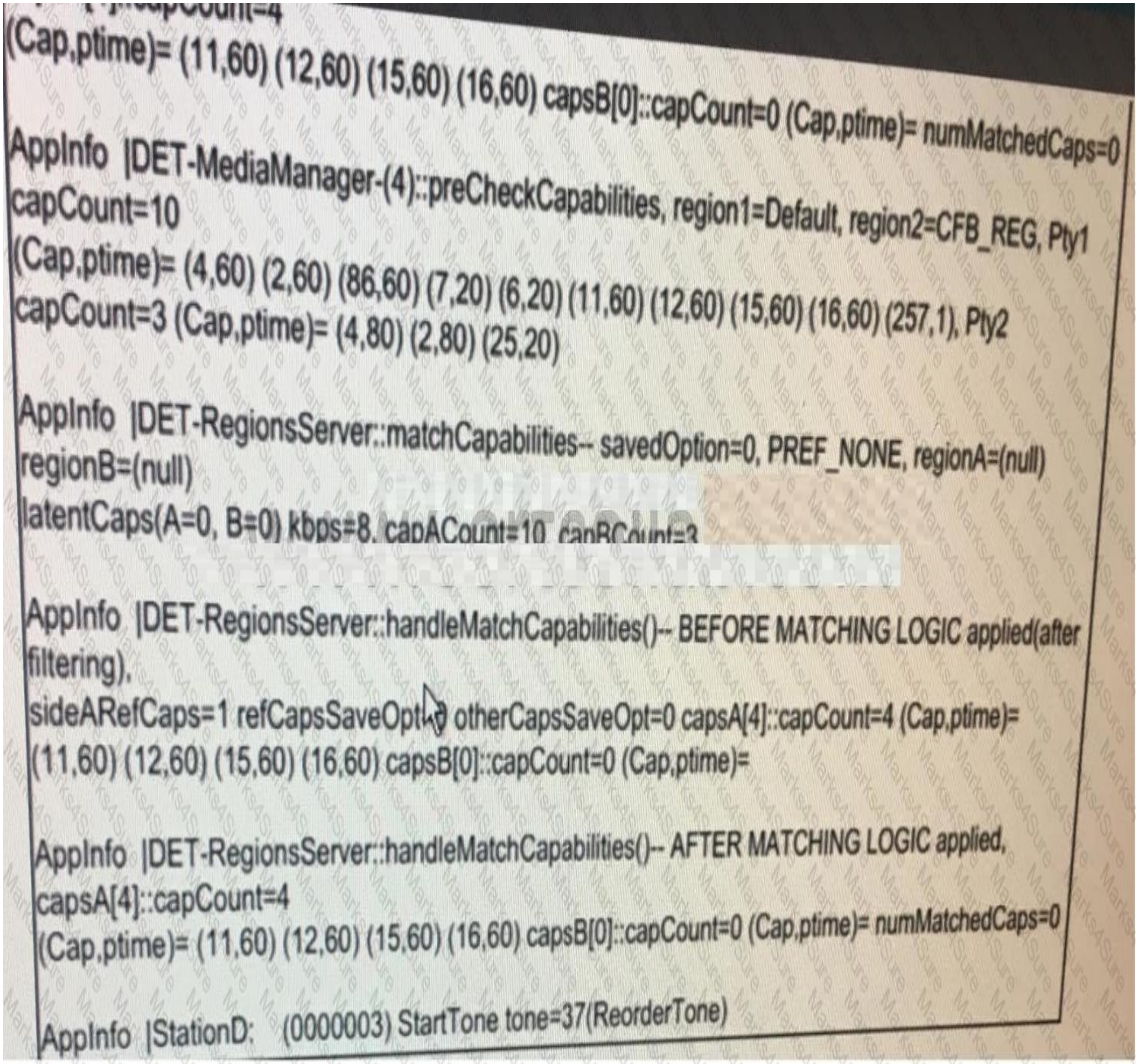
D. server model and local components (CPU, RAM, adapters, local storage) at the part number level

E. configuration settings and patch recommendations for VMware software

Answer: D

Question #:177 - (Exam Topic 1)

Refer to the exhibit.



```
(Cap,ptime)= (11,60) (12,60) (15,60) (16,60) capsB[0]::capCount=0 (Cap,ptime)= numMatchedCaps=0
AppInfo |DET-MediaManager-(4)::preCheckCapabilities, region1=Default, region2=CFB_REG, Pty1
capCount=10
(Cap,ptime)= (4,60) (2,60) (86,60) (7,20) (6,20) (11,60) (12,60) (15,60) (16,60) (257,1), Pty2
capCount=3 (Cap,ptime)= (4,80) (2,80) (25,20)

AppInfo |DET-RegionsServer::matchCapabilities-- savedOption=0, PREF_NONE, regionA=(null)
regionB=(null)
latentCaps(A=0, B=0) kbps=8, capACount=10 capBCount=3

AppInfo |DET-RegionsServer::handleMatchCapabilities()-- BEFORE MATCHING LOGIC applied(after
filtering),
sideARefCaps=1 refCapsSaveOpt=0 otherCapsSaveOpt=0 capsA[4]::capCount=4 (Cap,ptime)=
(11,60) (12,60) (15,60) (16,60) capsB[0]::capCount=0 (Cap,ptime)=

AppInfo |DET-RegionsServer::handleMatchCapabilities()-- AFTER MATCHING LOGIC applied,
capsA[4]::capCount=4
(Cap,ptime)= (11,60) (12,60) (15,60) (16,60) capsB[0]::capCount=0 (Cap,ptime)= numMatchedCaps=0

AppInfo |StationD: (0000003) StartTone tone=37(ReorderTone)
```

You have received UCM SDI trace from a client who is having issues with conference calls.

Based on the information in the trace file, what could be the possible cause of conference failure? (Choose three)

A. Conference Bridge only supports G711

B. Transcoder is missing in MRGL

C. Region relationship is null between Conference Bridge and phones

D. Region relationship is set to G729 between Conference Bridge and phones

E. Conference bridge capabilities count is 0

F. Media termination point is missing from MRGL

Answer: A B D

Question #:178 - (Exam Topic 1)

Refer to the exhibit.

```
voice service voip
 clid substitute name
 clid network-provided
 address-hiding
 mode border-element
 allow-connections sip to sip
 allow-connections sip to h323
 allow-connections h323 to sip
 allow-connections h323 to h323
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 fax protocol t38 version 3 ls-redundancy 0 hs-redundancy 0 fallback none
 no fax-relay sg3-to-g3
 modem passthrough nse codec g711ulaw
 sip
  session transport tcp
  min-se 360 session-expires 360
  ds0-num
  header-passing
  media flow-around
  pass-thru content sdp
  error-passthru
  registrar server expires max 600 min 60
  options-ping 90
  early-offer forced
  midcall-signaling passthru
  no call service stop
```

A user is on an outbound call through a Cisco Unified border Element gateway. When the user places the call on hold, the remote party hears silence. The Cisco Unified Communication Manager Cluster is using multicast on hold. The Cisco Unified Border Element Gateway is on the same subnet as the Cisco Unified Communication Manager Cluster.

Which two options will resolve this issue? (Choose two)

A. Media flow-through must be configured.

B. CCM-manager music-on-hold should be removed from the configuration.

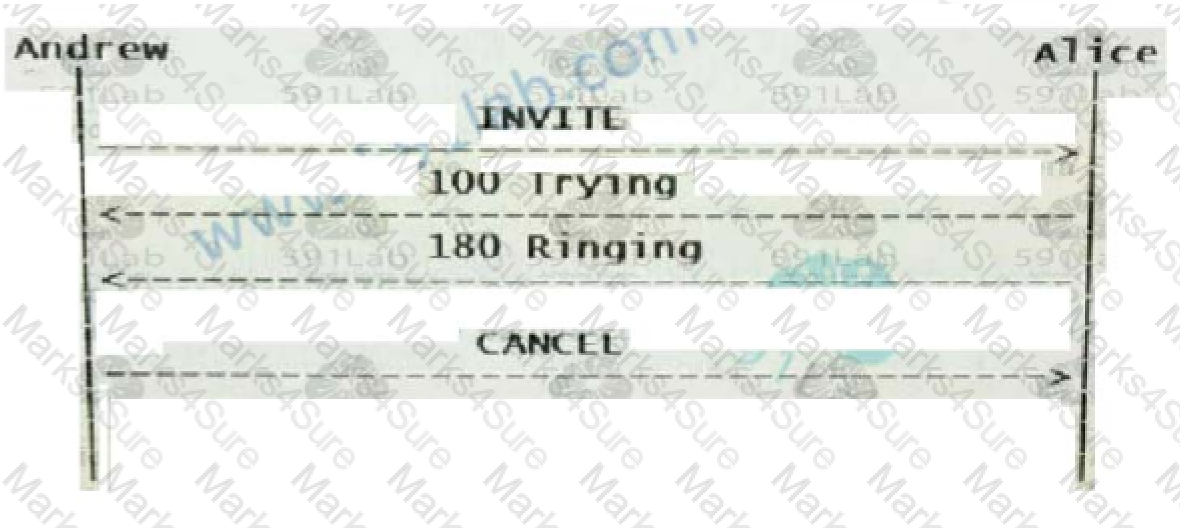
C. The session transport UDP command must be configured.

D. The Cisco unified border Element router must be set up for gateway-based MOH.

E. The pass-thru content sdp command should be removed.

Answer: A E

Question #:179 - (Exam Topic 1)



Refer to the exhibit. Which event is next in the SIP call flow?

A. A "487 Request Terminated" is sent by Alice

B. An ACK sent by Alice

C. A "199 Early Dialog Terminated" is sent by Alice

D. A "202 Accepted" is sent by Alice

E. A "200 OK" is sent by Alice

Answer: E

Question #:180 - (Exam Topic 1)

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. 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.

Answer: E

Question #:181 - (Exam Topic 1)

Which two parameters are requested in an Audit Connection message from a Cisco Unified CM to endpoint on a MGCP gateway? (Choose two)

A. Call ID

B. Capabilities

C. Bearer Information

D. Connection Parameters

E. Connection Mode

F. Connection ID

Answer: C F

Question #:182 - (Exam Topic 1)

In Cisco Unity Connection, 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

Answer: A B E

Question #:183 - (Exam Topic 1)

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.

Answer: C

Question #:184 - (Exam Topic 1)

An engineer is setting up a proxy TFTP between multiple Cisco communication Manager clusters.

Drag the step from the left to the correct order on the right to properly configure the certificates for the proxy TFTP. Not all options will be used.

Select and Place:

Export all certificates using Bulk Certificate management **in all** participant Clusters.

Import the certificates on each participant cluster using Bulk Certificate **Import.**

Restart the Cisco CallManager and Cisco Tomcat services in all participant servers.

Enable "Auto-Registration" in all participant clusters to allow phones **to register.**

Select "Consolidate" from each Publisher of the remote clusters.

Choose "Consolidate" from one Publisher of any participant cluster.

Configure **the** same Global Cluster **ID in Enterprise Parameters in all participant clusters.**

Step 1

Step 2

Step 3

Step 4

Answer:

Export all certificates using Bulk Certificate management **in all** participant Clusters.

Import the certificates on each participant cluster using Bulk Certificate **Import.**

Restart the Cisco CallManager and Cisco Tomcat services in all participant servers.

Enable "Auto-Registration" in all participant clusters to allow phones **to register.**

Select "Consolidate" from each Publisher of the remote clusters.

Choose "Consolidate" from one Publisher of any participant cluster.

Configure **the** same Global Cluster **ID in Enterprise Parameters in all participant clusters.**

Export all certificates using Bulk Certificate management **in all** participant Clusters.

Choose "Consolidate" from one Publisher of any participant cluster.

Import the certificates on each participant cluster using Bulk Certificate **Import.**

Restart the Cisco CallManager and Cisco Tomcat services in all participant servers.

Explanation

Export all certificates using Bulk Certificate management in all participant Clusters.

Choose "Consolidate" from one Publisher of any participant cluster.

Import the certificates on each participant cluster using Bulk Certificate Import.

Restart the Cisco CallManager and Cisco Tomcat services in all participant servers.

Question #:185 - (Exam Topic 1)

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.

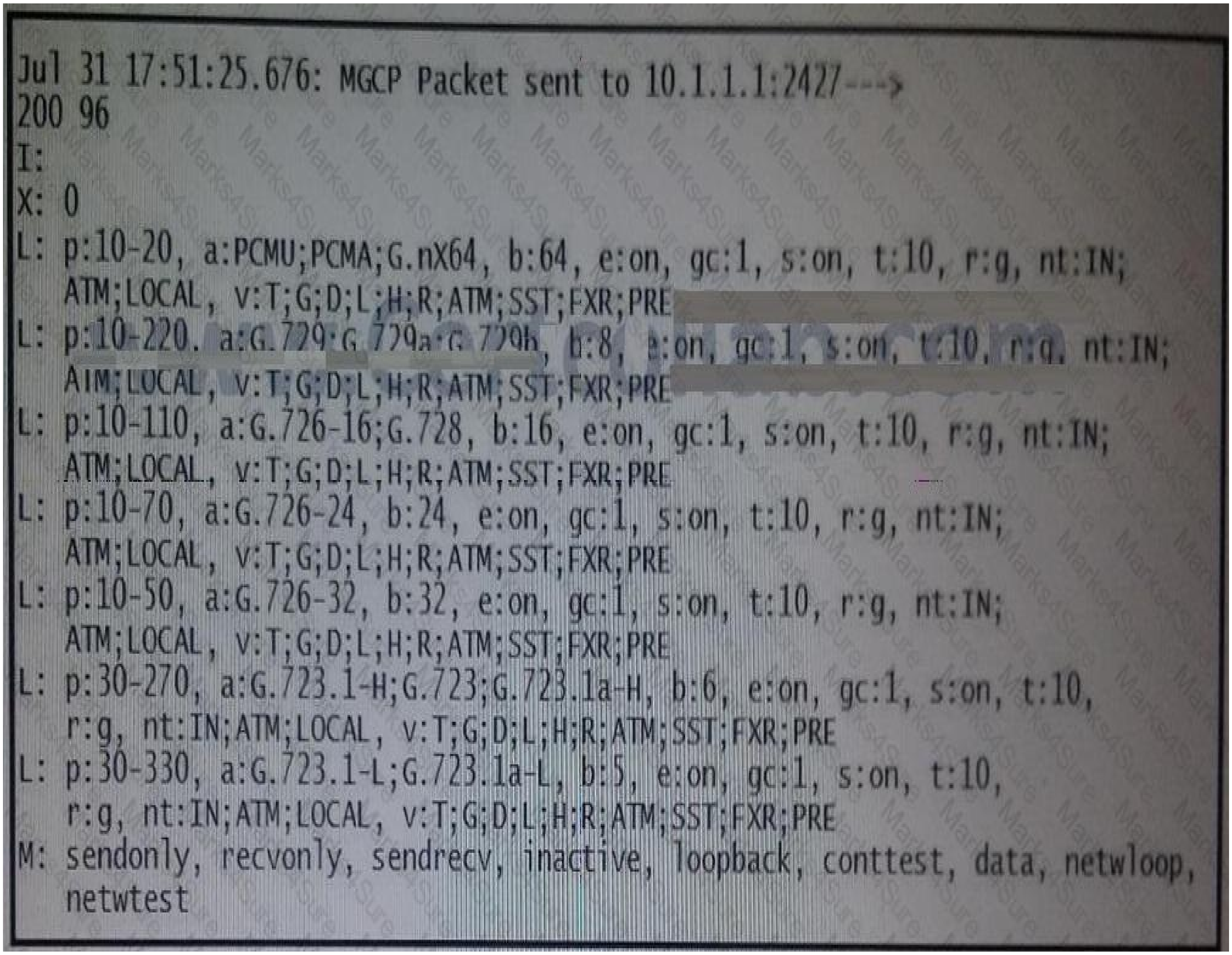
D. Cisco Unified Reporting.

E. Call Detail Records data collection internal is not a configurable parameter.

Answer: A

Question #:186 - (Exam Topic 1)

Exhibit:



```
Jul 31 17:51:25.676: MGCP Packet sent to 10.1.1.1:2427--->
200 96
I:
X: 0
L: p:10-20, a:PCMU;PCMA;G.nX64, b:64, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:10-220, a:G.729;G.729a;G.729b, b:8, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:10-110, a:G.726-16;G.728, b:16, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:10-70, a:G.726-24, b:24, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:10-50, a:G.726-32, b:32, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:30-270, a:G.723.1-H;G.723;G.723.1a-H, b:6, e:on, gc:1, s:on, t:10,
   r:g, nt:IN;ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:30-330, a:G.723.1-L;G.723.1a-L, b:5, e:on, gc:1, s:on, t:10,
   r:g, nt:IN;ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, conttest, data, netwloop,
   netwtest
```

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

Answer: C

Question #:187 - (Exam Topic 1)

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.

Answer: A E

Question #:188 - (Exam Topic 1)

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

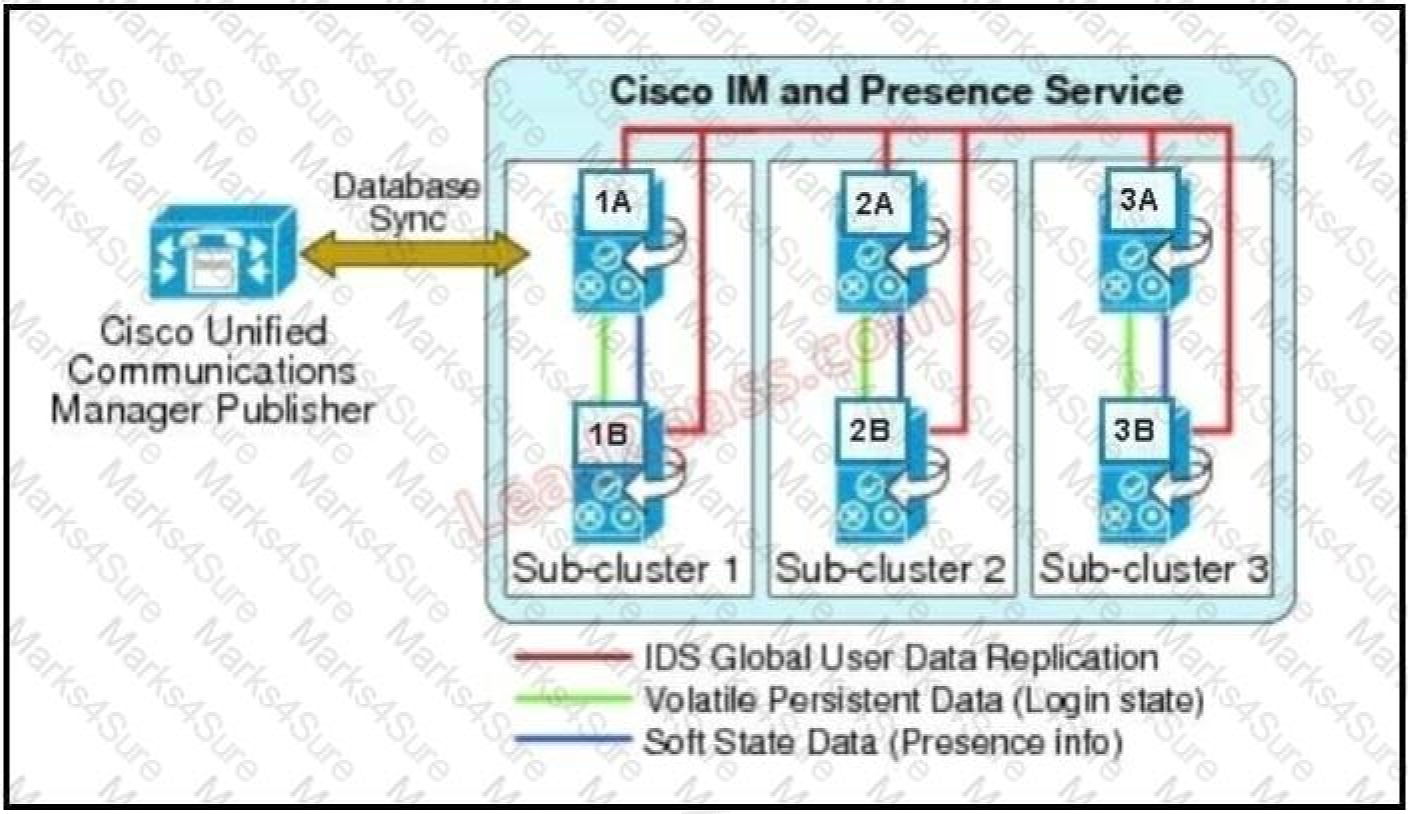
F. Park

G. Hold

Answer: A C E

Question #:189 - (Exam Topic 1)

Refer to the exhibit.



Refer to the exhibit. Which Cisco IM & Presence deployment is shown?

A. baic

B. hybrid

C. high availability

D. mixed

Answer: C

Question #:190 - (Exam Topic 1)

A Cisco collaboration engineer is troubleshooting unexpected SIP call disconnect. Which three responses corresponding to the 5xx range? (Choose Three)

A. Forbidden

B. Unauthorized

C. Request timeout

D. Service unavailable

E. Bad gateway

F. Internal server error

Answer: D E F

Question #:191 - (Exam Topic 1)

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

Answer: D

Question #:192 - (Exam Topic 1)

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

Answer: A C

Question #:193 - (Exam Topic 1)

Refer to the exhibit.

What happens to the USB e-token after the administrator fails to enter the correct password at the next attempt?

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.

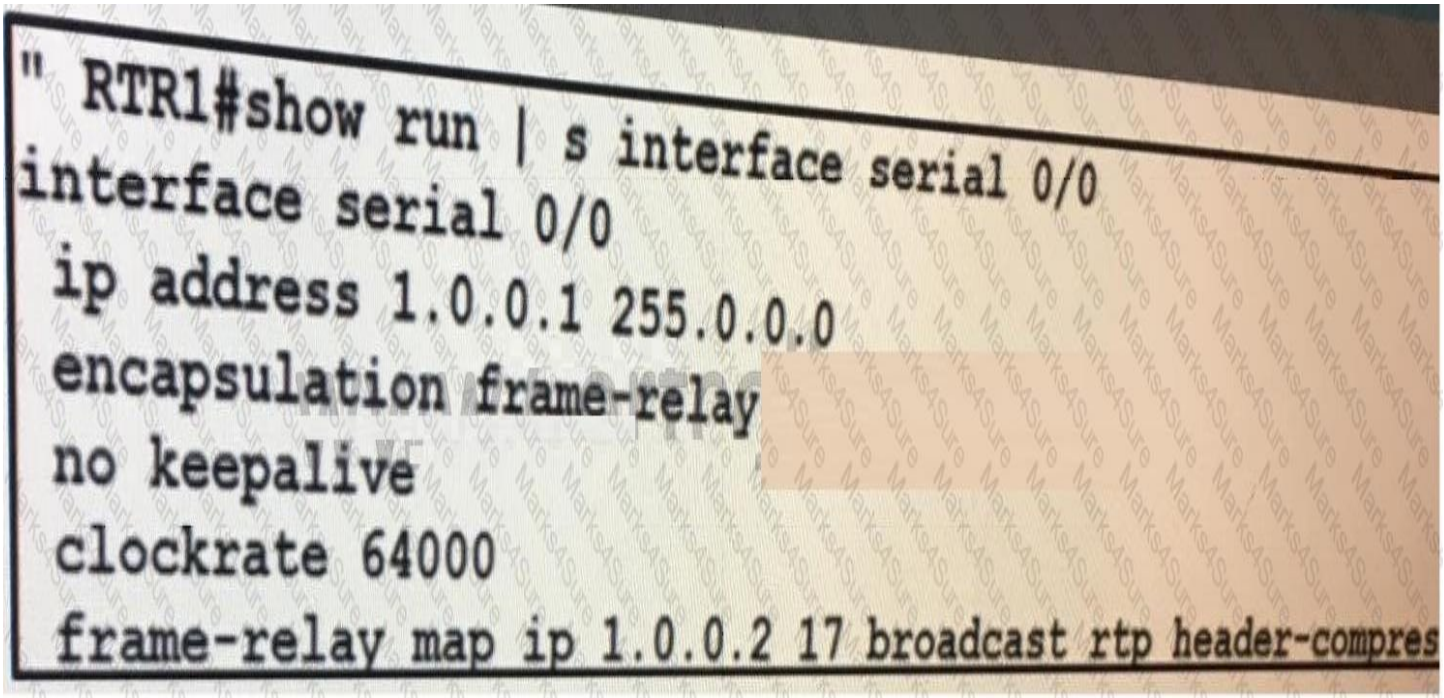
D. The token cannot be used on the same client PC again. It can be used with another Cisco CTL Client on a different PC.

E. The token is locked forever.

Answer: E

Question #:194 - (Exam Topic 1)

Refer to the exhibit.



An engineer is upgrading existing frame-relay network to MPLS using Ethernet. The amount of bandwidth from service provider will remain the same. What two issues the engineer must consider when changing from frame-relay to Ethernet for voice connectivity? (Choose two)

A. Overhead with Ethernet L2 is much smaller than Frame-Relay L2 headers

B. Using G711 codec, calls will consume slightly more bandwidth over Ethernet than frame-relay calls

C. Ability to use header compression will not be available when using ethernet

D. Using G711 codec, calls will consume slightly less bandwidth over Ethernet than frame-relay calls

E. Ethernet and MPLS will allow engineer to implement QoS which is not available on frame relay

Answer: B C

Question #:195 - (Exam Topic 1)

Refer to the exhibit.

A collaboration engineer is configuring device mobility. Which three occur to phone A1 when it moves to physical location B? (Choose three)

A. Phone A PSTN calls preserve home location dialing behavior.

B. Phone A inherits CSS from roaming device pool DP_B2.

C. Phone A PSTN calls Adopts roaming location dialing behavior.

D. Phone A retains CSS from Home device pool DP_A1.

E. Phone A retains media Resource group list from home device pool DP_A1.

F. Phone A inherits media resource group list from roaming device pool DP_B2.

Answer: B C E

Question #:196 - (Exam Topic 1)

Which two security services are provided by the Phone Proxy function on a Cisco ASA appliance? (Choose two.)

A. It provides internetworking to ensure that extended IP Phone traffic is encrypted as long as the Cisco Unified Communications Manager is in secure mode.

B. It manipulates the call signalling to ensure that all media is routed via the adaptive security appliance.

C. It requires a remote routing device with an IPsec VPN tunnel.

D. It intercepts and authenticates soft clients before they reach Cisco Unified Communications Manager clusters.

E. It supports encrypted TFTP operation of IP phone configuration files.

Answer: B D

Question #:197 - (Exam Topic 1)

Refer to the exhibit.

*****Exhibit is Missing*****

Which description of the event captured in the SIP message between a Cisco Unified Communication Manager Express router and Cisco Unity Express is true?

A. The MWI notification can be destined only to a SCCP IP phone

B. The MWI notification method used is subscribe-notify

C. The MWI notification can be destined only to a SIP IP phone

D. The MWI notification method used is outcall

E. The MWI notification method used is unsolicited

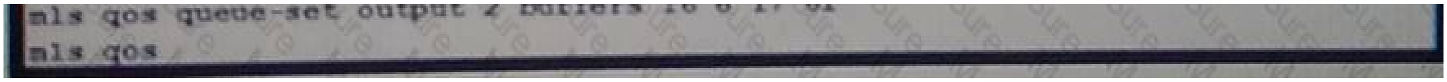
Answer: B

Question #:198 - (Exam Topic 1)

Refer to the exhibit.

```
interface GigabitEthernet1/0/13
 description Phone and PC port
 srr-queue bandwidth share 1 10 30 100
 srr-queue bandwidth shape 1 0 0 0
 queue-set 2
 mls qos trust device cisco-phone
 mls qos trust cos

mls qos map cos-dscp 0 8 16 26 32 46 48 56
mls qos srr-queue input bandwidth 90 10
mls qos srr-queue input threshold 1 8 16
mls qos srr-queue input threshold 2 34 66
mls qos srr-queue input buffers 67 33
mls qos srr-queue input cos-map queue 1 threshold 2  1
mls qos srr-queue input cos-map queue 1 threshold 3  0
mls qos srr-queue input cos-map queue 2 threshold 1  2
mls qos srr-queue input cos-map queue 2 threshold 2  4 6 7
mls qos srr-queue input cos-map queue 2 threshold 3  3 5
mls qos srr-queue input dscp-map queue 1 threshold 2  9 10 11 12 13 14 15
mls qos srr-queue input dscp-map queue 1 threshold 3  0 1 2 3 4 5 6 7
mls qos srr-queue input dscp-map queue 1 threshold 3  32
mls qos srr-queue input dscp-map queue 2 threshold 1  16 17 18 19 20 21 22 23
mls qos srr-queue input dscp-map queue 2 threshold 2  33 34 35 36 37 38 39 48
mls qos srr-queue input dscp-map queue 2 threshold 2  49 50 51 52 53 54 55 56
mls qos srr-queue input dscp-map queue 2 threshold 2  57 58 59 60 61 62 63
mls qos srr-queue input dscp-map queue 2 threshold 3  24 25 26 27 28 29 30 31
mls qos srr-queue input dscp-map queue 2 threshold 3  40 41 42 43 44 45 46 47
mls qos srr-queue output cos-map queue 1 threshold 3  5
mls qos srr-queue output cos-map queue 2 threshold 3  3 6 7
mls qos srr-queue output cos-map queue 3 threshold 3  2 4
mls qos srr-queue output cos-map queue 4 threshold 2  1
mls qos srr-queue output cos-map queue 4 threshold 3  0
mls qos srr-queue output dscp-map queue 1 threshold 3  40 41 42 43 44 45 46 47
mls qos srr-queue output dscp-map queue 2 threshold 3  24 25 26 27 28 29 30 31
mls qos srr-queue output dscp-map queue 2 threshold 3  48 49 50 51 52 53 54 55
mls qos srr-queue output dscp-map queue 2 threshold 3  56 57 58 59 60 61 62 63
mls qos srr-queue output dscp-map queue 3 threshold 3  16 17 18 19 20 21 22 23
mls qos srr-queue output dscp-map queue 3 threshold 3  32 33 34 35 36 37 38 39
mls qos srr-queue output dscp-map queue 4 threshold 1  8
mls qos srr-queue output dscp-map queue 4 threshold 2  9 10 11 12 13 14 15
mls qos srr-queue output dscp-map queue 4 threshold 3  0 1 2 3 4 5 6 7
mls qos queue-set output 1 threshold 1 138 138 92 138
mls qos queue-set output 1 threshold 2 138 138 92 400
mls qos queue-set output 1 threshold 3 36 77 100 318
mls qos queue-set output 1 threshold 4 20 50 67 400
mls qos queue-set output 2 threshold 1 149 149 100 149
mls qos queue-set output 2 threshold 2 118 118 100 235
mls qos queue-set output 2 threshold 3 41 68 100 272
mls qos queue-set output 2 threshold 4 42 72 100 242
mls qos queue-set output 1 buffers 10 10 26 54
mls qos queue-set output 2 buffers 16 6 17 61
```



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 charges 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.

Answer: C E

Question #:199 - (Exam Topic 1)

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".

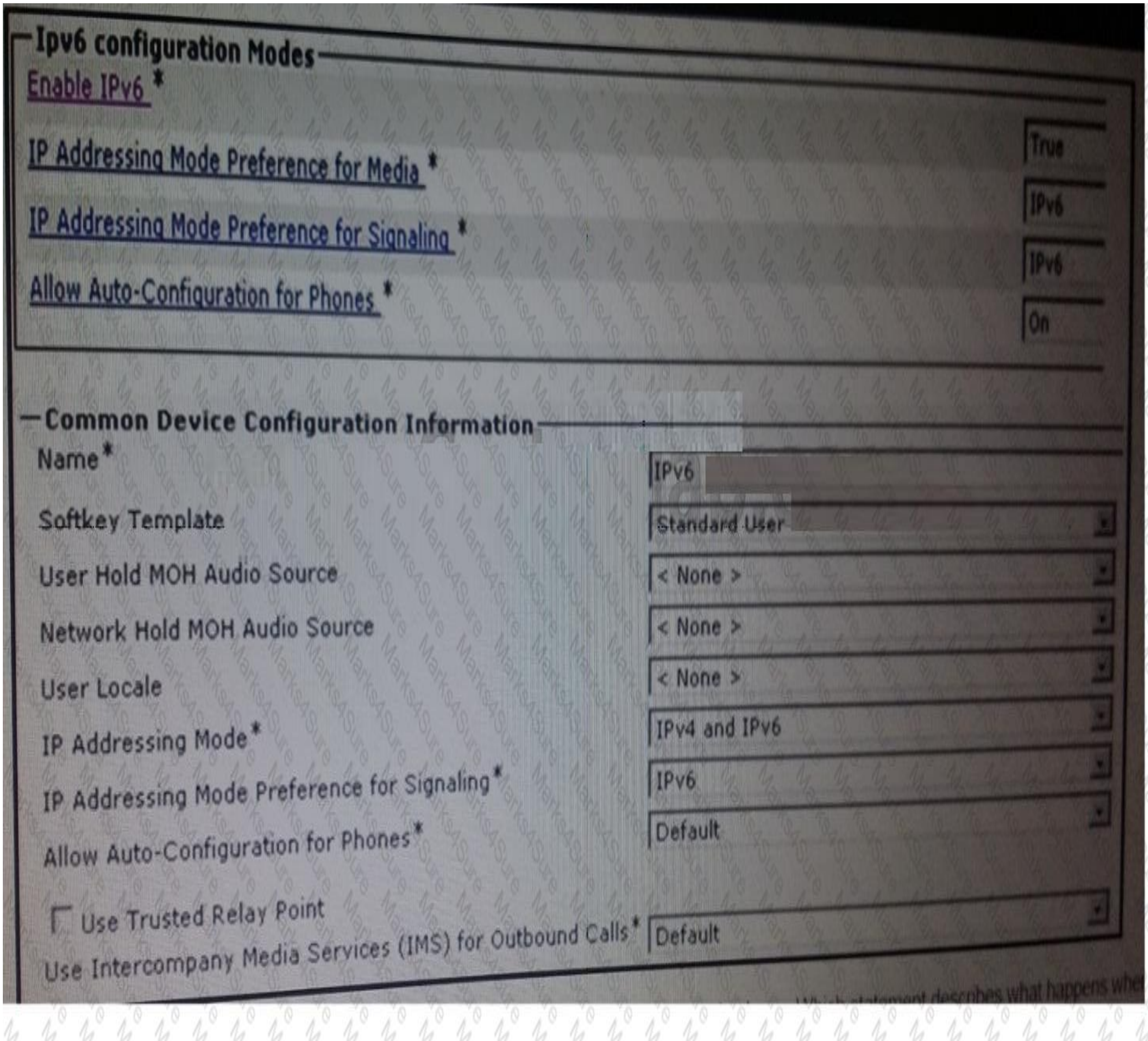
D. In the IM&P server Enterprise Parameters Configuration, enable the Phone Personalization parameter.

E. In the Cisco Unified CM Jabber Service Profile, uncheck "Make this the default service profile for the system".

Answer: A B

Question #:200 - (Exam Topic 1)

Refer to the exhibit.



The common device configuration has been assigned to an IPv4 only phone. Which statement describes what happens when the phone attempts to register with Cisco unified Communications manager?

A. The IP phone displays the message `registration rejected' because it cannot support IPv6

B. The IP phone invokes MTP to translate from ipv4 to IPv6

C. The IP phone ignores the settings and registers with the ipv4 address

D. The IP phone uses SLAAC to acquire an IPv6 address and override the limitation

Answer: A

Question #:201 - (Exam Topic 1)

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

Answer: C E

Question #:202 - (Exam Topic 1)

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

Answer: C

Question #:203 - (Exam Topic 1)

An Engineer is troubleshooting this situation

-MPLS connection between site A and site B is flapping in intervals of 30 seconds at least twice an hour

-Phones at site B failover to their survivable remote site,Telephony configured gateway.

-Phones fail back to their primary CUCM server as soon as the connection is reestablished.

-Phones failover again to SRST.

Which UCM setting allows the phones to stay in SRST for a longer period of tome before falling back to CUCM?

A. Keepalive timeout timer

B. Connection Monitor Duration

C. Station Keepalive interval

D. Maximum phone fallback queue depth

E. T302 Timer

Answer: B

Question #:204 - (Exam Topic 1)

Your customer reported that the Sync Agent service failed to start after a reinstallation of a Cisco this problem after you review these customer logs?

A. Add "appadmin" application user in Cisco UCM and IM & Presence

B. Add "appadmin" application user in Ciscon IM & Presence

C. Add "appaduser" application user in Cisco UCM

D. Add "appadmin" application user in Cisco UCM

E. Add "appaduser" application user in Cisco UCM and IM & Presence

Answer: D

Question #:205 - (Exam Topic 1)

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

A. iDivert

B. DND

C. Answer

D. NewCall

E. EndCall

F. CallBack

Answer: A B C

Question #:206 - (Exam Topic 1)

How does Call Detail Record Agent running on a Cisco Unified Communications Manager node determine if a

CDR flat file is ready to be transferred to a designated CDR Repository node?

A. The CDR Agent transfers all new CDR flat files upon receiving notification from the CDR Repository node

on the name of last file successfully received

B. The CDR Agent transfers all new CDR flat files generated after the last successful transfer

C. The CDR Agent transfers any CDR flat file before its deletion

D. The CDR Agent transfers all CDR flat files at a specific configurable time of day

E. The CDR Agent knows transfer eligibility from the name of a CDR flat file name

Answer: E

Question #:207 - (Exam Topic 1)

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

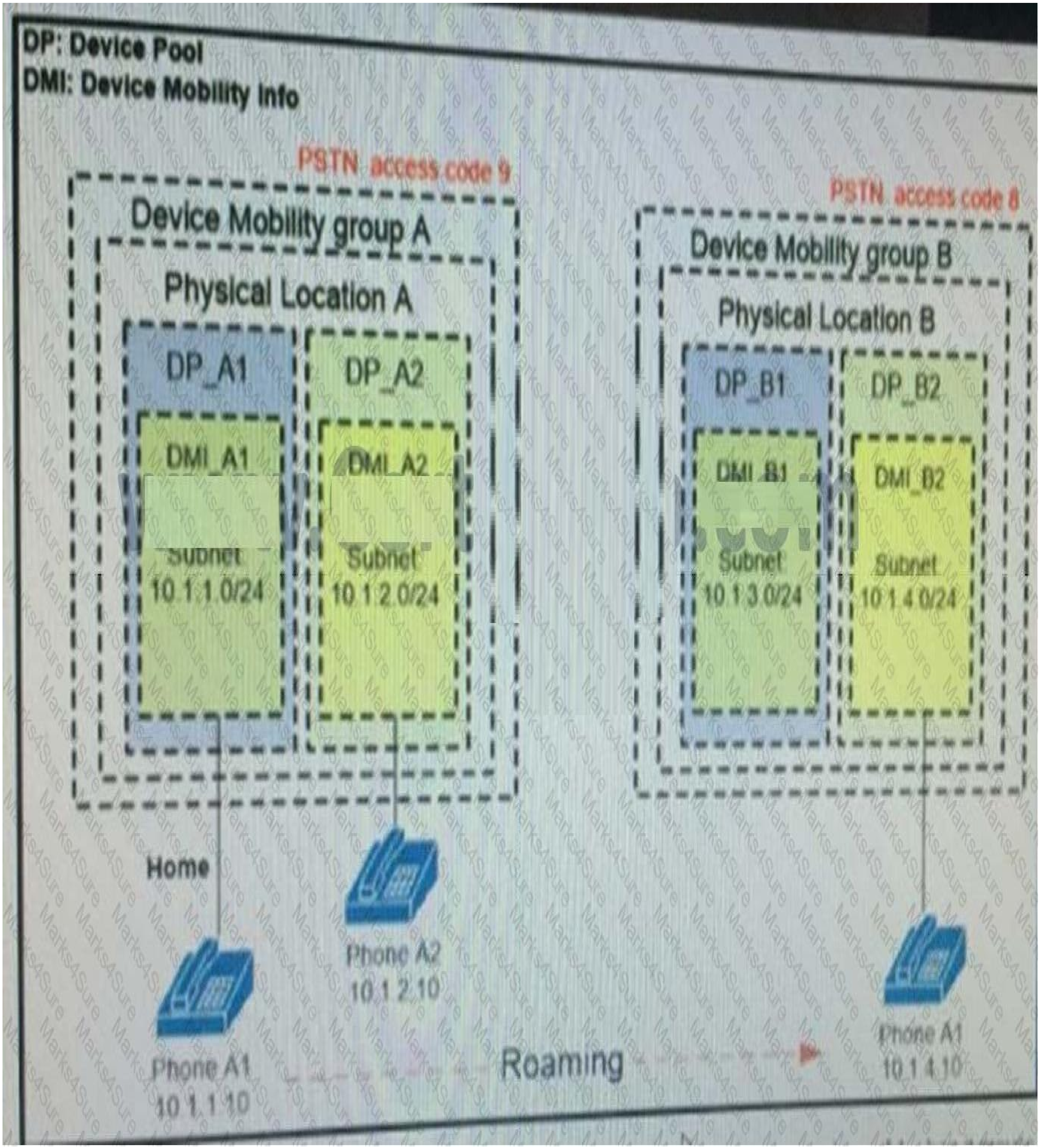
D. 77.6 kb/s

E. 71.3 kb

Answer: A

Question #:208 - (Exam Topic 1)

Refer to the exhibit.



A collaboration engineer is configuring device mobility. Which three events occur to phone A1 When it moves to physical location B? (Choose three)

A. Phone A PSTN calls preserve home location dialling behaviour

B. Phone A inherits CSS from roaming device pool DP_B2

C. Phone A PSTN calls Adopts roaming location dialling behaviour

D. Phone A retains CSS from Home device pool DP_A1

E. Phone A retains media Resource group list from home device pool DP_A1

F. Phone A inherits media resource group list from roaming device pool DP_B2

Answer: A D E

Question #:209 - (Exam Topic 1)

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.

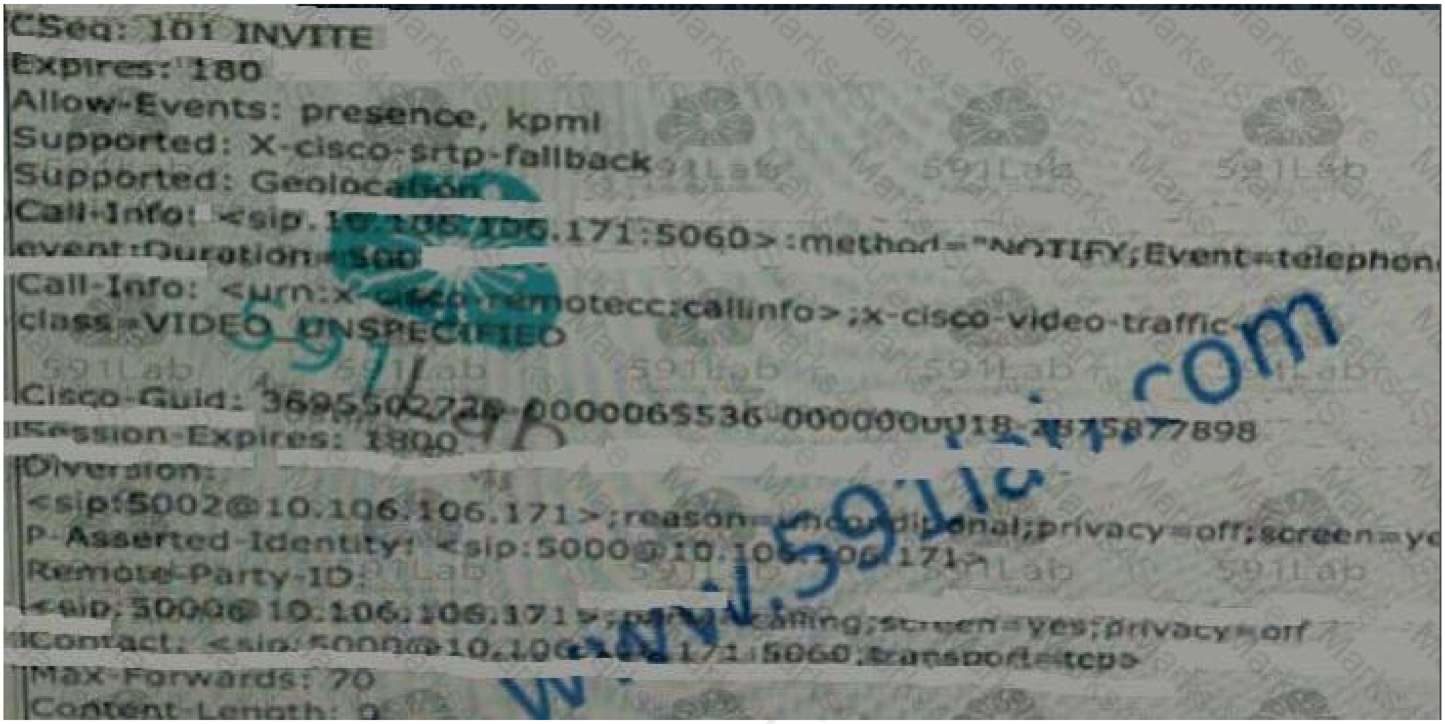
E. required bandwidth 250 kbps, given PPP transport, compressed RTP, and a 30-byte payload.

F. required bandwidth 650 kbps, given PPP transport, and a 20-byte payload.

Answer: B D

Question #:210 - (Exam Topic 1)

Refer to the exhibit.



```
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.106.106.171:5060>;method="NOTIFY;Event=telephon
event;Duration=500
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-
class=VIDEO_UNSPECIFIED
Cisco-Guid: 3695502720-0000065536-0000000018-2875877898
Session-Expires: 1800
Diversion:
<sip:5002@10.106.106.171>;reason=unconditional;privacy=off;screen=ye
P-Asserted-Identity: <sip:5000@10.106.106.171>
Remote-Party-ID:
<sip:5000@10.106.106.171>;party=calling;screen=yes;privacy=off
Contact: <sip:5000@10.106.106.171:5060;transport=tcp>
Max-Forwards: 70
Content-Length: 0
```

Which Cisco IOS SIP profile is valid for copying value from the "Diversion" header to the "From" header in a SIP INVITE message?

A. voice class sip-profiles 1

request INVITE sip-header Diversion copy "<sip:(.*)@.*" u02

request INVITE sip-header From copy "<sip:(.*)@.*" u01

request INVITE sip-header From modify "(.*)<sip:.*(@)(.*)" "\1<sip:\u01@\2"

B. voice class sip-profiles 1

request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01

request INVITE sip-header From copy "<sip:(.*)@.*" u02

request INVITE sip-header From modify "(.*)<sip:.*(@)(.*)" "\2<sip:\u01@\1"

C. voice class sip-profiles 1

request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01

request INVITE sip-header From copy "<sip:(.*)@.*" u02

request INVITE sip-header From modify "(.*)<sip:.*(@)(.*)" "\1<sip:\u02@\2"

D. voice class sip-profiles 1

request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01

request INVITE sip-header From copy "<sip:(.*)@.*" u02

request INVITE sip-header From modify "(.*)<sip:.*(@)(.*)" "\1<sip:\u01@\1"

E. voice class sip-profiles 1

request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01

request INVITE sip-header From copy "<sip:(.*)@.*" u02

request INVITE sip-header From modify "(.*)<sip:.*(@)(.*)" "\1<sip:\u01@\2"

Answer: E

Question #:211 - (Exam Topic 1)

What is a major disadvantage of virtual machines versus containers?

A. Operational Management

B. Security

C. Boot Time

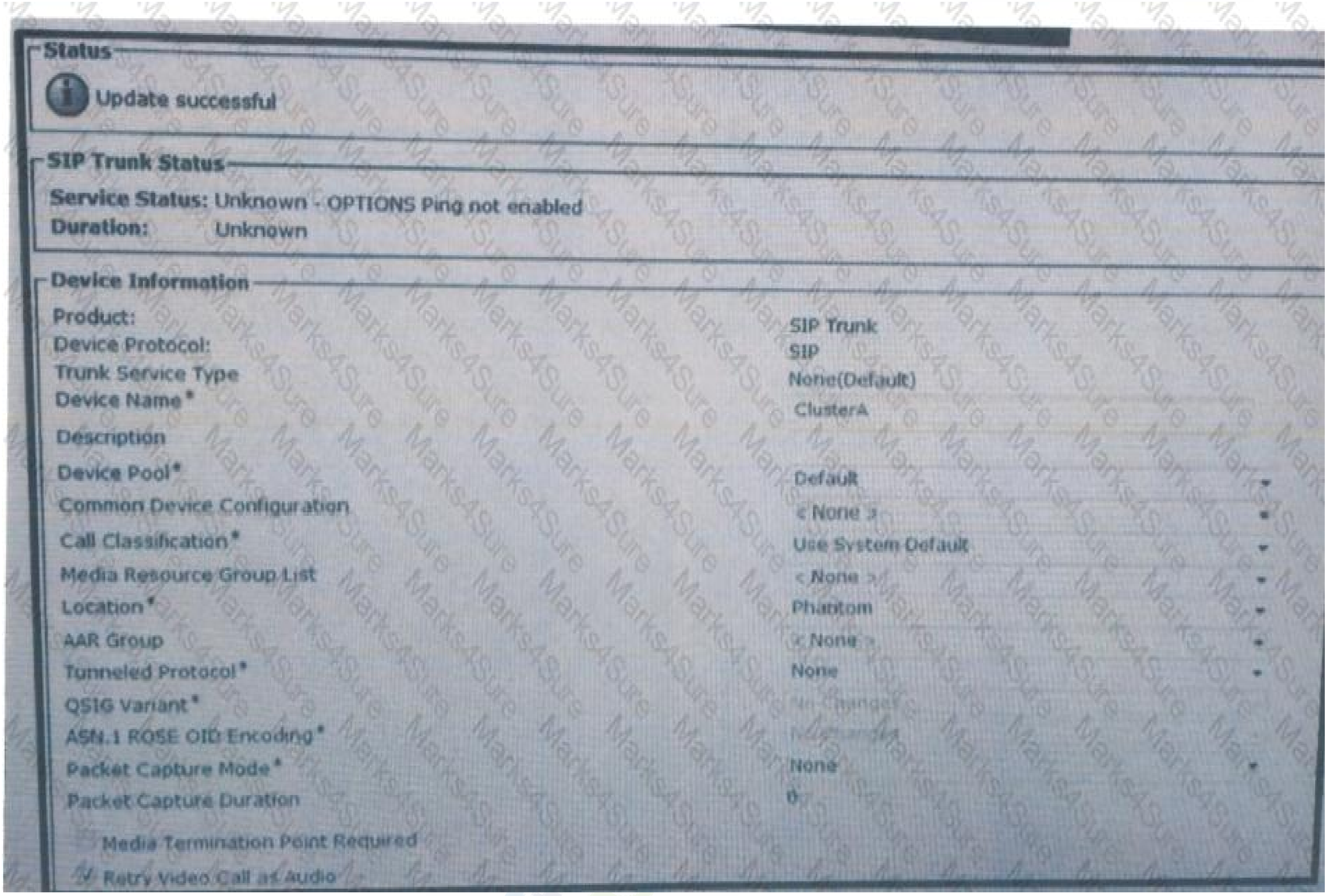
D. Limited management tools

E. Vendor lock-in

Answer: C

Question #:212 - (Exam Topic 1)

Refer to the exhibit.



A customer has two Cisco Unified Communication manager 9.X clusters that serve the same location. An engineer has attempted to set up Enhanced Location call admission control so that any call within a site between phones on the two clusters do not decrement the available bandwidth to and from that site. However, the real time monitoring tool currently shows bandwidth being used from the site to Hub_none. When a call is placed between phones at the site, which action must be taken to correct this situation?

A. The link between clusters must be a type of inter-cluster trunk instead of a sip trunk.

B. The hub_none location must have a link configuration to the phantom location.

C. The device pool names must match between clusters.

D. The Hub_none location must have a link configured to the shadow location.

E. The SIP trunks should be changed to use the shadow location.

Answer: E

Explanation

Shadow is a new system location created for intercluster Enhanced Location CAC. In order to pass location information across clusters, the SIP ICT needs to be assigned to the system location Shadow.

The system location Shadow:

- Is a valid location only for a SIP ICT. Devices other than SIP trunks assigned incorrectly to the Shadow location are treated as if assigned to the Hub_None location.
- Cannot have a link connecting to other user defined locations, so bandwidth cannot be deducted between the Shadow location and other user defined locations.
- Has no intra-location bandwidth capacities, so bandwidth cannot be deducted within the Shadow location.

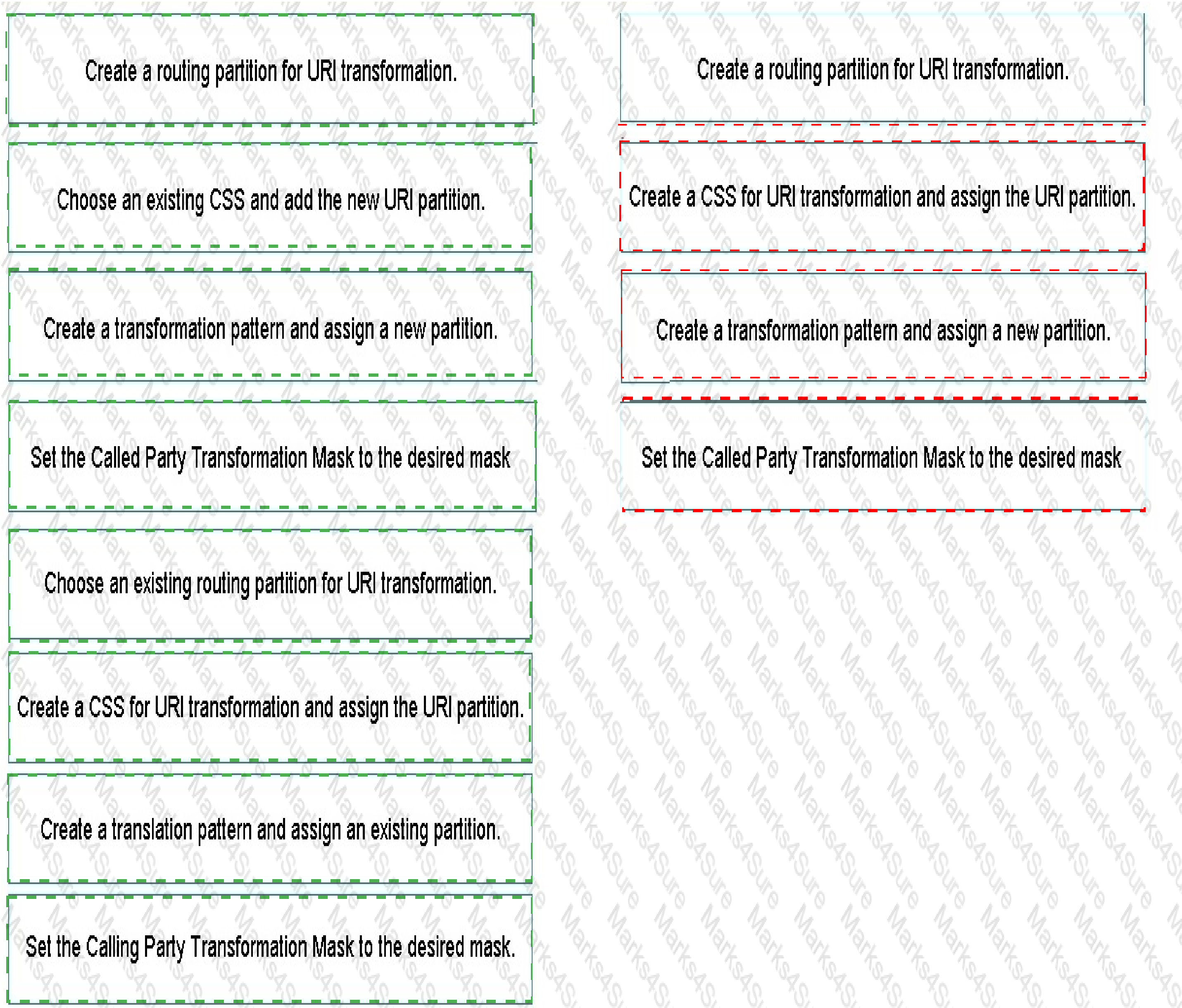
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmfeat/CUCM_BK_CEF0C471_00

Question #:213 - (Exam Topic 1)

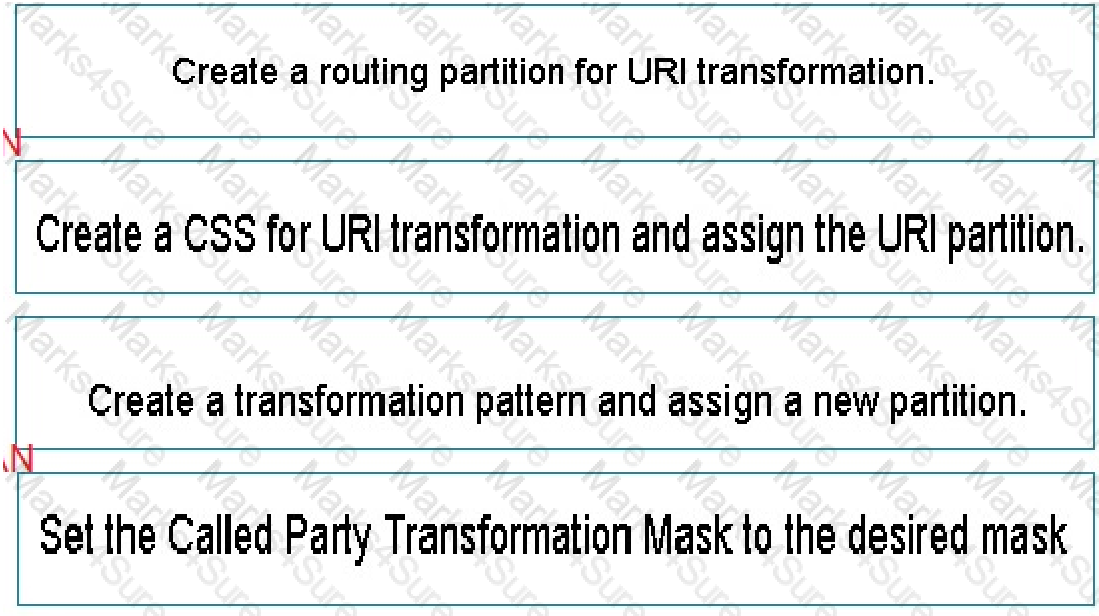
Drag the configuration steps on the left to the correct order for configuring digits transformation for URI dialing according to CISCO best practices on the right. Not all options will be used.


Create a routing partition for URI transformation.
Choose an existing CSS and add the new URI partition.
Create a transformation pattern and assign a new partition.
Set the Called Party Transformation Mask to the desired mask
Choose an existing routing partition for URI transformation.
Create a CSS for URI transformation and assign the URI partition.
Create a translation pattern and assign an existing partition.
Set the Calling Party Transformation Mask to the desired mask.
Step 1
Step 2
Step 3
Step 4

Answer:



Explanation



http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/9x/uc9x/dialplan.html

Question #:214 - (Exam Topic 1)

Refer to the exhibit.

Which option describes the security encryption status of this active call on a Cisco IP phone?

A. Encrypted call media but unencrypted call signalling

B. Encrypted call signalling and media

C. Encrypted call signalling but unencrypted call media

D. Unencrypted call signalling and media

E. Not enough information provided to answer this QUESTION NO:

Answer: C

Question #:215 - (Exam Topic 1)

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. CTIGWUser

Answer: D

Explanation

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/7x/uc7_0/directry.html

Default Application Users for Unified CM	
Application User	**Used by:**
CCMAdministrator	Unified CM Administration (default "super user")
CCMQRTSecureSysUser	Cisco Quality Reporting Tool
CCMQRTSysUser	
CCMSysUser	Cisco Extension Mobility
IPMASecureSysUser	Cisco Unified Communications Manager Assistant
IPMASysUser	
WDSecureSysUser	Cisco WebDialer
WDSysUser	

Question #:216 - (Exam Topic 1)

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 internal is not a configurable parameter.

Answer: B

Question #:217 - (Exam Topic 1)

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

A. Closed

B. Holiday

C. Alternate

D. Internal

E. Busy

Answer: C

Question #:218 - (Exam Topic 1)

Which three issues prevent a customer's existing Jabber client from seeing the present status of a newly provisioned and registered Jabber client in its contact list? (Choose three.)

A. Owner user ID is not set on the newly provisioned client.

B. Incoming calling search space on SIP trunk to IM & P.

C. Primary DN is not set in end user configuration of the newly provisioned client

D. Subscriber calling search space is not defined on newly provisioned client.

E. PC cannot resolve the FQDN of IM & P.

F. IM & P incoming ACL blocking inbound status.

G. Subscriber calling search space on SIP trunk to IM & P.

Answer: E F G

Explanation

No Presence Information After Login

Problem

You receive no Presence information after login.

Solution

Complete these steps in order to resolve this issue:

- Make sure that the DNS server the PC is pointed to can resolve the fully qualified name of the CUPS server.

The host entry will not suffice, you must resolve via DNS.

- Check the SUBSCRIBE CSS on the SIP trunk to CUP.

This CSS must include the partitions of the devices you are trying to receive status on.

- The CUP SIP proxy incoming access control list (ACL) is not allowing incoming SIP presence messages to reach the presence engine. As a test, set the incoming ACL to **ALL** and reset the SIP proxy and presence engine. Log in again to the CUPC and try to reconfigure the incoming ACL properly.

http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-presence/97443-cups-cupc-ts

Question #:219 - (Exam Topic 1)

Refer to the exhibit.

```
dial-peer voice 1 voip
 description incoming from PSTN
 incoming called-number [2-9]..[2-9]......
dial-peer voice 2 voip
 description outbound to CUCM
 destination-pattern [2-9]..[2-9]......
 session protocol sipv2
 session target ipv4:10.10.10.10
```

A carrier delivers a SIP call to cisco Unified CM through a Cisco Unified border Elements with The Invite destination different than "To" field. The Unified CM Administration engineer sees that the calls go to the invite destination instead of the "To" field Unified CM. Which option shows how the engineer correct that problem in the Cisco Unified border Elements router?

A)

```
voice class sip-profiles 10
 request INVITE peer-header sip To copy
   "sip:(.*)@" u01
 request INVITE sip-header SIP-Req-URI modify
   ".*@(.*)" "INVITE sip:\u01@\1"
voice class sip-copylist 1
 sip-header To
dial-peer voice 1 voip
 voice-class sip copy-list 1
dial-peer voice 2 voip
 voice-class sip profiles 10
```

B)

```
voice class sip-profiles 10
 request INVITE peer-header sip INVITE copy
   "sip:(.*)@" u01
 request INVITE sip-header To modify
   ".*@(.*)" "INVITE sip:\u01@\1"
voice class sip-copylist 1
 sip-header To
dial-peer voice 1 voip
 voice-class sip copy-list 1
dial-peer voice 2 voip
 voice-class sip profiles 10
```

C)

```
voice class sip-profiles 10
 request INVITE peer-header sip To copy
   "sip:(.*)@" u01
 request INVITE sip-header SIP-Req-URI modify
   ".*@(.*)" "INVITE sip:\u01@\1"
voice class sip-copylist 1
 sip-header To
dial-peer voice 2 voip
 voice-class sip profiles 10
 voice-class sip copy-list 1
```

D)

```
voice class sip-profiles 10
 request INVITE sip-header To copy
   "sip:(.*)@" u01
 request INVITE sip-header SIP-Req-URI modify
   ".*@(.*)" "INVITE sip:\u01@\1"
voice class sip-copylist 1
 sip-header To
dial-peer voice 1 voip
 voice-class sip copy-list 1
dial-peer voice 2 voip
 voice-class sip profiles 10
```

A. Exhibit A

B. Exhibit B

C. Exhibit C

D. Exhibit D

Answer: A

Question #:220 - (Exam Topic 1)

Refer to the exhibit.

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP 172.16.100.50:5060;branch=z9hG4bKc37e7b85b2
From: "Agent C" <sip:31051531000172.16.100.50>;tag=184-1c9cfaa5-5c1b-49be-840b-296a0e488c30-33493072
To: <sip:9131051511110172.16.100.90>;tag=BF0008-18BE
Date: Wed, 11 Mar 2015 05:25:25 GMT
Call-ID: e1817d00-4ff1d18b-b5-326410ac0172.16.100.50
CSeq: 104 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Remote-Party-ID: "PSTN CALLER" <sip:51511110172.16.100.90>;party=called;screen=no;privacy=off
Contact: <sip:9131051511110172.16.100.90:5060;transport=tcp>
Supported: replaces
Call-Info: <sip:172.16.100.90:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Supported: sdp-anat
Server: Cisco-SIPGateway/IOS-15.x
Session-Expires: 1800;refresher=uac
Require: timer
Supported: timer
Content-Type: application/sdp
Content-Length: 205

v=0
o=CiscoSystemsSIP-GW-UserAgent 676 6894 IN IP4 172.16.100.90
s=SIP Call
c=IN IP4 172.16.100.90
t=0 0
m=audio 7852 RTP/AVP 0 8 9 18 4
c=IN IP4 69.85.125.25
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:4 G723/8000
a=ptime:20
```

Which three pieces of information can be derived from this sip massage? (Choose Three)

A. The call will have no audio

B. Only OOB DTMF will be supported

C. G722 codec will be chosen

D. The B2BUA uses IP 172.16.100.50

E. This is a flow-around configuration

F. The call will last only 30 minutes

Answer: B D F

Question #:221 - (Exam Topic 1)

Refer to the exhibit.

G.711 A-law Codec Enabled *	Disabled
G.711 mu-law Codec Enabled *	Enabled for All Devices
G.722 Codec Enabled *	Disabled
iLBC Codec Enabled *	Enabled for All Devices
iSAC Codec Enabled *	Enabled for All Devices
Default Intraregion Max Audio Bit Rate *	64 kbps (G.722, G.711)
Default Interregion Max Audio Bit Rate *	64 kbps (G.722, G.711)
Default Intraregion Max Video Call Bit Rate (Includes Audio) *	384
Default Interregion Max Video Call Bit Rate (Includes Audio) *	384
Use Video BandwidthPool for Immersive Video Calls *	True
Default Intraregion and Interregion Link Loss Type *	Low Loss
Default Audio Codec List between Regions *	Factory Default low loss
Default Audio Codec List within Region *	Factory Default low loss
Accept Audio Codec Preferences in Received Offer *	Off

What is the preferred audio codec for all calls between different regions?

A. iLBC codec

B. G711ulaw codec

C. iSAC codec

D. low loss configured codec

Answer: B

Question #:222 - (Exam Topic 1)

Which three options are interior device types in Cisco Unified Communication Manager logical partitioning policies? (choose three)

A. SIP IP phones

B. Media Termination Points

C. SIP Trunk to Cisco Unity Connection

D. MGCP gateway with FXO ports

E. MGCP gateway with FXS ports

F. MGCP gateway with Q.SIG PRI

Answer: A E F

Question #:223 - (Exam Topic 1)

Which route pattern is matched in Cisco Unified Communication Manager Version 11.0 when a user dials 2001?

A. 200X configured with urgent priority

B. 20[02-4]1 configured with urgent priority

C. 200! Configured with urgent priority

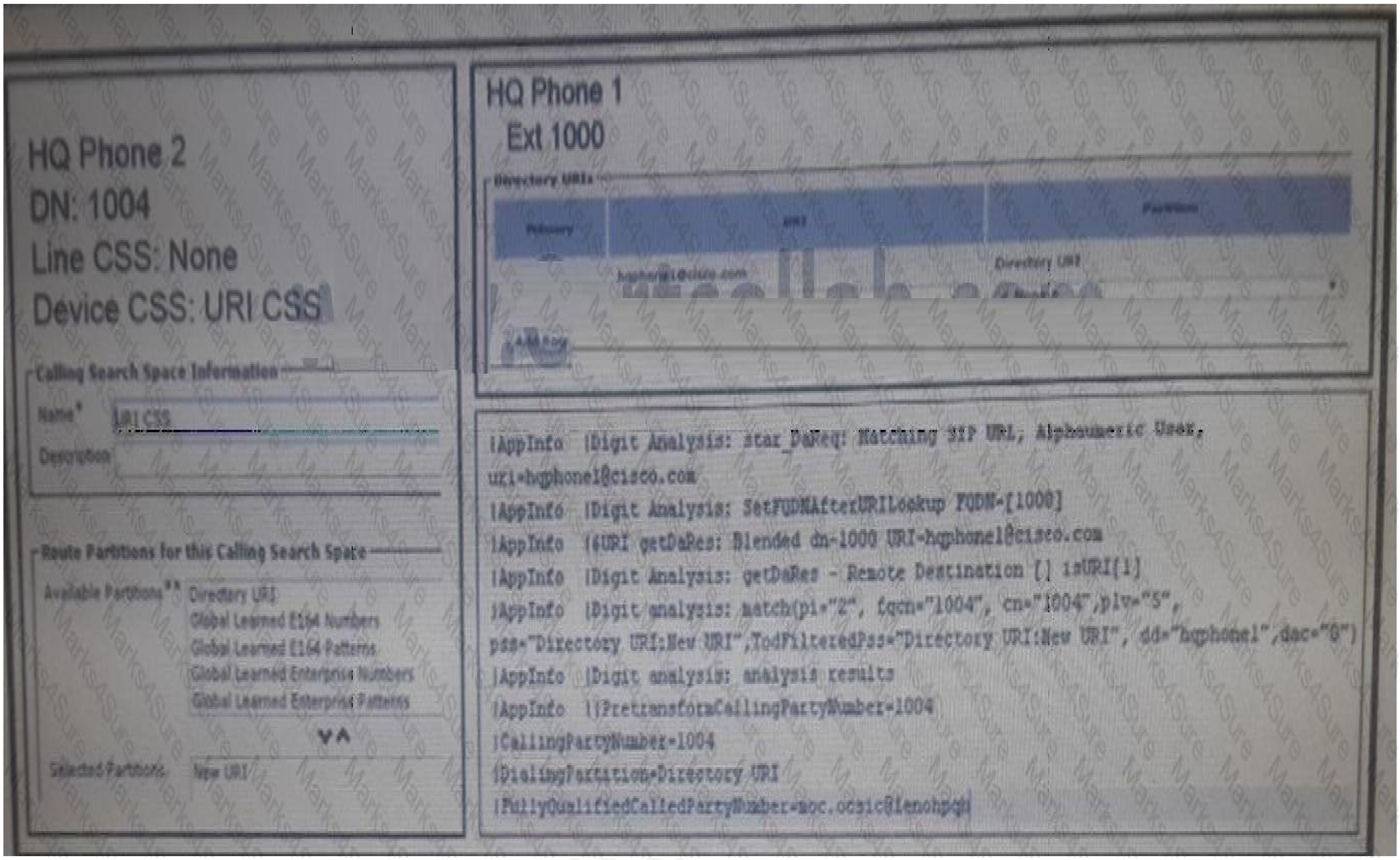
D. 20[*2-4]1 configured with urgent priorit

E. 20[1-4]1 configured with nonurgent priority

Answer: C

Question #:224 - (Exam Topic 1)

Refer to the exhibit.



A Cisco collaboration engineer has been asked to remove the ability HQ phone 2 to dial HQ phone 1 by URI dialling. After removing the partition assigned to hqphone1@cisco.com HQ phone 2's CSS, HQ phone 2 is still able to reach HQ phone 1. Why is the HQ phone 1 still reachable using URI dialling?

A. Directory URI Alias partition has been defined in Enterprise parameters

B. Phone needs to be reset for changes to take effect

C. Directory URI partition cannot be deleted therefore still will be reachable

D. CSS Changes failed to be applied after hitting save due to Database replication issue

Answer: A

Question #:225 - (Exam Topic 1)

A client wants to play and compose voice messages from Microsoft Outlook. What is required for this functionality?

A. single inbox synchronisation with send and draft messages

B. single inbox with ViewMail

C. single inbox with mailboxes larger than 2 GB

D. single inbox user message delivery with folder deletion

Answer: B

Question #:226 - (Exam Topic 1)

Which two actions does the Cisco Unified IP phone use the initial Trust list to perform? (Choose two.)

A. Decrypt secure XML files

B. Encrypt RTP traffic for ip phones that are not registered to the same call manager cluster

C. Download background image files

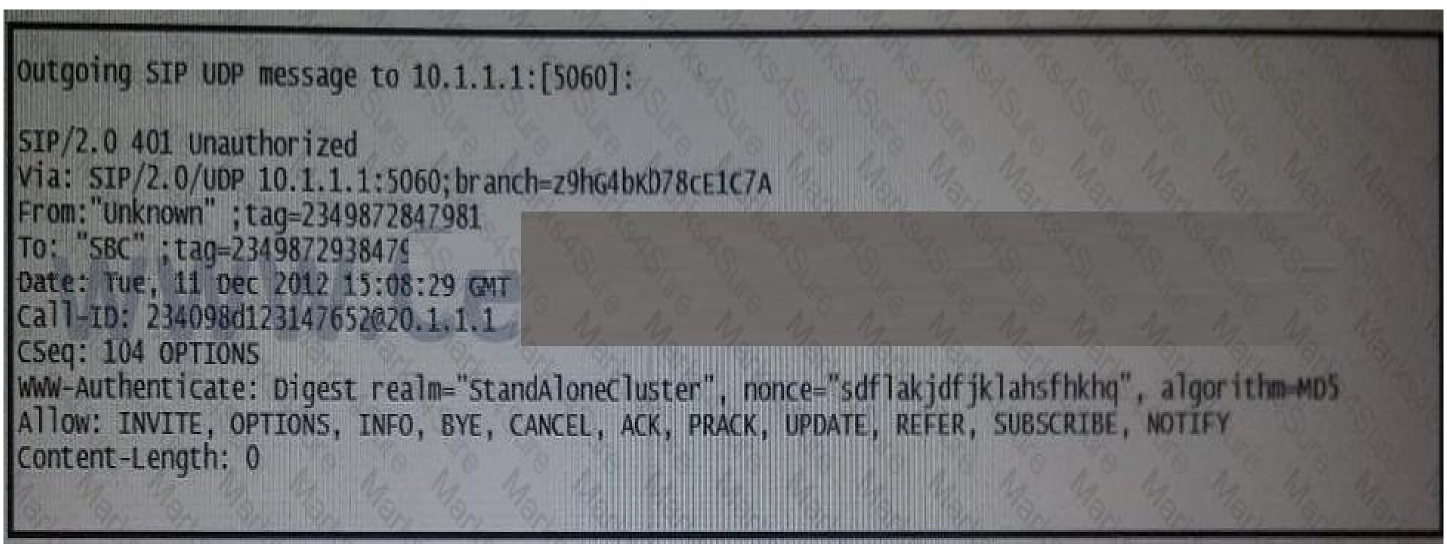
D. Authenticate their configuration file signature

E. Talk securely to CAPF which is a prerequisite to support configuration files encryption

Answer: D E

Question #:227 - (Exam Topic 1)

Refer to the exhibit.


```
Outgoing SIP UDP message to 10.1.1.1:[5060]:

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.1.1.1:5060;branch=z9hG4bKD78cE1C7A
From:"Unknown" ;tag=2349872847981
To: "SBC" ;tag=234987293847
Date: Tue, 11 Dec 2012 15:08:29 GMT
Call-ID: 234098d123147652@20.1.1.1
CSeq: 104 OPTIONS
WWW-Authenticate: Digest realm="StandAloneCluster", nonce="sdflakjdfjklahsfhkhq", algorithm=MD5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Content-Length: 0
```

The exhibit shows an outgoing SIP 401 response message from Cisco Unified

Communications Manager to a SIP VoIP service provider gateway. Which action can the Cisco Unified

Communication Manager Systems administrator use to change the response to "200 OK"?

A. Disable OPTIONS ping on Cisco Unified Communications manager sip trunk,

B. Create an SIP response alias to force outgoing 401 messages to "200 OK"

C. Make sure the gateway IP address of the SIP VoIP service provider is defined correctly in Cisco Unified Communications Manager SIP trunk

D. Enable OPTIONS ping on Cisco Unified Communications Manager SIP trunk

E. Disable digest authentication on Cisco Unified Communications Manager SIP trunk.

Answer: E

Question #:228 - (Exam Topic 1)

Refer to the exhibit.

```
1087: NOT 13:10:18.180262 VPNC: VPN cert chain trusted
1088: DBG 13:10:18.181643 VPNU: SM wakeup - chld=0 tmr=0 io=1 res=0
1089: NOT 13:10:18.183502 VPNC: Using URL addr = (https://209.165.200.225/phonevpn)
1090: NOT 13:10:18.184080 VPNC: Host name = (209.165.200.225)
1091: NOT 13:10:18.184840 VPNC: Parsing host name from certificate...
1092: NOT 13:10:18.185512 VPNC: hostID not found in subject name
1093: ERR 13:10:18.186303 VPNC: hostIDCheck failed!!!
1094: ERR 13:10:18.188052 VPNC: ssl_state_cb: TLSv1: write: alert: fatal:unknown CA
1095: ERR 13:10:18.188968 VPNC: alert_err: SSL write alert: code 48, unknown CA
1096: ERR 13:10:18.189991 VPNC: create_ssl_connection: SSL_connect ret -1 error 1
1097: ERR 13:10:18.191394 VPNC: SSL: SSL_connect: SSL_ERROR_SSL (error 1)
1098: ERR 13:10:18.192406 VPNC: SSL: SSL_connect: error:14090086:SSL
routines:SSL3_GET_SERVER_CERTIFICATE:certificate verify failed
1099: ERR 13:10:18.193416 VPNC: create_ssl_connection: SSL setup failure
1100: ERR 13:10:18.195227 VPNC: do_login: create_ssl_connection failed
1101: NOT 13:10:18.196442 VPNC: vpn_stop: de-activating vpn
1102: NOT 13:10:18.197296 VPNC: vpn_set_auto: auto -> auto
1103: NOT 13:10:18.197904 VPNC: vpn_set_active: activated -> de-activated
1104: NOT 13:10:18.198711 VPNC: set_login_state: LOGIN: 1 (TRYING) --> 3 (FAILED)
1105: NOT 13:10:18.199577 VPNC: set_login_state: VPNC : 1 (LoggingIn) --> 3 (LoginFailed)
1106: NOT 13:10:18.200518 VPNC: vpnc_send_notify: notify type: 1 [LoginFailed]
```

A phone VPN failed to establish a VPN with the Cisco ASA. The support engineer downloaded the console logs and analyzed them. When two steps resolve this issue? (Choose two)

A. Configure user and password authentication instead of certificate only.

B. Uncheck the enable Host ID check checkbox under the VPN profile in Cisco Unified CM.

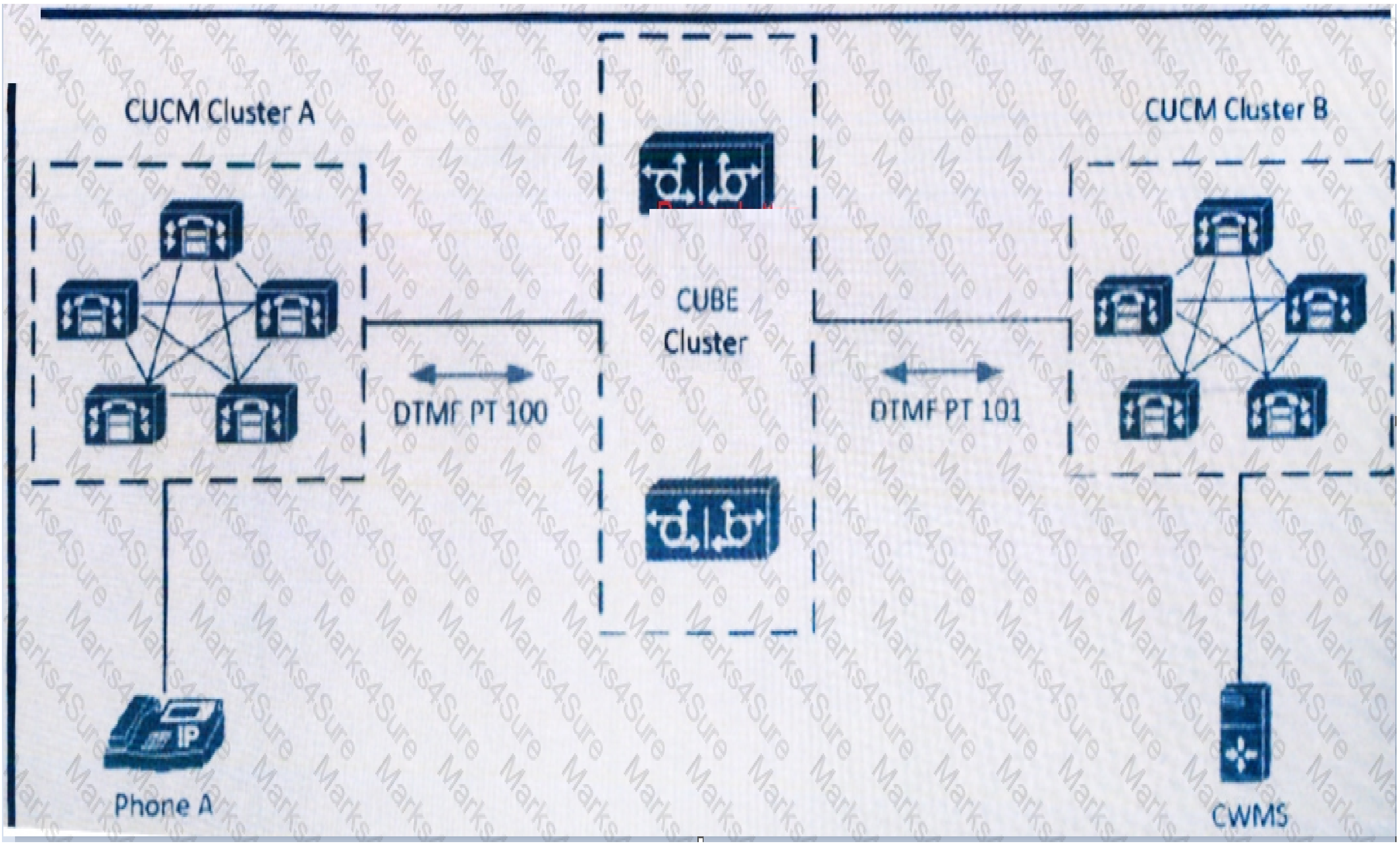
C. Reset the Cisco Unified CM TFTP service to allow caching of the new certificate.

D. Delete the current certificate so the phone can download a new one.E .Register the phone internally to download the new configuration.

Answer: B

Question #:229 - (Exam Topic 1)

Refer to the exhibit.



A CUBE Cluster is working in HSRP box-to-box failover model. When the phone A calls Cisco WebEx meeting server to start a conference session, no DTMF tones are recognized. Which configuration change will fix this problem when configured on both CUBEs?

A. Voice-class sip asymmetric payload dtmf in dial-peer configuration

B. Dtmf-relay rtp-nte digitdrop in the dial-peer configuration

C. Media flow-around under voice service voip configuration

D. Modem relay nse payload-type101 underglobal sip configuration

E. Asymmetric payload full configured under global sip configuration

Answer: E

Explanation

Symmetric and Asymmetric Calls

Cisco UBE supports dynamic payload type negotiation and interworking for all symmetric and asymmetric payload type combinations. A call leg on Cisco UBE is considered as symmetric or asymmetric based on the payload type value exchanged during the offer and answer with the endpoint:

• A symmetric endpoint accepts and sends the same payload type.

• An asymmetric endpoint can accept and send different payload types.

The Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP-to-SIP Calls feature is enabled by default for a symmetric call. An offer is sent with a payload type based on the dial-peer configuration. The answer is sent with the same payload type as was received in the incoming offer. When the payload type values negotiated during the signaling are different, the Cisco UBE changes the Real-Time Transport Protocol (RTP) payload value in the VoIP to RTP media path.

To support asymmetric call legs, you must enable The Dynamic Payload Type Interworking for DTMF and Codec Packets for SIP-to-SIP Calls feature. The dynamic payload type value is passed across the call legs, and the RTP payload type interworking is not required. The RTP payload type handling is dependent on the endpoint receiving them.

Configuring global SIP asymmetric payload support.

Example:

Router(conf-serv-sip)# asymmetric payload full

The dtmf and dynamic-codecs keywords are internally mapped to the full keyword to provide asymmetric payload type support for audio and video codecs, DTMF, and NSEs.

Question #:230 - (Exam Topic 1)

A CUCM engineer has deployed Type B SIP Phones on a remote site and no SIP dial rules were deployed for these phones. How Will CUCM receive the DTMF after the phone goes off- hook and the button are pressed?

A. Each digit will be received by CUCM in a SIP NOTIFY message as soon as they are pressed

B. The first digit will be received in a sip invite and subsequent digits will be received using NOTIFY message as soon as they are pressed.

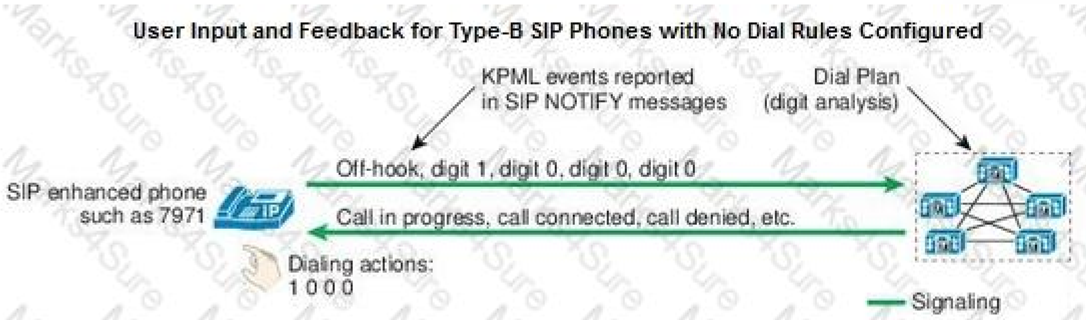
C. Each digit bill be received by CUCM in a SIP INVITE as soon as the dial soft key has been pressed.

D. All digits will be received by CUCM in a SIP INVITE as soon as the dial soft key has been pressed

Answer: A

Explanation

Type-B IP telephones offer functionality based on the Key Press Markup Language (KPML) to report user key presses. Each one of the user input events will generate its own KPML-based message to Unified CM. From the standpoint of relaying each user action immediately to Unified CM, this mode of operation is very similar to that of phones running SCCP.



Every user key press triggers a SIP NOTIFY message to Unified CM to report a KPML event corresponding to the key pressed by the user. This messaging enables Unified CM's digit analysis to recognize partial patterns as they are composed by the user and to provide the appropriate feedback, such as immediate reorder tone if an invalid number is being dialed.

In contrast to Type-A IP phones running SIP without dial rules, Type-B SIP phones have no Dial key to indicate the end of user input. A user dialing 1000 would be provided call progress indication (either ringback tone or reorder tone) after dialing the last 0 and without having to press the Dial key. This behavior is consistent with the user interface on phones running the SCCP protocol.

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/5x/50dialpl.html#wp1090653

https://supportforums.cisco.com/document/87236/working-concept-sccp-sip-phones-and-dial-rules

Question #:231 - (Exam Topic 1)

Refer to the exhibit.

```
voice service voip
 no ip address trusted authenticate
 rtcp keepalive
 mode border-element
 allow-connections sip to sip
 redundancy
 no supplementary-service sip refer
 signaling forward unconditional
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  min-se 360 session-expires 360
  header-passing
  error-passthru
  asserted-id pai
  midcall-signaling passthru
  privacy-policy passthru
  pass-thru headers unsupp
  pass-thru content unsupp
  no call service stop
```

Which two SIP packet handing behaviour will result with this cisco Unified Border Element

(CUBE) configuration? (Choose two)

A. Unsupported content/MIME pass-through

B. SIP Refer is not support when received on this CUBE

C. Privacy headers received on SIP message will be replaced with NON-privacy headers on this CUBE

D. P-Preferred identities

E. Mid-call codec changes

Answer: A E

Explanation

Configuring Cisco UBE for Unsupported Content Pass-through at the Global Level

To configure Unsupported Content Pass-through on an Cisco Unified Border Element at the global level, perform the steps in this section.

SUMMARY STEPS

1. enable

2. configure terminal

3. voice service voip

4. sip

5. pass-thru {content {sdp | unsupp} | headers {unsupp | list tag}}

6. exit

http://www.cisco.com/c/en/us/td/docs/ios/voice/cube/configuration/guide/15_0/vb_15_0_Book/vb-gw-sipsip.htm

Configuring Passthrough SIP Messages at the Global Level

Perform this task to configure passthrough SIP messages d at the global level:

SUMMARY STEPS

1. enable

2. configure terminal

3. voice service voip

4. sip

5. mid-call signaling passthru media-change

6. exit

http://www.cisco.com/en/US/docs/ios-xml/ios/voice/cube_proto/configuration/15-2mt/cube-midcall-reinvite.htm

Question #:232 - (Exam Topic 1)

A customer asks you to build a SIP/SIMPLE IM federation between a Cisco Unified Communication Manager IM & Presence environment and a Microsoft Lync 2013 environment. The Lync administrator insists that TLS is used for the federation and that the SSL certificate used by Cisco Unified Communication Manager IM & Presence is signed by the private CA of the enterprise. Which certificate must be in the Certificate Signing Request to accommodate the Lync administrator requirements?

A. cupxmpp

B. tomcat

C. cupxmpps2s

D. ipsec

E. cup

Answer: C

Question #:233 - (Exam Topic 1)

Refer to the exhibit.

```
application
service CCM http://172.16.100.50:8080/ccmivr/pages/IVRMainpage.vxml

dial-peer voice 4100 pots
service ccm
incoming called-number 4100


dial-peer voice 4101 voip
destination-pattern 4100
codec g729r8
dtmf-relay rtp-nte
session target ipv4:172.16.101.50
```

A collaboration engineer configured MVA for a company using an existing Cisco IOS voice gateway.

When testing inbound calls it is found that they are all failing.

Which two sets of configuration changes fix this problem? (Choose two)

A. Dial-peer voice 4100 potsServices ccmDestination-pattern 4100$

B. Dial-peer voice 4101 voipDtmf-relay h245-alphanumericSession target ipv4:172.16.100.50

C. Dial-peer voice 4101 voipDtmf-relay h245-alphanumericSession target ipv4:172.16.100.50Codec g711ulaw

D. Dial-peer voice 4100 potsservice CCM

E. Dial-peer voice 4100 potsservice CCMIncoming called-number ,T

F. Dial-peer voice 4101 VOIPDtmf-relay h245-signalSession target ipv4:172.16.100.50Codec g711alaw

Answer: A C

Question #:234 - (Exam Topic 1)

Which definition is included in a Cisco UC on UCS TRC?

A. Required RAID configuration When 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

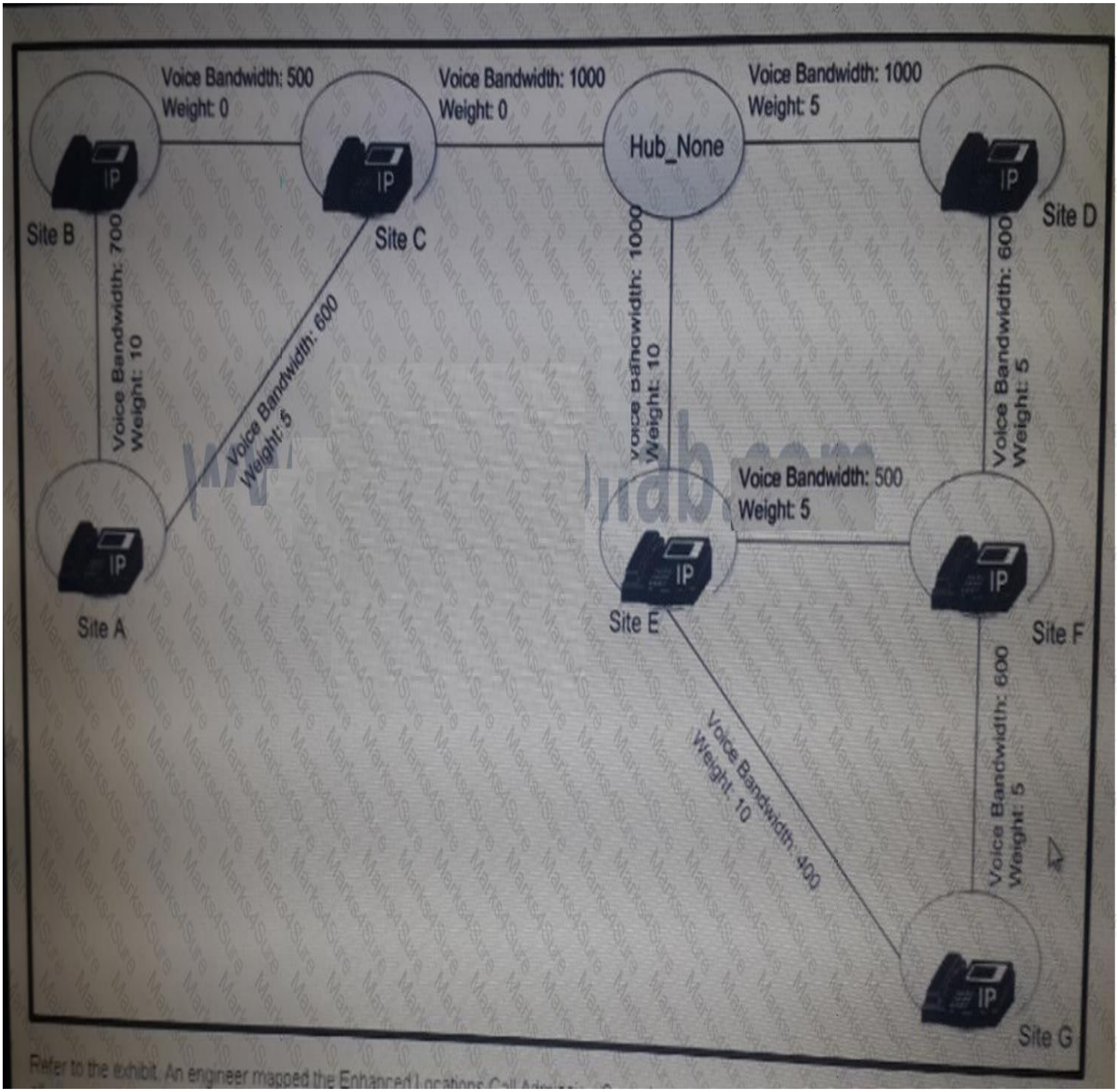
D. Server model and local components (CPU, RAM, adapters, local storage) by name only. Part numbers are not included because they change over time

E. Configuration settings and patch recommendations for VMware software

Answer: A

Question #:235 - (Exam Topic 1)

Refer to the exhibit.



An engineer mapped the enhanced location call admission control configuration to match the physical links bandwidth allowances. Assuming no other calls are consuming any bandwidth, how many G722 calls are allowed between site A and site G?

A. 5

B. 7

C. 12

D. 25

E. 37

Answer: B

Question #:236 - (Exam Topic 1)

Which three unified CM features are affected by application dial rules?

A. Device mobility

B. Manager auto-attendant

C. Extension mobility

D. Web dialer

E. Unified mobility

F. Manager assistant

Answer: D E

Question #:237 - (Exam Topic 1)

A Cisco collaboration architect is evaluating a list of codecs to use in a voice infrastructure. Which three facts are associated with iSAC and should be considered in the decision? (Choose three)

A. The codec has better quality with less bandwidth for sideband applications

B. The codec will not be supported in TDM voice gateways

C. The codec will adjust its bandwidth consumption to the network conditions

D. The codec will not be available for H.323 and MGCP devices

E. The codec will not support low complexity

F. The codec will not be supported by SCCP configured on DSPFARMS

Answer: A C E

Question #:238 - (Exam Topic 1)

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

Answer: B F

Question #:239 - (Exam Topic 1)

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

A. 331.2 kb/s

B. 261.6 kb/s

C. 238.4 kb/s

D. 347.8 kb/s

E. 274.7 kb/s

Answer: A

Question #:240 - (Exam Topic 1)

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
```

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

Answer: B C

Question #:241 - (Exam Topic 1)

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

Answer: B

Question #:242 - (Exam Topic 1)

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

A. A Location Bandwidth Manager Hub Group must be created for each cluster.

B. Links must be created to the Shadow location.

C. The location name must be the same on both clusters.

D. SIP ICT must use the Shadow location.

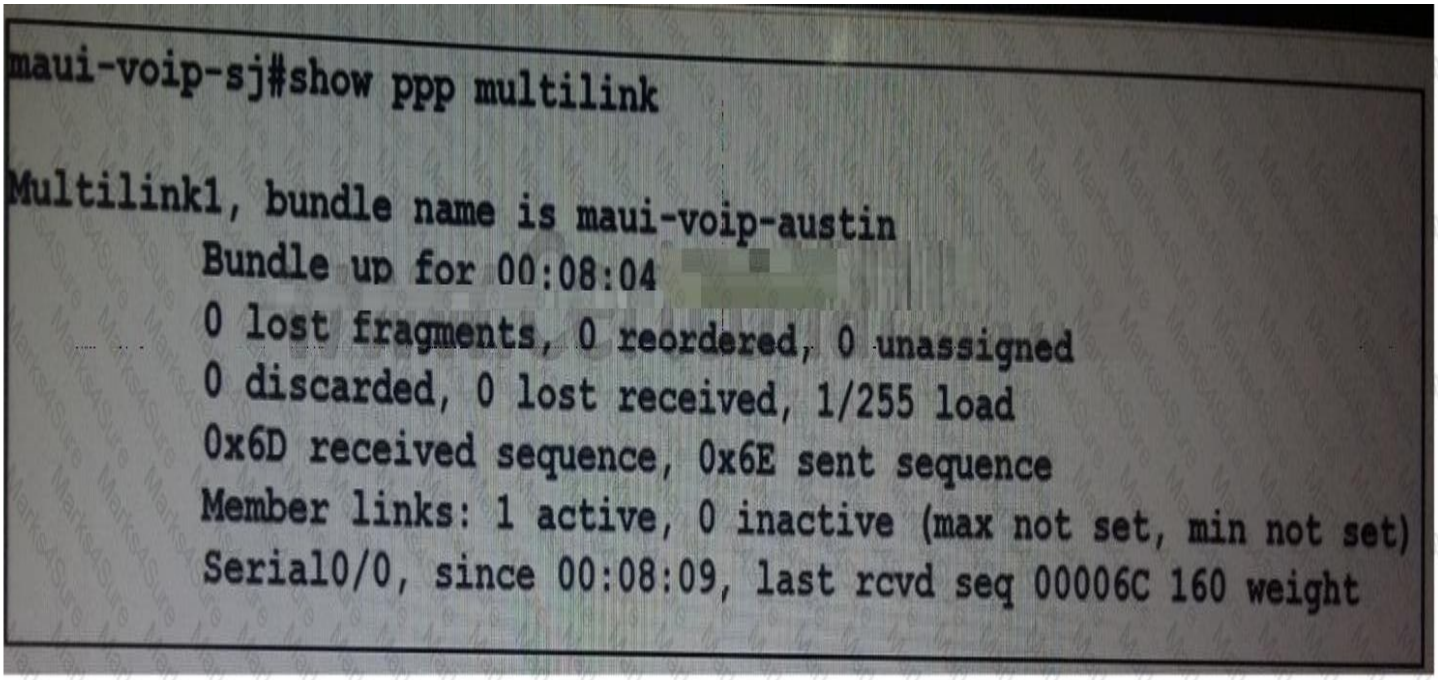
E. The Location Bandwidth Manager Service should be started on only two servers in each cluster.

F. The Cisco Unified Communications Manager version must be 8 .6 or higher

Answer: A C D

Question #:243 - (Exam Topic 1)

Refer to the exhibit.



Which two commands, when configured on a PPP multilink interface allow a fragment size of 160 bytes? (Choose two).

A. bandwidth 128

B. ppp multilink fragment-delay 40

C. ppp multilink fragment-delay 15

D. bandwidth 384

E. ppp multilink fragment-delay 10

F. bandwidth 512

G. ppp multilink fragment-delay 20

H. bandwidth 768

Answer: A E

Question #:244 - (Exam Topic 1)

The Video engineer wants to enable the LATM codec to allow video endpoint to communicate over audio with other IP devices. Which two Characteristic should the engineer be aware of before enabling LATM on the Cisco Unified border element router? (Choose two)

A. Dual tone Multifrequency interworking with LATM codec is not supported.

B. Codec transcoding between LATM and other codecs is not supported.

C. SIP UPDATE massage outlined in RFC3311 is not supported.

D. Box-to-Box High availability support feature is not supported.

E. Configure LATM under a voice class or dial peer is not supported.

F. Basic calls using flow-around or flow-through is not supported.

Answer: A B

Explanation

Restrictions for AAC-LD MP4A-LATM Codec Support on Cisco UBE

Cisco UBE does not support the following:

- Codec transcoding between MP4A-LATM and other codecs
- Dual-tone Multifrequency (DTMF) interworking with MP4A-LATM codec
- Non-SIP-SIP, that is, SIP to other service provider interface (SPI) interworking with MP4A-LATM codec

http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book.pdf

Question #:245 - (Exam Topic 1)

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.

Answer: A C

Explanation

Guidelines for Agent Phone Configuration

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

Choose Device > Phone in

Unified Communications Manager Administration. The Find and List Phones window is displayed.

- Enter search criteria to locate a specific phone and click Find. A list of phones that match the search criteria is displayed. Click the device name of the phone to which you want to add a directory number. The Phone Configuration window is displayed.
- In the Unified Communications Manager Administration Phone Configuration web page, select the required Association Information (on the left) to get to the Directory Number Configuration web page. On this page, make the following changes:
- In the Multiple Call/Call Waiting Settings section, set the Maximum Number of Calls to 2 (default is 4) for Cisco Unified IP Phones 7900 Series and 3 for Cisco Unified IP Phones 8961, 9951, and 9971.

Note:If you are using Cisco Finesse for your agent desktop, you must set the Maximum Number of Calls to 2 for all agent phones.

- In the Multiple Call/Call Waiting Settings section, set the Busy Trigger value to 1 (default is 2).
- In the Call Forward and Call Pickup Settings section, verify that you do not forward any Unified Communications Manager device to the Unified CCX extension of an agent.
- In the Call Forward and Call Pickup Settings section, verify that you do not configure the Unified CCX extension of an agent to forward to a Unified CCX route point.
- 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 Phone 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 Securities 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.
- Do not forward any Unified Communications Manager device to the Unified CCX extension of an

agent.

- Do not configure the Unified CCX extension of an agent to forward to a Unified CCX route point.
- Do not use characters other than the numerals 0 to 9 in the Unified CCX extension of an agent.
- Do not configure two lines on an agent phone with the same extension when both lines exist in different partitions.
- Do not assign a Unified CCX extension to multiple devices.
- 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.)
- To use Cisco Unified IP Phones 9900 Series, 8900 Series, and 6900 Series as agent devices, the RmCm application user in Unified Communications Manager needs to have "Allow device with connected transfer/conference" option assigned to itself.

Old Question

Question #:246 - (Exam Topic 1)

In addition to SIP triggers types can invoke applications on Cisco Utility Express? (Choose two.)

A. JTAPI

B. Cisco Unified CM Telephony

C. VoiceView

D. IMAP

E. Voice mail

F. HTTP

Answer: A F

Question #:247 - (Exam Topic 1)

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 Not Available Hunt Option settings on the Line Group Configuration page

B. Calls are handled according to the When Queue Is Full settings on the Hunt Pilot Configuration page.

C. Calls are handled according to the When Maximum Wait Time Is Met settings on the Hunt Pilot Configuration page.

D. Calls are handled according to the Forward Hunt No Answer settings on the Hunt Pilot configuration page.

E. Calls are handled according to the When Maximum Wait Time Is Met settings in Cisco Unified Communications Manager Service Parameters.

Answer: C

Question #:248 - (Exam Topic 1)

In a Network Function Virtualization reference architecture, which two statements about virtualized network functions are true? (choose two)

A. VNF performs the orchestration and lifecycle management of the software resources that supports the virtualized infraestructure

B. VNF functionality includes control and management of the compute, storage and network resources in the NFV framework

C. VNF is the totally of all hardware and software components that built up the VNF environment

D. VNF is a virtualization of a legacy network function

E. One VNF can be deployed over multiple VMs where each VM hosts a single component of the VNF

Answer: A

Question #:249 - (Exam Topic 1)

What does a weight represent in the enhanced location Call Admission Control mechanism on Cisco Unified Communications Manager?

A. It defines bandwidth that is available on a link.

B. It defines bandwidth that is available between locations.

C. It is the amount bandwidth allocation for different types of traffic.

D. It is used to provide relative priority of a location.

E. It is used to provide relative priority of a link between locations.

Answer: E

Question #:250 - (Exam Topic 1)

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.

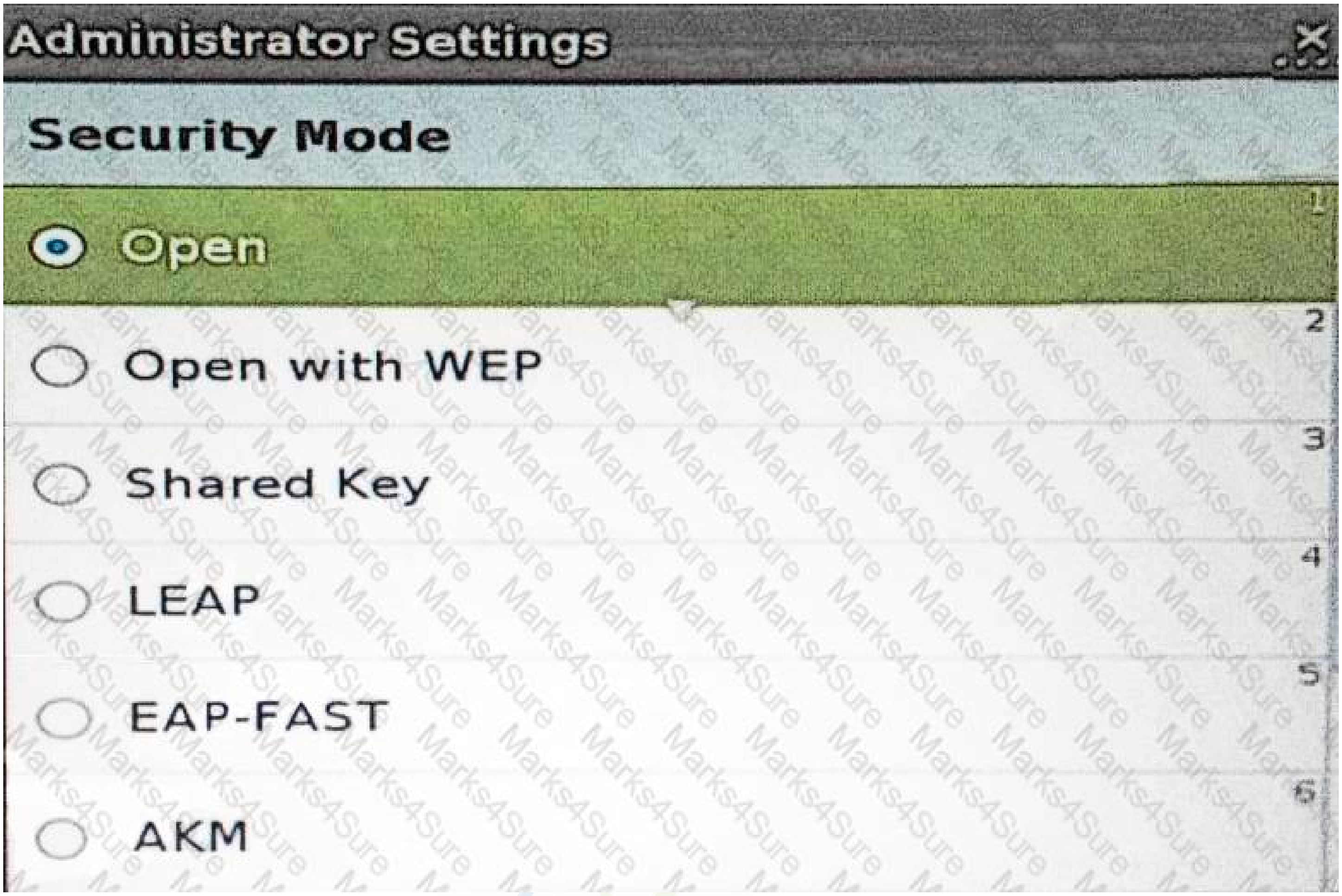
G. A server running Microsoft SQL server is required.

H. The server must have standard licensing.

Answer: B C D G

Question #:251 - (Exam Topic 1)

Refer to the exhibit.



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

Answer: B E F

Question #:252 - (Exam Topic 1)

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.

```
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
```

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

Answer: B C

Question #:253 - (Exam Topic 1)

A Cisco Unified Comunicator Manager Administrator is working on devices that are roaming within the company using device mobility. Which two configuration settings have priority over the device setting when using the roaming device pool? (choose two)

A. Physical Location

B. Device Mobility Calling Search Space

C. Calling Party Transformation CSS

D. Adjunct CSS

E. Media Resource Group List

F. Region

Answer: E F

Question #:254 - (Exam Topic 1)

Which four attributes are needed to determine the time to complete a TFTP file transfer process? (Choose four.)

A. Response Timeout

B. File type

C. File Size

D. round trip - time

E. network interface type

F. packet loss percentage

G. network throughput

Answer: A C D F

Question #:255 - (Exam Topic 1)

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.

Answer: C

Question #:256 - (Exam Topic 1)

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

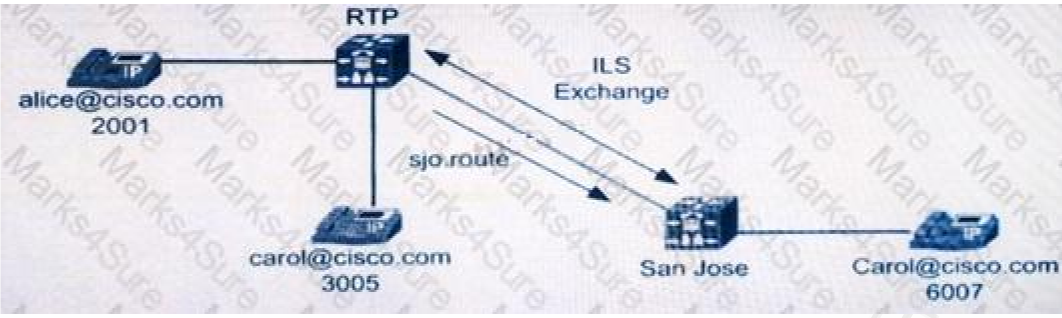
E. Cisco ASA Host ID check authentication

F. Cisco Unified CM user ID and password authentication

Answer: C E

Question #:257 - (Exam Topic 1)

Refer to the exhibit.



Which three events happen when Alice calls carol@cisco.com and the URI lookup policy on the Cisco Unified CM server has been set to case insensitive? (Choose three)

A. The RTP server routes the call to carol@cisco.com because remote URIs have priority

B. The RTP sever looks up to see if carol@cisco.com is associated to a local number

C. The San Jose server calls carol@cisco.com upon receiving the invite request

D. The San Jose server provide carol's directory URI using ILS exchange

E. The RTP server sends the call to carol@cisco.com because it has priority

F. The RTP server drops the call because it has two identical matches

Answer: B D E

Question #:258 - (Exam Topic 1)

Refer to the exhibit.

A reporting specialist found that calls answered by the phones are not being recorded.

Which two configuration changes can resolve this issue? (Choose two)

A. Enable automatic call recording

B. Assign a recording profile

C. Configure a CTI route point to the recorder

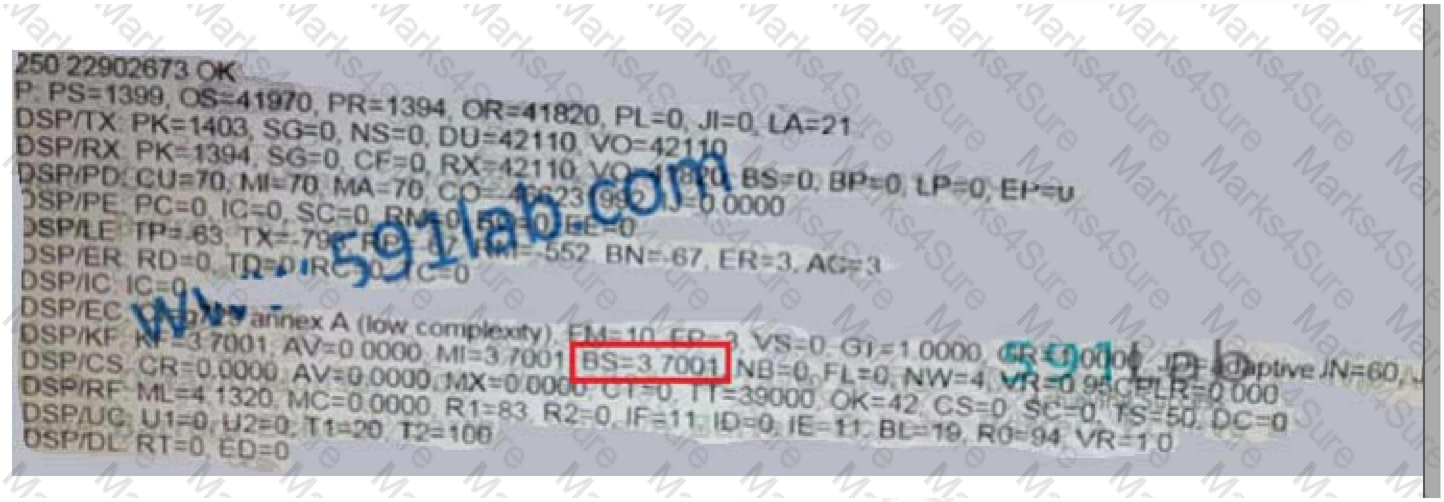
D. Activate built in bridge

E. Create a SIP trunk to the recording server

F. Manager assistant

Answer: A B

Question #:259 - (Exam Topic 1)



Which option is the MOS value in the Cisco IOS snippet?

A. 1.0000

B. 0.0000

C. 4.1320

D. 3

E. 3.7001

Answer: E

Question #:260 - (Exam Topic 1)

Cisco Unified CM User Options

------Permission Information---

Groups: Standard CTI Phone Administrator

Standard CTI Enabled

Roles: Standard CCM Admin Users

Standard CCM Phone Management

Standard CCM ReadOnly

Standard CTI EnabledRefer to the exhibit. An end user is trying to access a GUI page, but an "Access to requested resource is

denied, please contact administrator" error message is displayed.

Which setting resolves this issue and always end-user webpage access?

A. Standard Audit Users

B. Standard CCM Admin Users

C. Standard CAR Admin Users

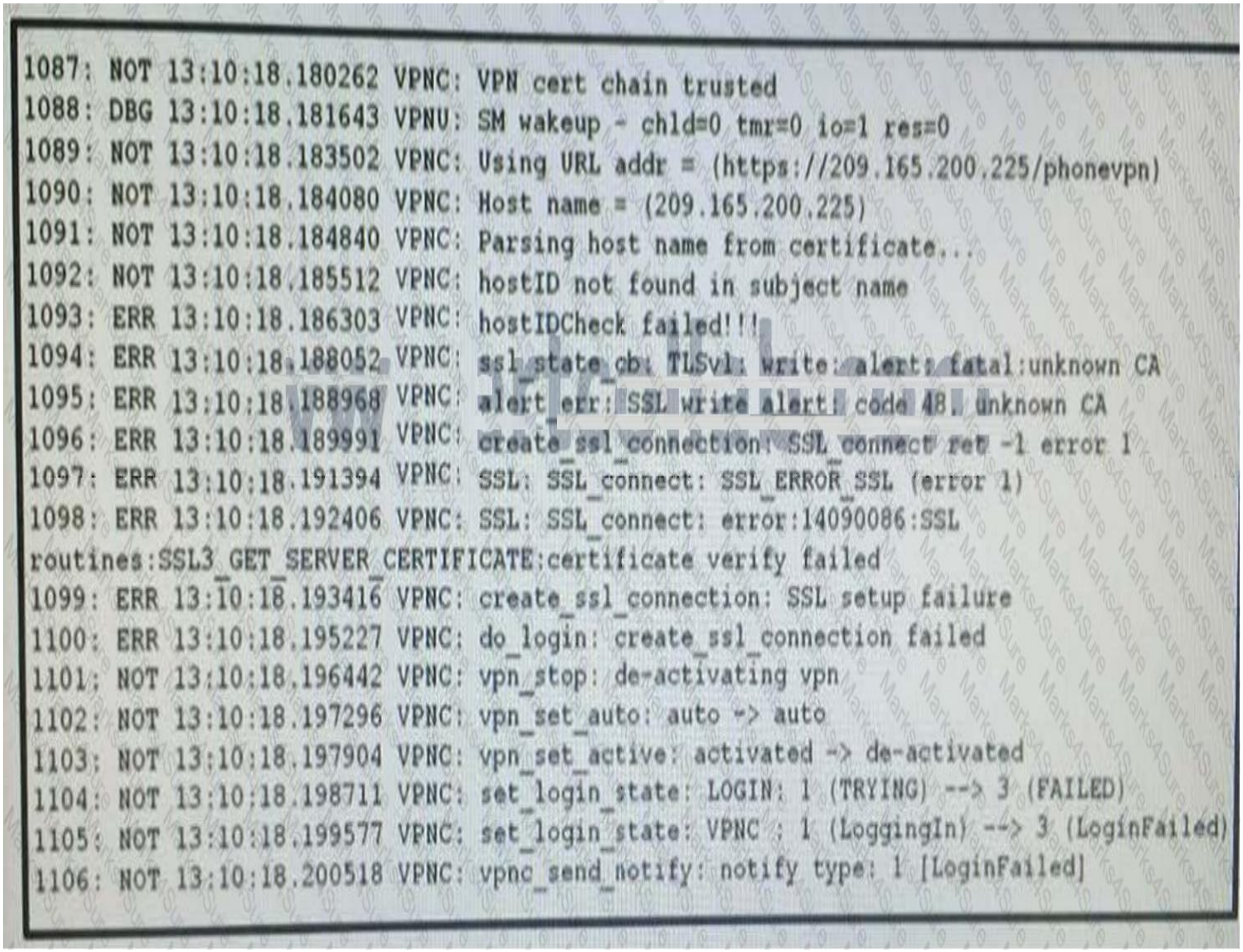
D. Standard CCM End Users

E. Standard CCM sSuper Users

Answer: D

Question #:261 - (Exam Topic 1)

Refer to the exhibit.



```
1087: NOT 13:10:18.180262 VPNC: VPN cert chain trusted
1088: DBG 13:10:18.181643 VPNU: SM wakeup - chld=0 tmr=0 io=1 res=0
1089: NOT 13:10:18.183502 VPNC: Using URL addr = (https://209.165.200.225/phonevpn)
1090: NOT 13:10:18.184080 VPNC: Host name = (209.165.200.225)
1091: NOT 13:10:18.184840 VPNC: Parsing host name from certificate...
1092: NOT 13:10:18.185512 VPNC: hostID not found in subject name
1093: ERR 13:10:18.186303 VPNC: hostIDCheck failed!!!
1094: ERR 13:10:18.188052 VPNC: ssl_state_cb: TLSv1: write: alert: fatal:unknown CA
1095: ERR 13:10:18.188968 VPNC: alert_err: SSL write alert: code 48, unknown CA
1096: ERR 13:10:18.189991 VPNC: create_ssl_connection: SSL_connect ret -1 error 1
1097: ERR 13:10:18.191394 VPNC: SSL: SSL_connect: SSL_ERROR_SSL (error 1)
1098: ERR 13:10:18.192406 VPNC: SSL: SSL_connect: error:14090086:SSL
routines:SSL3_GET_SERVER_CERTIFICATE:certificate verify failed
1099: ERR 13:10:18.193416 VPNC: create_ssl_connection: SSL setup failure
1100: ERR 13:10:18.195227 VPNC: do_login: create_ssl_connection failed
1101: NOT 13:10:18.196442 VPNC: vpn_stop: de-activating vpn
1102: NOT 13:10:18.197296 VPNC: vpn_set_auto: auto -> auto
1103: NOT 13:10:18.197904 VPNC: vpn_set_active: activated -> de-activated
1104: NOT 13:10:18.198711 VPNC: set_login_state: LOGIN: 1 (TRYING) --> 3 (FAILED)
1105: NOT 13:10:18.199577 VPNC: set_login_state: VPNC : 1 (LoggingIn) --> 3 (LoginFailed)
1106: NOT 13:10:18.200518 VPNC: vpnc_send_notify: notify type: 1 [LoginFailed]
```

A phone VPN failed to establish a VPN with the Cisco ASA. The support engineer downloaded the console logs and analysed them. When two steps resolve this issue? (Choose two)

A. Configure user and password authentication instead of certificate only

B. Uncheck the enable Host ID check checkbox under the VPN profile in Cisco Unified CM

C. Reset the Cisco Unified CM TFTP service to allow caching of the new certificate

D. Delete the current certificate so the phone can download a new one

E. Register the phone internally to download the new configuration

Answer: B E

Question #:262 - (Exam Topic 1)

A UCCX manager is monitoring several groups and has added a new team for the finance department. The manager can monitor all team members except those that have just been added in the finance department.

Which UCCX administration steps can resolve the issue?

A. Wizards> RmCm Wizards > Modify existing service queue

B. Subsystem> RmCm> Resources

C. Subsystem> RmCm> Contact service queue

D. Subsystem> Team> Assign supervisor and contact service queue

E. Tools> User management> Agent capability view

Answer: B

Question #:263 - (Exam Topic 1)

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 **utils cuc cluster activate** to re-establish intracluster.

D. A message left on the subscriber server during the outage may be lost during the cluster recovery.

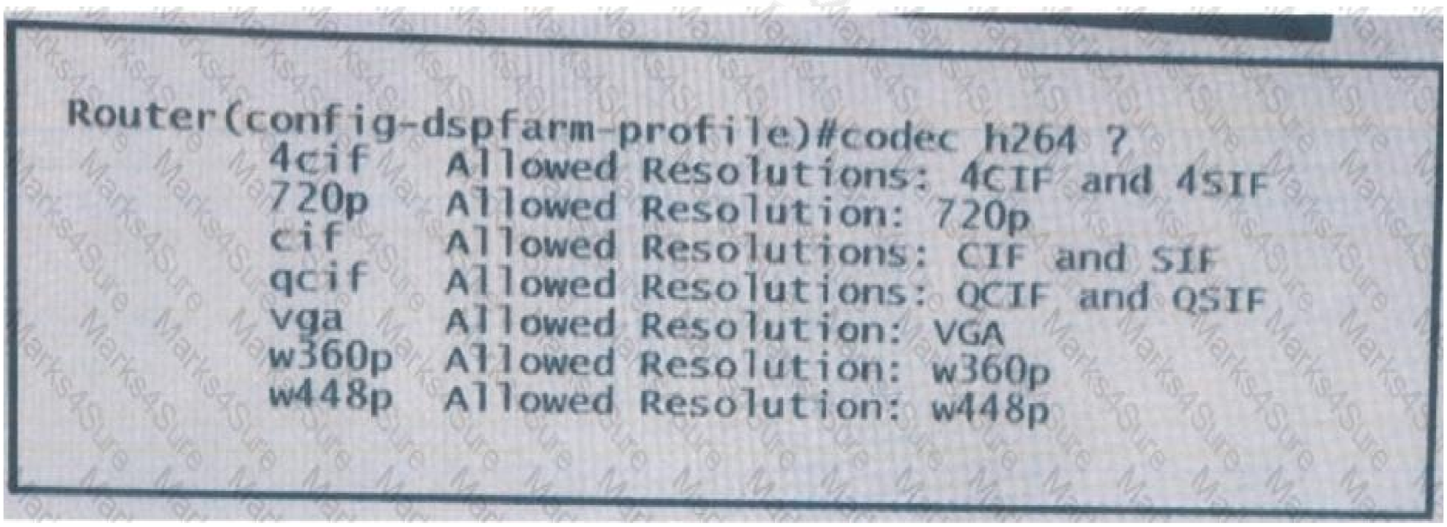
E. The replication between the nodes becomes defunct, requiring the administrator to run **utils cuc cluster renegotiate** to re-establish intracluster communication.

F. The Unity Connection Database can become corrupted, causing the need to reinstall the subscriber server.

Answer: A C

Question #:264 - (Exam Topic 1)

Exhibit:



```
Router(config-dspfarm-profile)#codec h264 ?
  4cif    Allowed Resolutions: 4CIF and 4SIF
  720p    Allowed Resolution: 720p
  cif     Allowed Resolutions: CIF and SIF
  qcif    Allowed Resolutions: QCIF and QSIF
  vga     Allowed Resolution: VGA
  w360p   Allowed Resolution: w360p
  w448p   Allowed Resolution: w448p
```

Which Cisco IOS multipoint video conferencing profile is being configured?

A. Homogeneous

B. guaranteed-audio

C. rendezvous

D. heterogeneous

E. guaranteed-video

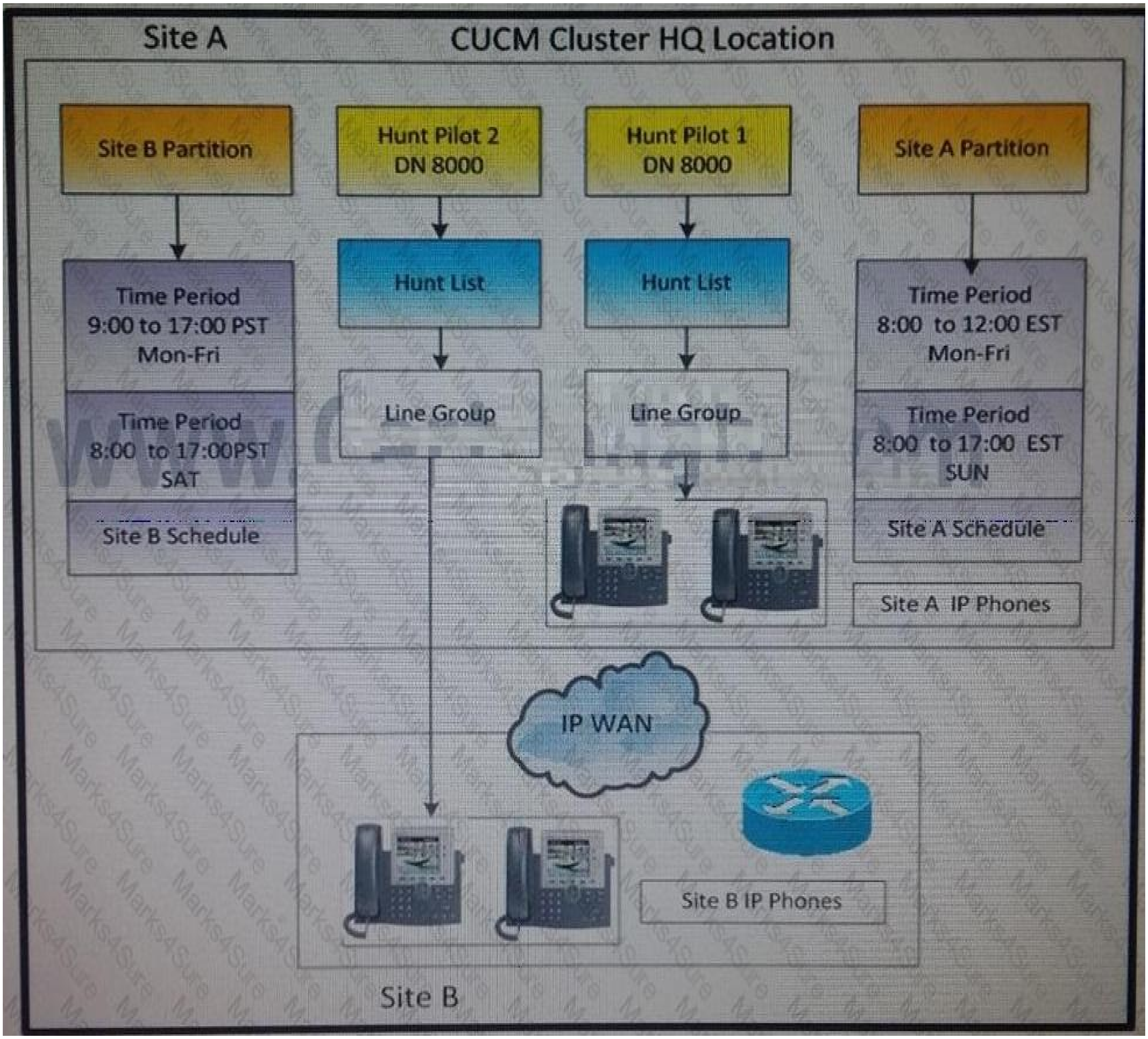
Answer: D

Explanation

Heterogeneous, which means that multiple codecs and multiple bit rates are supported

Question #:265 - (Exam Topic 1)

Refer to the exhibit.



Site A and site B have a 3-hour time difference. An administrator has time-of-day routing configured at site A and site B for all incoming calls to the main phone number, DN 8000. Two hunt pilots with the same DN are configured with the time periods of site A and B. Which Statement about the incoming calls is true?

A. Incoming calls to hunt pilot 8000 on Wednesday after 1500 PST are answered by the site B IP phone.

B. Incoming calls to hunt pilot 8000 on Saturday are answered by the site A IP phone.

C. Incoming calls to hunt pilot 8000 on Saturday after 17:00 PST are answered by the site A and site B IP phones.

D. Incoming calls to hunt pilot 8000 on Sunday are answered by the site B IP phone

Answer: A

Question #:266 - (Exam Topic 1)

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

Answer: C D G

Question #:267 - (Exam Topic 1)

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

Answer: B C

Question #:268 - (Exam Topic 1)

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

Answer: A B C

Question #:269 - (Exam Topic 1)

Which four Cisco Unified CM components can request media resource deallocation? (Choose four)

A. call dependency call control

B. unicast bridge control

C. device manager

D. music on hold control

E. line control

F. matrix control

G. call control

H. trusted relay point

Answer: A B D H

Question #:270 - (Exam Topic 1)

On 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.

Answer: D

Question #:271 - (Exam Topic 1)

The UCCX consultant is creating a new script must perform an action after the call is terminated between the agent and the caller used to perform this post call action?

A. The delay step to allow the call to continue after audio call termination.

B. the OnExcpetionClear step to allow the script to continue to the next command in the script

C. the Goto step to allow the script to continue to the next command in the script.

D. the CallSubFlow step to continue on to the next command.

Answer: C

Question #:272 - (Exam Topic 1)

VM´s are best suited for running what kind of workloads?

A. Monolithic Applications

B. Jobs that scale, but don't interact much

C. Cloud Native Applications

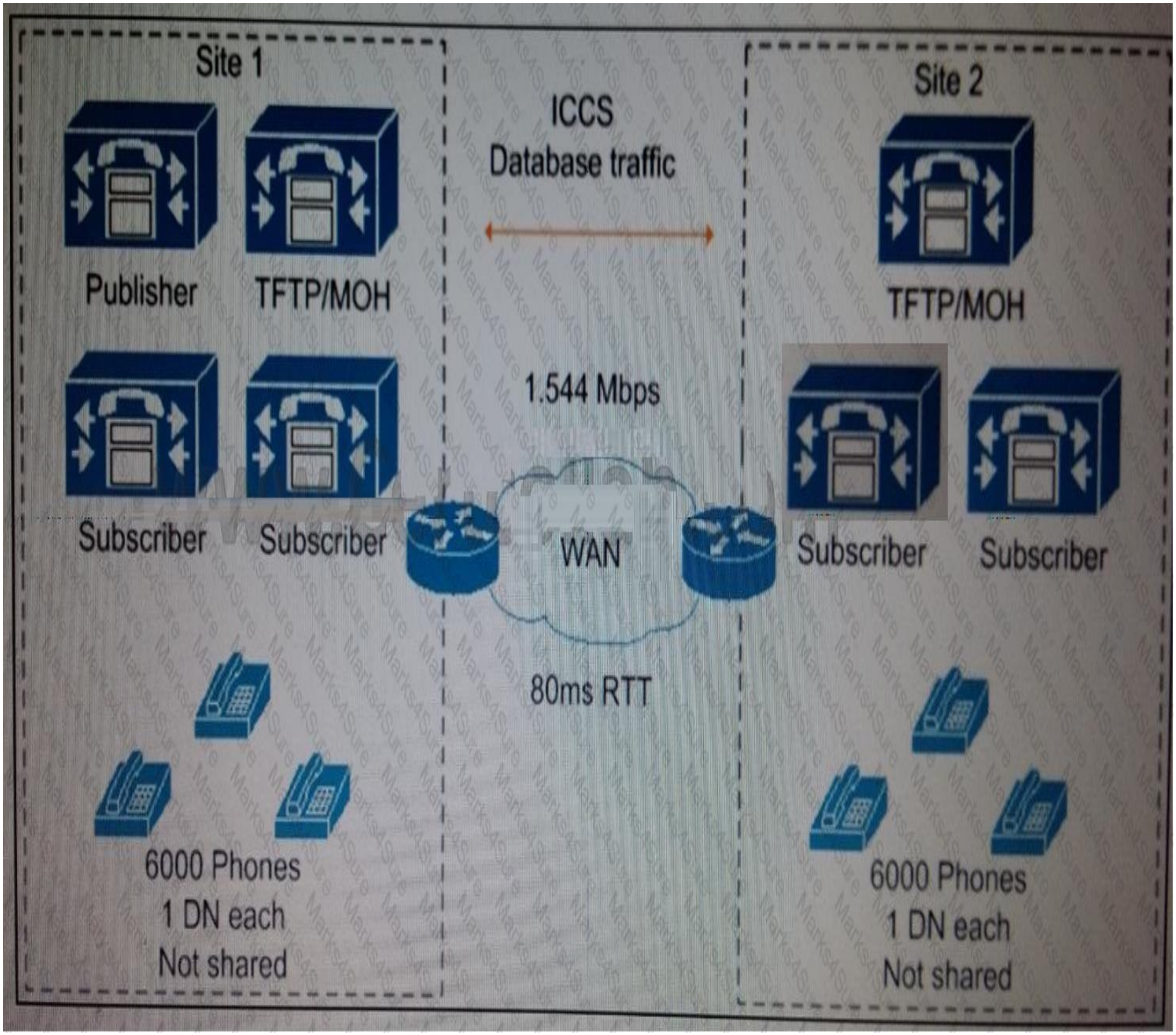
D. Micro Services

E. Cloud-Based applications

Answer: A

Question #:273 - (Exam Topic 1)

Refer to the exhibit.



Customer is planning to deploy a clustering over the WAN UCM topology with 2 subscribers at site 1 and 2 subscribers at site 2. How much bandwidth would be required between site 1 and site 2 to for database replication?

A. 1.544 Mbps

B. 3.088 Mbps

C. 4.632 Mbps

D. 6.176 Mbps

E. 7.772 Mbps

Answer: B

Question #:274 - (Exam Topic 1)

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

Answer: B

Question #:275 - (Exam Topic 1)

Refer to the exhibit.

```
!
ephone-dn 1
 number 1001
!
ephone-dn 2
 number 1002
!
ephone-dn 3
 number 1003
 ephone-hunt login
!
ephone-dn 4
 number 1004
!
ephone-dn 5
 number 1005
 ephone-hunt login
!
ephone-dn 6
 number 1006
!
ephone-hunt 1 peer
 list 1001,1002,1004,*
 hop 6
 final 1100
!
```

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

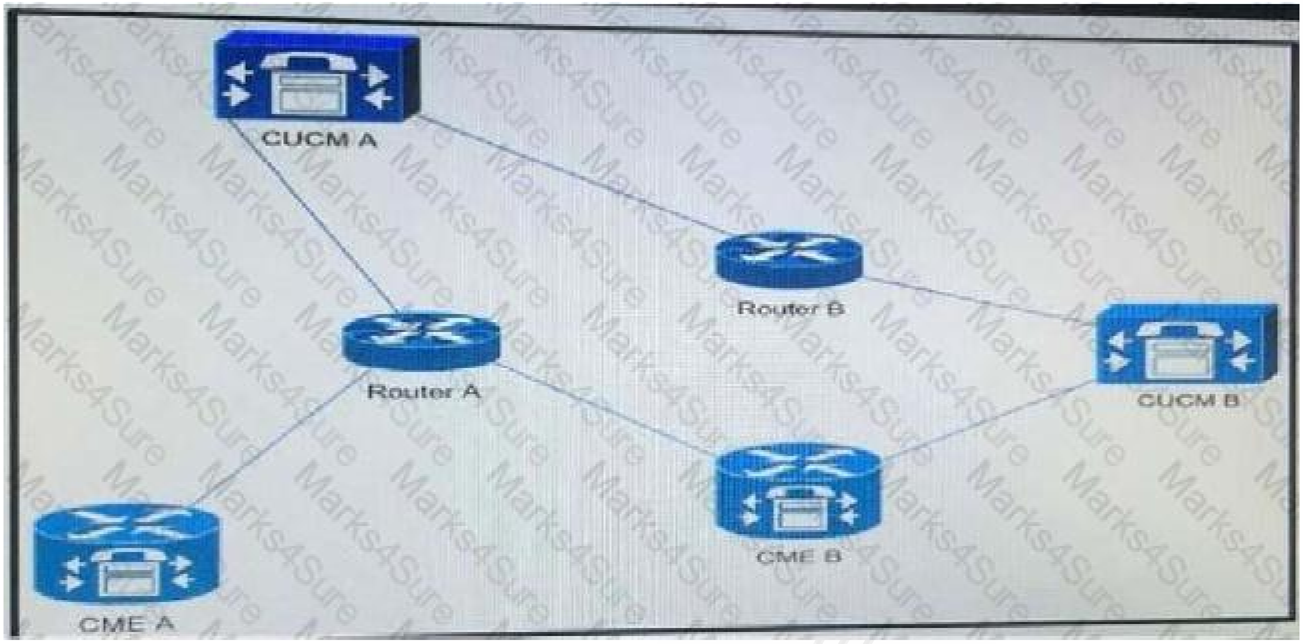
D. ephone-dn 4

E. ephone-dn 6

Answer: C

Question #:276 - (Exam Topic 1)

Refer to the exhibit.



A collaboration engineer is configuring dynamic call routing and DN learning between two Cisco UCM and Two

Cisco UCM express. What two configuration tasks will support this? (Choose two)

A. CME B should be configure as service advertisement forwarder only

B. CME B should be configure as service advertisement client and forwarder

C. Router A and CME B should be configure to use the same autonomous system number

D. Router A and CME B should be configure to use the same autonomous system number

E. Router B and CME B should be configure to use the same autonomous system number

F. CME B should be configured as service advertisement client only.

Answer: B C

Question #:277 - (Exam Topic 1)

A Jabber for window user is on a call with cisco telepresence EX90 endpoint at the same location. During the call, the video on the jabber for Windows application was high quality but the video on the EX90 was choppy and slow. When the administrator checked the service rate on the EX90 it showed 2048 Kbps. Which two configuration changes can fix this problem?

A. Lower the bit rate in the region configuration in communication manager between the endpoints.

B. Increase the location bandwidth for immersive video between the endpoints

C. Enable BFCP in the SIP profile for the jabber client

D. Enable H.263 on the EX90

E. Replace the camera for the jabber user with the precision HD USB camera

F. Increase the bandwidth between the jabber video client and the EX90

Answer: E F

Question #:278 - (Exam Topic 1)

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

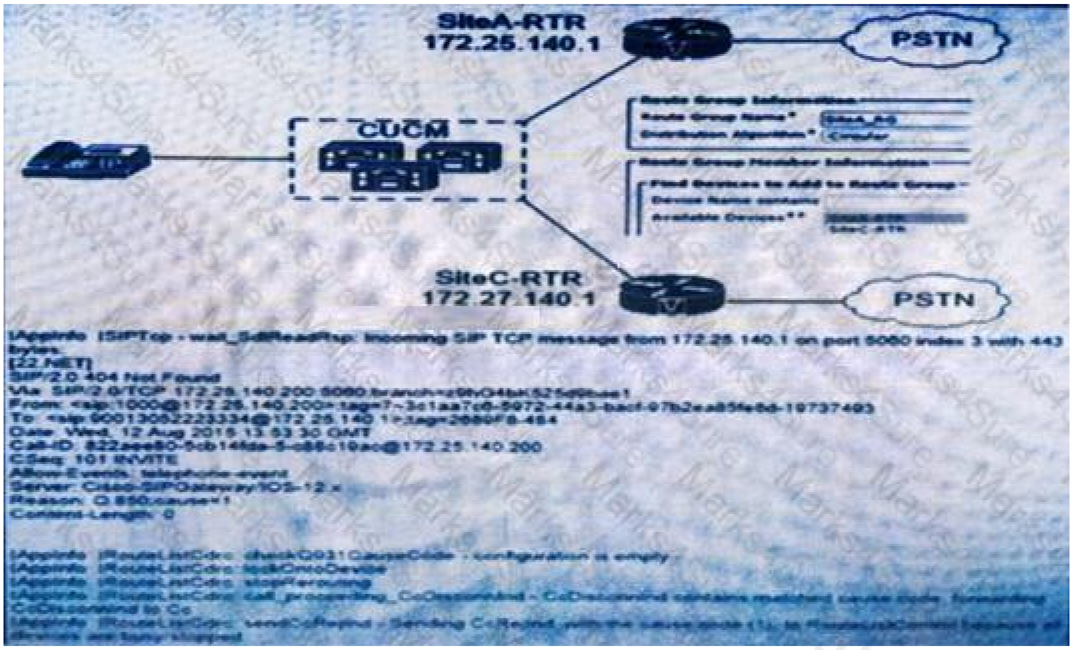
E. LLDP Power Priority

F. Day Display Not Active

Answer: A C

Question #:279 - (Exam Topic 1)

Refer to the exhibit.



A network engineer is troubleshooting a call routing issue where failed calls on the primary path

(SiteA-RTR) were not sent to the secondary path (SiteC-RTR).

Why is CUCM unable to extend the call setup through SiteC-RTR?

A. Stop routing on Q.931 Disconnect cause code is set to 27

B. Stop routing on Unallocated Number Flag is set to true

C. Stop Routing on User Busy Flag is set to true

D. Retry count for SIP Invite is set to 1

E. Retry count for SIP response is set to 1

Answer: B

Question #:280 - (Exam Topic 1)

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 manages any single digit from 0 to 9.

B. It manages any single digit from 0 to 9 or the asterisk (*) or pound (#) symbols.

C. It is a delimiter and has no significant dialling impact

D. It manages one or more digits from 0 to 9 or the asterisk (*) or pound (#) symbols.

E. It manages one or more digits from 0 to 9.

Answer: B

Question #:281 - (Exam Topic 1)

Refer to the exhibit.

```
dial-peer voice 1 voip
 description incoming from PSTN
 incoming called-number [2-9]..[2-9]......
dial-peer voice 2 voip
 description outbound to CUCM
 destination-pattern [2-9]..[2-9]......
 session protocol sipv2
 session target ipv4:10.10.10.10
```

A carrier delivers a SIP call to Cisco Unified CM through a Cisco Unified border Elements with the Invite destination different than "To" field. The Unified Cm Administration engineer sees that the calls go to the Invite destination instead of the "To" field Unified CM.

Which option shows how the engineer corrected that problem in the Cisco Unified border

Elements router?

O A.

```
voice class sip-profiles 10
 request INVITE peer-header sip To copy
  "sip:(.*)@" u01
 request INVITE sip-header SIP-Req-URI modify
  ".*@(.*)" INVITE sip :\u01@\1"
voice class sip-copylist 1
 sip-header To
dial-peer voice 1 voip
 voice-class sip copy-list 1
dial-peer voice 2 voip
 voice-class sip profiles 10
```

B.

```
voice class sip-profiles 10
 request INVITE peer-header sip INVITE copy
  "sip:(.*)@" u01
 request INVITE sip-header To modify
  ".*@(.*)" "INVITE sip:\u01@\1"
voice class sip-copylist 1
 sip-header To
dial-peer voice 1 voip
 voice-class sip copy-list 1
dial-peer voice 2 voip
 voice-class sip profiles 10
```

C.

```
voice class sip-profiles 10
 request INVITE peer-header sip To copy
  "sip:(.*)@" u01
 request INVITE sip-header SIP-Req-URI modify
  ".*@(.*)" "INVITE sip:\u01@\1"
voice class sip-copylist 1
 sip-header To
dial-peer voice 2 voip
 voice-class sip profiles 10
 voice-class sip copy-list 1
```

D.

```
voice class sip-profiles 10
 request INVITE sip-header To copy
  "sip:(.*)@" u01
 request INVITE sip-header SIP-Req-URI modify
  ".*@(.*)" "INVITE sip:\u01@\1"
voice class sip-copylist 1
 sip-header To
dial-peer voice 1 voip
 voice-class sip copy-list 1
dial-peer voice 2 voip
 voice-class sip profiles 10
```

A. Option A

B. Option B

C. Option C

D. Option D

Answer: A

Question #:282 - (Exam Topic 1)

Refer to the exhibit.

A collaboration engineer is automating the cluster installations with answer files. After

initializing the installation, installation wizard cannot recognize the files in the disk. What is the root cause of the issue?

A. The wizard can recognize only one file at a time

B. The file names are incorrectly formatted

C. The clusterinfo_clusterConfig.xml file is corrupted as it is only 1KB in size

D. The Secondarynode_platformConfig.xml is causing a conflict

Answer: B

Explanation

Explanation

Question #:283 - (Exam Topic 1)

Refer to the exhibit.

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

Answer: D

Question #:284 - (Exam Topic 1)

Refer to the exhibit.

```
Gatekeeper#show gatekeeper gw

GATEWAY TYPE PREFIX TABLE
=========================
Prefix: 1*
  Zone GK master gateway list:
    10.1.1.2:49392 HQGK_2
    10.1.1.1:50972 HQGK_1
```

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

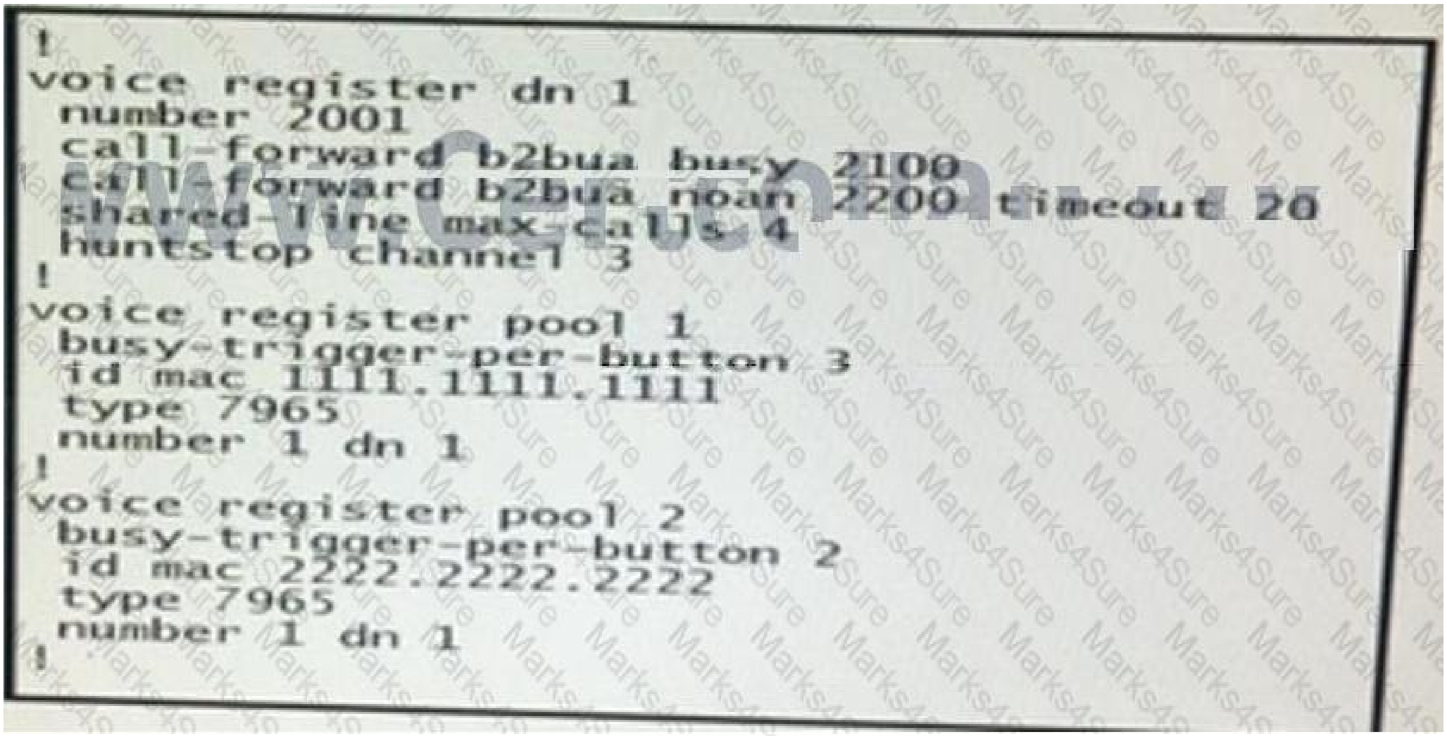
E. gw-type-prefix 1* default-technology

F. Zone remote HQGK_1 cciecollab.com 10.1.1.1

Answer: B C

Question #:285 - (Exam Topic 1)

Refer to the exhibit.



```
!
voice register dn 1
 number 2001
 call-forward b2bua busy 2100
 call-forward b2bua noan 2200 timeout 20
 shared-line max-calls 4
 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 the MAC address 11111.1111.1111, 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.Which option describes what will happen to the fourth incoming call?

A. Both phones will ring, but only IP phone 2 can answer the call.

B. Neither phone will ring and the call will be forwarded to 2100.

C. Both phones will ring and either phone can answer the call.

D. Neither phone will ring and the call will be forwarded to 2200.

E. Both phones will ring, but only IP phone 1 can answer the call.

Answer: B

Question #:286 - (Exam Topic 1)

Refer to the exhibit.

A Cisco collaboration engineer has been asked to remove the ability HQ phone 2 to dial HQ phone

1 by URI dialing. After removing the partition assigned to hqphone1@cisco.com HQ phone 2's CSS, HQ phone 2 is still able to reach HQ phone 1. Why is the HQ phone 1 still reachable using

URI dialing?

A. Directory URI Alias partition has been defined in Enterprise parameters.

B. Phone needs to be reset for changes to take effect.

C. Directory URI partition cannot be deleted therefore still will be reachable.

D. CSS Changes failed to be applied after hitting save due to Database replication issues.

Answer: A

Question #:287 - (Exam Topic 1)

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

Answer: A B C

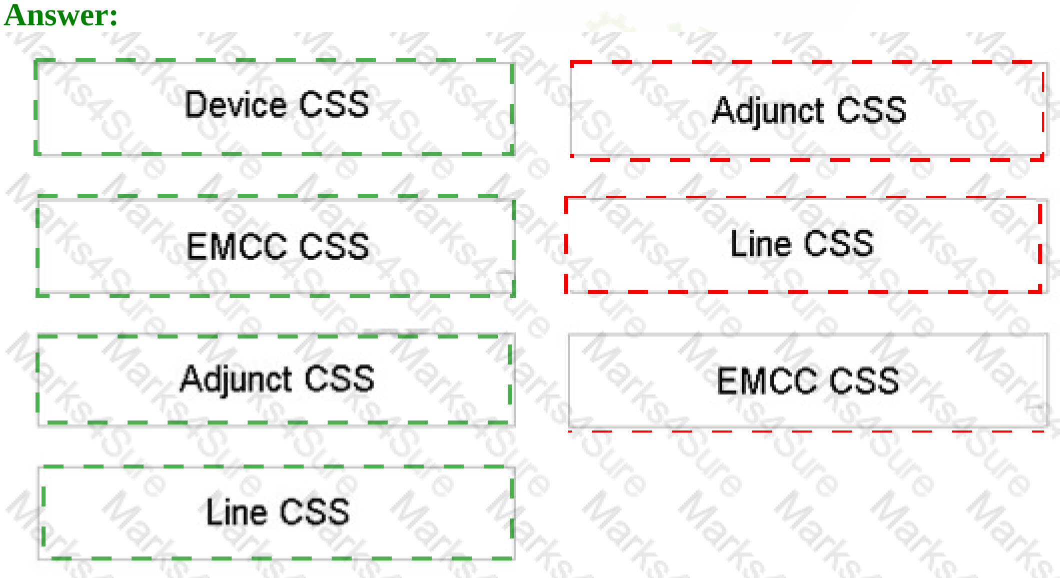
Question #:288 - (Exam Topic 1)

An engineer configuration EmCC needs to understand the priority order in which the home Cluster concatenates calling search space (CSS) when users login to the visiting Phones.

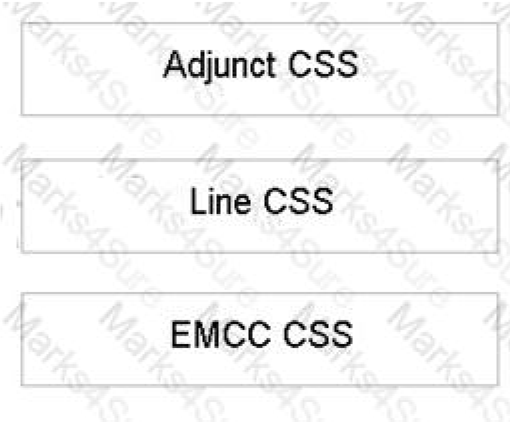
Drag the CSS on the left to the correct priority order on the right. Not all options will be used.

Priority 1 is the highest and priority 3 is the lowest.

Answer:



Explanation



Question #:289 - (Exam Topic 1)

Which two type of patterns can optionally be marked, with urgent priority in Cisco Unified Communication

Manager version 11.0?(choose two)

A. SIP Route Pattern

B. Intercom Directory Number

C. CTI Route Point Directory Number

D. Route Pattern

E. Meet-Me Number

F. Voice Mail Pilot Number

Answer: C D

Question #:290 - (Exam Topic 1)

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.

Answer: A E F

Question #:291 - (Exam Topic 1)

Refer to the Exhibit.

```
sccp local Loopback0
sccp ccm CCM-SUBSCRIBE01 identifier 2 version 7.0
sccp ccm CCM-SUBSCRIBE02 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 bind interface Loopback0
 associate ccm 1 priority 1
 associate ccm 2 priority 2
associate profile 2 register XD-REMOTE

dspfarm profile 2 transcode
 codec g729br8
 codec g729r8
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 422
 associate application SCCP
!
telephony-service
max-ephone 10
max-dn 15
ip source-address 10.1.1.2 port 2000
```

An Engineer is troubleshooting transcoding issue in a remote branch office. After a WAN outage all ip phones can register to a CME in SRST 2900 ISR Router.

However, the users reported that calls disconnect after pressing the answer softkey.

Which three configurations are necessary for successful media resource failover? (Choose three.)

A. SCCP CCM GrouplAssociate CCM 3 priority 1

B. SCCP SSM 10.1.1.2 identifier 3 version 7.0

C. Telephony-serviceSDSpfarm units 1SDSpfarm tag 1 XD-remote

D. Seep ccm 10.1.1.2 identifier 1 version 7.0

E. Associated profile 3 register XD-remote2

F. Seep ccm group 1Associate ccm 3 priority 3

Answer: B C F

Question #:292 - (Exam Topic 1)

Which two QoS guidelines are recommended for provisioning interactive video traffic? (Choose two.)

A. Latency should be no more than 4–5 seconds.

B. Overprovision interactive video queues by 20% to accommodate bursts.

C. Loss should be no more than 5%.

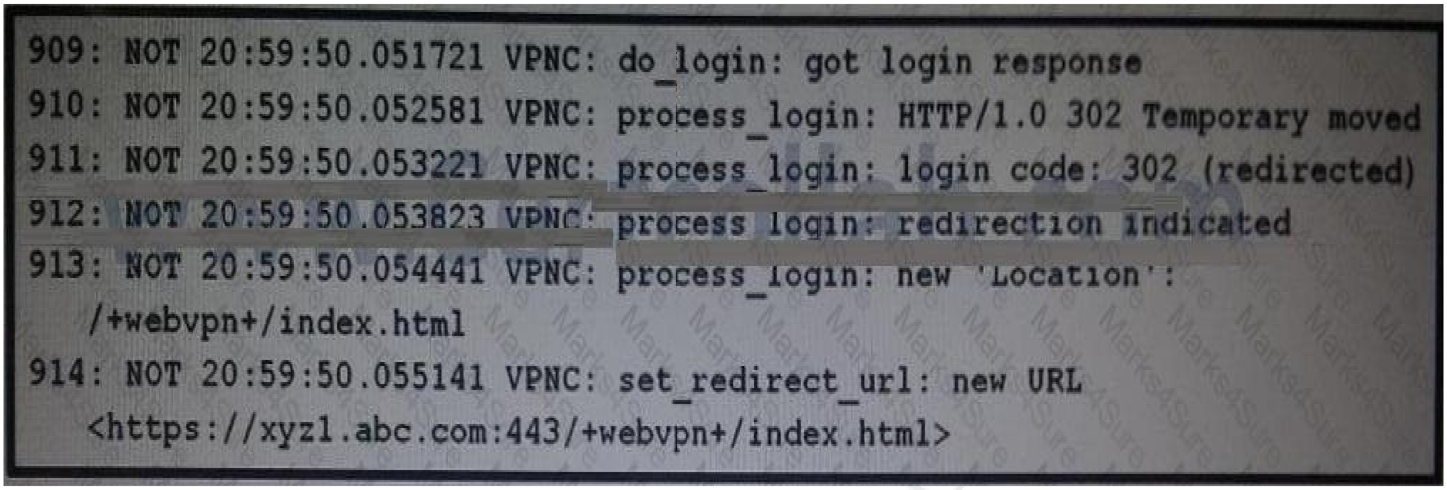
D. Interactive video should be marked to DSCP CS4.

E. Jitter should be no more than 30 ms.

Answer: B E

Question #:293 - (Exam Topic 1)

Refer to exhibit:



```
909: NOT 20:59:50.051721 VPNC: do_login: got login response
910: NOT 20:59:50.052581 VPNC: process_login: HTTP/1.0 302 Temporary moved
911: NOT 20:59:50.053221 VPNC: process_login: login code: 302 (redirected)
912: NOT 20:59:50.053823 VPNC: process_login: redirection indicated
913: NOT 20:59:50.054441 VPNC: process_login: new 'Location':
    /+webvpn+/index.html
914: NOT 20:59:50.055141 VPNC: set_redirect_url: new URL
    <https://xyz1.abc.com:443/+webvpn+/index.html>
```

A Cisco Unified CM engineer configured a phone VPN for remote users but the users cannot register the phones to the VPN which configuration changes fixes this problem?

A. Configure enable outside in the webVPN configuration on the Cisco ASA

B. Configure the split-tunnel-policy tunnel all attribute on the Cisco ASA

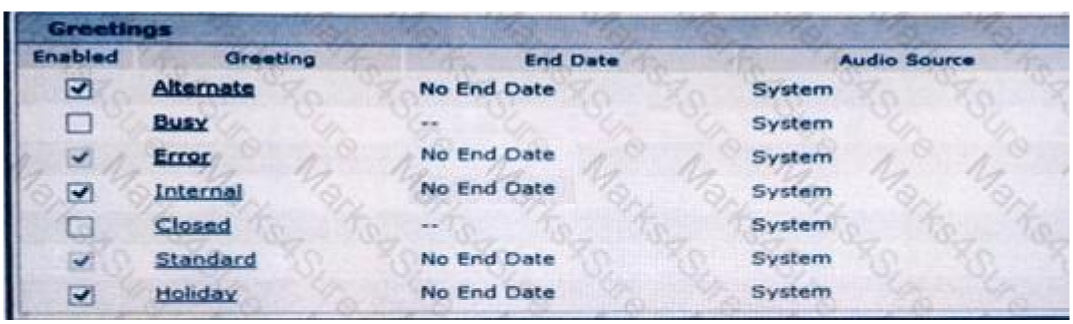
C. Configuration the ssl trust-point SSL outside on the Cisco ASA

D. Remove the Cisco ASA IP address from the VPN load-balancing configuration

Answer: D

Question #:294 - (Exam Topic 1)

Refer to the exhibit.



Greetings

Enabled	Greeting	End Date	Audio Source
☑	Alternate	No End Date	System
☐	Busy	--	System
☑	Error	No End Date	System
☑	Internal	No End Date	System
☐	Closed	--	System
☑	Standard	No End Date	System
☑	Holiday	No End Date	System

A voicemail administrator was asked to create a call handler for the sales department with the following requirements:

After creating the call handler and making some test calls only the default system greeting is heard.

Which four configuration changes are needed to company with this business request? (Choose four.)

A. disable the Alternate Greeting under Call Handler Greetings

B. create a new closed schedule and assign it to the sales Call Handler

C. record a new Greeting and assign it to the Alternate Greeting

D. record a new Greeting and assign it to the Holiday Greeting

E. record a new Greeting and assign it to the Internal Greeting

F. create a new holiday schedule to be used by the Holiday Greeting

G. disable the Internal Greeting under Call Handler Greeting

H. enable the Closed Greeting under Call Handler Greetings

Answer: C D E F

Question #:295 - (Exam Topic 1)

When neither the active or standby Location Bandwidth Manager in the configured LBM group is available, what will the Cisco Call Manager 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.

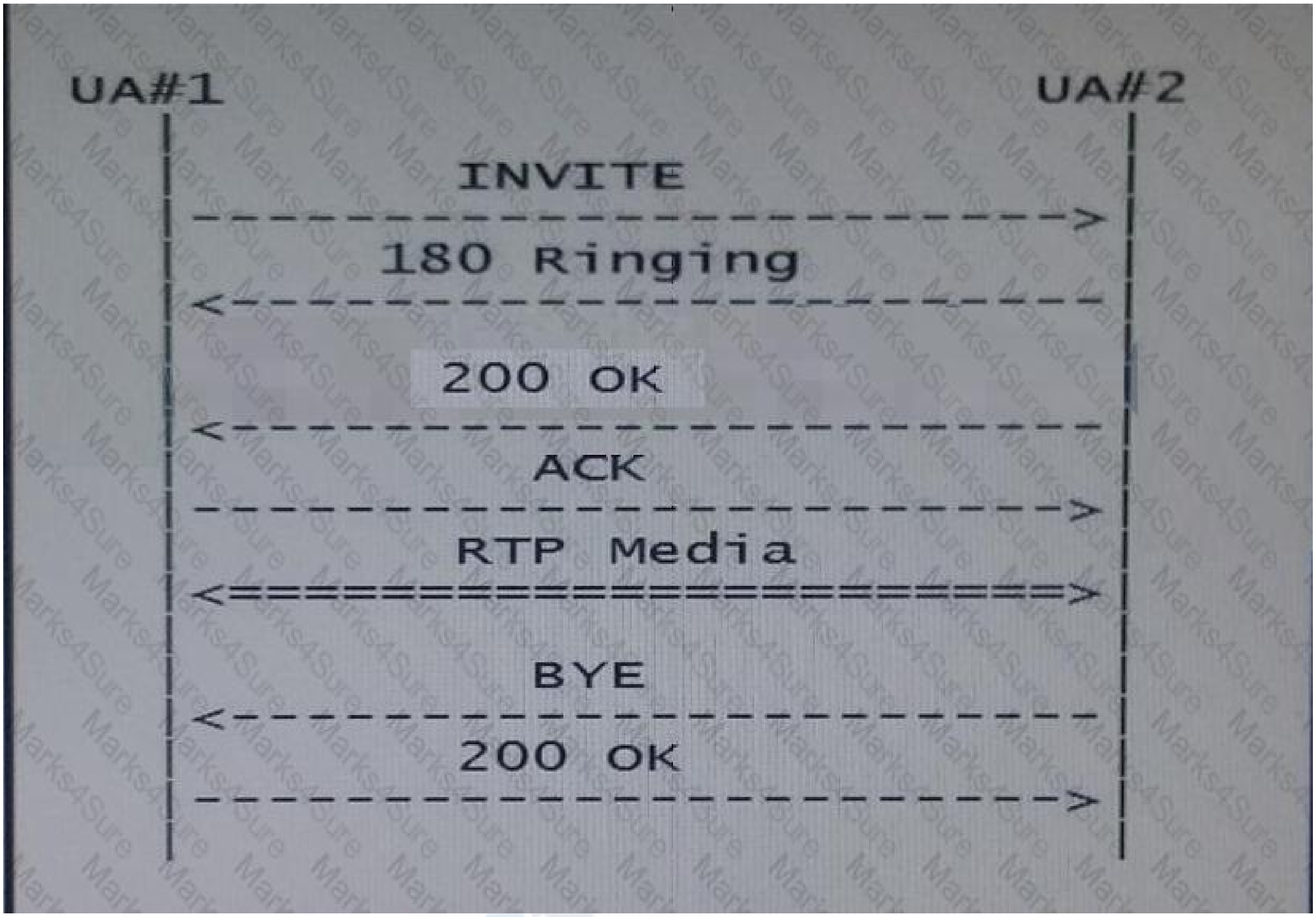
D. It will attempt to communicate with the local LBM service for location CAC decisions.

E. It will allow all calls until communication is re-established with any configured servers in the LBM group.

Answer: D

Question #:296 - (Exam Topic 1)

Refer to Exhibit:



How many SIP signalling dialog(s) took place in this SIP message exchange between two SIP user agents?

A. 1

B. 2

C. 3

D. 4

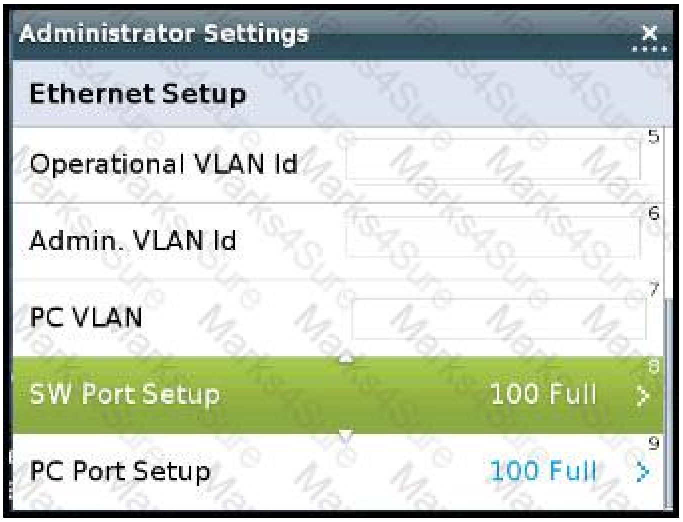
E. 5

F. 6

Answer: A

Question #:297 - (Exam Topic 1)

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

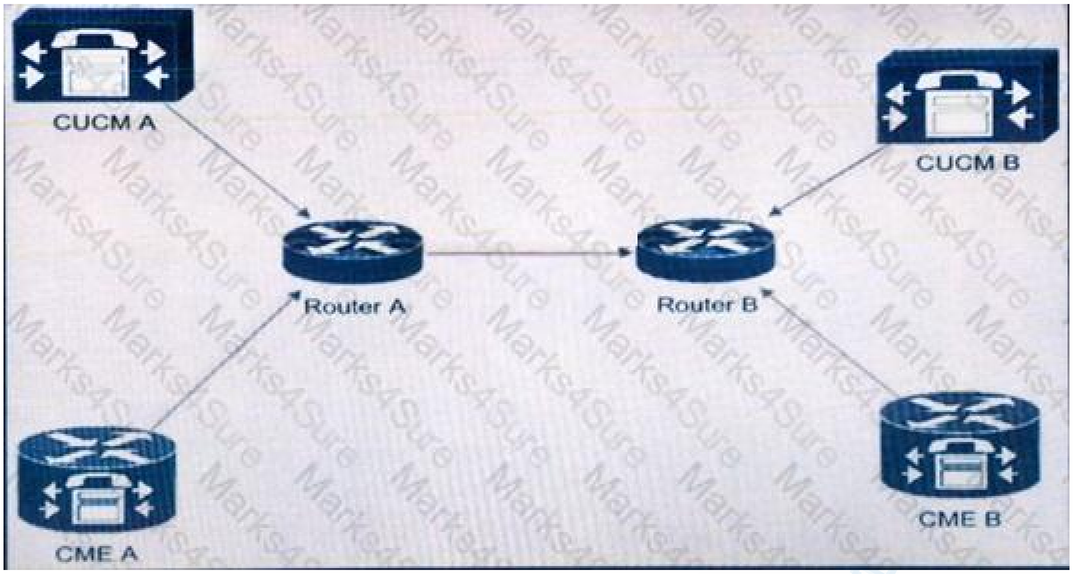
D. SW Port Setup

E. PC Port Setup

Answer: B D E

Question #:298 - (Exam Topic 1)

Refer to the exhibit.



An engineer is configuring dynamic Call routing and DN learning between two Cisco Unified CM and Two Cisco Unified CME systems. Which two configuration steps are required for all this feature to work? (Choose two)

A. Configure routers A and to use a different autonomous system number for DN routing

B. Configure routers A and to use EIGRP for IP Routing

C. Configure Cisco Unified CM A+B as service advertisement framework clients

D. Configure router A and to use OSPF for IP Routing

E. Configure Cisco Unified CME A+B as service advertisements forwarders

F. Configure routers A and to use the same autonomous system number for DN Routing

Answer: C F

Question #:299 - (Exam Topic 1)

The Cisco Unified Border Element is configured using high availability with the Hot Standby Routing Protocol.

```
CUBE_PSTN#show voice high-availability summary

======== Voice HA DB INFO ========
Number of calls in HA DB: 100
Number of calls in HA sync pending DB: 50
Number of calls in HA preserved session DB: 70
```

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 non-native calls is 70.

C. The number of native calls is 501 .

D. The number of calls preserved is 220.

E. The total number of active calls is 100.

Answer: A B

Explanation

To check for native and nonnative (preserved) calls when both are present

The numbers of calls on the system are shown as follows:

Total number of calls = "Number of calls in HA DB" + "Number of calls in HA sync pending DB". This is 100 + 50 = 150 in the example output below.

Total number of preserved (nonnative) calls = "Number of calls in HA preserved session DB". This is 70 in the example output below.

Total number of native calls (calls set up since the failover and therefore not preserved over the failover) is the difference in the previous two numbers. In this example, it is 150 - 70 = 80.

XFR-2#show voice high-availability summary

======== Voice HA DB INFO ========

Number of calls in HA DB: 100

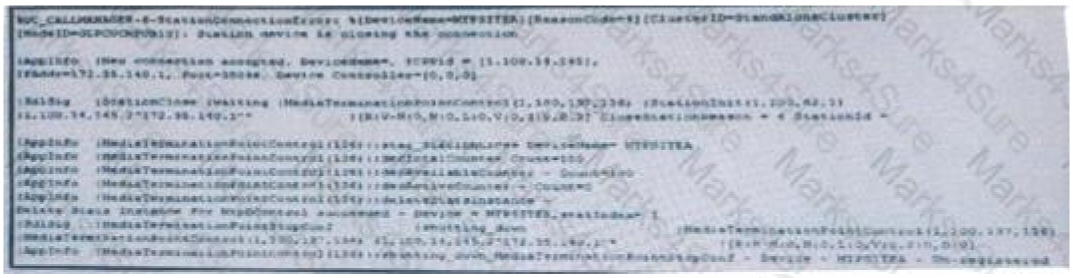
Number of calls in HA sync pending DB: 50

Number of calls in HA preserved session DB: 70

http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-border-element/112095-cube

Question #:300 - (Exam Topic 1)

Refer to the exhibit.



A isco collaboration engineer discovers that an instance of IOS image termination point (MTP) could not maintain stable registration with CUCM.

Callmanager traces are shown in the exhibit. What is the reason for the flapping registration?

A. The CCM version on IOS configuration does not match the CUCM version.

B. The IOS MTP is experiencing high CPU and is missing its keep-alives.

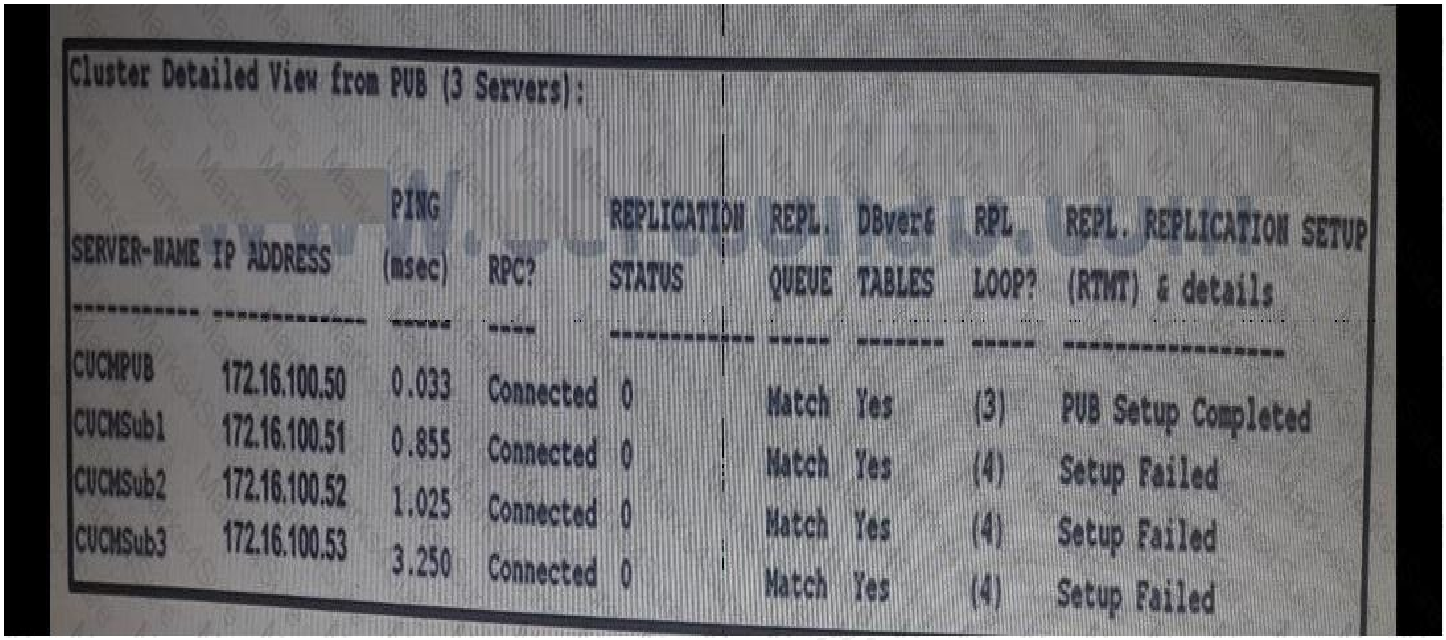
C. A Firewall is blocking port 2000 intermittently between IOS Device and CUCM.

D. Another IOS Media device is attempting to register with the same name.

Answer: D

Question #:301 - (Exam Topic 1)

Refer to the exhibit.



Cluster Detailed View from PUB (3 Servers):

SERVER-NAME	IP ADDRESS	PING (msec)	RPC?	REPLICATION STATUS	REPL. QUEUE	DBver& TABLES	RPL LOOP?	REPL. REPLICATION SETUP (RTMT) & details
CUCMPUB	172.16.100.50	0.033	Connected	0	Match	Yes	(3)	PUB Setup Completed
CUCMSub1	172.16.100.51	0.855	Connected	0	Match	Yes	(4)	Setup Failed
CUCMSub2	172.16.100.52	1.025	Connected	0	Match	Yes	(4)	Setup Failed
CUCMSub3	172.16.100.53	3.250	Connected	0	Match	Yes	(4)	Setup Failed

Users on a four-node CUCM cluster are reporting call problems when attempting to call out to internal extension and PSTN. An engineer troubleshooting issue found a replication of the cluster is in status 4. Which three steps will resolve the replication problem? (Choose Three)

A. run the command utils dbreplication dropadmindb on all subscribers

B. run the command utils dbreplication repairable all from the publisher

C. run the command utils dbreplication stop on the publisher

D. run the command utils dbreplication reset all from the publisher

E. run the command utils dbreplication repair all from the publisher

F. run the command utils dbreplication stop on all subscribers

Answer: C D F

Question #:302 - (Exam Topic 1)

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

Answer: C

Question #:303 - (Exam Topic 1)

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

Answer: A D

Question #:304 - (Exam Topic 1)

Refer to the exhibit.

Verify external database server reachability (pingable)		
Verify external database server connectivity (database connection check)		The following Cisco Unified IM and Presence Service node to external database server connections failed: ◦ 172.16.100.52 >>test (Persistent Chat)

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.

Answer: A E

Question #:305 - (Exam Topic 1)

Refer to the exhibit.

```
38587249.000 |18:49:37.585 |AppInfo  |MGCPHandler received msg from: 10.163.25.1
NTFY 416258700 aaln/S0/SU0/0@cisco.com MGCP 0.1
N: ca@10.163.34.5:2427
X: 0
O: H/ci("","",0)
```

Which three options are potential reasons the FXO port is not receiving Caller ID? (Choose three.)

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.

Answer: A C E

Question #:306 - (Exam Topic 1)

An engineer is troubleshooting Cisco Jabber Federation where the server cannot discover the external Lync domains using SRV records. Which two steps resolve this problem? (Choose two)?

A. Configure the port number as 5065.

B. Configure the destination pattern as "lyncdomainname.com".

C. Set the Next Hop value as the IP address of the Microsoft Lync Server.

D. Configure the destination pattern as ".com.lyncdomainname*"

E. Configure the port number as 5060.

F. Set the protocol to TLS.

G. Set the Next Hop value as the IP address of the Lync Domain servers.

H. Set the protocol to TCP.

Answer: F G

Question #:307 - (Exam Topic 1)

Which of the below characteristics of RPL is true?

A. RPL can send only messages in secure mode

B. RPL uses hello message to send routing updates to its neighbors

C. RPL is an IPv6 link-state routing protocol

D. RPL is designed for lossy networks

Answer: D

Question #:308 - (Exam Topic 1)

Which definition is included in a Cisco UC on UCS TRC?

A. Required RAID configuration, when the TRC uses external storage.

B. Configuration settings and patch recommendations for VMware software.

C. Configuration of virtual - to - physical network interface mapping.

D. Design and installation of blade server chassis and switching.

E. Server model and local components (CPU, RAM, Adapters, Local Storage) at the orderable part number level.

Answer: E

Question #:309 - (Exam Topic 1)

A Call is made between two desk phones enabled with a single number each that are registered to a Cisco Unified CM cluster. The device pool for each device has a local route group defined.

When the call is placed to exit the system, which device pool controls the destination gateway?

A. destination RDP

B. Source Phone

C. Source RDP

D. Destination phone

Answer: B

Question #:310 - (Exam Topic 1)

You want your SDN controller to build a topology map of your network. Which protocol can be used to export IGP information from the network to the SDN controller?

A. VXLAN

B. JSON

C. BGP- LS

D. OpenFlow

E. PCEP

Answer: C

Question #:311 - (Exam Topic 1)

Refer to the exhibit.

assume that the bottom logical partition policy entry in Cisco Unified Communication

Manager was provisioned last. How is the call treated when an IP phxx GeoLocation places a call to a MGCP gateway with FXO ports in the San Jose GeoLocation?

A. The call is allowed because only the top policy entry matches the call flow

B. The call is allowed because the first listed logical partition policy takes precedence when multiple matches exist

C. The call is denied because the call flow matches neither policy entries

D. The call is denied because the last added logical partition policy takes precedence when multiple matches exist

E. The call is allowed because the call flow matches neither policy entries

Answer: D

Question #:312 - (Exam Topic 1)

Which option describes how NVF, OpenStack and KVM relate to each other? (Choose two.)

A. OpenStack and KVM can be used to provide NVF

B. OpenStack and KVM are not related to NVF

C. NVF and KVM are based on OpenStack

D. OpenStack and NVF enable KVM.

Answer: B

Question #:313 - (Exam Topic 1)

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.

Answer: B E

Question #:314 - (Exam Topic 1)

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

Answer: A

Question #:315 - (Exam Topic 1)

Refer to the exhibit.

A customer has two Cisco Unified Communication manager 9.X clusters that serve the same location. An engineer has attempted to set up Enhanced Location call admission control so that any call within a site between phones on the two clusters do not decrement the available bandwidth to and from that site. However, the real time monitoring tool currently shows bandwidth being used from the site to Hub_none. When a call is placed between phones at the site, which action must be taken to correct this situation?

A. The link between clusters must be a type of inter-cluster trunk instead of a sip trunk.

B. The hub_none location must have a link configuration to the phantom location.

C. The device pool names must match between clusters.

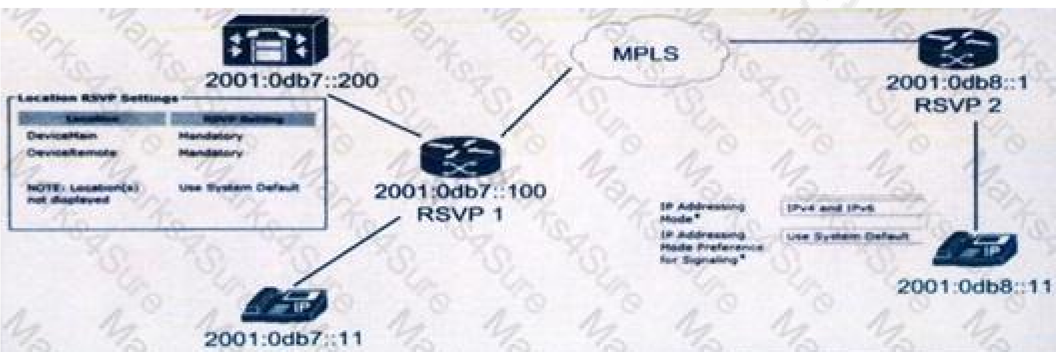
D. The Hub_none location must have a link configured to the shadow location.

E. The SIP trunks should be changed to use the shadow location.

Answer: E

Question #:316 - (Exam Topic 1)

Refer to the exhibit.



A collaboration engineer is troubleshooting a cluster that has been configured to use RSVP. The calls are being rejected and the caller receives a busy tone. What is the root cause of this problem?

A. The RSVP Agents are only using an Ipv6 address.

B. IP Addressing mode preference for signaling is set to use System Default.

C. RSVP relationship between Main and Remote is set to Mandatory

D. IP Addressing Mode is set to use Ipv4 and Ipv6

Answer: A

Question #:317 - (Exam Topic 1)

A Cisco collaboration engineer is troubleshooting unexpected SIP call disconnect. Which three responses corresponding to the 5xx range (Choose three.)

A. Forbidden

B. Unauthorized

C. Request timeout

D. Service unavailable

E. Bad gateway

F. Internal server error

Answer: D E F

Question #:318 - (Exam Topic 1)

Refer to the Exhibit.

```
sccp local Loopback0
sccp ccm CCM-SUBSCRIBER01 identifier 2 version 7.0
sccp ccm CCM-SUBSCRIBER02 identifier 1 version 7.0
sccp
!
sccp ccm group 1
 bind interface Loopback0
 associate ccm 1 priority 1
 associate ccm 2 priority 2
associate profile 2 register XD-REMOTE

dspfarm profile 2 transcode
 codec g729br8
 codec g729r8
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 422
 associate application SCCP
!
telephony-service
max-ephone 10
max-dn 15
ip source-address 10.1.1.2 port 2000
```

An Engineer is troubleshooting transcoding issue in a remote branch office after a WAN outage. All IP phones can register to a CME in SRST 2900 ISR Router however, the users reported that calls disconnect after pressing the answer soft key. Which three configuration are necessary for successful media resource failover? (Choose three)

A. SCCP ccm group 1

   Associate ccm 3 priority 1

B. SCCP ccm 10.1.1.2 identifier 3 version 7.0

C. Telephony-service

   Sdspfarm units 1

   Sdspfarm tag 1 XD-REMOTE

D. Sccp ccm 10.1.1.2 identifier 1 version 7.0

E. Associate profile 3 register XD-REMOTE2

F. Sccp ccm group 1

   Associate ccm 3 priority 3

Answer: B C F

Question #:319 - (Exam Topic 1)

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.

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.

Answer: D

Question #:320 - (Exam Topic 1)

A collaboration engineer is troubleshooting an MOH problem on a Cisco IOS SIP gateway. While searching through a debug ccsip message output, which three parameters in the SIP messages can be used to determine if the call was placed on hold? (Choose three)

A. OPTIONS WITH 301 CALLHOLD

B. INVITE WITH a=INACTIVE

C. INVITE WITH a=SENDONLY

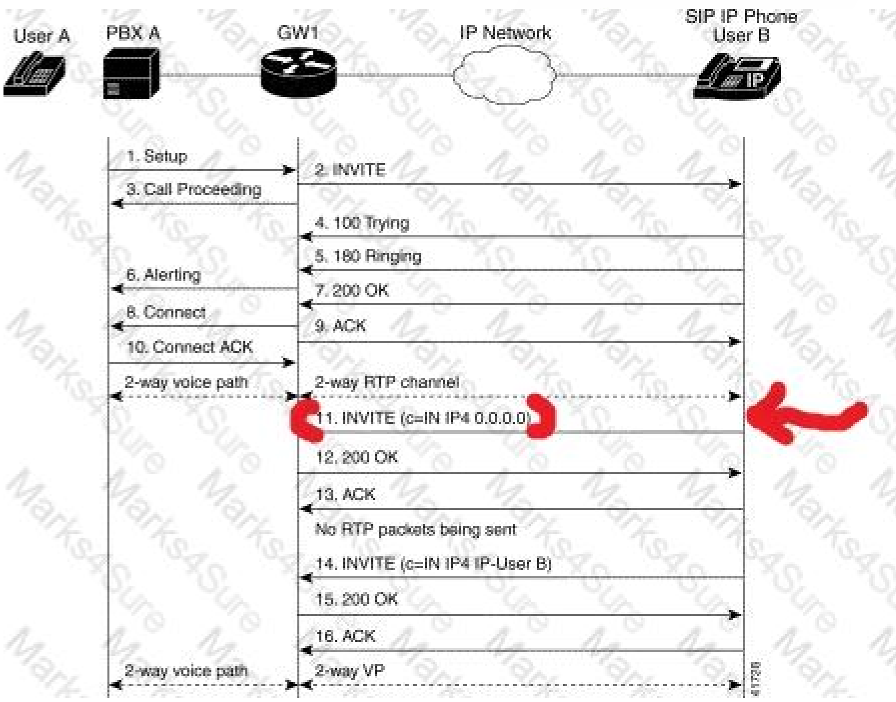
D. OPTION WITH c=INACTIVE

E. c=IN IP4 0.0.0.0

F. BYE WITH A = CALLHOLD

Answer: B C E

Explanation



When a call is on hold you get something like this:

v=0

o=CiscoSystemsCCM-SIP 919861 2 IN IP4 10.10.30.14

s=SIP Call

c=IN IP4 0.0.0.0

t=0 0

m=audio 30120 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=ptime:20

a=inactive

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

The connection parameter shows 0.0.0.0, When the call is taking off hold you, the connection parameter should indicate the ip address where media is sent to. So it will have a real value. (This is usual sent in the ACK.) cucm still sends a DO in the re-INVITE and the far end sends a 200 Ok with SDP. CUCM then sends ACK with SDP

This is how the whole call hold/transfer/call resume works

1. CUCM sends invite (re-invite) with an inactive SDP (a=inactive) to indicate a break in media path

2. CUCM sends a Delayed offer to insert MOH or to resume Media stream

NB: CUCM expects a sendrecv offer with SDP to the DO. (NB:if cucm gets an inactive offer SDP in the 200 OK instead of providing a send-recv offer SDP, the media path remains in an inactive state and causes calls to dropcall will drop),CUCM sends an ACK with sendonly to the 200 OK

3.CUCM establishes newcall leg with the intended transfered destination..Once this call is connected

4.CUCM receives a new Transfer instruction from the transferring phone to connect the held party

5. Next CUCM sends a re-invite with an inactive SDP to indicate a break in media path to MOH (in attempt to complete transfer)

6.Next CUCM sends an inactive SDP to indicate a break in media path between transfering party & transfered party to remove Xferring party from call

7. Next CUCM sends a DO re-invite to connect the transfered party. The far end then sends 200 OK with the required SDP to connect the call

Question #:321 - (Exam Topic 1)

A Cisco Unified Contact Centre Express Manager wants to add database integration to self - service interactive voice response application. Which four types of licensing and database servers support this requirement? (Choose four)

A. Oracle Database Service

B. UCCX Standard License

C. UCCX Enhanced License

D. Microsoft SQL Server

E. UCCX Premium License

F. SAP SQL Server

G. Sybase Adaptive

H. Postgress SQL Server

Answer: A D E G

Question #:322 - (Exam Topic 1)

Which meaning of the absence of the Allow header in a SIP message is true?

A. The sender does not support any SIP method

B. It is coming from an intermediate SIP entry

C. The sender supports all standard based SIP methods

D. The sender is not providing any information on what method it supports

Answer: D

Question #:323 - (Exam Topic 1)

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.

Answer: C E

Question #:324 - (Exam Topic 1)

Which two 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

Answer: A B C

Question #:325 - (Exam Topic 1)

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

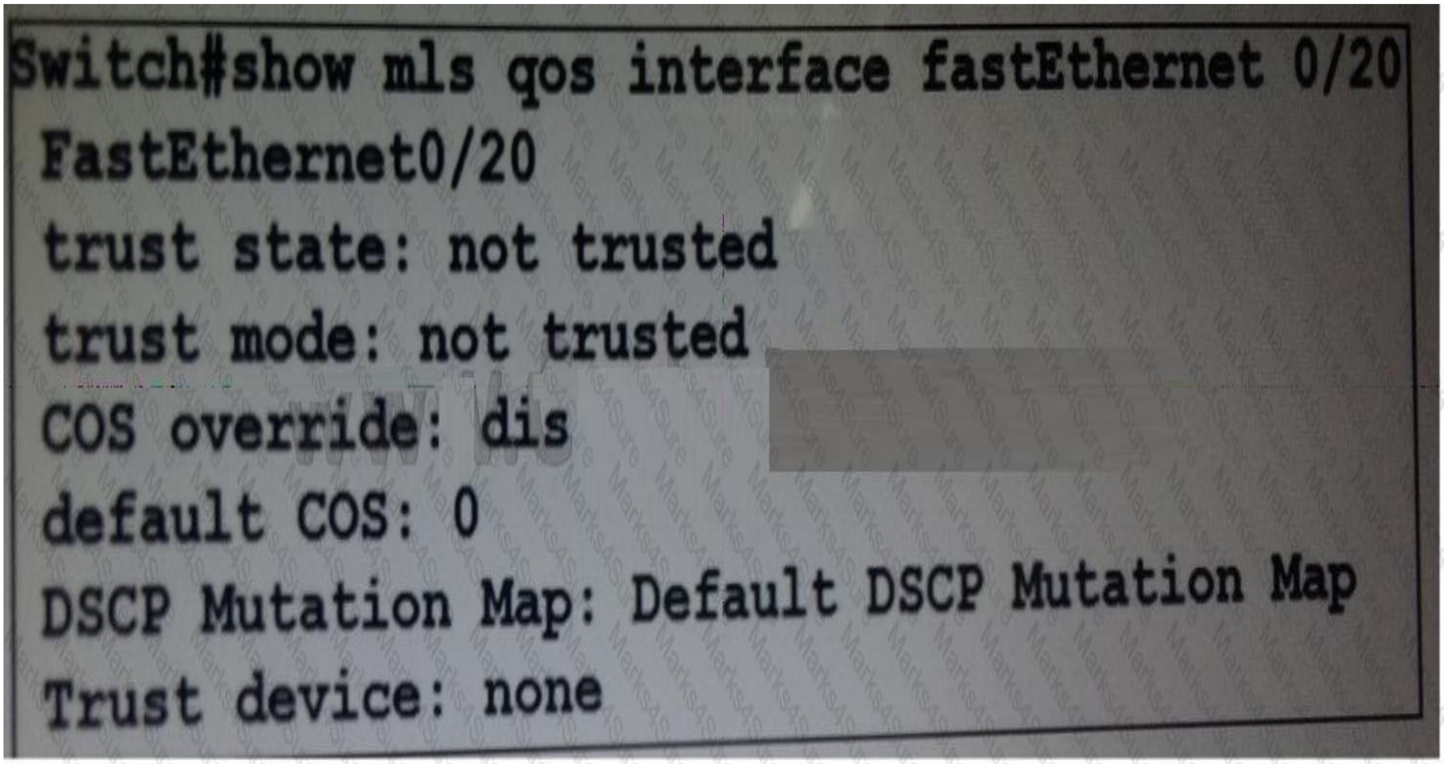
D. CfwdAll

E. There is no mandatory softkey in the Off Hook call state.

Answer: E

Question #:326 - (Exam Topic 1)

Refer to the exhibit.



A VoIP engineer must configure QoS for the switch ports. Which two set of commands configure switch port to remap every incoming packet aside from the default QoS? (Choose two).

A. FastEthernet 0/20

No mls qos trust device cisco-phone

B. FastEthernet 0/20

No mls qos trust cos

C. FastEthernet 0/20

mls qos cos 3

D. FastEthernet 0/20

Mls qos cos override

E. FastEthernet 0/20

No switchport priority extend trust

F. FastEthernet 0/20

Switchport priority extend cos

Answer: C D

Question #:327 - (Exam Topic 1)

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. authorization

E. permanently deleted awaiting

Answer: B

Question #:328 - (Exam Topic 1)

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 inbound calls can be handled simultaneously between ephone 1 and ephone 2 before a user busy tone is returned?

A. 6

B. 7

C. 8

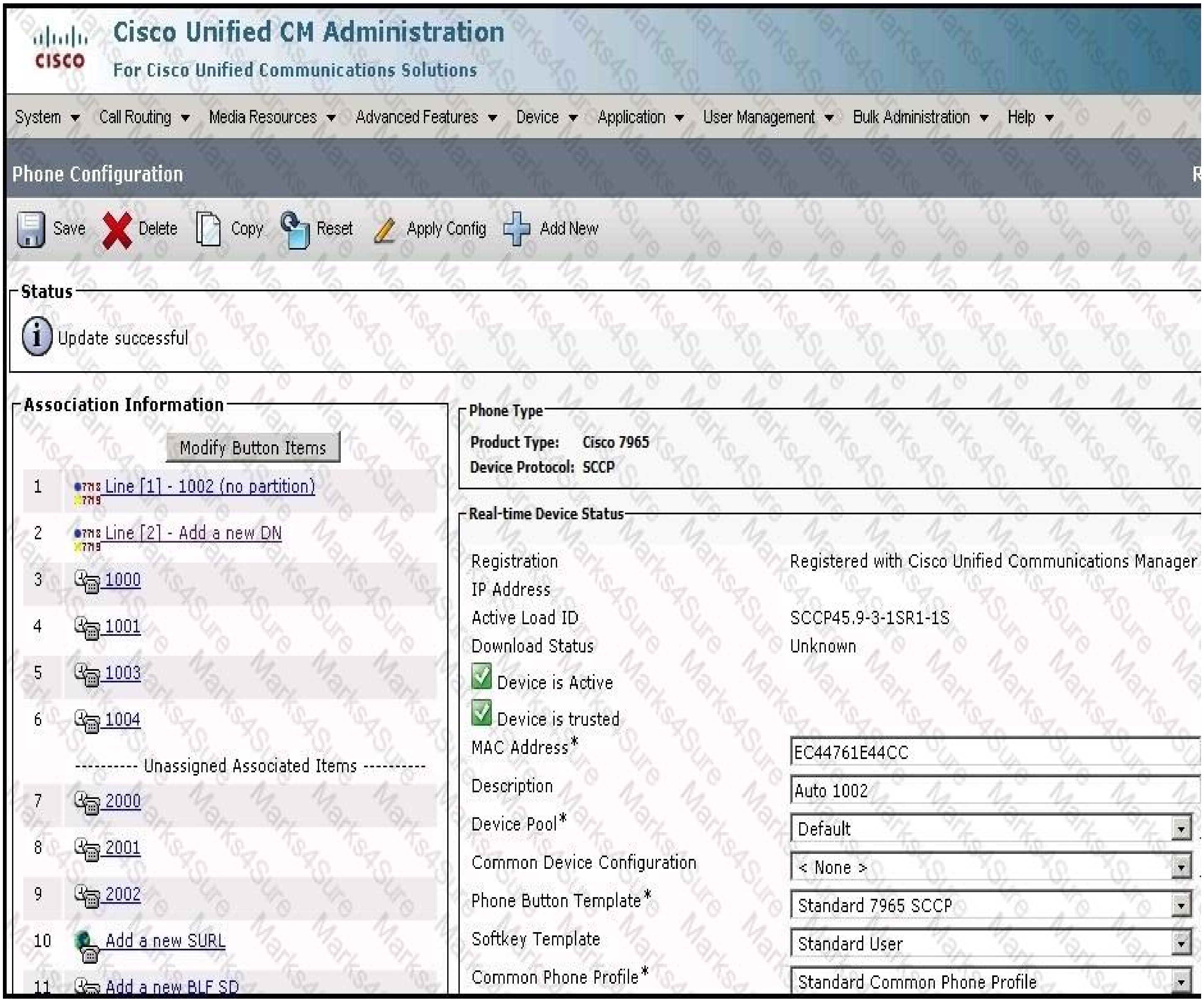
D. 9

E. 11

Answer: A

Question #:329 - (Exam Topic 1)

Refer to Exhibit:



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 soft key.

D. Press 6 on the phone keypad, followed by the Dial soft key.

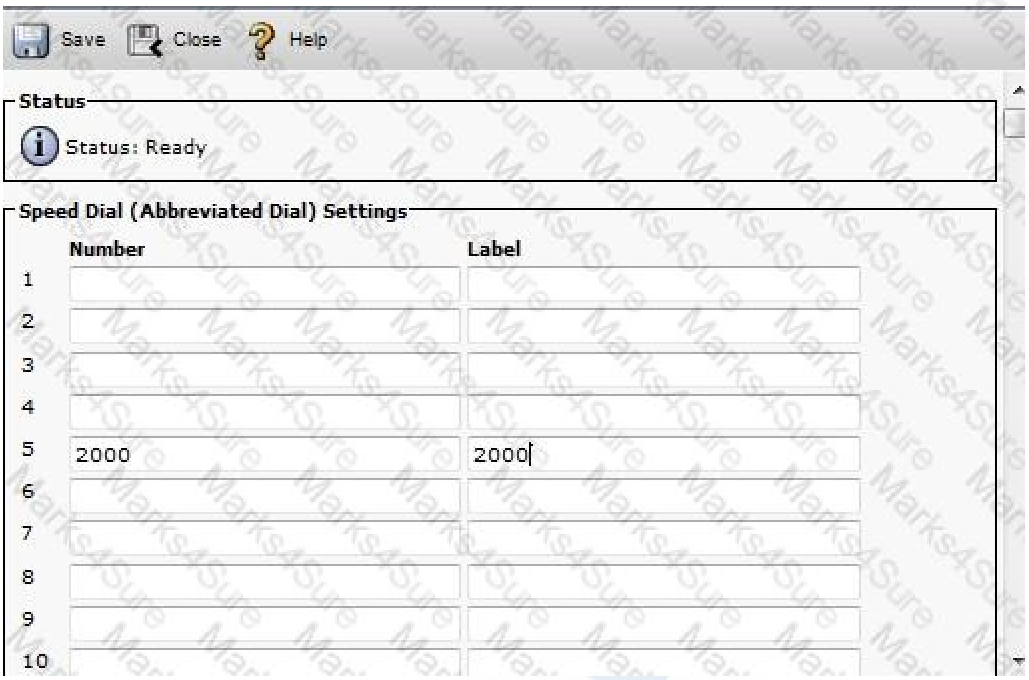
E. Press 5 on the phone keypad, followed by the AbbrDial soft key.

Answer: E

Explanation

Configure these settings for the speed-dial numbers that you access with abbreviated dialing. When the user configures up to 99 speed-dial entries, part of the speed-dial entries can get assigned to the speed-dial buttons on the IP phone; the remaining speed-dial entries get used for abbreviated dialing. When a user starts dialing digits, the AbbrDial softkey displays on the phone, and the user can access any speed-dial entry by entering the appropriate index (code) for abbreviated dialing.

Configuration for Dialing 5 on the phone keypad, followed by the AbbrDial softkey.

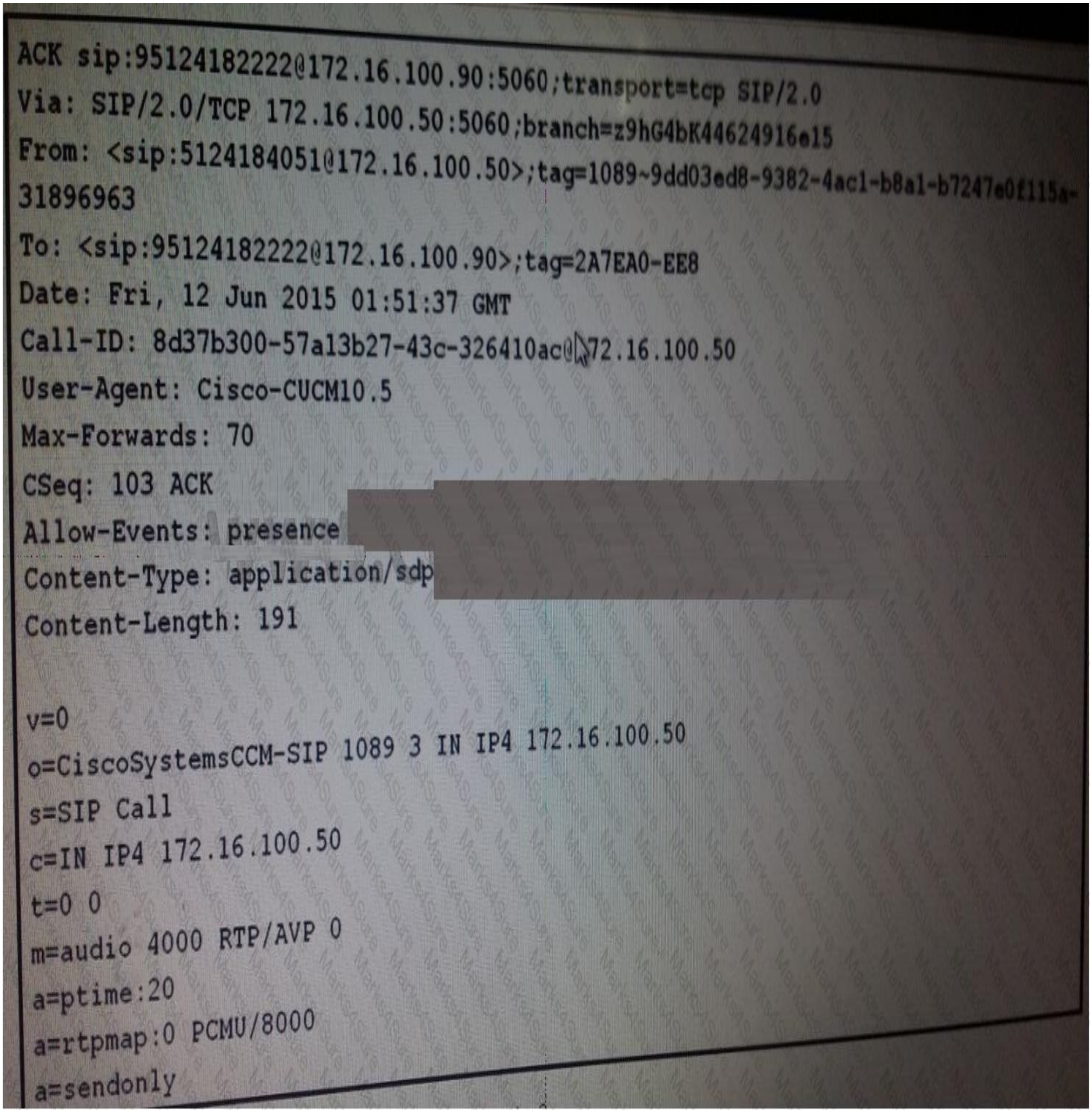


and it showing on the phone page as

Implemented and tested

Question #:330 - (Exam Topic 1)

Refer to the exhibit.



```
ACK sip:95124182222@172.16.100.90:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 172.16.100.50:5060;branch=z9hG4bK44624916e15
From: <sip:5124184051@172.16.100.50>;tag=1089~9dd03ed8-9382-4ac1-b8a1-b7247e0f115a-31896963
To: <sip:95124182222@172.16.100.90>;tag=2A7EA0-EE8
Date: Fri, 12 Jun 2015 01:51:37 GMT
Call-ID: 8d37b300-57a13b27-43c-326410ac@172.16.100.50
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 103 ACK
Allow-Events: presence [redacted]
Content-Type: application/sdp
Content-Length: 191

v=0
o=CiscoSystemsCCM-SIP 1089 3 IN IP4 172.16.100.50
s=SIP Call
c=IN IP4 172.16.100.50
t=0 0
m=audio 4000 RTP/AVP 0
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendonly
```

Which call status is this SIP message indicating?

A. Caller is in conference using MUTE

B. Caller is being call forwarded

C. Caller is on hold

D. Caller is transferred to voicemail

E. Caller is hearing ring back

Answer: C

Question #:331 - (Exam Topic 1)

Which two steps are valid to use when deploying a new identity process to Cisco Unified Communications Manger subscribers? (Choose two)

A. Mount the floppy disk image with the configuration file from the AFG

B. Import the subscriber virtual, machines with the appropriate OVA template

C. Run the command utils set-import on the template VM.

D. Perform a skip install

E. Mount the Cisco Unified Communications Manager ISO file.

Answer: B E

Question #:332 - (Exam Topic 1)

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 Postgress 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.

Answer: B C D G

Question #:333 - (Exam Topic 1)

Refer to the exhibit.

IP phone A makes a call to IP phone B and RTP voice stream is successfully established

between the endpoints. What option describes how to accomplish this call on phones with incompatible IP addressing versions?

A. phone A makes a call to IP phone B and RTP IPv4 and IPv6

B. Set IP addressing mode preference for signalling to IPv4

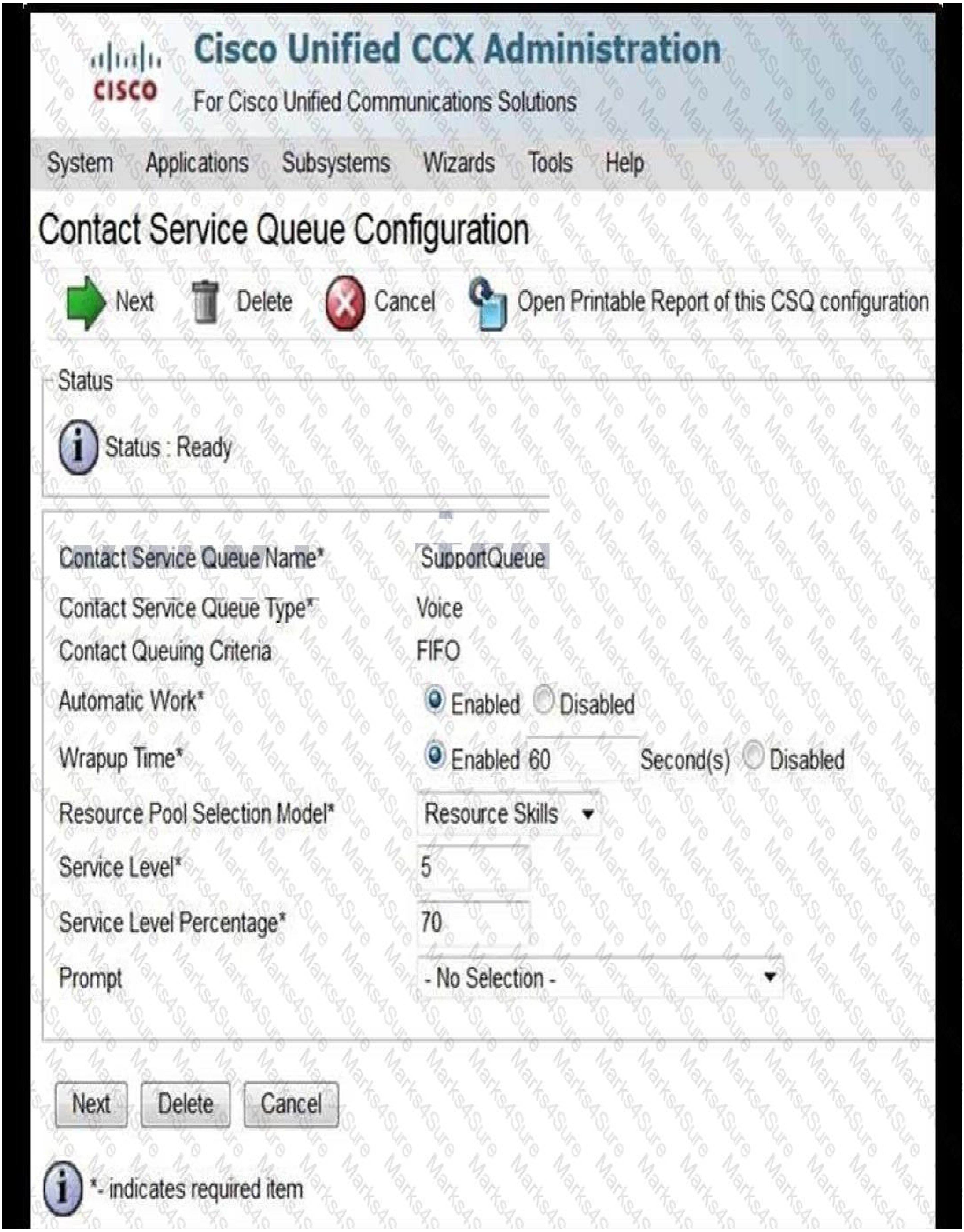
C. Assign a media resource group with available MTPs

D. Enable alternative network address types

Answer: C

Question #:334 - (Exam Topic 1)

Refer to the exhibit.


cisco
Cisco Unified CCX Administration
For Cisco Unified Communications Solutions
System Applications Subsystems Wizards Tools Help
Contact Service Queue Configuration
Next Delete Cancel Open Printable Report of this CSQ configuration
Status
Status : Ready
Contact Service Queue Name* SupportQueue
Contact Service Queue Type* Voice
Contact Queuing Criteria FIFO
Automatic Work* Enabled Disabled
Wrapup Time* Enabled 60 Second(s) Disabled
Resource Pool Selection Model* Resource Skills
Service Level* 5
Service Level Percentage* 70
Prompt - No Selection -
Next Delete Cancel
*- indicates required item

A Cisco collaboration engineer is writing a report to summarize the call distribution characteristics in a Cisco Unified Contact Centre 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).

Answer: C E F

Question #:335 - (Exam Topic 1)

Refer to the exhibit.

```
Jan 10 05:55:35.130: MGCP Packet sent to 10.1.1.2:2427--->
NTFY 217738192 *@MGCP-gateway.cisco.com MGCP 0.1
X: 0
O:
<---

Jan 10 05:55:35.130: MGCP Packet received from 10.1.1.2:2427 --->
200 217738192
<---
```

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 it will 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

Answer: C

Question #:336 - (Exam Topic 1)

---Unique Boundary

ContentType: application/x-931

ContentDisposition: signal, handling-option

Content Length: 46

0x0x0xEUROx70x0x70x0x0xEUROx70x70x0x0x70x70x71

---Unique Boundary

ContentType: application/qtd

ContentDisposition: signal, handling-option

Refer to the exhibit. Which set of Cisco IOS commands removes the following information from SIP invite SDP?

A. Gateway(config)# no signaling forward unconditional

B. Gateway(config)# signaling forward unconditional

C. Gateway(config)# voice service voip

Gateway(config)# sip

Gateway(config)# signaling forward unconditional

D. Gateway(config)# voice service voip

Gateway(config)# signaling forward unconditional

E. Gateway(config)# voice service voip

Gateway(config)# sip

Gateway(config)# no signaling forward unconditional

F. Gateway(config)# voice service voip

ContentType: application/x-931

Answer: F

Question #:337 - (Exam Topic 1)

Refer to the exhibit.

```
voice service voip
  no ip address trusted authenticate
  allow-connections sip to sip
  sip
  no update-callerid
 !
 voice register global
  mode cme
  max-dn 10
  max-pool 10
 !
 voice register dn 1
  number 5000 name Phone A
 !
 voice register dn 2
  number 5001 name Phone B
 !
 voice register dn 3
  number 5002 name Phone C
 !
 voice ragister pool 1
  id mac 0000.1111.112A
  type 8841 number 1 dn 1
 !
 voice register pool 1
  id mac 0000.1111.112B
  type 8841 number 1 dn 2
 !
 voice register pool 1
  id mac 0000.1111.112C
  type 8841 number 1 dn 3
 !
```

An engineer is trying to provision CUCME with three 8841 phones. However all phone fail to register. Which two changes in the configuration would allow the phones to register? (Choose two)

A. The registrar server command must be added under the voice register global configuration

B. The IP address trusted authenticate command must be added under voice service voip

C. The source-address command must be added under the voice register global configuration

D. The local SIP proxy address must be configuration under the sip-ua configuration

E. The registrar server command must be added under the sip section of voice service voip

Answer: C E

Question #:338 - (Exam Topic 1)

Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 huntstop channel 10
!
voice register pool 1
 id mac 1111.1111.1111
 type 7965
 number 1 dn 1
!
```

How many calls, inbound and outbound combined, are supported on the IP phone?

A. 1

B. 2

C. 8

D. 12

E. 50

Answer: E

Question #:339 - (Exam Topic 1)

Refer to the exhibit.

------Permission Information---

Groups: Standard CTI Allow Call Monitoring

Standard CTI Allow Control of Phones supporting C

Standard CTI Enabled

Roles: Standard CTI Allow Call Monitoring

Standard CTI Allow Control of Phones supporting C

Standard CTI Enabled

A VoIP engineer has enabled mixed mode on the Cisco Unified Communication Manager

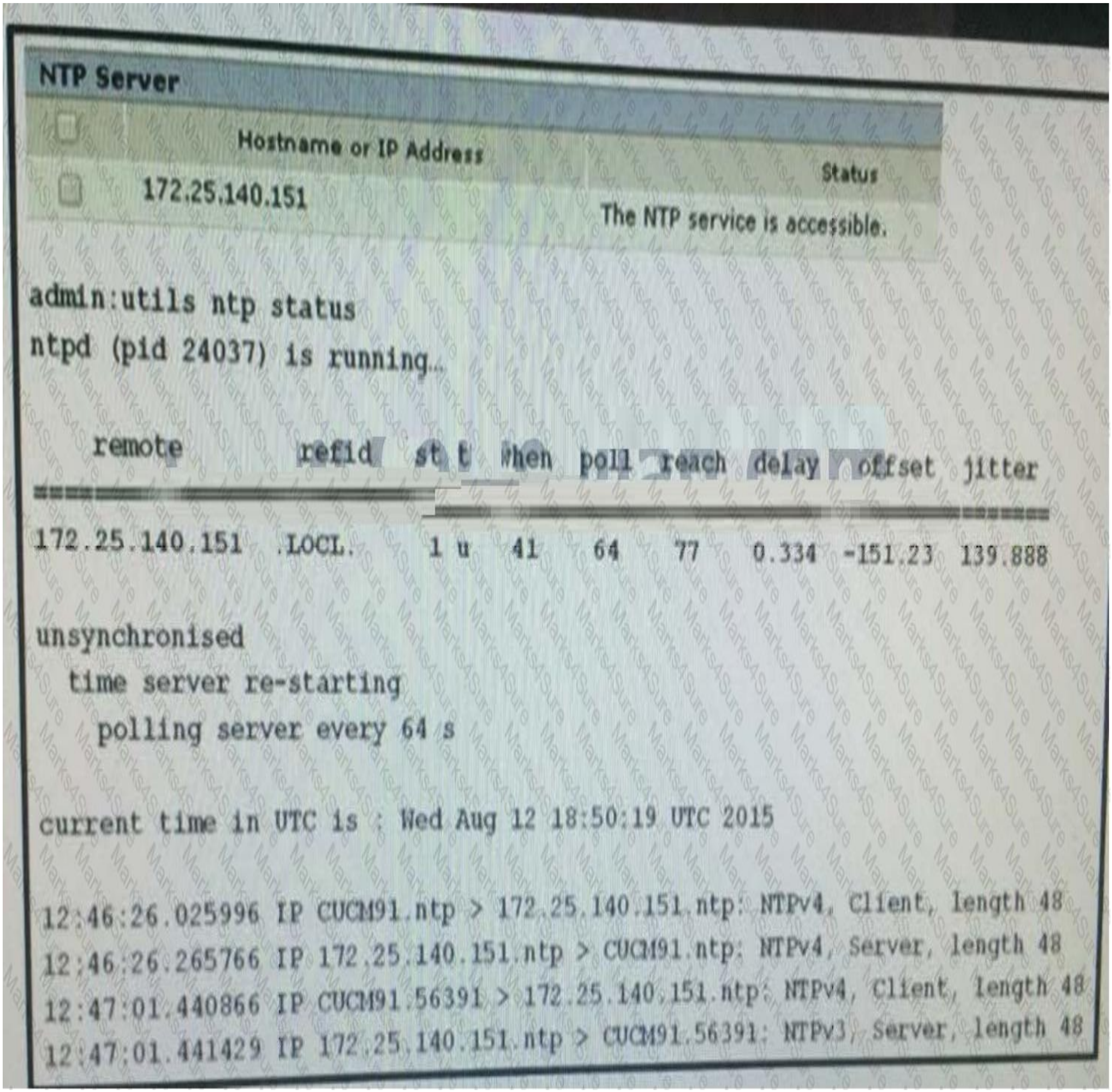
to secure CTI Application communications. Which two additional roles enable secure transmission of RTP and signaling for CTI applications? (choose two)

A. Standard CTI Allow Control of All Devices
B. Standard CTI Secure Connection
C. Standard Confidential Access Level Users
D. Standard CTI Secure Control of Phones Supporting Connected Xfer and Conf
E. Standard CTI Allow Reception of SRTP Key Material
F. Standard CTI Allow Control of Phones Supporting Rollover Mode

Answer: B C

Question #:340 - (Exam Topic 1)

Refer to the exhibit.



A network engineer is troubleshooting a NTP synchronized issue in CUCM. Why is NTP Unsynchronized?

A. The NTP server used is a Windows based NTP server

B. The IOS Command NTP server 172.25.140.151 version 3 is advertising NTPv3

C. The NTP server stratum is higher than four

D. A firewall is blocking NTP port 123

Answer: A

Question #:341 - (Exam Topic 1)

Which two header fields do a SIP user Agent Server use to indicate that the message needs Provisional Response Acknowledgement? (choose two)

A. Allow: 100rel

B. Cseq:[Numeric value that uniquely identifies the response message]

C. Rack:[Numeric value that uniquely identifies the response message]

D. Require: 100rel

E. Rseq:[Numeric value that uniquely identifies the response message]

F. Allow-Events: 100rel

Answer: D E

Question #:342 - (Exam Topic 1)

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

A. traffic shape rate 70000000 8750000 8750000

B. traffic shape rate 100000000 70000000 70000000

C. police cir 70000000 confirm - action transmit exceed - action drop

D. police 70000000 13125000 confirm - action transmit exceed - action drop

Answer: A

Question #:343 - (Exam Topic 1)

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? (Choose two.)

A. Closed

B. Busy

C. Alternate

D. Error

E. Internal

Answer: D

Question #:344 - (Exam Topic 1)

Within infrastructure as a Service, which two components are managed by the provider? (Choose two.)

A. applications

B. data

C. networking

D. servers

E. runtime

Answer: A D

Question #:345 - (Exam Topic 1)

Where can a Cisco Unified CM administrator define Call Detail Records data collection interval (Choose two.)

A. Cisco Unified Reporting

B. Cisco Unified CM Administration Enterprise Parameters

C. Cisco Unified Serviceability

D. Cisco Unified CM Administration Service Parameters

E. Call Detail Records data collection interval is not a configurable parameter.

Answer: B

Question #:346 - (Exam Topic 1)

You are configuring a Cisco Unified Communication Manager IM & Presence environment to support XMPP federation. Which set of protocol and port must you ensure that the firewall opens up to allow for this support?A. udp/5289

B. tcp/8443

C. tcp/5269

D. udp/8443

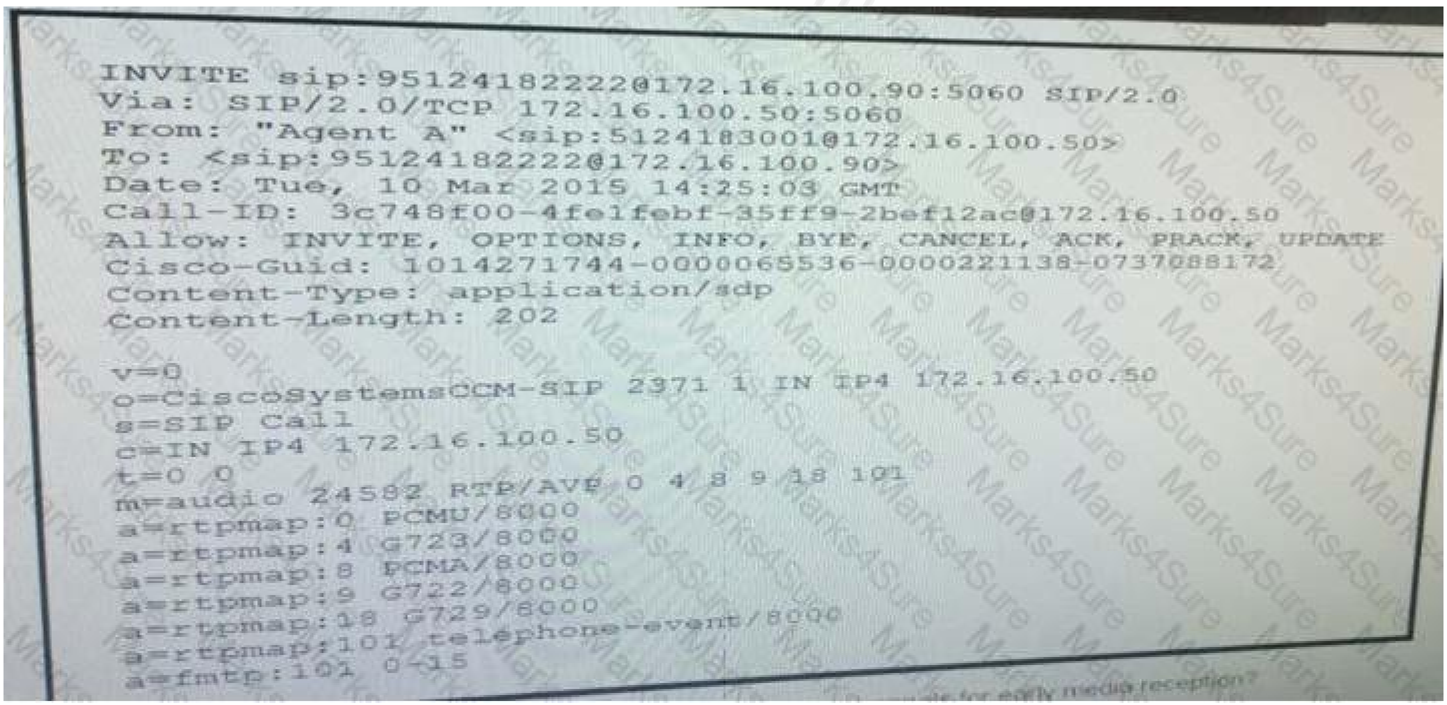
E. udp/5222

F. tcp/5222

C

Question #:347 - (Exam Topic 1)

Refer to the exhibit.



```
INVITE sip:9512418222@172.16.100.90:5060 SIP/2.0
Via: SIP/2.0/TCP 172.16.100.50:5060
From: "Agent A" <sip:5124183001@172.16.100.50>
To: <sip:9512418222@172.16.100.90>
Date: Tue, 10 Mar 2015 14:25:03 GMT
Call-ID: 3c748f00-4fe1febf-35ff9-2bef12ac@172.16.100.50
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE
Cisco-Guid: 1014271744-0000065536-0000221138-0737088172
Content-Type: application/sdp
Content-Length: 202

v=0
o=CiscoSystemsCCM-SIP 2371 1 IN IP4 172.16.100.50
s=SIP Call
c=IN IP4 172.16.100.50
t=0 0
m=audio 24582 RTP/AVP 0 4 8 9 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Which SIP message will trigger the calling device to open channels for early media reception?

A. 180 Ringing

B. ACK

C. INVITE

D. 183 session-progress

E. 200 OK

Answer: D

Question #:348 - (Exam Topic 1)

An engineer notices that two Cisco unity 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 indicators can 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 utils cuc cluster activate to re-establish intracluster communication.

D. A message left on the subscriber server during the outage may be lost during the cluster recovery.

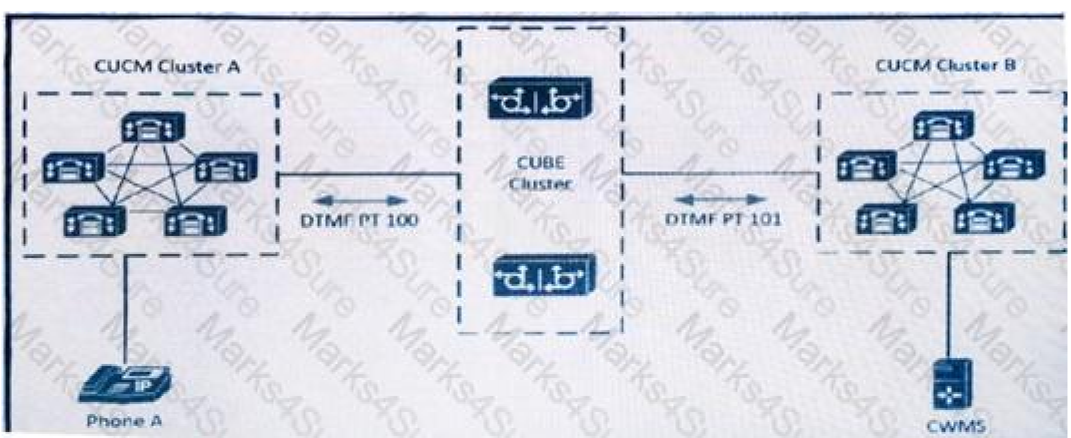
E. The replication between the nodes becomes defunct, requiring the administrator to run utils cuc cluster renegotiate to re-establish intracluster communication.

F. The Unity Connection Database can become corrupted, causing the need to reinstall the subscriber server.

Answer: A C

Question #:349 - (Exam Topic 1)

Refer to the exhibit.



A CUBE Cluster is working in HSRP box-to-box failover model. When the phone A calls Cisco WebEx meeting server to start a conference session, no DTMF tones are recognized. Which configuration change will fix this problem when configured on both CUBEs?

A. Voice-class sip asymmetric payload dtmf in dial-peer configuration

B. Dtmf-relay rtp-nte digitdrop in the dial-peer configuration

C. Media flow-around under voice service voip configuration

D. Modem relay nse payload-type101 underglobal sip configuration

E. Asymmetric payload full configured under global sip configuration

Answer: E

Question #:350 - (Exam Topic 1)

Three calls are active on ephone 1. Assume ephone 2 will remain idle.

```
!
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 additional calls can be placed from ephone 1?

A. 0

B. 1

C. 2

D. 3

E. 5

Answer: C

Question #:351 - (Exam Topic 1)

Which three statements about blending addressing in ILS/URI dialing are true? (choose three)

A. The destination endpoint must use the directory URI and the directory number for its response

B. The destination endpoint can use either the directory URI or the directory number for its response, both reach the same destination

C. Cisco Unified Communication manager inserts the directory URI and the directory number of the sending party in outgoing SIP invites but not for the responses to SIP invites

D. Cisco Unified Communication manager inserts the directory URI and the directory number of the sending party in outgoing responses to SIP invites

E. Cisco Unified Communication manager inserts the directory URI of the sending party in outgoing SIP invites

Answer: B D E

Question #:352 - (Exam Topic 1)

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 temporary 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

Answer: D

Question #:353 - (Exam Topic 1)

A customer is trying to configure Cisco Unified Communication Manager IM & Presence to support XMPP federation and asks you which DNS SRV record(s) must be created in the public DNS server.

Assume that the Cisco Unified Communication Manager IM & Presence is using "Default Domain" IM Address Scheme with domain customer.com. Which DNS SRV record(s) must be created?

A. _xmpp-client _tls.customer.com

B. _xmpp-server _tcp.customer.com

C. _xmpp-client _tcp.customer.com

D. _xmpp-server _tls.customer.com

E. _xmpp-sever _udp.customer.com

Answer: B

Question #:354 - (Exam Topic 1)

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

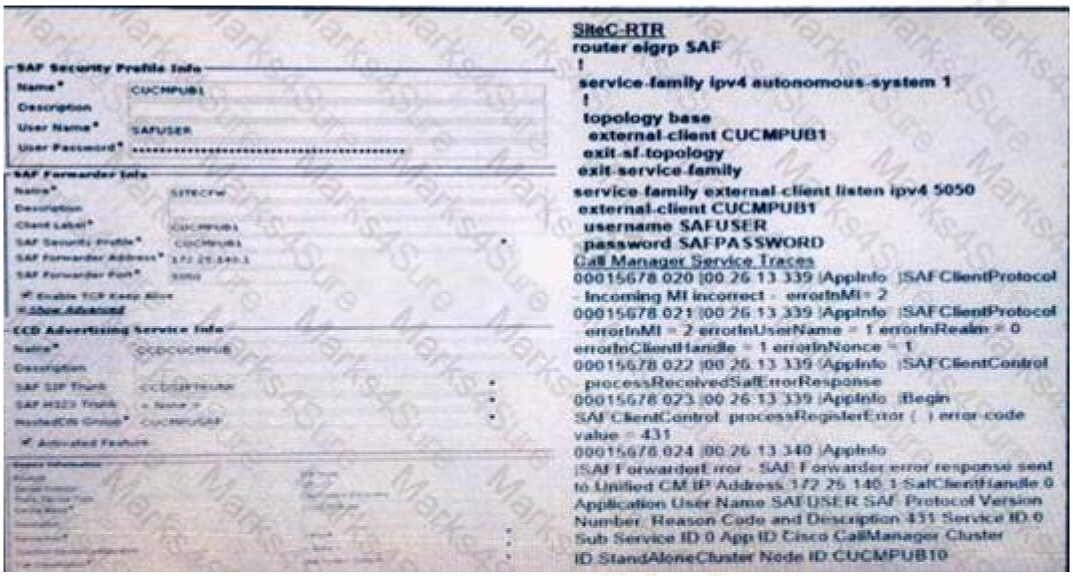
D. Cisco CallManager

E. Cisco TFTP

Answer: C

Question #:355 - (Exam Topic 1)

Refer to the exhibit.



A collaboration Engineer using the SHOW EIGRP service-family External-client IOS Command, noticed that a CUCM failed to register as an external SAF Client on a Cisco IOS router named SiteC-CRTR. The engineer has collected snippets of the IOS configuration screenshots and

CUCM trace shown in the exhibit.

What is the reason for the registration failure?

A. Password mismatch between CUCM and Router SAF configuration.

B. sf-interface loopback0 command missing under service-family ipv4 autonomous-system 1.

C. SIP trunk IP address pointing to a different address than SAF Forwarder address.

D. Name mismatch between SAF Forwarder name info field and external-client name on router.

E. ip multicast-routing command missing on router configuration.

Answer: B

Question #:356 - (Exam Topic 1)

Which three statements below are correct regarding tenant partitioning in Cisco Unity Connection? (choose three)

A. Each tenant can be associated only with one phone system and Cisco Unified Communication Manager

B. Overlapping Extension within a tenant is not supported

C. Ports are shareable between multiple tenants as long as you have the correct license

D. Voicemail networking such as Diginet and VPIM is supported

E. Custom keypad mappings are shared among all tenants

Answer: A B E

Question #:357 - (Exam Topic 1)

Which four protocols does Cisco Jabber support with Cisco IM & Presence?

A. XML

B. HTTP

C. JSON

D. XXML

E. AXL

F. REST

Answer: A B E F

Question #:358 - (Exam Topic 1)

Refer to the exhibit.

Which description of the event captured in the SIP message between a Cisco Unified Communication Manager Express Router and Cisco Unity Express is true?

A. The Cisco UCM Express router forwarded the SIP message, originated from a registered SIP IP Phone, to Cisco Unity Express to request MWI information

B. A SIP Ip phone initiated this message to the Cisco UCM Express router to request MWI information

C. The Cisco UCM Express router initiated this SIP message in response to an earlier solicitation from Cisco Unity Express for MWI information

D. A SIP Ip phone initiated this SIP message to Cisco Unity Express for MWI to request MWI information

E. The Cisco UCM EXpress router initiated this SIP message to Cisco Unity Express for message notification

Answer: A

Question #:359 - (Exam Topic 1)

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

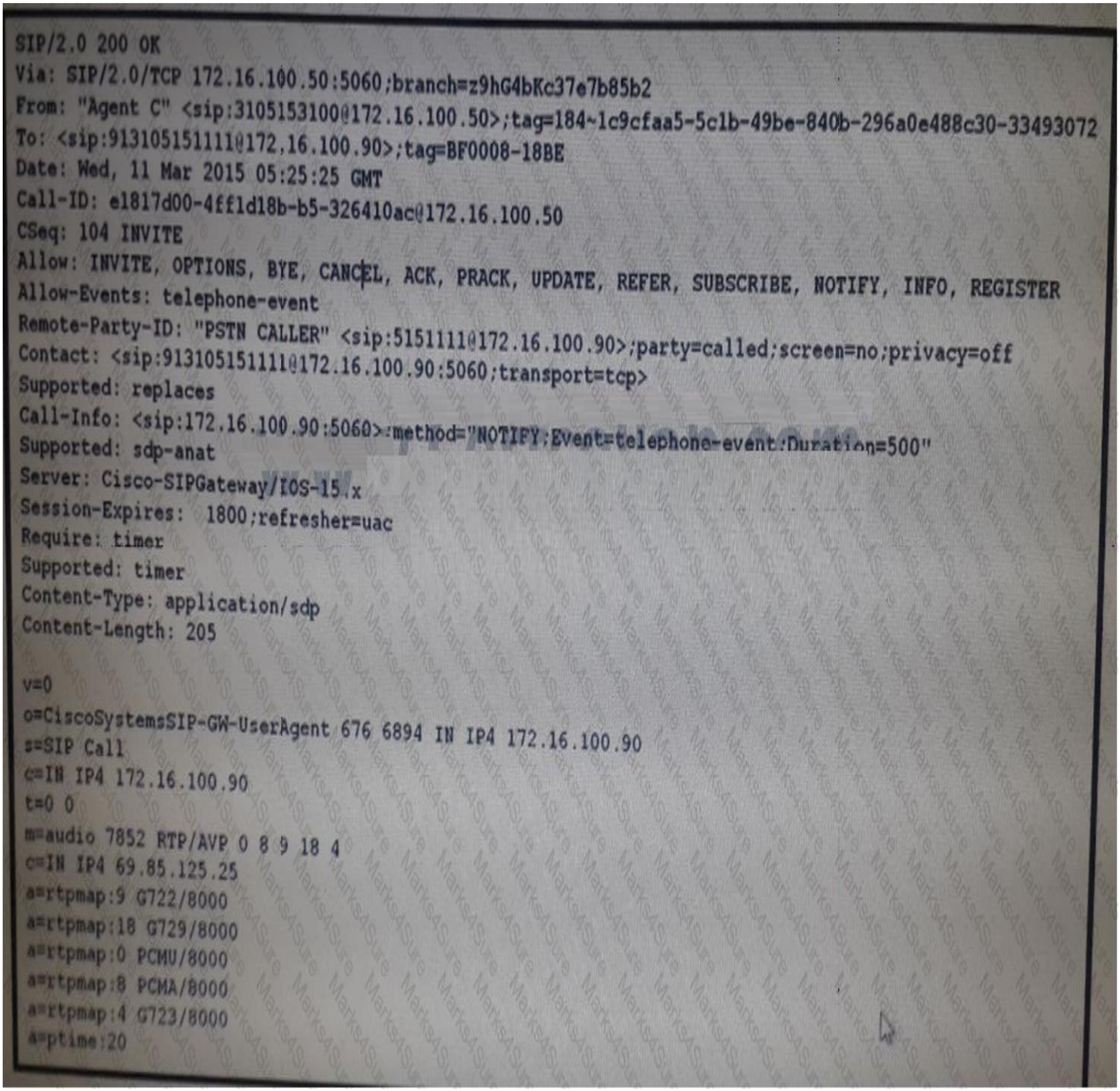
D. Video conferencing resources are reserved dynamically by Cisco IOS and cannot be changed.

E. This command is used for voice resource reservation only.

Answer: A

Question #:360 - (Exam Topic 1)

Refer to the exhibit.



```
SIP/2.0 200 OK
Via: SIP/2.0/TCP 172.16.100.50:5060;branch=z9hG4bKc37e7b85b2
From: "Agent C" <sip:3105153100@172.16.100.50>;tag=184~1c9cfaa5-5c1b-49be-840b-296a0e488c30-33493072
To: <sip:913105151111@172,16.100.90>;tag=BF0008-18BE
Date: Wed, 11 Mar 2015 05:25:25 GMT
Call-ID: e1817d00-4ff1d18b-b5-326410ac@172.16.100.50
CSeq: 104 INVITE
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
Allow-Events: telephone-event
Remote-Party-ID: "PSTN CALLER" <sip:5151111@172.16.100.90>;party=called;screen=no;privacy=off
Contact: <sip:913105151111@172.16.100.90:5060;transport=tcp>
Supported: replaces
Call-Info: <sip:172.16.100.90:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Supported: sdp-anat
Server: Cisco-SIPGateway/IOS-15.x
Session-Expires:  1800;refresher=uac
Require: timer
Supported: timer
Content-Type: application/sdp
Content-Length: 205

v=0
o=CiscoSystemsSIP-GW-UserAgent 676 6894 IN IP4 172.16.100.90
s=SIP Call
c=IN IP4 172.16.100.90
t=0 0
m=audio 7852 RTP/AVP 0 8 9 18 4
c=IN IP4 69.85.125.25
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:4 G723/8000
a=ptime:20
```

Which three pieces of information can be derived from this sip message? (Choose Three)

A. The call will have no audio

B. Only OOB DTMF will be supported

C. G722 codec will be chosen

D. The B2BUA uses IP 172.16.100.50

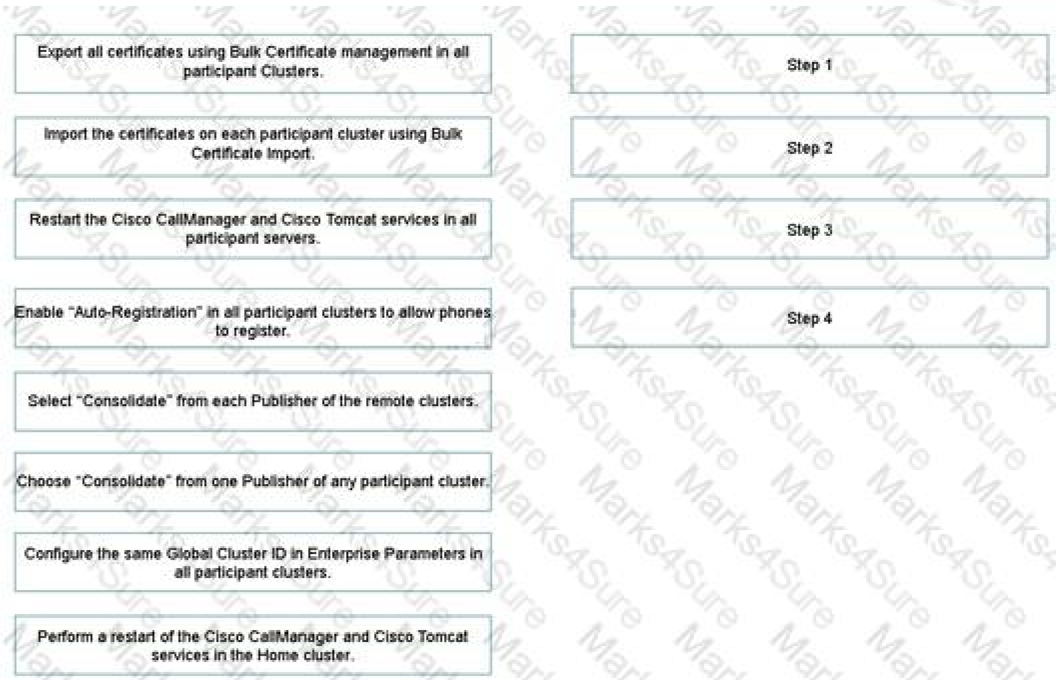
E. This is a flow-around configuration

F. The call will last only 30 minutes

Answer: B D F

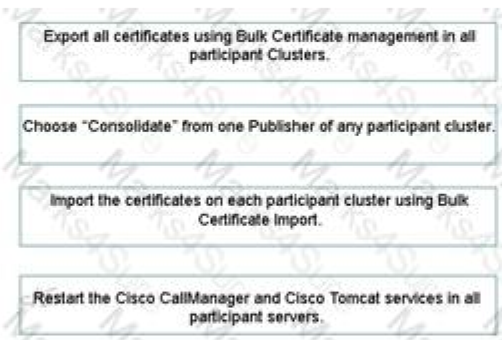
Question #:361 - (Exam Topic 1)

An engineer is setting up a proxy TFTP between multiple Cisco communication Manager clusters. Drag the step from the left to the correct order on the right to properly configure the certificates for the proxy TFTP. Not all options will be used.



Answer:

Explanation



Export all certificates using Bulk Certificate management in all participant clusters. Choose "Consolidate" from one Publisher of any participant cluster. Import the certificates on each participant cluster using Bulk Certificate. Import Restart the Cisco CallManager and Cisco Tomcat services in all participant servers.

Question #:362 - (Exam Topic 1)

An engineer wants to configure a Cisco IOS router to allow for up to four simultaneous conferences in CME mode. Which configuration meets the requirement?

A. dspfram profile 1 conference

Codec G711ulaw

Maximum sessions 4

Maximum conference- participants 8

Telephony-service

Sdspfarm units 4

Sdspfarm tag 4 hwcfb

Conference hardware

B. dspfram profile 1 conference

Codec G711ulaw

Maximum conference-participants 8

Telephony- service

Sdspfarm units 4

Sdspfarm tag 4 hwcfb

Conference hardware

Max-conference-participants 8

C. dspfram profile 1 conference

Codec G711ulaw

Maximum sessions 4

Maximum conference- participants 8

Telephony-service

Sdspfarm units 1

Sdspfarm tag 1 hwcfb

Conference hardware

Max- conferences 2

D. dspfram profile 1 conference

Codec G711ulaw

Maximum sessions 4

Maximum conference- participants 8

Telephony-service

Sdspfarm units 1

Sdspfarm tag 1 hwcfb

Conference hardware

Max- conferences 4

Answer: D

Explanation

dspfram profile 1 conference--> Configure dspfarm conference parameters

Codec G711ulaw --> Configure codecs that are allowed to participate – by default all codecs are supported

Maximum sessions 4 --> Configure maximum number of conferences,here 4

Maximum conference- participants 8 --> Configure max conference participants per conference, here 8

Telephony-service

Sdspfarm units 1 --> Configure maximum number of profiles

Sdspfarm tag 1 hwcfb --> Associate profile name and enable

Conference hardware --> Enable CME hardware conference

Max- conferences 4 --> maximum number of three-party conferences that are supported simultaneously by the Cisco CallManager Express (Cisco CME) router

max-conferences

To set the maximum number of three-party conferences that are supported simultaneously by the Cisco CallManager Express (Cisco CME) router, use the **max-conferences** command in telephony-service configuration mode. To reset this number to the default, use the **no** form of this command.

max-conferences *max-conference-number* **[gain** -6 **|** 0 **|** 3 **|** 6**]**

no max-conferences

Syntax Description

max-conference number	Maximum number of three-party conferences that are supported simultaneously by the router. This number is platform-dependent, and the default is half the maximum for each platform. The following are the maximum values for this argument: • Cisco 1700 series, Cisco 2600 series, Cisco 2801—8 • Cisco 2811, Cisco 2821, Cisco 2851, Cisco 3600 series, Cisco 3700 series—16 • Cisco 3800 series—24 (requires Cisco IOS Release 12.3(11)XL or higher) **Note** Each individual Cisco IP phone can host a maximum of one conference at a time. You cannot create a second conference on the phone if you already have an existing conference on hold.
gain	(Optional) Increases the sound volume of VoIP and public switched telephony network (PSTN) parties joining a conference call. The allowable decibel units are -6 db, 0 db, 3 db, and 6 db. The default is -6 db.

Command Default

Default is half the maximum number of simultaneous three-party conferences for each platform.

Usage Guidelines

This command supports three-party conferences for local and on-net calls only when all conference participants are using the G.711 codec. Conversion between G.711 mu-law and A-law is supported. Mixing of the media streams is supported by the Cisco IOS processor. The maximum number of simultaneous conferences is limited to the platform-specific maximums.

The **gain** keyword's functionality is applied to inbound audio packets, so conference participants can more clearly hear a remote PSTN or VoIP caller joining their call. Note that this functionality cannot discriminate between a remote VoIP/foreign exchange office (FXO) source, which requires a volume gain, and a remote VoIP/IP phone, which does not require a volume gain and may therefore incur some sound distortions.

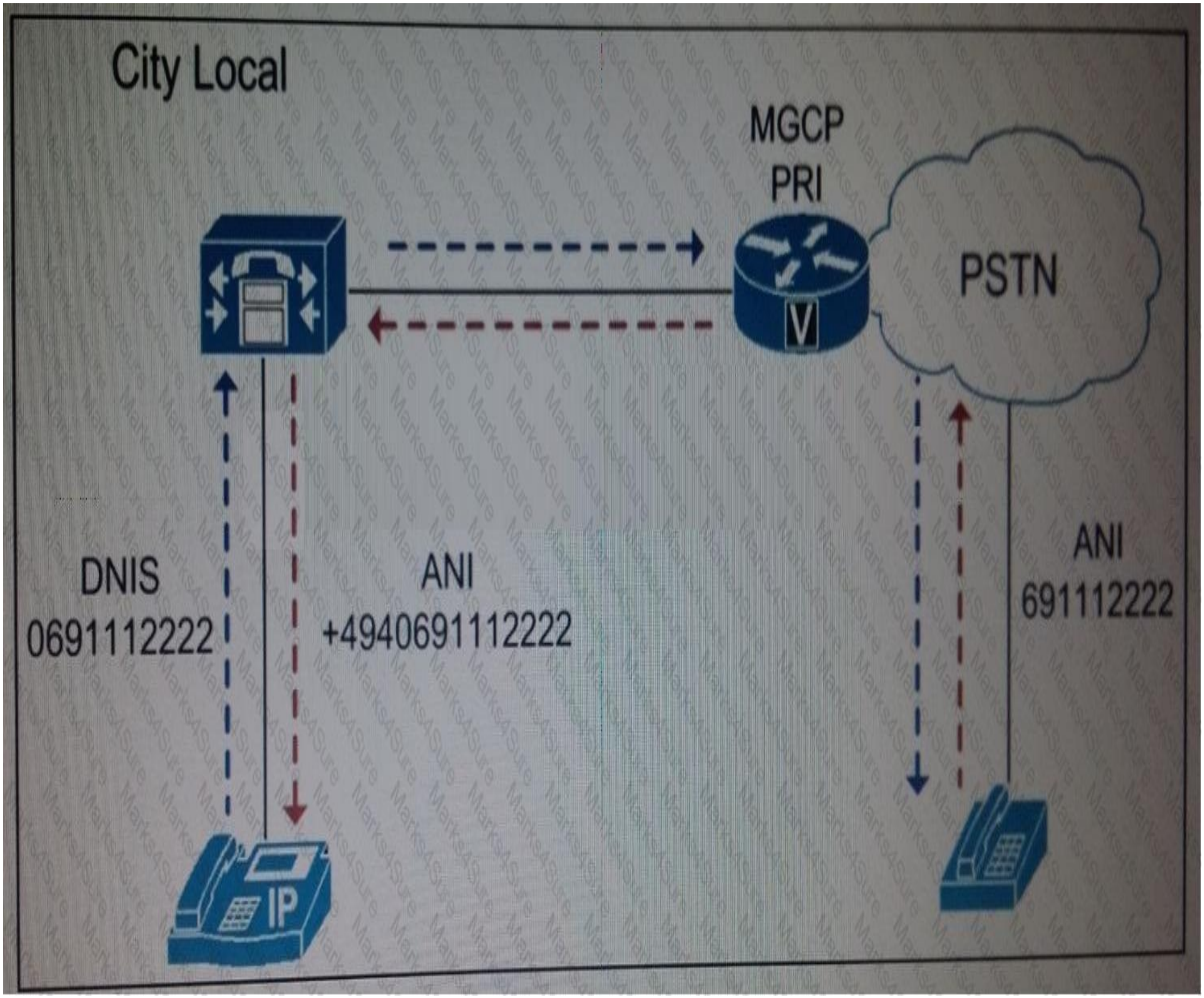
Examples

The following example sets the maximum number of conferences for a Cisco IP phone to 4 and configures a gain of 6 db for inbound audio packets from remote PSTN or VoIP calls joining a conference:

```
Router(config)# telephony-service
Router(config-telephony)# max-conferences 4 gain 6
```

Question #:363 - (Exam Topic 1)

Refer to the exhibit.



A n engineer is working with globalization and has these requirements,

- The ANI to be presented in the E.164 format to the phone on Cisco unified Communications manager

- Allow redialling from the call history without manually manipulating the digit string

Which three configuration steps in Cisco unified Communications manager are needed to meet this requirement? (Choose three).

A. set the prefix digits with 0 and calling number type subscriber

B. set the calling party transformation pattern with \+4940.! And discard digits PreAt

C. set the calling party transformation pattern with \+4940.! And discard digits PreDot

D. set the subscriber number prefix with +4940

E. set the prefix digits with 0 and calling number type national

F. set the national number prefix with +4940

Answer: A C D

Question #:364 - (Exam Topic 1)

Which OpenStack component can be used to create on object-based storage strategy?

A. Glance

B. Swift

C. Nova

D. Neutron

Answer: B

Question #:365 - (Exam Topic 1)

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 multi PPP link with cRTP enabled? (Choose two.)

A. 80.4 kb/s

B. 91.2 kb/s

C. 69.6 kb/s

D. 78.4 kb/s

E. 6 kb/s 2.4

Answer: C

Question #:366 - (Exam Topic 1)

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 SIP Rel1xx Options parameter to Send PRACK if 1xx Contains SDP.

B. Set the SIP Rel1xx parameter to Disabled.

C. Check the Fall Back to Local RSVP check box

D. Set the SIP Rel1xx Options parameter to Send PRACK for all 1xx Messages

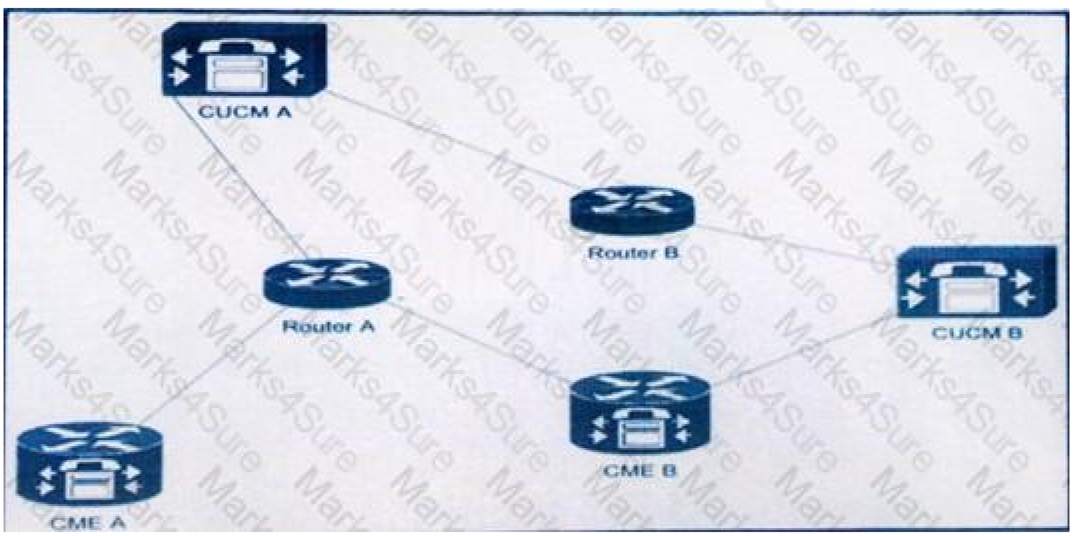
E. Set the RSVP over SIP parameter to Local RSVP

F. Set the RSVP over SIP parameter to E2E

Answer: A F

Question #:367 - (Exam Topic 1)

Refer to the exhibit.



A collaboration engineer is configuring dynamic call routing and DN learning between two Cisco UCM and Two Cisco UCM express. What two configuration tasks will support this? (Choose two)

A. should be configure as service advertisement forwarder only

B. should be configure as service advertisement client and forwarder

C. Router A and should be configure to use the same autonomous system number

D. Router A and should be configure to use the same autonomous system number

E. Router and should be configure to use the same autonomous system number

F. should be configured as service advertisement client only.

Answer: B C

Question #:368 - (Exam Topic 1)

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.

Answer: A B

Question #:369 - (Exam Topic 1)

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

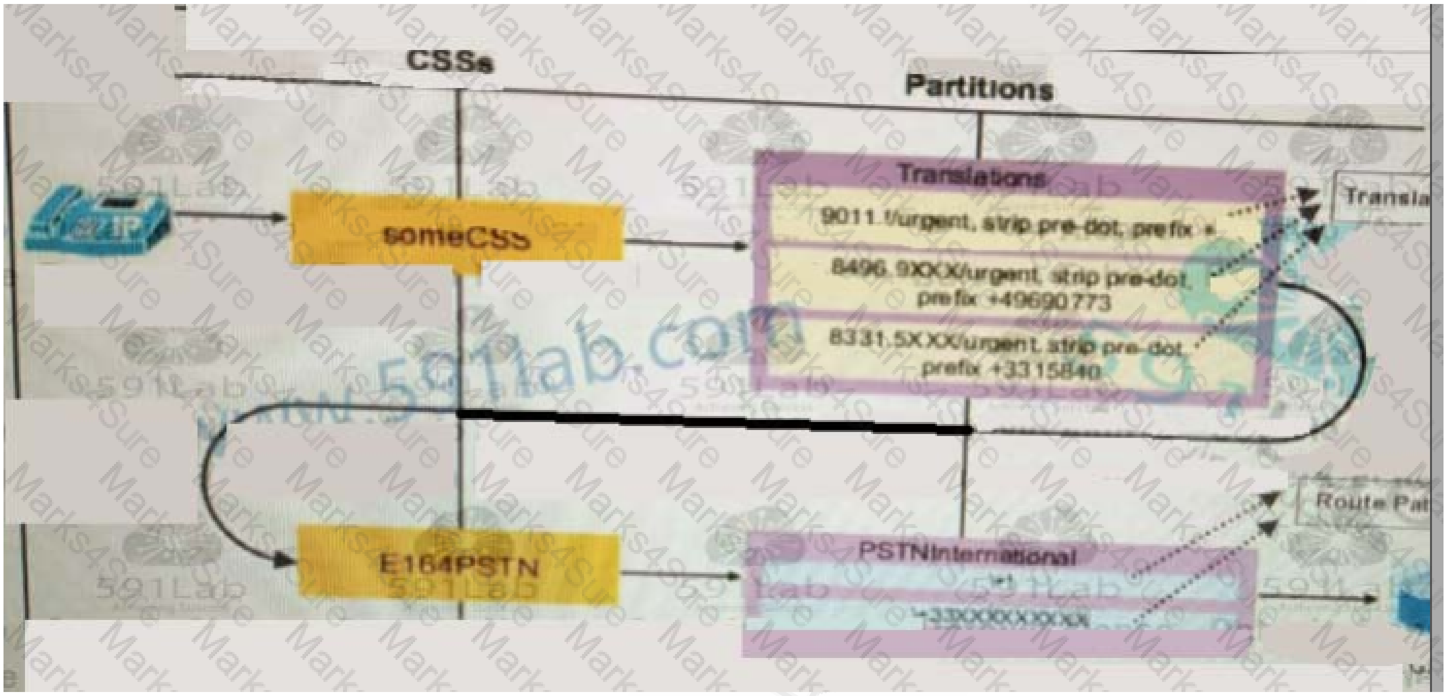
E. Ground-start Signaling Disconnect

F. Tone-based Supervisory Disconnect

Answer: D E F

Question #:370 - (Exam Topic 1)

Refer to the exhibit.



which statement describes the expected behavior when the IP phone registered on a Cisco Unified Communication Manager version 11.0 server dials 83315235?

A. The call must wait for inter-digit timeout before being routed toward the gateway, unless the "Do Not Wait for inter-digit Timeout on Subsequent Hops" is set on all the listed translation patterns

B. The call must wait for inter-digit timeout before being routed toward the gateway, unless the "Do Not Wait for inter-digit Timeout on Subsequent Hops" is set on all the listed translation patterns and route patterns

C. The call fails because there are no matching patterns

D. The call must wait for inter-digit timeout before being routed toward the gateway, unless the "Do Not Wait for inter-digit Timeout on Subsequent Hops" is set on all the listed route patterns

E. The call is routed immediately toward the gateway without inter-digit timeout

Answer: E

Question #:371 - (Exam Topic 1)

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 2nd 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 NO:

Answer: B

Question #:372 - (Exam Topic 1)

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

Answer: C

Question #:373 - (Exam Topic 1)

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 NO:.

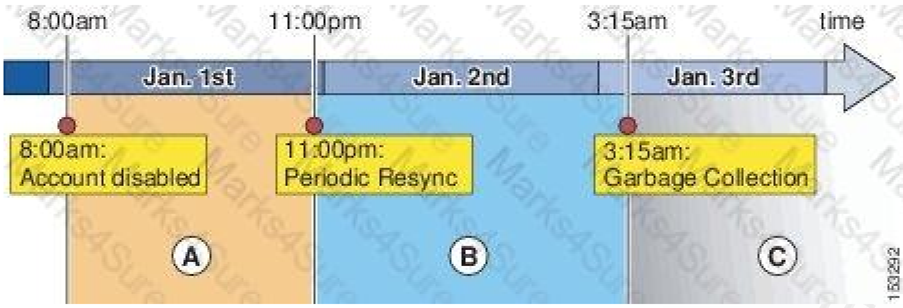
Answer: C

Explanation

Account Synchronization with Active Directory

Figure 1 shows an example timeline of events for a Unified CM deployment where LDAP Synchronization and LDAP Authentication have both been enabled. The re-synchronization is set for 11:00 PM daily.

Figure 1 Change Propagation with Active Directory



After the initial synchronization, the creation, deletion, or disablement of an account will propagate to Unified CM according to the timeline shown in Figure 17-7 and as described in the following steps:

1. At 8:00 AM on January 1, an account is disabled or deleted in AD. From this time and during the whole period A, password authentication (for example, Unified CM User Options page) will fail for this user because Unified CM redirects authentication to AD. However, PIN authentication (for example, Extension Mobility login) will still succeed because the PIN is stored in the Unified CM database.

2. The periodic re-synchronization is scheduled for 11:00 PM on January 1. During that process, Unified CM will verify all accounts. Any accounts that have been disabled or deleted from AD will at that time be tagged in the Unified CM database as inactive. After 11:00 PM on January 1, when the account is marked inactive, both the PIN and password authentication by Unified CM will fail.

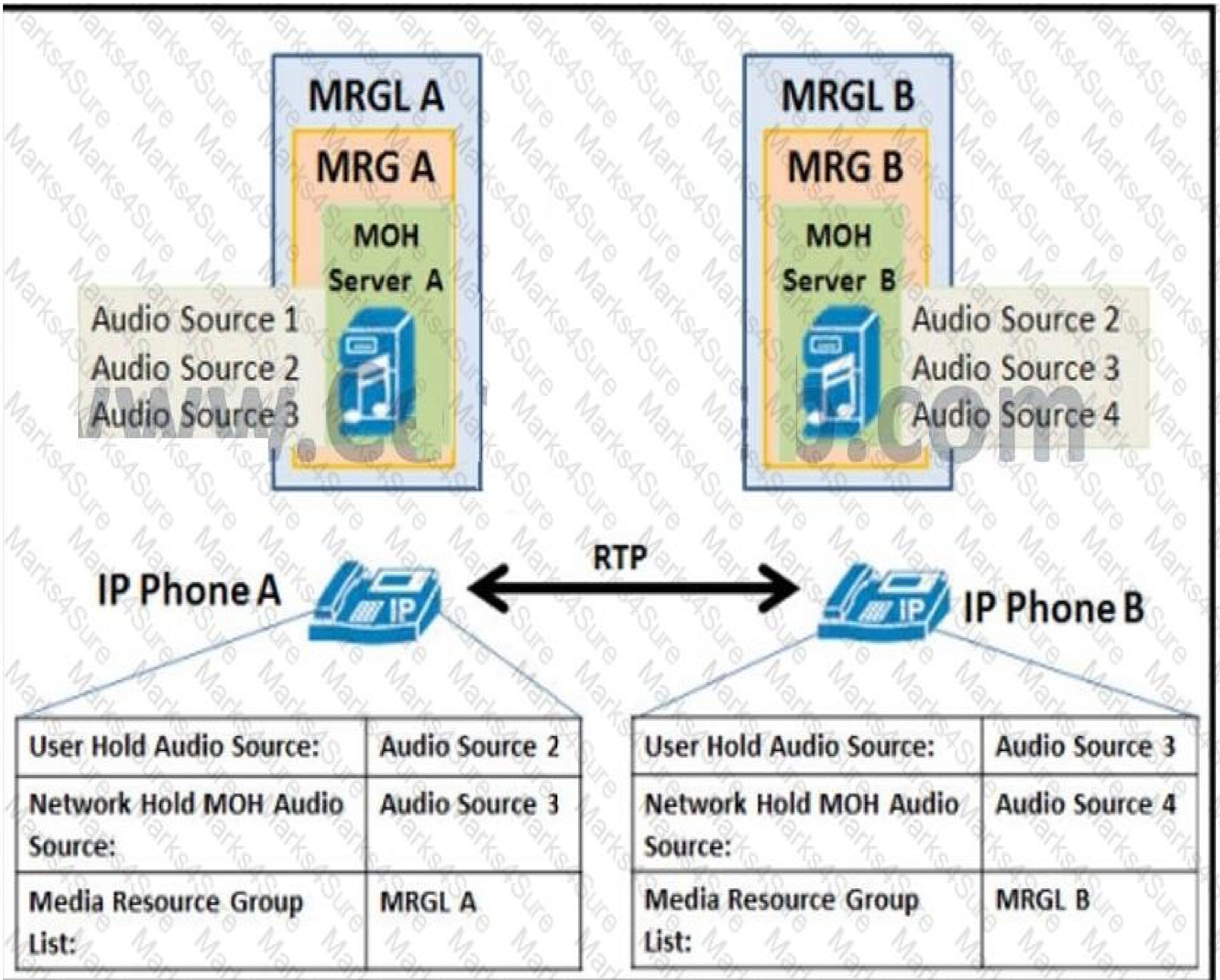
3. Garbage collection of accounts occurs daily at the fixed time of 3:15 AM. This process permanently deletes user information from the Unified CM database for any record that has been marked inactive for over 24 hours. In this example, the garbage collection that runs at 3:15 AM on January 2 does not delete the account because it has not been inactive for 24 hours yet, so the account is deleted at 3:15 AM on January 3. At that point, the user data is permanently deleted from Unified CM.

If an account has been created in AD at the beginning of period A, it will be imported to Unified CM at the periodic re-synchronization that occurs at the beginning of period B and will immediately be active on Unified CM.

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/8x/uc8x/directry.html

Question #:374 - (Exam Topic 1)

Refer to 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.

Answer: E

Question #:375 - (Exam Topic 1)

A Cisco Unified CM cluster is being set up for call control discover using the service advertising framework. An engineer discovers that patterns are not being learned by the cluster. Which two items must be checked in an attempt to resolve the issue? (Choose two)

A. The CCD block patterns are not preventing remote patterns from being entered into the local cache.

B. The hostedDN group on the cluster matches the patterns that should be learned.

C. The CCD advertising service is activated in Cisco unified CM serviceability.

D. A CCD route partition has been assigned for learned patterns.

E. The CCD requesting service is activated in Cisco unified CM serviceability.

F. The Sip trunk is enabled for call control discover.

Answer: A F

Question #:376 - (Exam Topic 1)

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

Answer: E

Question #:377 - (Exam Topic 1)

Which header field does a SIP User Agent Client use to advertise that it supports Provisional Response Acknowledgement?

A. Supported: 100rel

B. Cseq: PRACK

C. Require: 100rel

D. Allow: PRACK

E. Accept: 100rel

Answer: C

Question #:378 - (Exam Topic 1)

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.

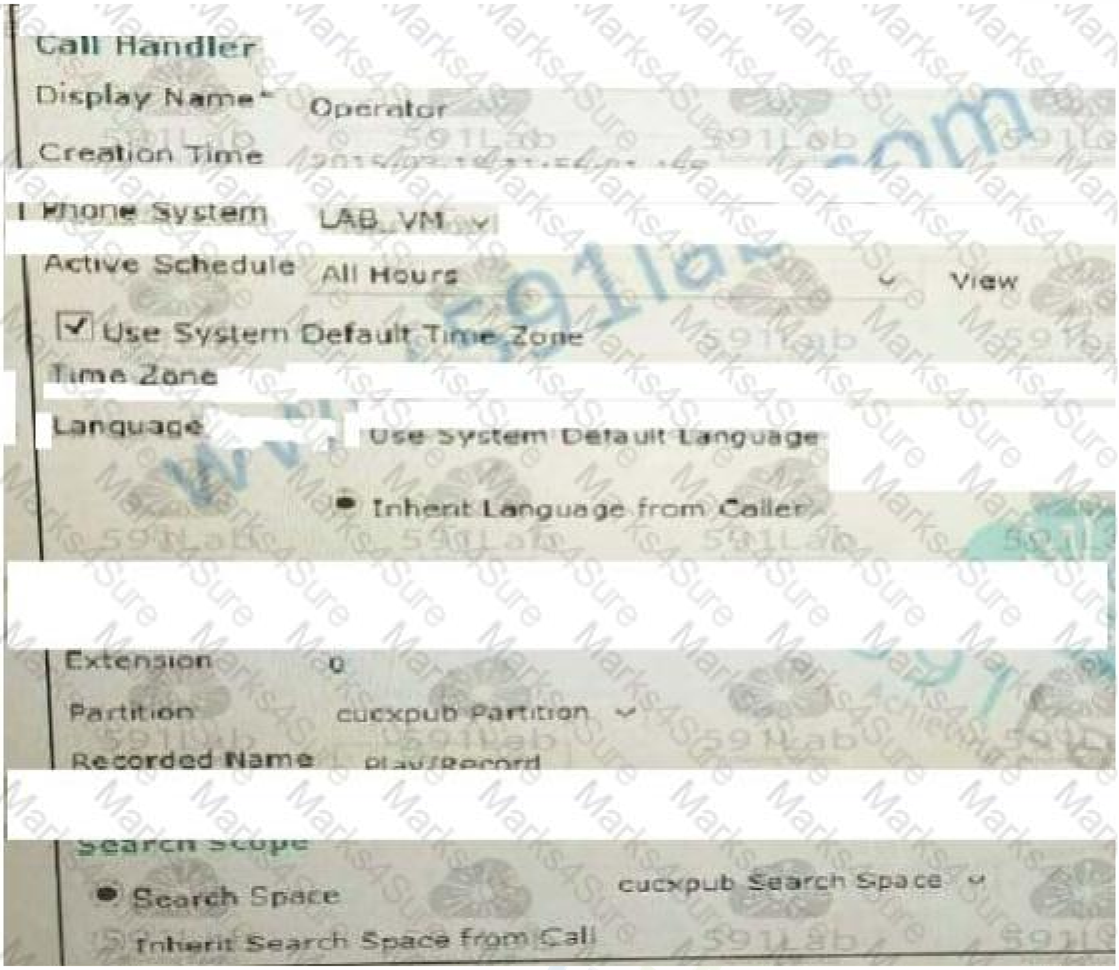
D. Enable Prepend Digits to Dialled Extensions and configure complete extension number on the Edit Caller input page for the selected key.

E. Reduce Caller input timeout in Cisco Unity Connection Enterprise Parameters.

Answer: C

Question #:379 - (Exam Topic 1)

Refer to the exhibit.



A unity administrator is assigned to determine why the Corporate System users cannot call certain Call Handlers. What configuration element should be changed to allow the users to reach this call handler?

A. The Phone System

B. The Active Schedule

C. The Search Space

D. The Partition

Answer: D

Question #:380 - (Exam Topic 1)

Refer to the exhibit.

```
class-map match-all Control
   match ip dscp cs3
   match ip dscp af31
   match access-group name Voice
class-map match-all RTP
   match ip dscp ef
   match access-group name Voice
   match precedence 5
!
policy-map Voice-out
   class RTP
      priority percent 60
   class Control
      bandwidth remaining percent 5
   class class-default
      bandwidth remaining percent 95

ip access-list extended Voice

permit upd any 10.10.25.0 255.255.0.0 range 16384 32767

interface Serial 0/0
   ip address 10.11.25.1 255.255.255.0
   duplex auto
   speed auto
   service-policy out Voice-out

interface GigabitEthernet 0/1

ip address 10.10.25.1 255.255.255.0

duplex auto

speed auto
```

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.

Answer: B D

Question #:381 - (Exam Topic 1)

Multiple users report that when they try to login to Cisco Unified Communications Extension Mobility Manager this error is retrieved?

"Error:-[26] Busy, please try again"

Which description of the cause is true?

A. The intracluster Multiple Login Behaviour is set to Multiple Logins Not Allowed.

B. the Validate IP Address parameter is set to false, so the IP is not found in the trust list.

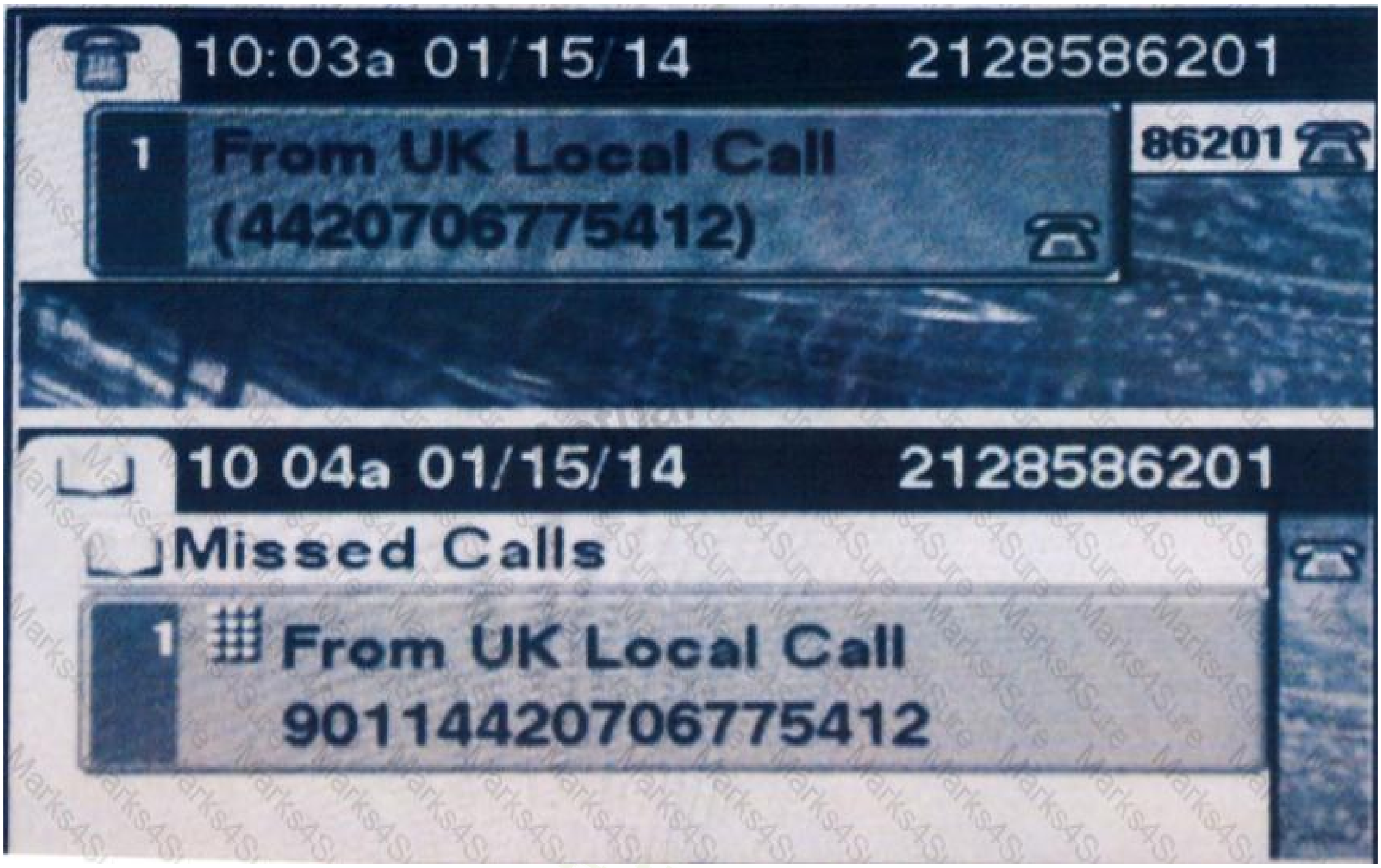
C. The number of concurrent log in and logout requests is higher than the Maximum Concurrent Requests service parameter.

D. the Extension Mobility Cache size has been lowered to 1000 and no more logins are allowed beyond this number.

Answer: C

Question #:382 - (Exam Topic 1)

Refer to the exhibit.



A CUCM engineer is working with Globalization and localization on H323 gateway. Which four configuration changes are needed to achieve the result on the exhibit? (Choose four)

A. Create a CSS and PT for calling party transformation pattern.

B. Create a transformation profile and add 9011 in the international number prefix field.

C. Assign a transformation profile in the incoming transformation profile setting in the E 164 transformation number prefix field.

D. Assign the calling party transformation CSS to the device pools in the cluster.

E. Uncheck the use device pool calling party transformation CSS on all the phones.

Answer: A B C E

Question #:383 - (Exam Topic 1)

Which statement describes DTMF processing on a Cisco Unified Contact Centre Express with supported SIP based on agent IP phones that are registered to Cisco Unified Communications Manager?

A. Cisco Unified CCX receives the DTMF digits in the RTP payload based on RFC 2833

B. Cisco Unified CCX receives the DTMF digits via SIP NOTIFY messages

C. Cisco Unified CCX receives the DTMF digits via SIP INFO messages

D. Cisco Unified CCX receives the DTMF digits JTAPI messages

E. Cisco Unified CCX receives the DTMF digits as part of the audio encoding in the RTP stream.

Answer: D

Question #:384 - (Exam Topic 1)

When using LDAP synchronization to automatically create and synchronize Cisco Unified Communication Manager end users, which two LDAP attributes can be synchronized to directory URI? (choose two)

A. msRTCSIPprimaryuseraddress

B. mail

C. msRTCIPuseraddress

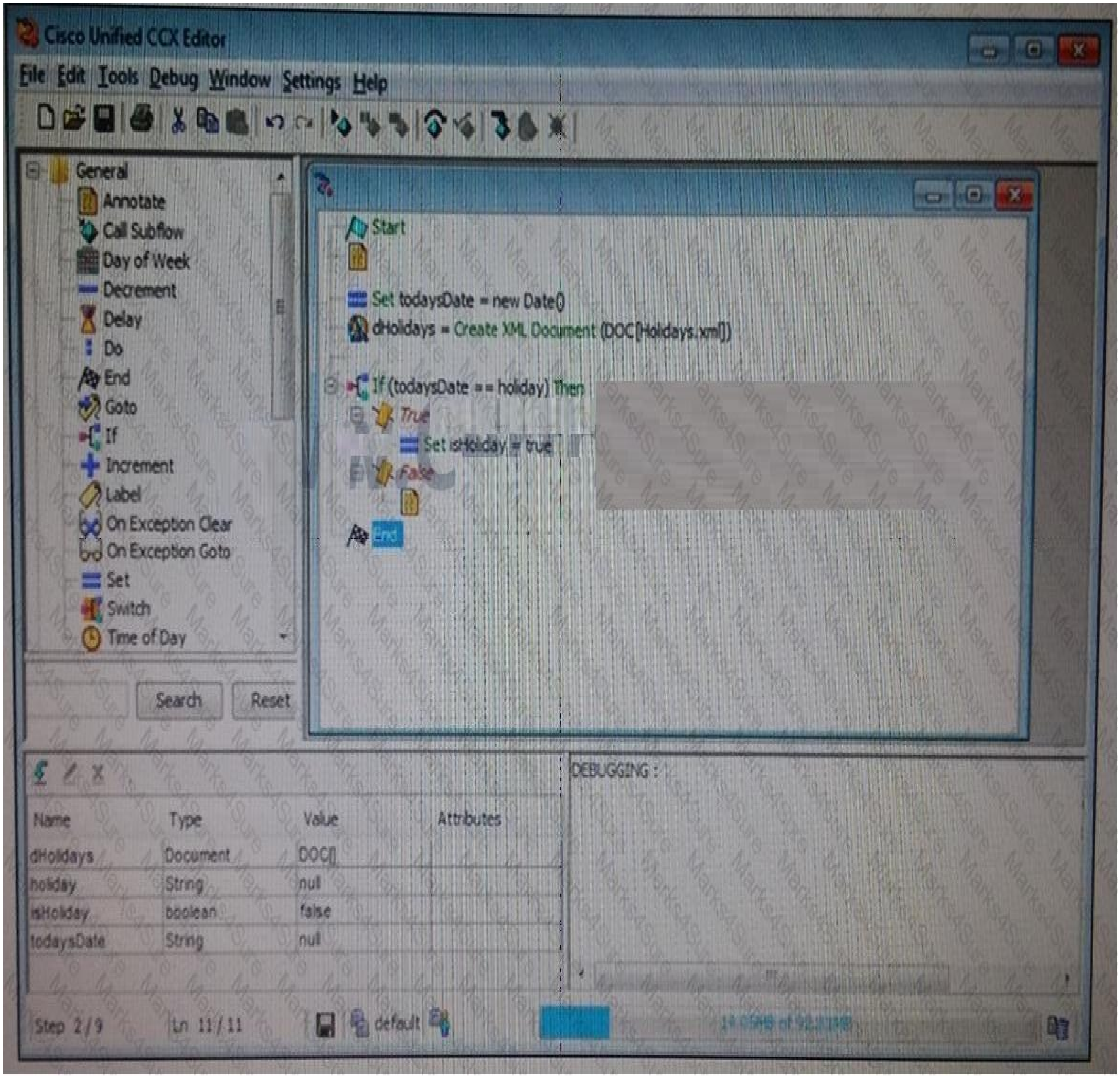
D. email

E. emailaddress

Answer: A B

Question #:385 - (Exam Topic 1)

Exhibit:



A UCCX engineer is configuring a script that allows the clients to listen to audio based on a holiday schedule.

Which step allows the script to play prompts from an external VXML application?

A. Voice browser step

B. Play prompt step

C. Generic recognition step

D. Call subflow step

Answer: A

Question #:386 - (Exam Topic 1)

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. Trnsfer

C. Join

D. RmLstC

E. Select

F. Confrn

Answer: A F

Question #:387 - (Exam Topic 1)

```
INFO        [main]sync.SyncUtil-Got[AppUserUsername]=[appadmin]
platformConfig.xml
INFO [main]sync.SyncUtil-Query CUCM DB:select a plod,a name， c.cred
application
INFO [main]sync.SyncUtil-Readming UCM DB connection status from file
ERROR [main]sync.SyncUtil – Unable to find the appluser in the cuc
updating CUP
```

Your customer reported that the Sync Agent service failed to start after a reinstallation of a Cisco this problem after you review these customer logs?

A. Add "appadmin" application user in Cisco UCM and IM & Presence

B. Add "appadmin" application user in Ciscon IM & Presence

C. Add "appaduser" application user in Cisco UCM

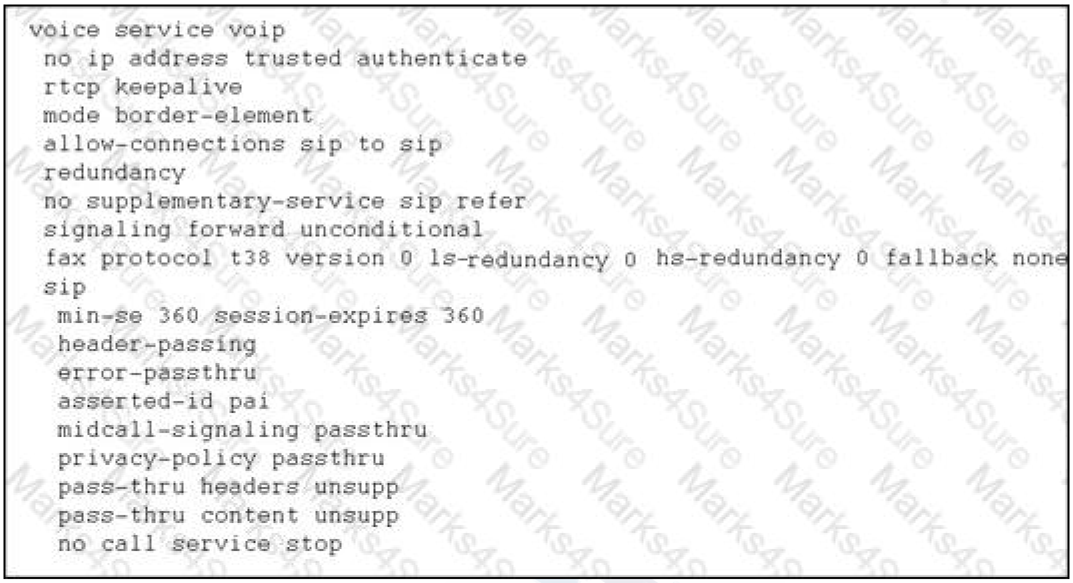
D. Add "appadmin" application user in Cisco UCM

E. Add "appaduser" application user in Cisco UCM and IM & Presence

Answer: D

Question #:388 - (Exam Topic 1)

Refer to the exhibit.



```
voice service voip
 no ip address trusted authenticate
 rtcp keepalive
 mode border-element
 allow-connections sip to sip
 redundancy
 no supplementary-service sip refer
 signaling forward unconditional
 fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
 sip
  min-se 360 session-expires 360
  header-passing
  error-passthru
  asserted-id pai
  midcall-signaling passthru
  privacy-policy passthru
  pass-thru headers unsupp
  pass-thru content unsupp
  no call service stop
```

Which two SIP packet handling behavior will result with this CISCO Unified Border Element (CUBE) configuration? (Choose two)

A. Unsupported content/MIME pass-through

B. SIP Refer is not supported when received on this CUBE

C. Privacy headers received on SIP massage will be replaced with NON-privacy headers on this CUBE

D. P-Preferred identities

E. Mid-call codec changes

Answer: A E

Question #:389 - (Exam Topic 1)

Which three services must be stopped to change the IM&Presence service default domain setting of DOMAIN NOT SET? (Choose three.)

A. Cisco SIP Proxy

B. Cisco ASL Web Service

C. Cisco XCP Router

D. Cisco XCP Authentication Service

E. Cisco Intercluster Sync Agent

F. Cisco Presence Engine

Answer: A C F

Question #:390 - (Exam Topic 1)

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

Answer: A

Question #:391 - (Exam Topic 1)

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

B. SEP111122223333.cnf.xml.enc.sgn

C. SEP111122223333.cnf.xml.sgn.enc

D. SEPDefault.cnf.xml.enc.sgn

E. SEP111122223333.xml.enc

Answer: B

Question #:392 - (Exam Topic 1)

Which two options are Power Save configuration parameters for Cisco 9971 IP Phones? (Choose two.)

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

Answer: C F

Question #:393 - (Exam Topic 1)

Which two search scope options are removed from a directory handler when you check "Voice enabled" Check box? (Choose two.)

A. Class of service

B. Partition

C. Search space

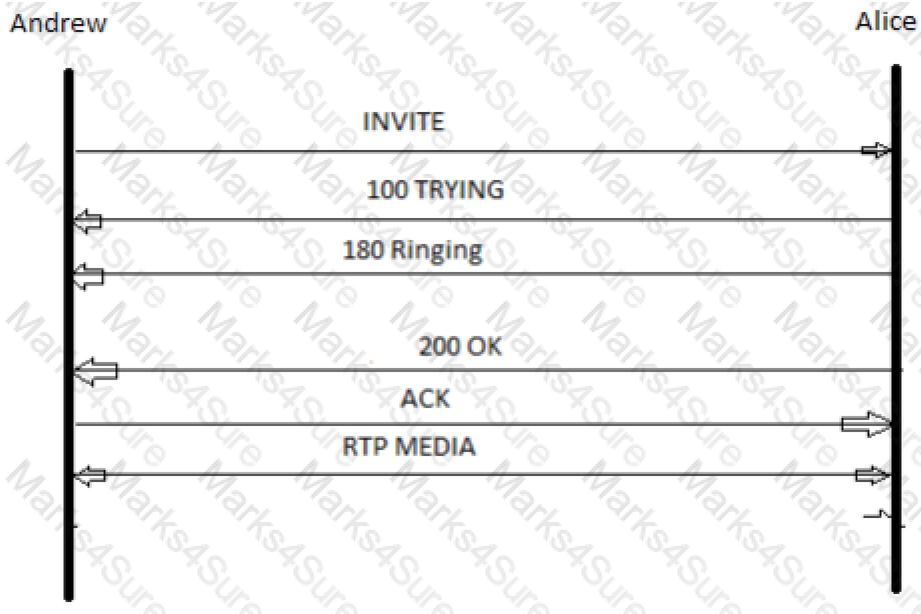
D. System Distribution list

E. Phone system

Answer: A D

Question #:394 - (Exam Topic 1)

Refer to the exhibit.



Which event is next in the SIP call flow when Andrew hangs up the call?

A. A DISCONNECT is generated by Andrew

B. A BYE is generated by Andrew

C. A NOTIFY is generated by Andrew

D. A CANCEL is generated by Andrew

E. A "487 Request Terminated" is generated by Andrew

Answer: E

Question #:395 - (Exam Topic 1)

Refer to Exhibit:

```
909: NOT 20:59:50.051721 VPNC: do_login: got login response
910: NOT 20:59:50.052581 VPNC: process_login: HTTP/1.0 302 Temporary moved
911: NOT 20:59:50.053221 VPNC: process_login: login code: 302 (redirected)
912: NOT 20:59:50.053823 VPNC: process_login: redirection indicated
913: NOT 20:59:50.054441 VPNC: process_login: new 'Location':
    /+webvpn+/index.html
914: NOT 20:59:50.055141 VPNC: set_redirect_url: new URL
    <https://xyz1.abc.com:443/+webvpn+/index.html>
```

A cisco Unified CM engineer configured a phone VPN for remote users but the users cannot register the phones to the VPN. Which configuration change fixes this problem?

A. Configure enable outside in the webVPN configuration on the Cisco ASA

B. Configure the split-tunnel-policy tunnel all attribute on the Cisco ASA

C. Configure the ssl trust-point SSL outside on the Cisco ASA

D. Remove the Cisco ASA IP address from the VPN load-balancing configuration

Answer: D

Question #:396 - (Exam Topic 1)

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.

Answer: C E

Question #:397 - (Exam Topic 1)

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 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. permanently deleted awaiting

D. authorization

E. delete pending

Answer: B

Question #:398 - (Exam Topic 1)

What is the maximum number of configurable speed dial entries for a Cisco Unified 9971 IP Phone?

A. 4

B. 199

C. 50

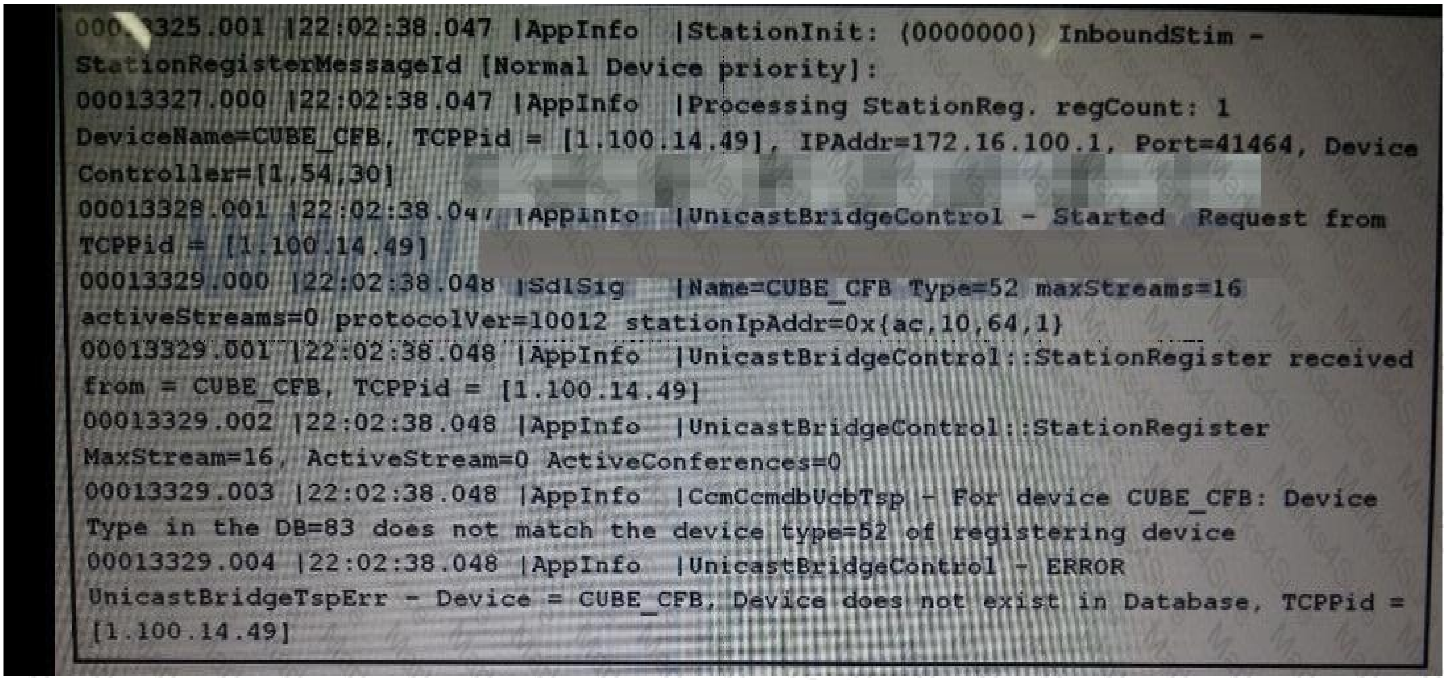
D. 3

E. 2

Answer: B

Question #:399 - (Exam Topic 1)

A CUCM administrator has configured a Cisco IOS CFB using a Cisco 2921 router and encounters an issue.



```
000.3325.001 |22:02:38.047 |AppInfo |StationInit: (0000000) InboundStim -
StationRegisterMessageId [Normal Device priority]:
00013327.000 |22:02:38.047 |AppInfo |Processing StationReg. regCount: 1
DeviceName=CUBE_CFB, TCPPid = [1.100.14.49], IPAddr=172.16.100.1, Port=41464, Device
Controller=[1,54,30]
00013328.001 |22:02:38.047 |AppInfo |UnicastBridgeControl - Started Request from
TCPPid = [1.100.14.49]
00013329.000 |22:02:38.048 |SdlSig |Name=CUBE_CFB Type=52 maxStreams=16
activeStreams=0 protocolVer=10012 stationIpAddr=0x{ac,10,64,1}
00013329.001 |22:02:38.048 |AppInfo |UnicastBridgeControl::StationRegister received
from = CUBE_CFB, TCPPid = [1.100.14.49]
00013329.002 |22:02:38.048 |AppInfo |UnicastBridgeControl::StationRegister
MaxStream=16, ActiveStream=0 ActiveConferences=0
00013329.003 |22:02:38.048 |AppInfo |CcmCcmdbUcbTsp - For device CUBE_CFB: Device
Type in the DB=83 does not match the device type=52 of registering device
00013329.004 |22:02:38.048 |AppInfo |UnicastBridgeControl - ERROR
UnicastBridgeTspErr - Device = CUBE_CFB, Device does not exist in Database, TCPPid =
[1.100.14.49]
```

What is the root cause?

A. The CFB sent the registration message twice

B. The CFB protocol version is not compatible with this CUCM version

C. CUCM has not received the first keepalive message from CFB

D. Device type in CUCM does not match the one configured on Cisco IOS

E. CUBE_CFB name was entered incorrectly in CUCM database

Answer: D

Question #:400 - (Exam Topic 1)

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

Answer: C

Question #:401 - (Exam Topic 1)

Which method does a Cisco unified 9971 Phone use to send keep-alive messages to Cisco unified

Communications manager?

A. Sccp StationServerReq

B. SIP Notify with events set to keep-alive

C. SIP register with expires set to zero

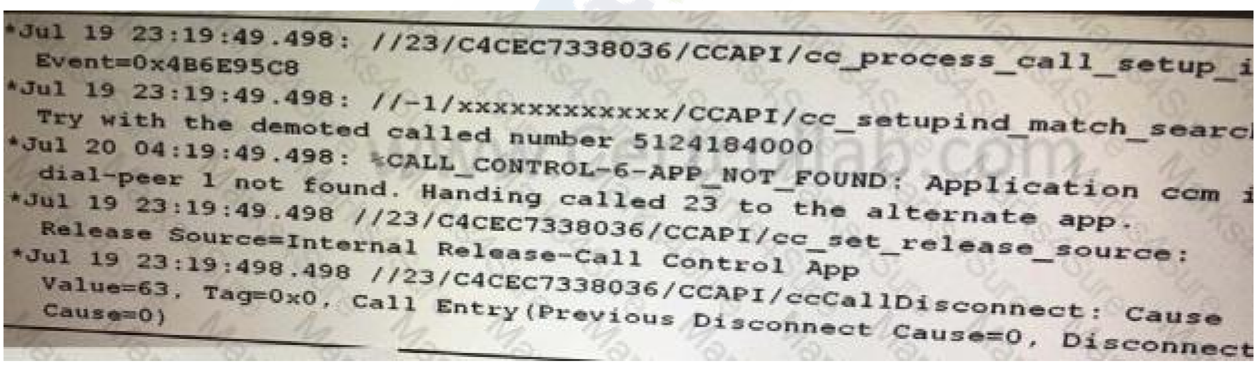
D. Sccp stationregister

E. SIP options

Answer: C

Question #:402 - (Exam Topic 1)

Refer to the exhibit.



```
*Jul 19 23:19:49.498: //23/C4CEC7338036/CCAPI/cc_process_call_setup_i
  Event=0x4B6E95C8
*Jul 19 23:19:49.498: //-1/xxxxxxxxxxxx/CCAPI/cc_setupind_match_searc
  Try with the demoted called number 5124184000
*Jul 20 04:19:49.498: %CALL_CONTROL-6-APP_NOT_FOUND: Application ccm i
  dial-peer 1 not found. Handing called 23 to the alternate app.
*Jul 19 23:19:49.498 //23/C4CEC7338036/CCAPI/cc_set_release_source:
  Release Source=Internal Release-Call Control App
*Jul 19 23:19:498.498 //23/C4CEC7338036/CCAPI/ccCallDisconnect: Cause
  Value=63, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disconnect
  Cause=0)
```

An engineer is debugging inbound MVA calls that stopped working after a power outage. Initial inspection did not show any configuration changes. Based on the debug output, what is the cause of this issue?

An engineer is debugging inbound MVA calls that stopped working after a power outage. Initial inspection did not show any configuration changes. Based on the debug output, what is the cause of this issue?

A. Application ccm should be capitalized to fix this problem

B. The alternate call control application that is configured is releasing the call internally

C. Unified CM Publisher server is not reachable

D. Unified CM service parameter Enable Mobile Voice Access defaulted to false after the outage

Answer: B

Question #:403 - (Exam Topic 1)

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 arrived 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.

Answer: E

Explanation

SCCP: Configuring a Virtual SNR DN

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, 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.

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmesnr.html#

Question #:404 - (Exam Topic 1)

Which two statements about Cisco Unified Communications manager mixed mode cluster are true? (choose two)

A. Security-related settings other than device security mode, such as the SRST allowed check box, get ignored

B. The device security mode configures the security capability for your standalone server or a cluster

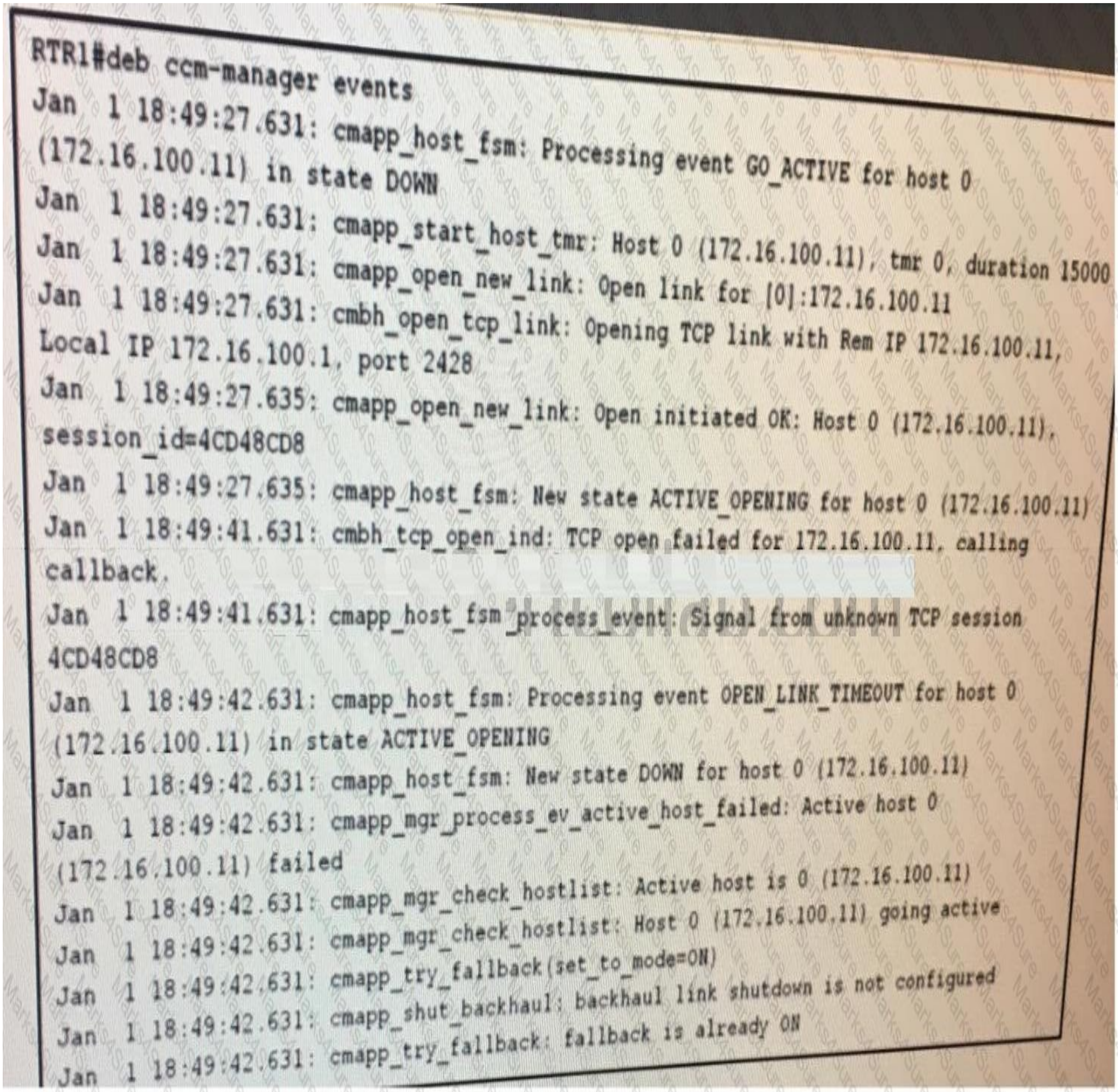
C. Auto-registration does not work when you configure mixed mode

D. The phone matches nonsecure connections with Cisco Unified Communications Manager even if the device security mode specifies authenticated or encrypted

Answer: A C

Question #:405 - (Exam Topic 1)

Refer to the exhibit.



```
RTR1#deb ccm-manager events
Jan  1 18:49:27.631: cmapp_host_fsm: Processing event GO_ACTIVE for host 0
(172.16.100.11) in state DOWN
Jan  1 18:49:27.631: cmapp_start_host_tmr: Host 0 (172.16.100.11), tmr 0, duration 15000
Jan  1 18:49:27.631: cmapp_open_new_link: Open link for [0]:172.16.100.11
Jan  1 18:49:27.631: cmbh_open_tcp_link: Opening TCP link with Rem IP 172.16.100.11,
Local IP 172.16.100.1, port 2428
Jan  1 18:49:27.635: cmapp_open_new_link: Open initiated OK: Host 0 (172.16.100.11),
session_id=4CD48CD8
Jan  1 18:49:27.635: cmapp_host_fsm: New state ACTIVE_OPENING for host 0 (172.16.100.11)
Jan  1 18:49:41.631: cmbh_tcp_open_ind: TCP open failed for 172.16.100.11, calling
callback.
Jan  1 18:49:41.631: cmapp_host_fsm_process_event: Signal from unknown TCP session
4CD48CD8
Jan  1 18:49:42.631: cmapp_host_fsm: Processing event OPEN_LINK_TIMEOUT for host 0
(172.16.100.11) in state ACTIVE_OPENING
Jan  1 18:49:42.631: cmapp_host_fsm: New state DOWN for host 0 (172.16.100.11)
Jan  1 18:49:42.631: cmapp_mgr_process_ev_active_host_failed: Active host 0
(172.16.100.11) failed
Jan  1 18:49:42.631: cmapp_mgr_check_hostlist: Active host is 0 (172.16.100.11)
Jan  1 18:49:42.631: cmapp_mgr_check_hostlist: Host 0 (172.16.100.11) going active
Jan  1 18:49:42.631: cmapp_try_fallback(set_to_mode=ON)
Jan  1 18:49:42.631: cmapp_shut_backhaul: backhaul link shutdown is not configured
Jan  1 18:49:42.631: cmapp_try_fallback: fallback is already ON
Jan  1 18:49:42.631: cmapp_try_
```

Which option correctly describes the output shown?

A. Fallback to the PRI worked successfully when the connection to CUCM was lost

B. The ccm switchback uptime-delay was incorrectly set to immediate instead of graceful

C. Redundant CUCM server will be tried after 15 seconds has passed

D. ccm-manager fallback-mgcp command was missing from the configuration

Answer: A

Question #:406 - (Exam Topic 1)

Refer to the exhibit.

```
voice service voip
  no ip address trusted authenticate
  allow-connections sip to sip
  sip
  no update-callerid
 !
 voice register global
  mode cme
  max-dn 10
  max-pool 10
 !
 voice register dn 1
  number 5000 name Phone A
 !
 voice register dn 2
  number 5001 name Phone B
 !
 voice register dn 3
  number 5002 name Phone C
 !
 voice ragister pool 1
  id mac 0000.1111.112A
  type 8841 number 1 dn 1
 !
 voice register pool 1
  id mac 0000.1111.112B
  type 8841 number 1 dn 2
 !
 voice register pool 1
  id mac 0000.1111.112C
  type 8841 number 1 dn 3
 !
```

An engineer is trying to provision an CUCME with three 8841 phones. However all phones fail to register. Which two changes in the configuration would allow the phones to register? (Choose two.)

A. The registrar server command must be added under the VOICE register global configuration.

B. The IP address trusted authenticate command must be added under voice service voip.

C. The source-address command must be added under the voice register global configuration.

D. The local SIP proxy address must be configuration under the sip-ua configuration.

E. The registrar server command must be added under the sip section of voice service voip.

Answer: C E

Question #:407 - (Exam Topic 1)

A Cisco Unified CM user is set up with one remote destination profile that has two remote destination numbers.

The first remote destination number is the user's mobile phone in country A and the second is a mobile phone located in country B All outbound calls are centralized from the gateways at country A. The user reports that inbound calls are property routed to the mobile phone as long as the user is in country A, but inbound calls are not routed successfully to the user's second remote destination number while the user is in country B . What two actions are required for calls to be successfully routed to country B? (Choose two.)?

A. The rerouting calling search space assigned to the user's remote destination profile must have access to international calls.

B. the Enable Mobile Connect option must be selected under the user's second remote destination number.

C. the Enable Mobile Voice Access option must be selected under the end user page.

D. The value of Remote Destination Limit should be changed to 2 instead of the default value of 4, under the end user page.

E. The value of Maximum Wait Time for Desk Pickup should be changed to 20000 instead of the default of 10000, under the end user page.

Answer: A B

Question #:408 - (Exam Topic 1)

Which two search scope option are removed from a directory handler when you check "Voice enabled" Check box? (Choose two)

A. Class of service

B. Partition

C. Search space

D. System Distribution list

E. Phone system

Answer: A D

Question #:409 - (Exam Topic 1)

Which procedure uses H.225 messages to exchange H.245 Master-Salve Determination information?

A. H.323 early media

B. H.245 terminal capability set

C. H 225 tunneling

D. H.245 tunneling

E. H.323 Fast Connect

Answer: D

Question #:410 - (Exam Topic 1)

The Video engineer wants to enable the LATM codec to allow video endpoint to communicate over audio with other IP devices. Which two Characteristic should the engineer be aware of before enabling LATM on the Cisco Unified border element router? (Choose two)

A. Dual tone Multifrequency interworking with LATM codec is not supported.

B. Codec transcoding between LATM and other codecs is not supported.

C. SIP UPDATE massage outlined in RFC3311 is not supported.

D. Box-to-Box High availability support feature is not supported.

E. Configure LATM under a voice class or dial peer is not supported.

F. Basic calls using flow-around or flow-through is not supported.

Answer: A B

Question #:411 - (Exam Topic 1)

Which four presence statuses are supported in Cisco IM & Presence federation with Microsoft Lync Server?

A. offline

B. busy

C. available

D. on the pone

E. away

F. do not disturb

G. in a meeting

Answer: A B C E

Question #:412 - (Exam Topic 1)

A call is made between two desk phones enabled with single number reach that are registered to a Cisco Unified CM cluster. The device pool for each device has a local route group defined. When the call is place to exit the system, which device pool controls the destination gateway? (Choose two.)

A. Source phone

B. Source RDP

C. Destination phone

D. Destination RDP

Answer: A

Question #:413 - (Exam Topic 1)

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.

Answer: C

Question #:414 - (Exam Topic 1)

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.

Answer: D

Question #:415 - (Exam Topic 1)

Refer to the Exhibit.

```
INVITE sip:951241822220172.16.100.90:5060 SIP/2.0
Via: SIP/2.0/TCP 172.16.100.50:5060:branch=z9hG4bK12af194b3e2624
From: "Agent A" <sip:51241830010172.16.100.50>:tag=1001843-825a8e37-305d-403f-9a76-ee5343b3b431-61539586
To: <sip:951241822220172.16.100.90>
Date: Tue. 10 Mar 2015 14:25:03 GMT
Call-ID: 3c748f00-4felfebf-35ff9-2bef12ac0172.16.100.50
Supported: timer.resource-priority.replaces
Min-SE: 1800
User-Agent: Cisco-CUCM9.1
Allow: INVITE. OPTIONS. INFO. BYE. CANCEL. ACK. PRACK. UPDATE. REFER. SUBSCRIBE. NOTIFY
Cseq: 101 INVITE
Expires: 100
Allow-Events: presence
Supported: X-cisco-srtp-fallback
Cisco-Guid: 1014271744-0000065536-0000221138-0737088172
Session-Expires: 1800
P-Asserted-Identity: "Agent A" <sip:51241830010172.16.100.50>
Remote-Party-ID: "Agent A" <sip:51241830010172.16.100.50>:party=calling:screen=yes:privacy=off
Contact: "Agent A" <sip:51241830010172.16.100.50.5060:transport=tcp>:isFocus
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 202

v=0
o=CiscoSystemsCCM-SIP 2371 1 IN IP4 172.16.100.50
s=SIP Call
c=IN IP4 172.16.100.50
t=0 0
m=audio 24582 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

An agent initiated a video call but was establish as audio only. The support engineer collected and analyzed the Cisco Unified CM traces. Which two options caused this problem? (Choose two)

A. A hardware MTP was assigned to the call.

B. SIP Notify DTMF was requested and negotiated.

C. MTP Required was checked on the SIP Trunks.

D. Use Trusted Relay Point is set on one of the phones.

E. MRGL assigned to phones with Trusted Relay Point.

Answer: D E

Question #:416 - (Exam Topic 1)

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

Answer: D

Question #:417 - (Exam Topic 1)

A Cisco Unified CM engineer configured a phone VPN for remote users but the users cannot register the phones to the VPN.

```
909: NOT 20:59:50.051721 VPNC: do_login: got login response
910: NOT 20:59:50:052581 VPNC: process_login: HTTP/1.0 302 Temporary moved
911: NOT 20:59:50.053221 VPNC: process_login: login code: 302 (redirected)
912: NOT 20:59:50.053823 VPNC: process_login: redirection indicated
913: NOT 20:59:50.054441 VPNC: process_login: new 'Location':
   /+webvpn+/index.html
914: NOT 20:59:50.055141 VPNC: set_redirect_url: new URL
   <https://xyz1.abc.com:443/+webvpn+/index.html>
```

Which configuration changes fix this problem?

A. Configure enable outside in the webVPN configuration on the Cisco ASA

B. Configure the split-tunnel-policy tunnel all attribute on the Cisco ASA

C. Configuration the ssl trust-point SSL outside on the Cisco ASA

D. Remove the Cisco ASA IP address from the VPN load-balancing configuration

Answer: D

Question #:418 - (Exam Topic 1)

O: 103

Assume that your customer domain is customer.com. Which external DNS SRV record is needed to facilitate service discovery so that the Cisco Jabber client can automatically find the appropriate servers to connect to while off-net?

A. _cisco-uds_tcp customer.com

B. _cuplogin_tcp customer.com

C. _collab-edge_tls customer.com

D. _cisco-uds_tls customer.com

E. _cuplogin_tcp customer.com

F. _collab-edge_tcp customer.com

Answer: A

Question #:419 - (Exam Topic 1)

Assume that a Cisco Unified Communication Manager IM & Presence environment is properly configured with database. How long is the archived data maintained?

A. 12 months

B. 36 months

C. 24 months

D. 60 months

E. The maintenance period is dictated by the data purging policy of the external database administrator

Answer: E

Question #:420 - (Exam Topic 1)

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

Answer: C

Question #:421 - (Exam Topic 1)

Which option is the main advantage of using Cisco UCS Director within a data centre environment?

A. it uses auto discovery to detect, manage, and provision system components.

B. It uses automated workflows for resource provisioning

C. It Supports, service chaining for physical and virtual services.

D. It supports inventory and fault management across multiple Cisco devices

Answer: A

Question #:422 - (Exam Topic 1)

A company is deploying new Cisco Unified Communications infrastructure. The CTO has decided that high

availability for voicemail system is mandatory. The enterprise has primary and backup data centre that is geographically dispersed. Cisco Unified Communications infrastructure support for Unity connection applications? (Choose two)

A. Transcoder

B. Disk space

C. Latency

D. Bandwidth

E. Memory

Answer: C D

Question #:423 - (Exam Topic 1)

Which two descriptions of TFTP option 66 and 150 for Cisco IP phones are true? (choose two)

A. Option 150 and 60 are standard based

B. Option 150 supports only TFTP server IP addresses, but option 66 supports only TFTP server DNS names

C. Option 150 supports multiple entries of TFTP server IP addresses, but option 66 supports only one entry of either TFTP server or DNS names

D. Option 150 is Cisco proprietary, but option 66 is standard based

E. Option 66 is Cisco proprietary, but 150 is standard based

F. Option 150 supports multiple entries of TFTP server IP addresses, but option 66 supports only one entry of TFTP server DNS names

Answer: C D

Question #:424 - (Exam Topic 1)

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

Answer: D

Question #:425 - (Exam Topic 1)

Exhibit:

```
!
class-map match-any signal
 match ip dscp cs3
class-map match-any rtp
 match ip dscp ef
!
policy-map VoIP
 class rtp
   bandwidth percent 33
   compress header ip rtp
 class signal
   bandwidth percent 5
 class class-default
   fair-queue
!
interface serial 0/1/0
 service-policy output VoIP
!
```

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

Answer: E

Question #:426 - (Exam Topic 1)

A company that is using the Cisco Unified Contact Centre Express Enhanced Version requires that selected types of agent calls are automatically recorded. Which call required operation can be used to satisfy this requirement?

A. Instruct supervisors use the Record button on Cisco Agent Desktop to trigger recording.

B. Configure the Cisco Agent Desktop workflow to trigger recording.

C. Recording is not supported on the Cisco Unified CCX Enhanced Version. It is supported only on the premium version

D. Instruct agents to use the Record button on Cisco IPPA to trigger recording

E. Instruct supervisors to use the Record button on Cisco Supervisor Desktop to trigger recording

Answer: B

Question #:427 - (Exam Topic 1)

A network engineer is configuring iLBC on SIP dial peer and is asked to comply with this parameters13.33 KBPS bit rate payload size four times the default value

A. Router(config-dial-peer)#Codec ilbc mode 20 bytes 114

B. Router(config-dial-peer)#Codec ilbc mode 30 bytes 200

C. Router(config-dial-peer)#Codec ilbc mode 20 bytes 228

D. Router(config-dial-peer)#Codec ilbc mode 30 bytes 100

Answer: B

Question #:428 - (Exam Topic 1)

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

B. SEP111122223333.cnf.xml.enc.sgn

C. SEP111122223333.cnf.xml.sgn.enc

D. SEPDefault.cnf.xml.enc.sgn

E. SEP111122223333.xml.enc

Answer: B

Question #:429 - (Exam Topic 1)

Refer to the exhibit.

```
%Q.931 is backhauled to CCM MANAGER 0x0003 on DSL 0. Layer 3 output may not apply

ISDN Serial0/0/0:23 interface
        dsl 0, interface ISDN Switchtype = primary-ni
        L2 Protocol = Q.921 0x0000  L3 Protocol(s) = CCM MANAGER 0x0003
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Active dsl 0 CCBs = 0
    The Free Channel Mask:  0x807FFEEF
    Number of L2 Discards = 0, L2 Session ID = 117
ISDN Serial0/2/1:23 interface
        dsl 1, interface ISDN Switchtype = primary-ni
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Active dsl 1 CCBs = 0
    The Free Channel Mask:  0x00000000
    Number of L2 Discards = 0, L2 Session ID = 0
Total Allocated ISDN CCBs = 0
```

Which two statements about the show command output are true? (Choose two)

A. T1 0/2/1 terminates Q.921 signalling to a Cisco Unified Communications Manager server

B. T1 0/2/1 terminates Q.931 signalling on the gateway

C. T1 0/0/0 terminates Q.931 signalling to a Cisco Unified Communications Manager server

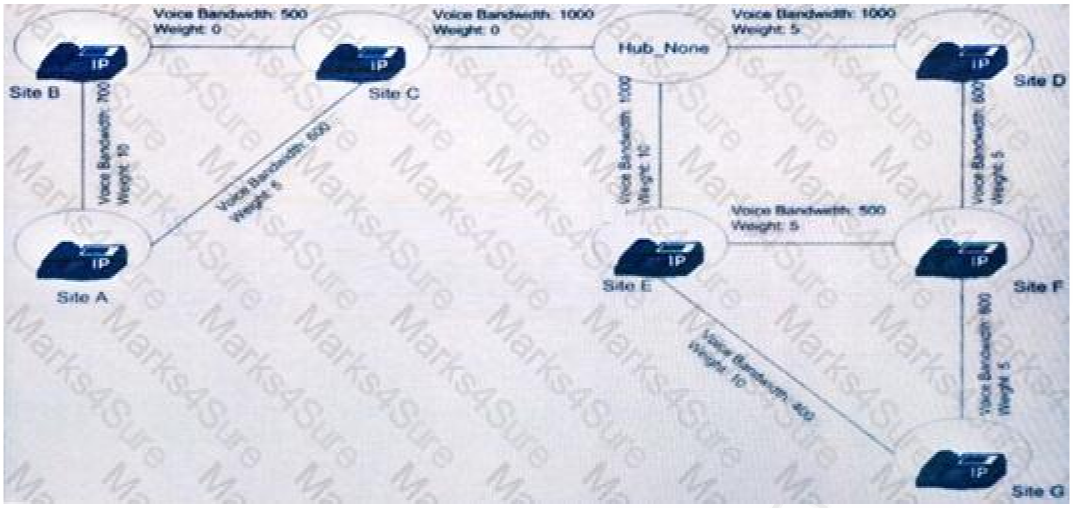
D. T1 0/0/0 terminates SIP signalling to a Cisco Unified Communications Manager server

E. T1 0/0/0 terminates Q.921 signalling on the gateway

Answer: C E

Question #:430 - (Exam Topic 1)

Refer to the exhibit.



An engineer mapped the enhance location call admission control configuration to match the physical links and bandwidth allowances. Assuming no other calls are consuming any bandwidth, how many G722 calls are allowed between site A and site G?

A. 5

B. 7

C. 12

D. 25

E. 37

Answer: B

Question #:431 - (Exam Topic 1)

A jabber for window user is on a call with Cisco Telepresence EX90 endpoint at the same location.

During the call, the video on the jabber for Windows application was quality but the video on the EX90 was choppy and slow. When the administration checked the service rate on the EX90, it showed 2048 KBps. Which two configuration changes can fix this problem?

A. Lower the bit rate in the region configuration in communication manager between the endpoints.

B. Increase the location bandwidth for immersive video between the endpoints.

C. Enable BFCP in the SIP profile for the jabber client.

D. Enable H.263 on the EX90.

E. Replace the camera for the jabber user with the precision HD USB camera.

F. Increase the bandwidth between the jabber video client and the EX90.

Answer: B

Question #:432 - (Exam Topic 1)

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".

Answer: A B

Question #:433 - (Exam Topic 1)

According to RFC 3261, which SIP response is not final?

A. 2xx

B. 1xx

C. 3xx

D. 5xx

E. 6xx

F. 4xx

Answer: B

Question #:434 - (Exam Topic 1)

Which factor is most important when using heartbeats to maintain the high availability status of an application

A. bandwidth

B. routing

C. latency

D. round-trip time

Answer: C

Question #:435 - (Exam Topic 1)

Refer to the exhibit.

```
application
service CCM http://172.16.100.50:8080/ccmivr/pages/IVRMainpage.vxml

dial-peer voice 4100 pots
service ccm
incoming called-number 4100


dial-peer voice 4101 voip
destination-pattern 4100
codec g729r8
dtmf-relay rtp-nte
session target ipv4:172.16.101.50
```

A collaboration engineer configured MVA for a company using an existing Cisco IOS voice gateway.

When testing inbound calls it is found that they are all failing.

Which two sets of configuration changes fix this problem? (Choose two)

A. Dial-peer voice 4100 pots

Services ccm

Destination-pattern 4100$

B. Dial-peer voice 4101 voip

Dtmf-relay h245-alphanumeric

Session target ipv4:172.16.100.50

C. Dial-peer voice 4101 voip

Dtmf-relay h245-alphanumeric

Session target ipv4:172.16.100.50

Codec g711ulaw

D. Dial-peer voice 4100 pots

service CCM

E. Dial-peer voice 4100 pots

service CCM

Incoming called-number .T

F. Dial-peer voice 4101 VOIP

Dtmf-relay h245-signal

Session target ipv4:172.16.100.50

Codec g711alaw

Answer: C E

Explanation

Please note that, current scenario we are talking only inbound calls.Tricky question

http://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmana

Configuration Example

application

service

mva http://10.106.103.149:8080/ccmivr/pages/IVRMainpage.vxml

dial-peer voice 5050 pots

service mva

incoming called-number 5050

no digit-strip

direct-inward-dial

!

dial-peer voice 3001 voip

destination-pattern 5050

session target ipv4:10.106.103.149

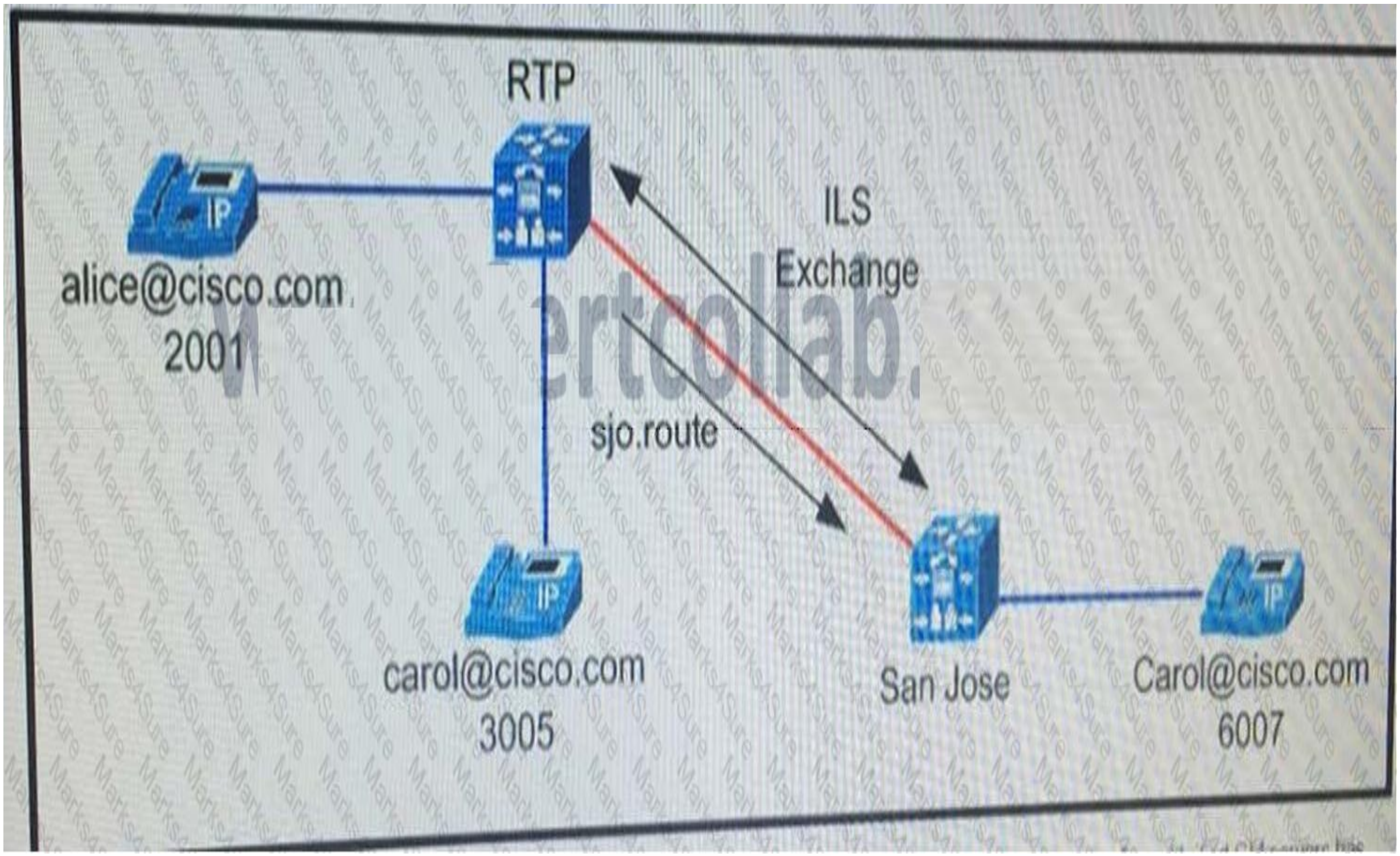
dtmf-relay h245-alphanumeric

codec g711ulaw

no vad

Question #:436 - (Exam Topic 1)

Refer to the exhibit.



Which three events happen When Alice calls carol@cisco.com and the URI lookup policy on the Cisco Unified CM server has been set to case insensitive? (Choose three)

A. The RTP server routes the call to carol@cisco.com because remote URIs have priority

B. The RTP sever looks up to see if carol@cisco.com is associated to a local number

C. The San Jose server calls carol@cisco.com upon receiving the invite request

D. The San Jose server provide carol's directory URI using ILS exchange

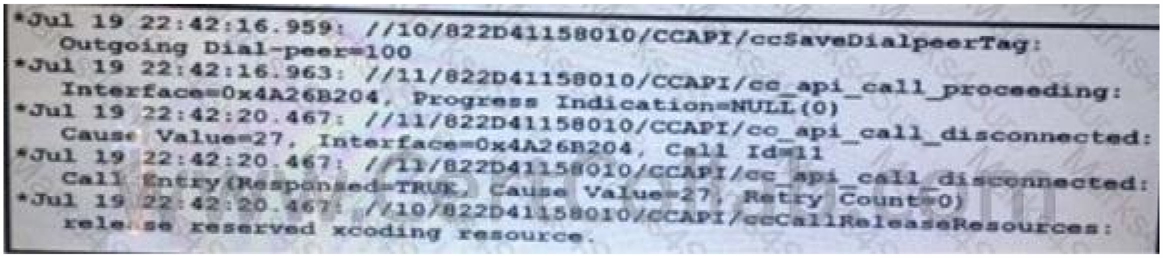
E. The RTP server sends the call to carol@cisco.com because it has priority

F. The RTP server drops the call because it has two identical matches

Answer: B D E

Question #:437 - (Exam Topic 1)

Refer to the exhibit.



```
*Jul 19 22:42:16.959: //10/822D41158010/CCAPI/ccSaveDialpeerTag:
   Outgoing Dial-peer=100
*Jul 19 22:42:16.963: //11/822D41158010/CCAPI/cc_api_call_proceeding:
   Interface=0x4A26B204, Progress Indication=NULL(0)
*Jul 19 22:42:20.467: //11/822D41158010/CCAPI/cc_api_call_disconnected:
   Cause Value=27, Interface=0x4A26B204, Call Id=11
*Jul 19 22:42:20.467: //11/822D41158010/CCAPI/cc_api_call_disconnected:
   Call Entry(Responsed=TRUE, Cause Value=27, Retry Count=0)
*Jul 19 22:42:20.467: //10/822D41158010/CCAPI/ccCallReleaseResources:
   release reserved xcoding resource.
```

A network engineer is troubleshooting failed calls a CUBE to five-server CUCM cluster.

Which option describes the root cause?

A. A request was sent to CUCM server that does not belong to Cisco Unified CM group

B. Progress indication was not received

C. A second dial peer for load balancing and redundancy was not available

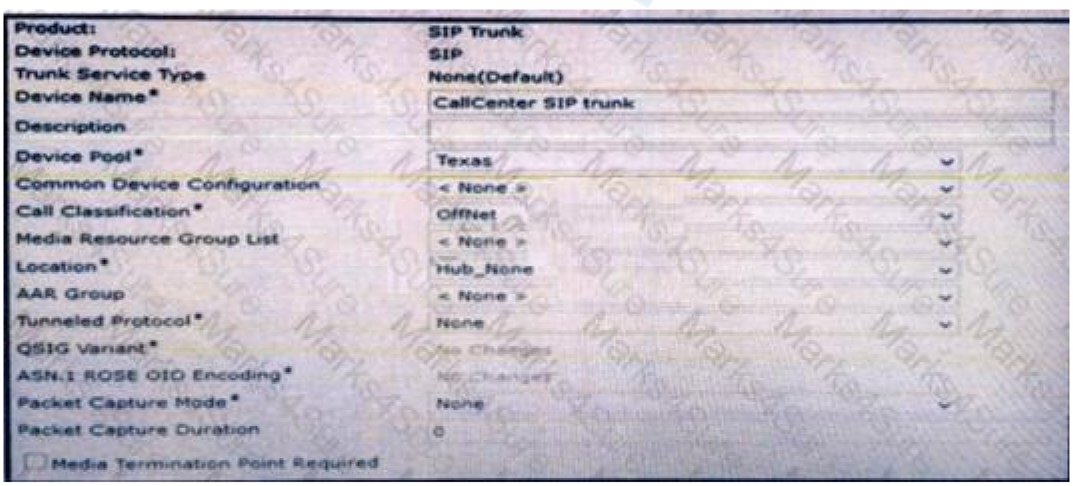
D. Run on all active nodes option was not checked

E. Transcoding resource was not allocated

Answer: C

Question #:438 - (Exam Topic 1)

Refer to the exhibit.



Cisco Unified CM users report that they hear dead air during call transfer but bi-directional audio resumes after the transferees answer the call. The transferees are located across a SIP trunk. A collaboration engineer is checking the sip trunk configuration on the Cisco Unified CM.

Which two configuration changes fix this problem? (Choose two)

A. Assign a Media Resource Group List to the SIP Trunk.

B. Place a check mark on Media Termination Point Required.

C. Make sure there is an Annunciator Resource available on the MRGL.

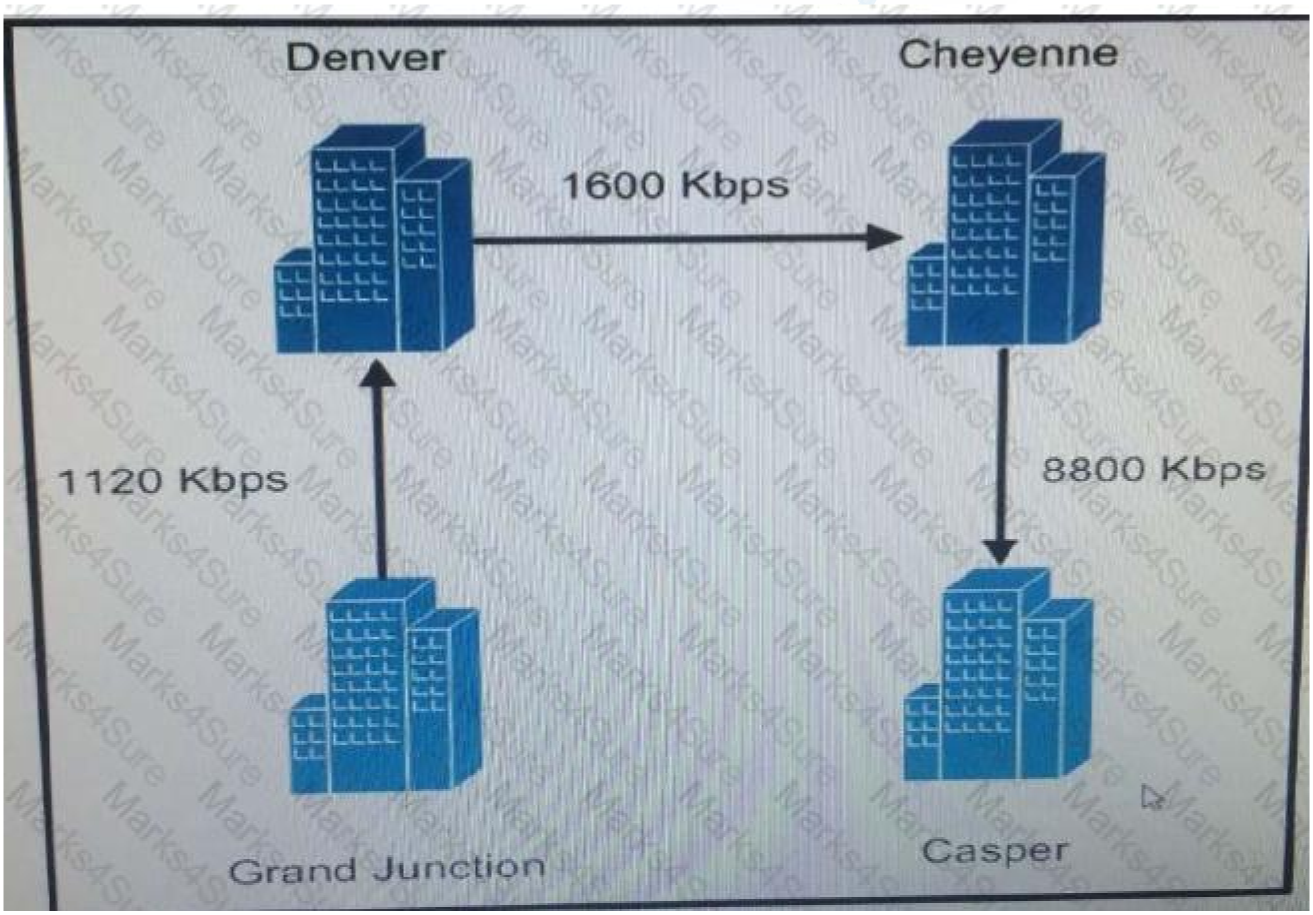
D. Modify the call Classification on the SIP trunk to OnNet.

E. Change the “Send H225 User Info” service parameter to “Use ANN for Ringback”.

Answer: A B

Question #:439 - (Exam Topic 1)

Refer to the exhibit.



A collaboration engineer configures Cisco Unified CM location using G.711 and iLBC for each site. The bandwidth for each link is shown. Which two options represent the maximum concurrent number of calls supported from Grand Junction to Casper for each Codec? (Choose two)

A. 20 G.711 calls

B. 18 G.711 calls

C. 36 iLBC calls

D. 42 iLBC calls

E. 11 G.711 calls

F. 51 iLBC calls

Answer: C E

Question #:440 - (Exam Topic 1)

Which two statements describe specs-based virtualization support model for third party servers running Cisco UC applicationns?(Choose two)

A. VMware vSphere is mandatory

B. Generic server hardware must be validated by Cisco

C. VMware vCenter is mandatory

D. Cisco TAC supports the server when all spec-based rules are followed

E. Only direct-attached storage is allowed

F. CIsco UC virtualization Foundation software is mandatory

Answer: A C

Question #:441 - (Exam Topic 1)

Refer to the exhibit.

```
Received:
INVITE sip:+15552341234@10.41.11.21:5060 SIP/2.0
Via: SIP/2.0/TCP 10.41.11.10:5060;branch=z9hG4bK4bb12565f6120d
Via: SIP/2.0/TCP 10.10.20.20:5060;branch=z9hG4bK2d4790;received=192.0.2.111
Via: SIP/2.0/TCP 10.50.40.30:5060;branch=z9hG4bK2d6014;received=192.0.2.20
From: "Test" <sip:5559874321@customer.com>;tag=15269661~58798a82-6f11-4d0c-9024-092b2761f953-71117600
To: <sip:+15552341234@10.41.11.21>
Cisco-Guid: 2960899712-0000065536-0000112086-0202057994
P-Asserted-Identity: "Test" <sip:5559874321@customer.com>
Remote-Party-ID: "Test" <sip:5559874321@customer.com>;party=calling;screen=yes;privacy=off
Contact: <sip:5559874321@10.41.11.10:5060;transport=tcp>
```

```
voice translation-rule 1
 rule 1 /\+/ //
!
voice translation-rule 2
 rule 1 /\+1/ //
!
voice translation-rule 3
 rule 1 /\+1/ /91/
!
voice translation-profile 1
 translate called 1

voice translation-profile 2
 translate called 2

voice translation-profile 3
 translate called 3
```

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 3

dial-peer voice 2 voip

incoming uri via 2

translation-profile incoming 2

dial-peer voice 3 voip

incoming called-number [2-9]..[2-9]……

translation-profile incoming 1

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 1

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

host dns:customer.com

dial-peer voice 1 voip

incoming uri via 1

translation-profile incoming 2

dial-peer voice 2 voip

incoming uri via 2

translation-profile incoming 1

dial-peer voice 3 voip

incoming uri via 3

incoming called-number +1T

translation-profile incoming 1

F. 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

host dns:customer.com

dial-peer voice 1 voip

incoming uri via 1

translation-profile outgoing 3

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 2

Answer: B

Question #:442 - (Exam Topic 1)

Refer to the exhibit.

```
v=0
o=BroadWorks 12628880 1 IN IP4 172.31.200.196
s=-
c=IN IP4 172.31.200.196
t=0 0
a=sqn: 0
a=cdsc: 1 image udptl t38
a=cpar: a=T38FaxVersion:0
a=cpar: a=T38FaxUdpEC:t38UDPRedundancy
m=audio 10086 RTP/AVP 18 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=fmtp:18 annexb=no
```

Which three facts can be determined about the audio parameters of this call from this session description protocol? (Choose Three)

A. The DTMF relay will be RFC2833

B. The codec will be G711

C. The codec will be G729

D. VAD will be disabled for this call

E. VAD will be enabled for this call

F. The call will be a T38 fax call

Answer: A C D

Question #:443 - (Exam Topic 1)

Refer to the exhibit.

```
Router(config-dspfarm-profile)#codec h264 ?
       4cif      Allowed Resolutions: 4CIF and 4SIF
       720p      Allowed Resolution: 720p
       cif       Allowed Resolutions: CIF and SIF
       qcif      Allowed Resolutions: QCIF and QSIF
       vga       Allowed Resolution: VGA
       w360p     Allowed Resolution: w360p
       w448p     Allowed Resolution: w448p
```

Which Cisco IOS multipoint video conferencing profile is being configured?

A. guaranteed-video

B. guaranteed-audio

C. rendezvous

D. heterogeneous

E. guaranteed-video

Answer: D

Topic 2, Cisco Collaboration Infrastructure

Question #:444 - (Exam Topic 2)

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

Answer: E

Explanation

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 #:445 - (Exam Topic 2)

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 spec-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 spec-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 spec-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.

Answer: E

Explanation

VMware vCenter is

- **optional** when deploying on UConUCSTestedReferenceConfigurationhardware
- **mandatory** when deploying on UConUCSSpecs-basedandThird-partyServerSpecs-basedhardware.
- vCenter Statistics Level 4 logging is mandatory so that Cisco TAC is able to provide effective support.
- Clickhere 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 enablement 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).

Reference: http://docwiki.cisco.com/wiki/Unified_Communications_VMware_Requirements

Question #:446 - (Exam Topic 2)

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

Answer: C D

Explanation

Cisco EnergyWise Neighbor Discovery Process

The Cisco EnergyWise neighbor discovery process is the mechanism by which domain members discover each other and populate their Cisco EnergyWise neighbor tables. Cisco EnergyWise queries can subsequently be distributed to all domain members using the neighbor relationships to monitor and control the power usage of devices within a domain. Cisco EnergyWise domain members automatically discover their neighbors through one of two mechanisms:

- Cisco EnergyWise UDP broadcast packet
- 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.

Reference:
http://www.cisco.com/en/US/docs/solutions/Enterprise/Borderless_Networks/Energy_Management/energywisec
ml?referring_site=smartnavRD#wp555927

Question #:447 - (Exam Topic 2)

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.

Answer: B C

Reference:
http://docwiki.cisco.com/wiki/UC_Virtualization_Supported_Hardware#UC_on_UCS_Tested_Reference_Confi

Question #:448 - (Exam Topic 2)

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

Answer: A

Explanation

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 CNA / NIC / HBA) is design-dependent and not included in TRC definition.
- 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/FC/FCoE 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 2100/2200, 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 #:449 - (Exam Topic 2)

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

Answer: D

Explanation

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. clickhere).
  - ClickherefordetailedCiscoUCSserverdocumentation 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. ClickherefordetailedCiscoUCSserverdocumentation.
- 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/FC/FCoE 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 2100/2200, 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.

Reference:
http://docwiki.cisco.com/wiki/UC_Virtualization_Supported_Hardware#UC_on_UCS_Tested_Reference_Confi ations

Question #:450 - (Exam Topic 2)

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

Answer: B C

Explanation

The two file types that we get are Announcement Audio Files and Ringer Files.

Question #:451 - (Exam Topic 2)

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.

Answer: C F

Explanation

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 1. Summary of Distribution Models

Peer Firmware Sharing

Load Server

Traditional TFTP

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

• Proven 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 file copy to load server**

• **No failback to TFTP on error**

• More prone to user error

• High-bandwidth requirements

• Multiple requests for same file

• High CPU usage on TFTP server

Reference: http://www.cisco.com/en/US/prod/collateral/voicesw/ps6882/ps6884/white_paper_c11-583891.pdf

Question #:452 - (Exam Topic 2)

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.

Answer: D

Explanation

Voice traffic cannot recapture lost packets. Rather than retransmitting a lost network connection, the phone resets and attempts to reconnect its network connection.

Reference:
http://www.cisco.com/en/US/docs/voice_ip_comm/cuipph/6921_6941_6961/7_1_2/english/admin/guide/6921trl

Question #:453 - (Exam Topic 2)

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.

Answer: D

Explanation

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.

Reference:http://angryciscoguy.com/jello/cisco-answer-file-generator-to-the-rescue/

Question #:454 - (Exam Topic 2)

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

Answer: A D E F

Explanation

Four attributes that are needed to determine the time to complete TFTP file transfer process is:

- File Size
- Round-trip time
- Packet loss percentage
- Response timeout

Reference: http://www.cisco.com/en/US/prod/collateral/voicesw/ps6882/ps6884/white_paper_c11-583891_ps10451_Products_White_Paper.html

Question #:455 - (Exam Topic 2)

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

Answer: D

Explanation

Cisco Prime LMS 4.1 uses TCP port 43440.

Question #:456 - (Exam Topic 2)

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

Answer: A C

Reference:
http://www.cisco.com/en/US/docs/switches/lan/energywise/phase2_5/ios/configuration/guide/one_ent.html

Question #:457 - (Exam Topic 2)

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

Answer: A

Explanation

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 #:458 - (Exam Topic 2)

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.

Answer: C F

Explanation

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 software 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 #:459 - (Exam Topic 2)

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

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.

Question #:460 - (Exam Topic 2)

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.

Answer: B

Reference: http://www.cisco.com/en/US/docs/solutions/Enterprise/Border less_Networks/Energy_Management/energywisedg.html?referring_site=smartnavRD#wp554384

Question #:461 - (Exam Topic 2)

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

Answer: C E

Explanation

We get option for Identity Trust list Files and Mobile Voice Access audio files.

Topic 3, Telephony Standards and Protocols

Question #:462 - (Exam Topic 3)

Which two types of devices on Cisco Unified Communications Manager support iSAC? (Choose two.)

A. MGCP

B. SIP

C. SCCP

D. Music on Hold server

E. H.323

Answer: B C

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 Annunciator does not support iSAC. MGCP devices are not supported.

Question #:463 - (Exam Topic 3)

To which SIP response class do the SIP response codes 300 to 399 belong?

A. Provisional

B. Client Failure

C. Server Failure

D. Successful

E. Redirection

Answer: E

Explanation

Redirection — further action needs to be taken in order to complete the request. That is what this class implies.

Question #:464 - (Exam Topic 3)

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

Answer: C

Explanation

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 #:465 - (Exam Topic 3)

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

Answer: C

Explanation

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 #:466 - (Exam Topic 3)

Refer to the exhibit.

```
Call-Info: <sip: address>; method="NOTIFY;Event=telephone-event;Duration=msec"
```

Which DTMF relay method is advertised when the originating SIP gateway sends an INVITE message with a Call-Info header shown?

A. RFC 2833

B. SIP INFO

C. SIP NOTIFY

D. SIP KPML

E. In-band audio

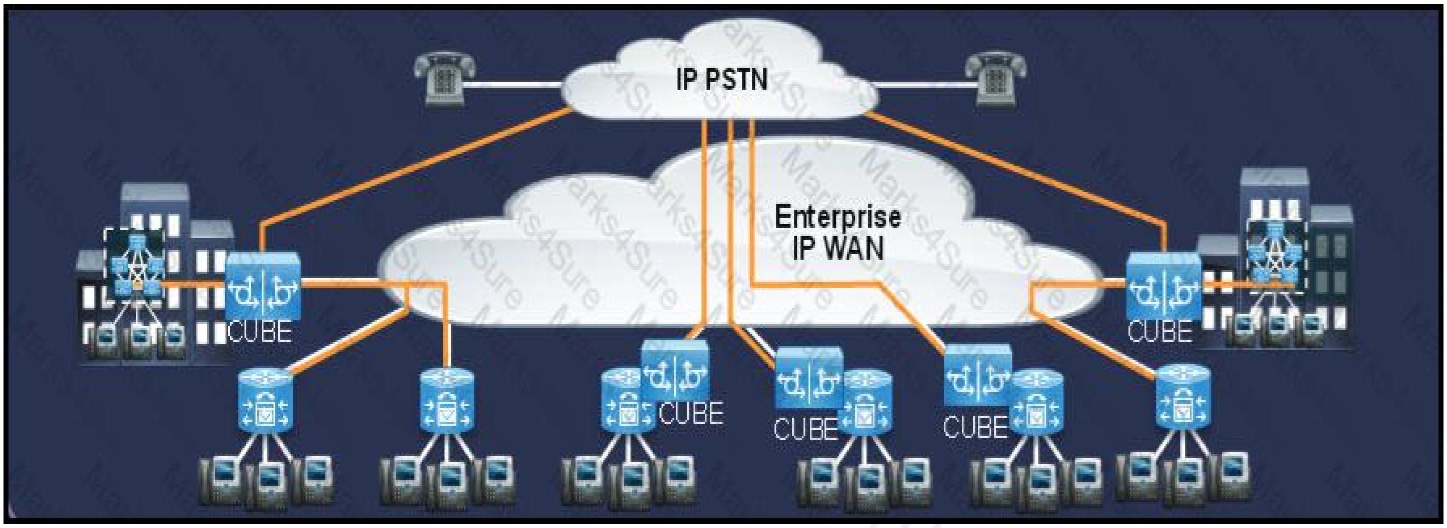
Answer: C

Explanation

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 #:467 - (Exam Topic 3)

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

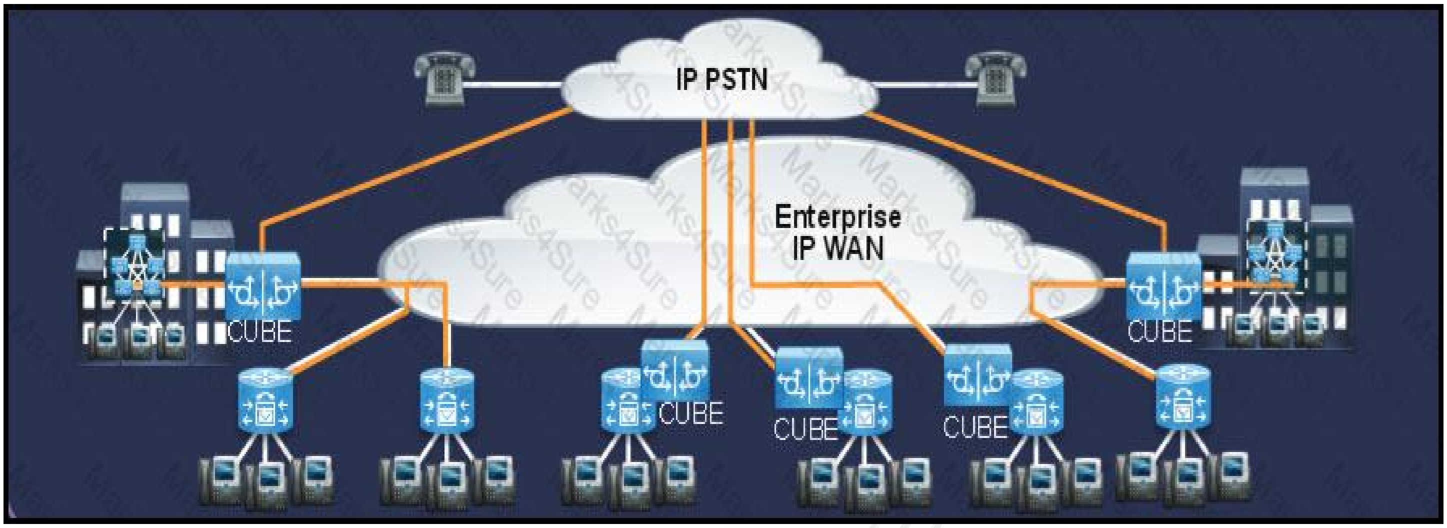
E. distributed

Answer: C

Explanation

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



Reference:

http://www.cisco.com/c/dam/en/us/products/collateral/unified-communications/unified-border-element/cis_4583

Question #:468 - (Exam Topic 3)

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

Answer: D

Explanation

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.

Question #:469 - (Exam Topic 3)

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.

Answer: B

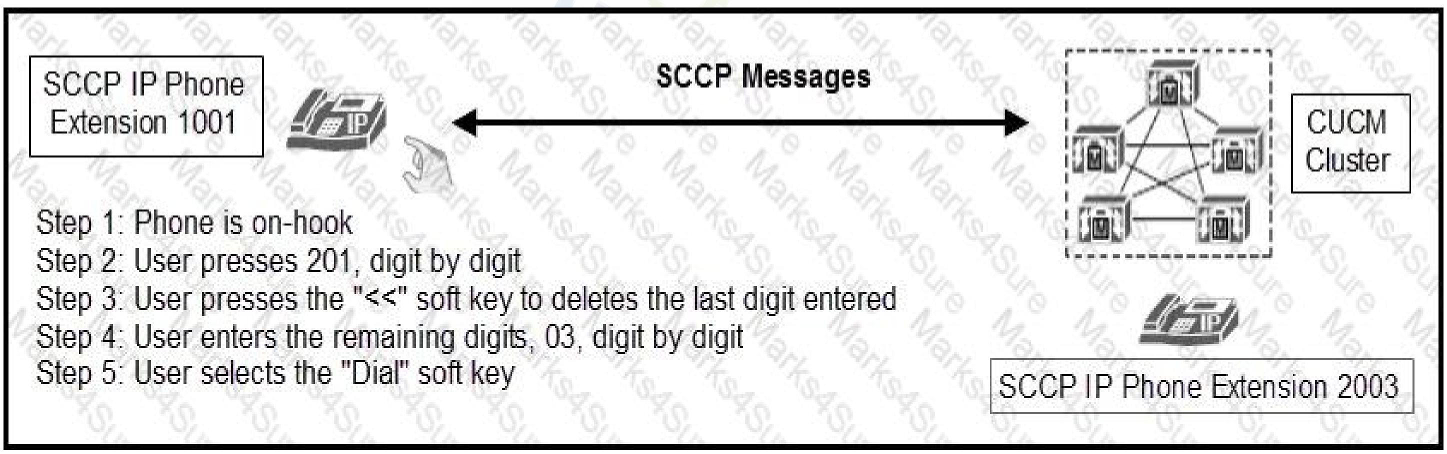
Explanation

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 #:470 - (Exam Topic 3)

Refer to the exhibit.



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.

Answer: C

Explanation

After the user delete phone stop the digit by digit dialing and send it as a whole setup.

Question #:471 - (Exam Topic 3)

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.

Answer: B

Explanation

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 #:472 - (Exam Topic 3)

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

Answer: A

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 #:473 - (Exam Topic 3)

Refer to the exhibit.

```
Jul 31 18:36:33.201: MGCP Packet sent to 10.1.1.1:2427--->
250 125 OK
P: PS=76, OS=12160, PR=75, OR=12000, PL=0, JI=7, LA=0
```

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.

Answer: D

Explanation

See MGCP packet debugging examples and their meanings at the Reference link below.

Reference: Sample of Debug MGCP Packets

http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-ca

Question #:474 - (Exam Topic 3)

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.

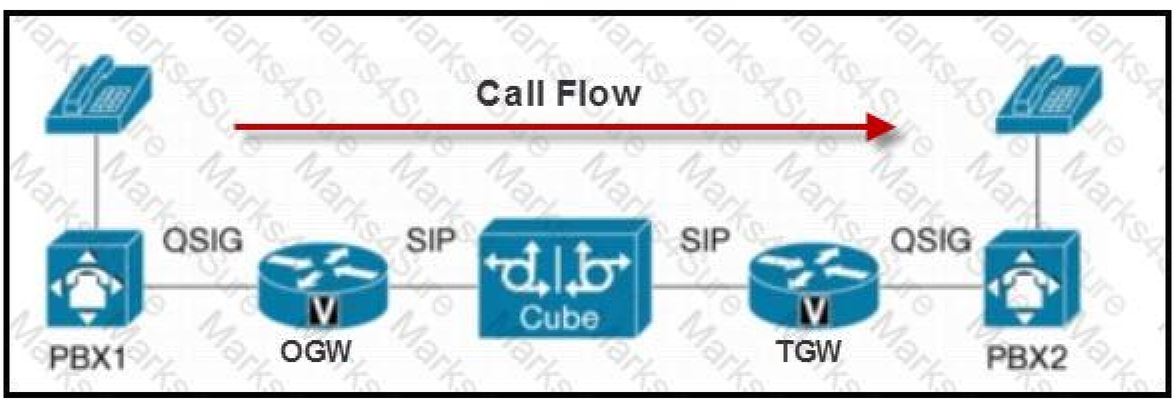
Answer: E

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 #:475 - (Exam Topic 3)

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

Answer: D

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.

Reference:
http://www.cisco.com/c/en/us/solutions/collateral/enterprise-networks/empowered-branch-solution/white_paper_c11_459092.html

Question #:476 - (Exam Topic 3)

Refer to the exhibit.

```
Jul 31 17:51:25.676: MGCP Packet sent to 10.1.1.1:2427--->
200 96
I:
X: 0
L: p:10-20, a:PCMU;PCMA;G.nX64, b:64, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:10-220, a:G.729;G.729a;G.729b, b:8, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:10-110, a:G.726-16;G.728, b:16, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:10-70, a:G.726-24, b:24, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:10-50, a:G.726-32, b:32, e:on, gc:1, s:on, t:10, r:g, nt:IN;
   ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:30-270, a:G.723.1-H;G.723;G.723.1a-H, b:6, e:on, gc:1, s:on, t:10,
   r:g, nt:IN;ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
L: p:30-330, a:G.723.1-L;G.723.1a-L, b:5, e:on, gc:1, s:on, t:10,
   r:g, nt:IN;ATM;LOCAL, v:T;G;D;L;H;R;ATM;SST;FXR;PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, conttest, data, netwloop,
   netwtest
```

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.

Answer: C

Explanation

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 #:477 - (Exam Topic 3)

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

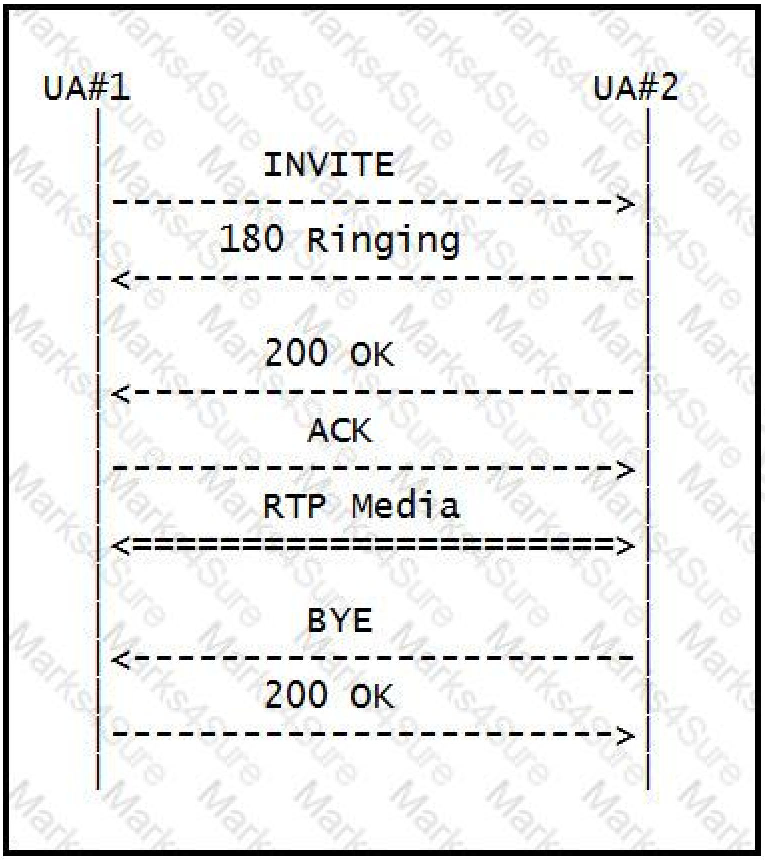
Answer: C

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.dataTypeandreverseLogicalChannelParameters.dataType components, respectively. The semantics of temporalSpatialTradeOffCapability 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 #:478 - (Exam Topic 3)

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

Answer: A

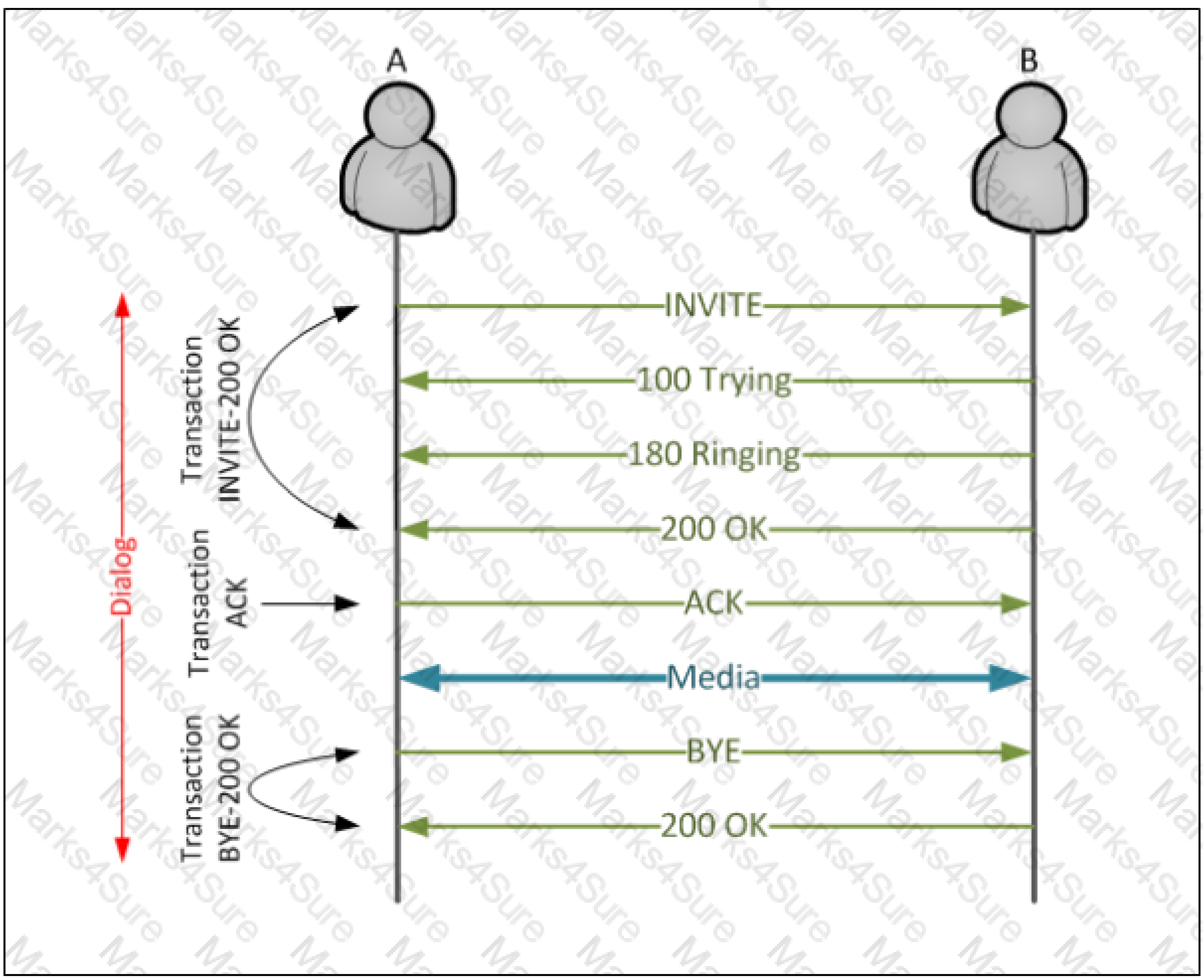
Explanation

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 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/

Question #:479 - (Exam Topic 3)

Which RAS message is used between two gatekeepers to resolve an alias address?

A. GRQ

B. ARQ

C. IRQ

D. LRQ

E. RRQ

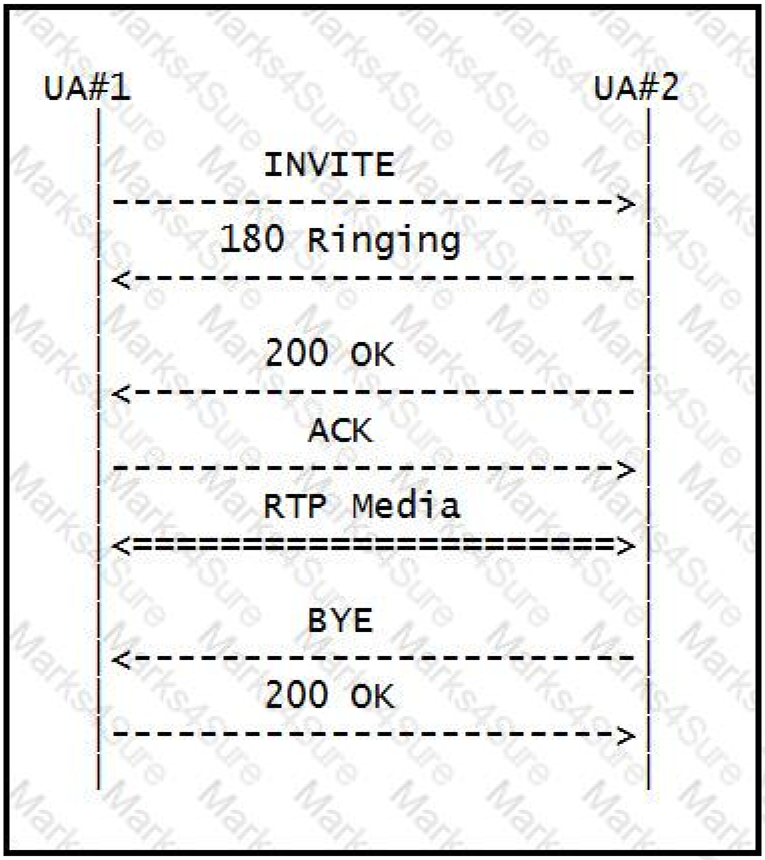
Answer: D

Explanation

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 #:480 - (Exam Topic 3)

Refer to the exhibit.



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

Answer: C

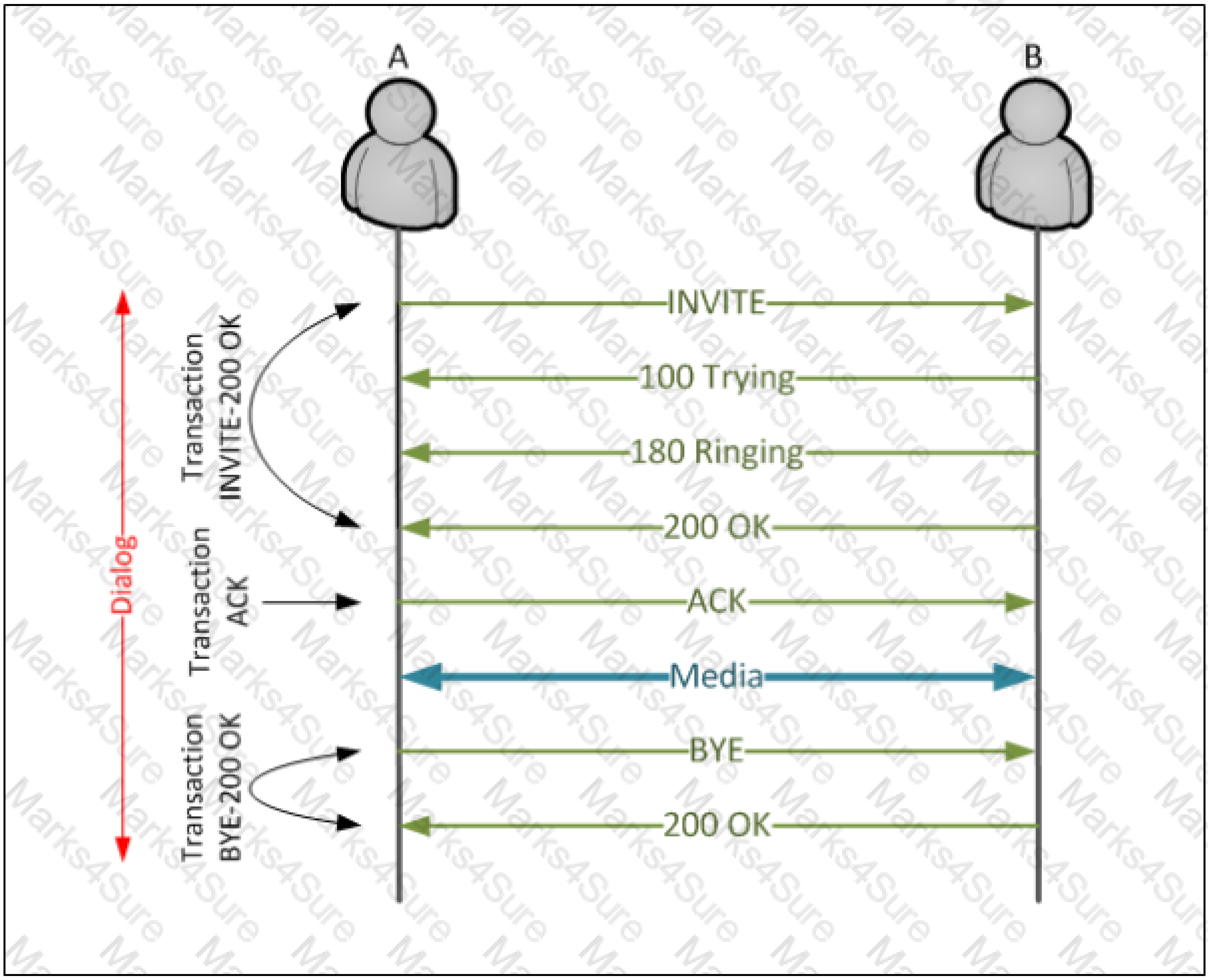
Explanation

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 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/

Question #:481 - (Exam Topic 3)

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

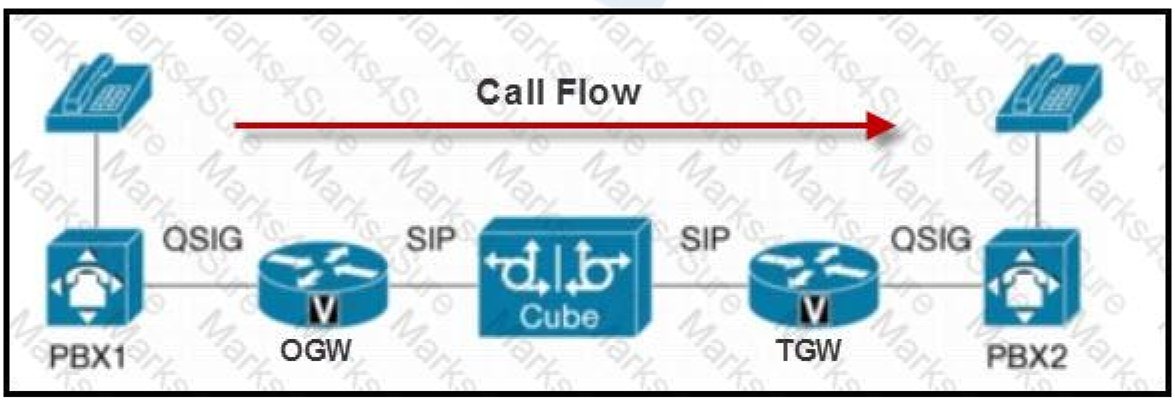
Answer: B

Explanation

The H.245 protocol is a media control protocol that is a part of 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 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 #:482 - (Exam Topic 3)

Refer to the exhibit.



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

A. Content-TypE. application/sdp

B. Content-TypE. application/qsig

C. Content-TypE. message/ISUP

D. Content-TypE. message/external-body

E. Content-TypE. application/x-q931

Answer: B

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:

http://www.cisco.com/c/en/us/solutions/collateral/enterprise-networks/empowered-branch-solution/white_paper_

Question #:483 - (Exam Topic 3)

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.

Answer: C

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 Annunciator does not support iSAC. MGCP devices are not supported.

Question #:484 - (Exam Topic 3)

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

Answer: C E

Explanation

Upon receiving an OpenReceiveChannelmessage, 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.

Question #:485 - (Exam Topic 3)

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

Answer: B

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 #:486 - (Exam Topic 3)

To which SIP response category does 301 Moved Permanently belong?

A. Provisional

B. Successful

C. Redirection

D. Client Failure

E. Server Failure

Answer: C

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 #:487 - (Exam Topic 3)

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

A. Ringing

B. Call is Being Forwarded

C. Session in Progress

D. Unknown Number

E. Call Does not Exist

Answer: B

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 that 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.

Reference:

http://en.wikipedia.org/wiki/List_of_SIP_response_codes

Question #:488 - (Exam Topic 3)

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

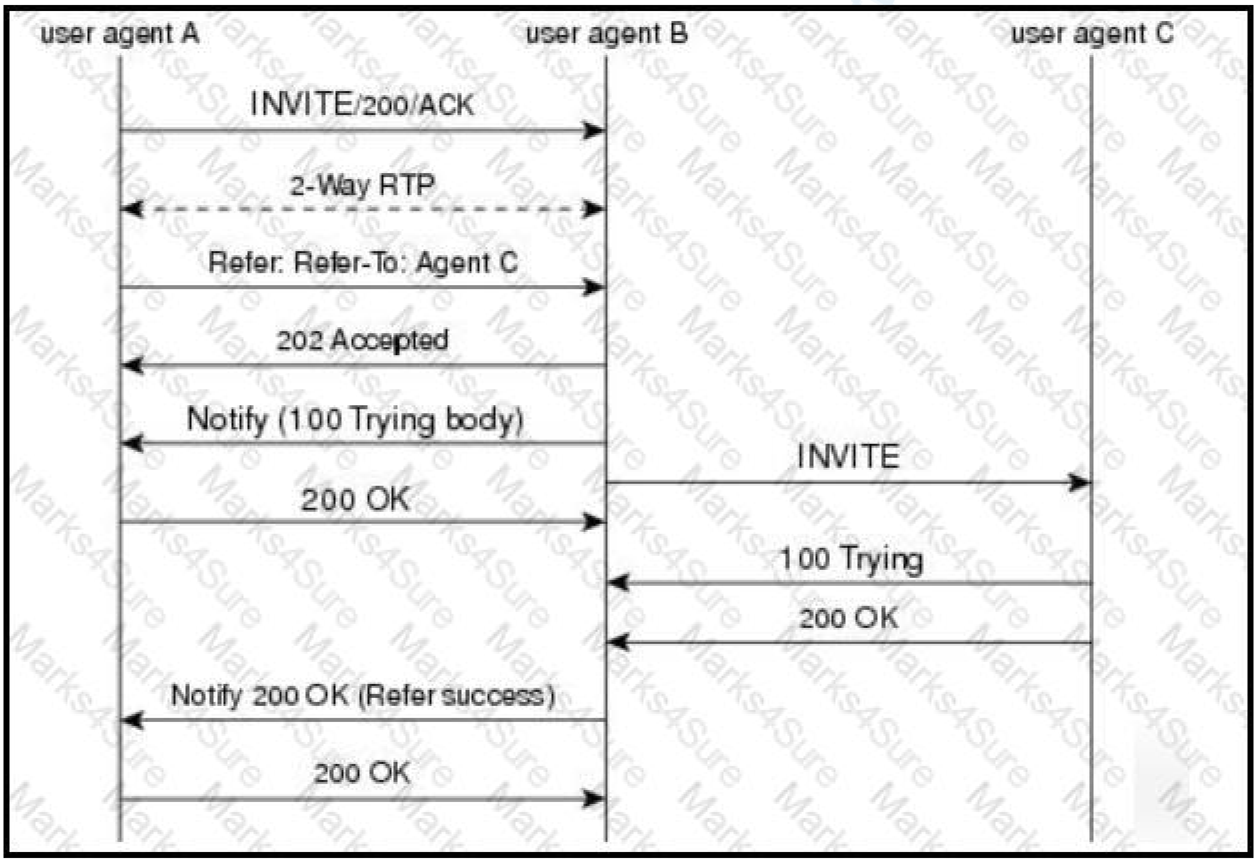
Answer: C

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 #:489 - (Exam Topic 3)

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

Answer: B

Explanation

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.

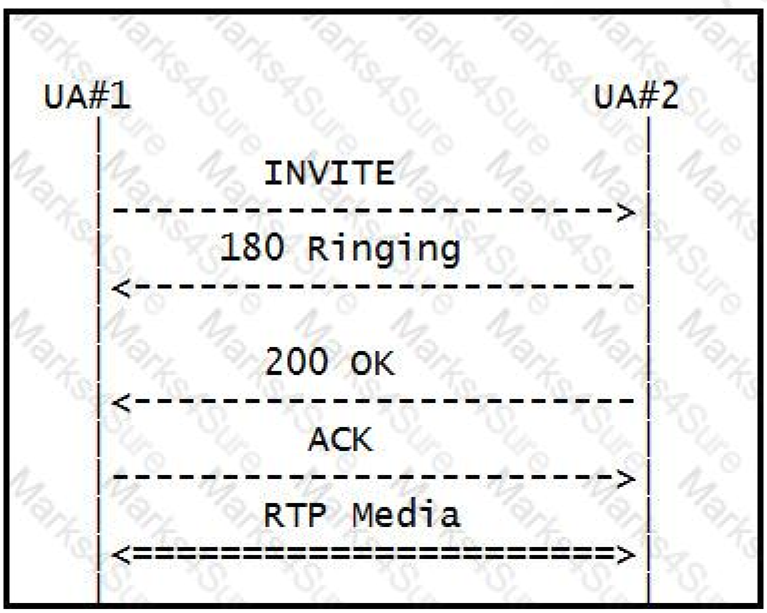
• Final-Recipient—User agent introduced into a call with the recipient.

Reference:

http://www.cisco.com/c/en/us/td/docs/ios_xr_sw/iosxr_r3-4/sbc/configuration/guide/sbc_c34/sbc34stx.pdf

Question #:490 - (Exam Topic 3)

Refer to the exhibit.



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

Answer: C

Explanation

In Early offer, SIP Send SDP in the invite, the other node will send the SDP in the 200 message.

Question #:491 - (Exam Topic 3)

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

Answer: C D

Explanation

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 [8]:

:

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 #:492 - (Exam Topic 3)

Refer to the exhibit.

```
Jul 31 18:12:16.640: MGCP Packet sent to 10.1.1.1:2427--->
200 108 OK
I: 2

v=0
c=IN IP4 10.1.1.254
m=audio 18630 RTP/AVP 0 100
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
a=X-cpar: a=fmtp:100 192-194,200-202
a=X-cap: 2 image udptl t38
```

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.

Answer: E

Question #:493 - (Exam Topic 3)

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

Answer: A B

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.

Question #:494 - (Exam Topic 3)

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

Answer: C D

Explanation

Connection info is optional field in SDP whether Media attributes decide the codec and media type for that call.

Question #:495 - (Exam Topic 3)

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

Answer: B

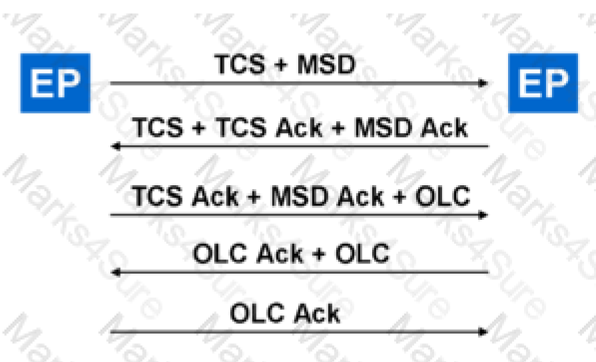
Explanation

A typical H.245 exchange looks similar to below figure

After this exchange of messages, the two endpoints (EP) in this figure would be transmitting audio in each direction. The number of message exchanges is numerous, each has an important purpose, but nonetheless takes time.

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 #:496 - (Exam Topic 3)

Refer to the exhibit.

```
Jan 19 17:02:49.998: MGCP Packet received from 10.1.1.1:2427--->
CRCX 83 S0/SU1/DS1-0/2@Router1.CiscoCustomer.com MGCP 0.1
C: D0000000027aef6f000001F500000003
X: 17
L: p:20, a:PCMU, s:off, t:b8, fxr/fx:t38
M: recvonly
R: D/[0-9ABCD*#]
Q: process,loop
```

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 an T1 Controller 0/1/2.

Answer: A

Explanation

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 #:497 - (Exam Topic 3)

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.

Answer: A

Explanation

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 #:498 - (Exam Topic 3)

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

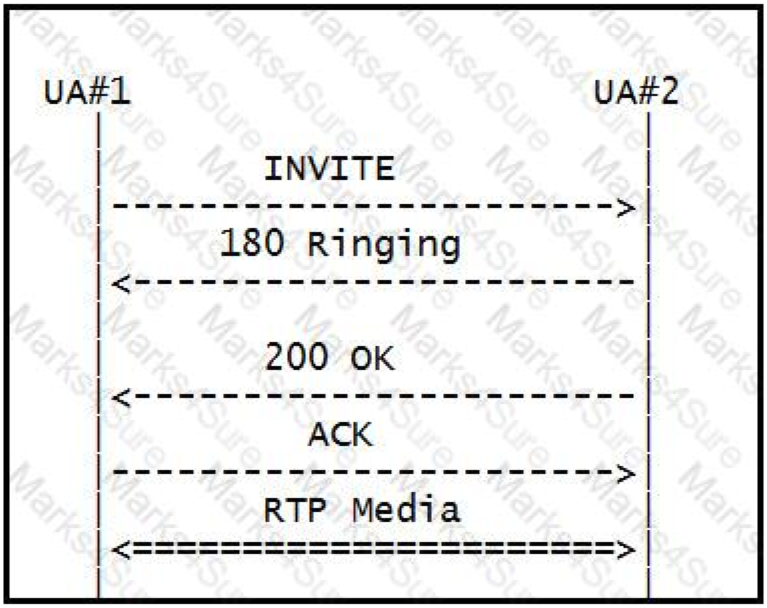
Answer: B

Explanation

The Refer method always begins within the context of an existing call and starts with theoriginator. The originator sends a Refer request to therecipient(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 #:499 - (Exam Topic 3)

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

Answer: D

Explanation

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 #:500 - (Exam Topic 3)

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

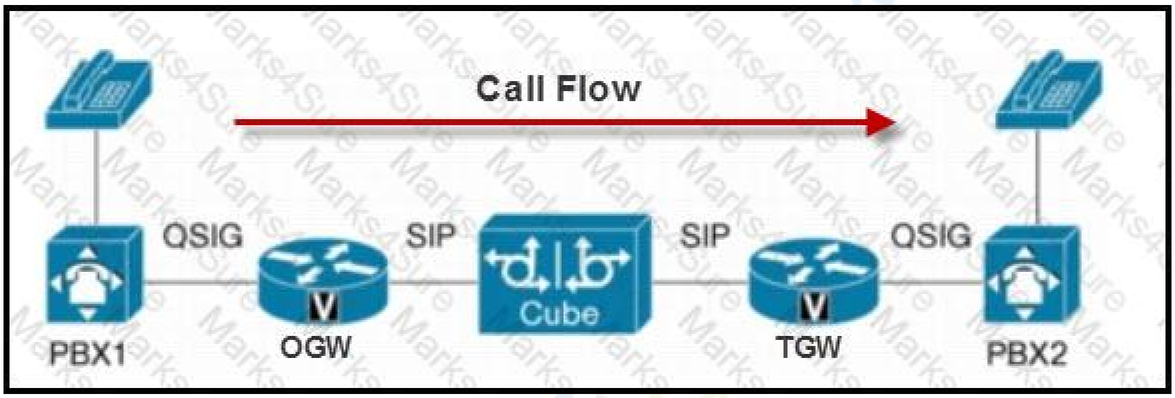
Answer: B F

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 #:501 - (Exam Topic 3)

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

Answer: A E

Explanation

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.

Reference: http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/sip/configuration/15-mt/sip-config-15-mt-book/voi-sip-tdm.html

Question #:502 - (Exam Topic 3)

Which two data frame lengths are supported by iLBC? (Choose two.)

A. 10 milliseconds

B. 20 milliseconds

C. 30 milliseconds

D. 40 milliseconds

E. 50 milliseconds

F. 60 milliseconds

Answer: B C

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, H323, and MGCP devices.

Question #:503 - (Exam Topic 3)

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

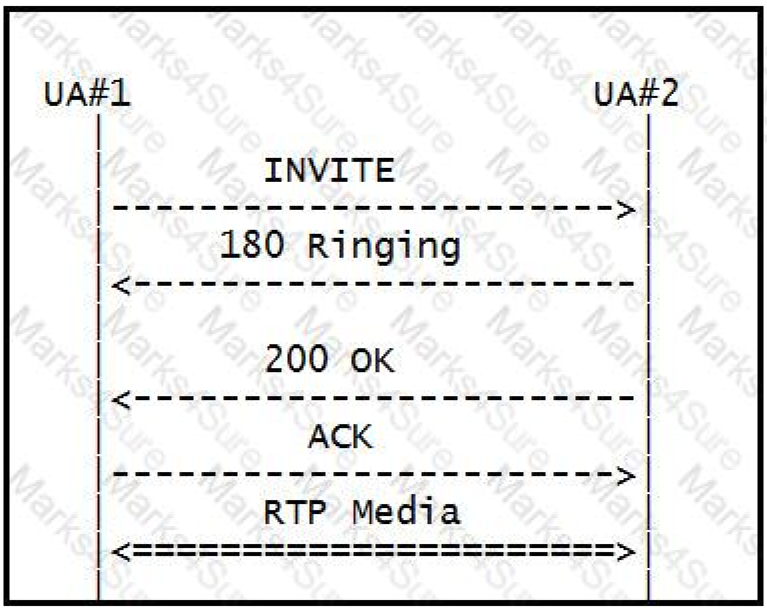
Answer: A D

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 #:504 - (Exam Topic 3)

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

Answer: C

Explanation

200 OK Indicates the request was successful.

Question #:505 - (Exam Topic 3)

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

Answer: C

Explanation

200 OK Indicates the request was successful. Whether other options state the request is still in progress or request is initiated.

Question #:506 - (Exam Topic 3)

Which SIP request method enables reliability of SIP 1xx response types?

A. ACK

B. PRACK

C. OPTIONS

D. CANCEL

E. REGISTER

Answer: B

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 exponentialbackoff. 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

Question #:507 - (Exam Topic 3)

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.

Answer: C D

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 #:508 - (Exam Topic 3)

Refer to the exhibit.

```
Jul 31 18:12:16.640: MGCP Packet sent to 10.1.1.1:2427--->
200 108 OK
I: 2

v=0
c=IN IP4 10.1.1.254
m=audio 18630 RTP/AVP 0 100
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=X-sqn:0
a=X-cap: 1 audio RTP/AVP 100
a=X-cpar: a=rtpmap:100 X-NSE/8000
a=X-cpar: a=fmtp:100 192-194,200-202
a=X-cap: 2 image udptl t38
```

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.

Answer: E

Explanation

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.

Topic 4, Cisco Unified Communications Manager (CUCM)

Question #:509 - (Exam Topic 4)

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.

Answer: B

Explanation

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 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

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” settings) 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_

Question #:510 - (Exam Topic 4)

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

Answer: D

Explanation

(2xx) Successful completion: The requested transaction was executed normally.

Question #:511 - (Exam Topic 4)

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

Answer: B E

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.

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/session_mgmt/deploy/8_5_1/overview.html

Question #:512 - (Exam Topic 4)

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 MRGL to 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 MRGL 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.

Answer: A

Question #:513 - (Exam Topic 4)

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.

Answer: A B

Explanation

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.

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

Question #:514 - (Exam Topic 4)

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

Answer: B

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 #:515 - (Exam Topic 4)

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 forward 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.

Answer: E

Explanation

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 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

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" settings) 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

Question #:516 - (Exam Topic 4)

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

Answer: A B

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 #:517 - (Exam Topic 4)

You are assisting a customer to troubleshoot a SIP early-offer problem with a SIP service provider. You have enabled Cisco CallManager trace and set the debug trace level to Detailed for SIP Call Processing trace on their standalone Cisco Unified Communications Manager 9.1 system. Using the RTMT tool, your customer has remote browsed to the Cisco UCM and asked you which trace file to download.

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

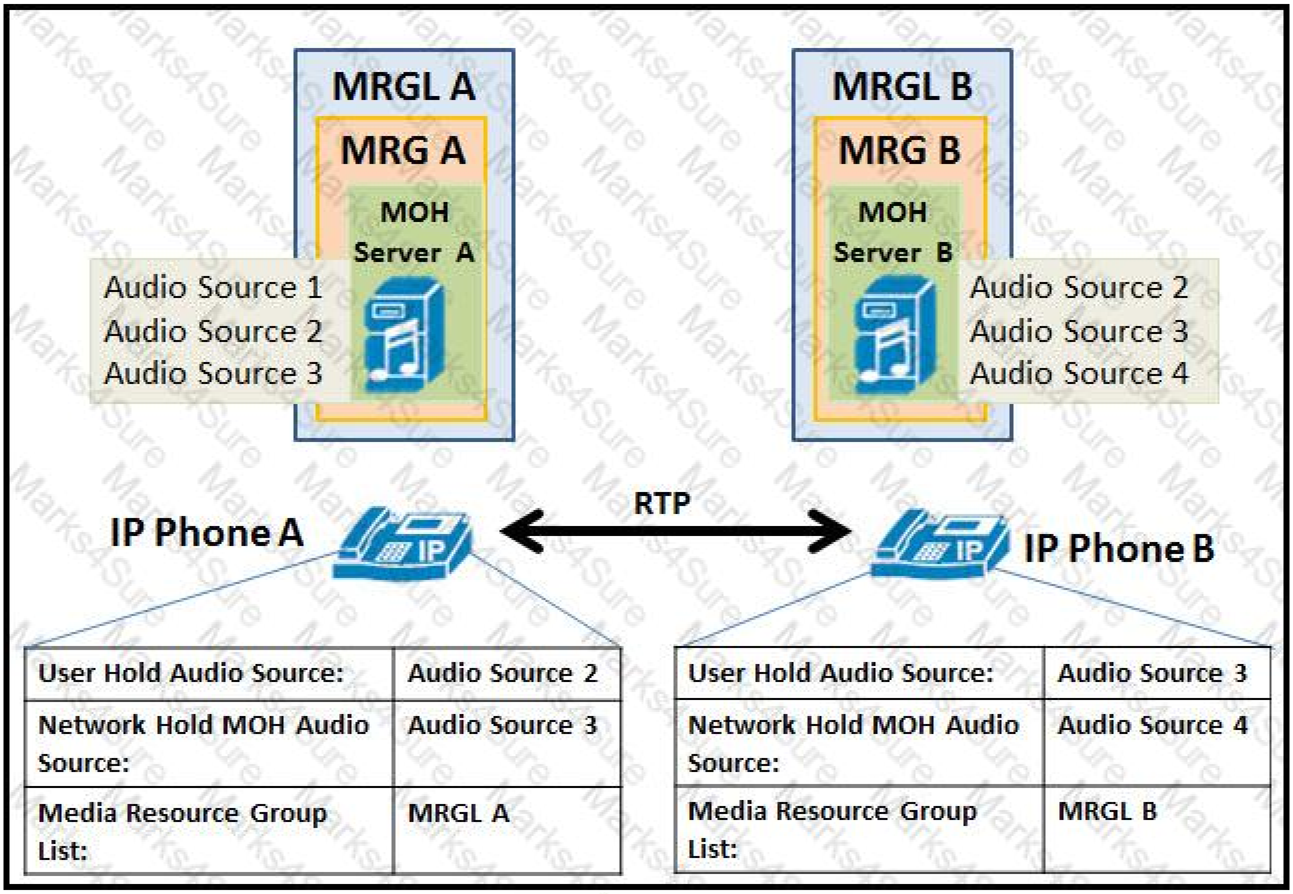
Answer: A

Explanation

SDL files log SIP messages from CCM.

Question #:518 - (Exam Topic 4)

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.

Answer: C

Explanation

Because audio source 2 is in top of the MRGL List and it will be selected locally first.

Question #:519 - (Exam Topic 4)

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.

Answer: E

Explanation

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.**

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/cac.html

Question #:520 - (Exam Topic 4)

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

Answer: A

Explanation

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

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/8x/uc8x/trunks.html

Question #:521 - (Exam Topic 4)

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

Answer: C

Explanation

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 #:522 - (Exam Topic 4)

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

Answer: D E

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 “<Extension> has become available but is on DND-R”. Callback available notification will be sent only after the terminating side disables DND Call Reject.

Question #:523 - (Exam Topic 4)

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.

Answer: B

Explanation

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 #:524 - (Exam Topic 4)

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

Answer: D

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 #:525 - (Exam Topic 4)

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.

Answer: B

Explanation

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 large 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.

- Default: 200.
- Minimum: 1.
- Maximum: 5000.

Question #:526 - (Exam Topic 4)

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. Subscriber Number

D. Regional Significant Number

E. Local User Code

F. National Numbering Plan

Answer: B C

Explanation

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 #:527 - (Exam Topic 4)

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

Answer: C

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:

•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.

•Remote Failover Deployment Model

Remote failover allows you 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.

Reference:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42models.html

Question #:528 - (Exam Topic 4)

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

Answer: B

Explanation

G.728—Low-bit-rate codec that video endpoints support.It support kilobits per second

Question #:529 - (Exam Topic 4)

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

Answer: C

Explanation

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_BK_CD2F83FA_00_

Question #:530 - (Exam Topic 4)

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

Answer: B

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.

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/7x/uc7_0/models.html

Question #:531 - (Exam Topic 4)

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

Answer: B

Explanation

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 #:532 - (Exam Topic 4)

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

Answer: E

Explanation

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 #:533 - (Exam Topic 4)

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.

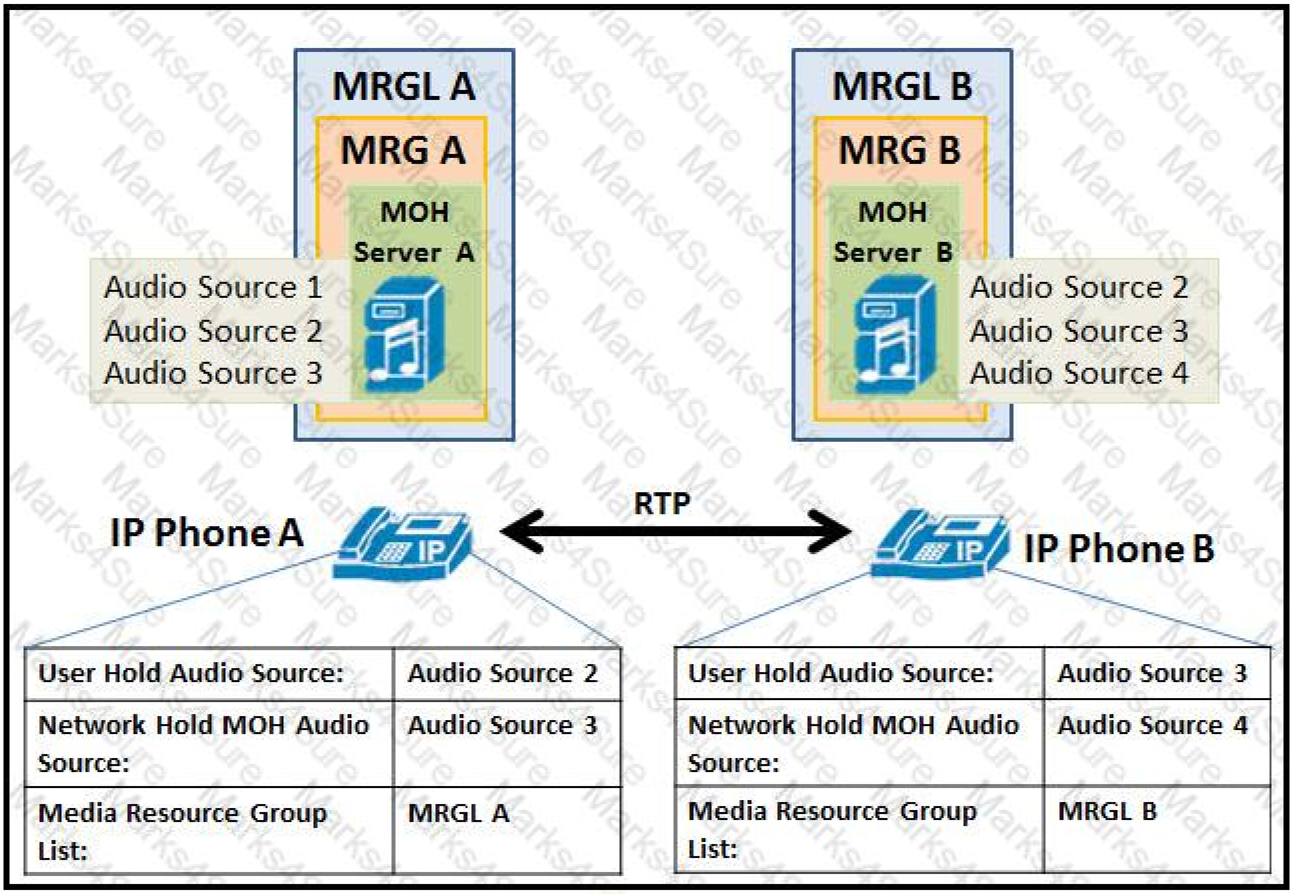
Answer: B D

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 #:534 - (Exam Topic 4)

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 B presses the Hold softkey?

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.

Answer: B

Explanation

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

Question #:535 - (Exam Topic 4)

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

Answer: B

Explanation

Hold Reversion and Intercom

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

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/7_1_2/ccmfeat/fsgd-712-cm/fsdnd.html#wp

Question #:536 - (Exam Topic 4)

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.

E. It is used to provide the relative priority of a location.

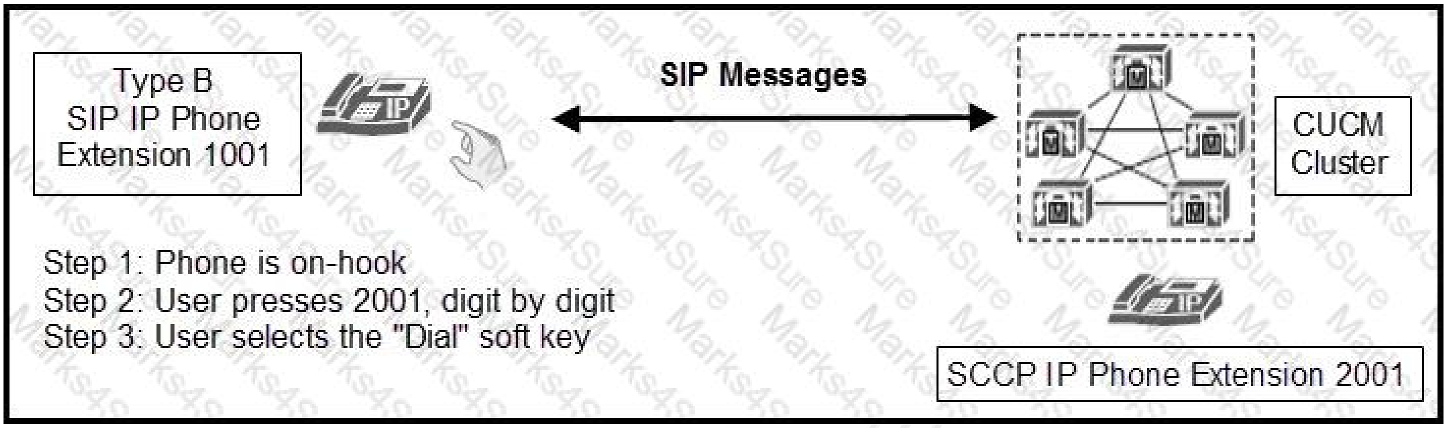
Answer: D

Explanation

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 #:537 - (Exam Topic 4)

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.

Answer: D

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 #:538 - (Exam Topic 4)

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

Answer: E

Explanation

Cisco Unified Communications Manager supports dialing using directory URIs for call addressing. Directory URIs look like email addresses and follow the username@host 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 #:539 - (Exam Topic 4)

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.

Answer: D

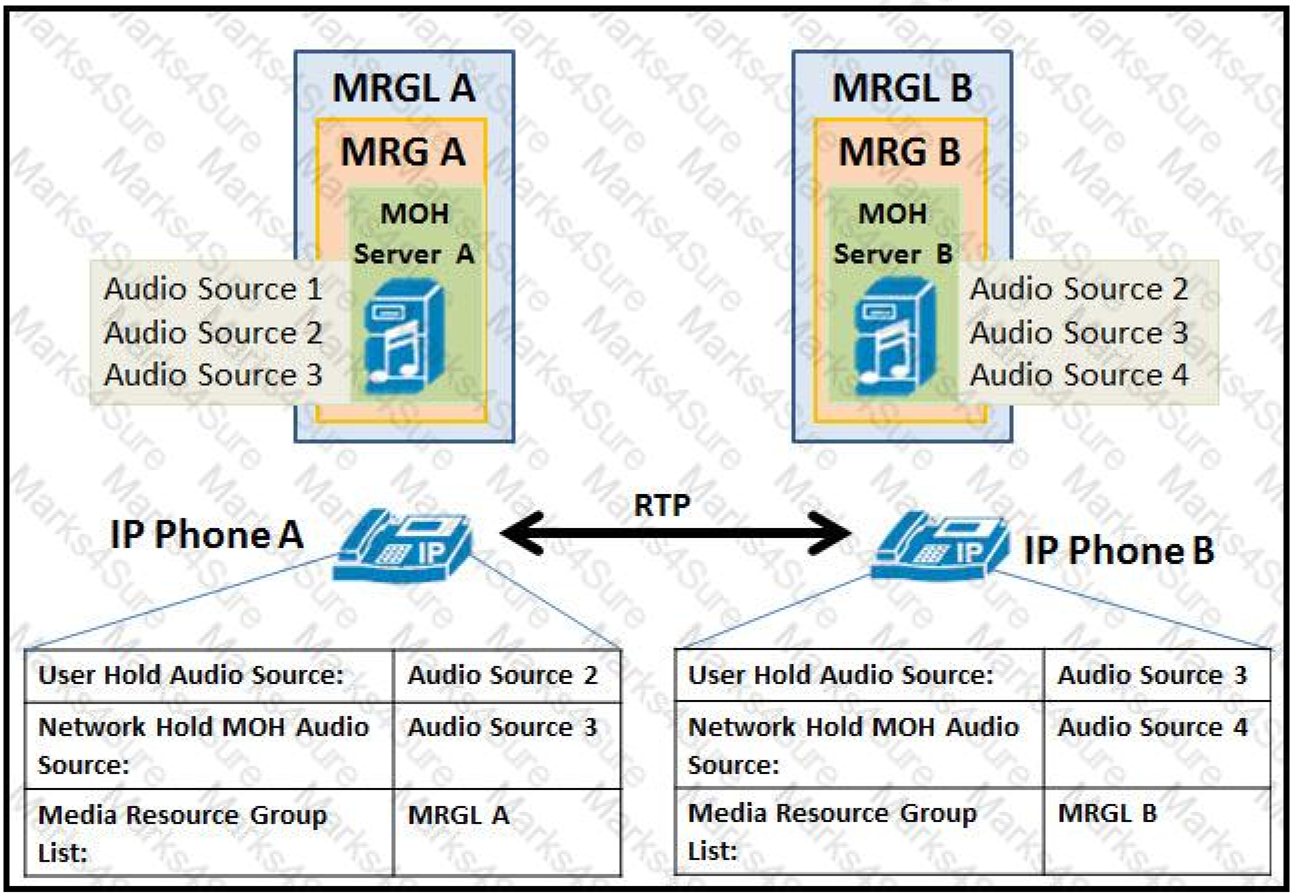
Explanation

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/ccmcfg/CUCM_BK_A34970C5_00_n-guide-91/CUCM_BK_A34970C5_00_admin-guide-91_chapter_0100011.html

Question #:540 - (Exam Topic 4)

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 tone on-hold beep tones.

E. IP phone A user hears no on-hold music or beep tones.

Answer: E

Question #:541 - (Exam Topic 4)

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

E. visiting cluster's LBM hub

Answer: B

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 #:542 - (Exam Topic 4)

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 machines.

D. The configuration file should be placed inside an ISO file and mounted on the virtual machine.

Answer: B

Reference: http://www.cisco.com/c/en/us/td/docs/voice_
ip_comm/cucm/install/9_0_1/CUCM_BK_I87B437D_00_installing-cucm-90/CUCM_BK_I87B437D_00_instal

Question #:543 - (Exam Topic 4)

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

Answer: B

Explanation

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 #:544 - (Exam Topic 4)

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

Answer: C F

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.

Reference:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42dvmobl.html

Question #:545 - (Exam Topic 4)

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

Answer: D

Explanation

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 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.

Reference: http://www.cisco.com/c/en/us/td/doc s/voice_ip_comm/cucm/admin/6_1_1/ccmfeat/cmfsgd611/fsdnd.html

Question #:546 - (Exam Topic 4)

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

Answer: C

Explanation

All messages to and from a skinny device are preceded by either the words StationD or StationInit. StationD messages are sent from call manager to IP phone. Skinny message transmission such as this between the IP phone and CallManger occurs for every action undertaken by the IP phone, includinginitialization, registration, on-hook, off-hook, dialing of the digits, key press on the phone, and so much more.

Question #:547 - (Exam Topic 4)

Which statement about the effective path in the Enhanced Location Call Admission Control mechanism on Cisco Unified Communications Manager is true?

A. It is a 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.

Answer: C

Explanation

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 #:548 - (Exam Topic 4)

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 support 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.

Answer: A B

Explanation

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.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/bat/9_1_1/CUCM_BK_C271A69D_00_cucm-bulk

Question #:549 - (Exam Topic 4)

When IP phone A was provisioned in a Cisco Unified Communications Manager, 2001 was configured as the directory number for its first line. Also, bob@cisco.com 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, bobby@cisco.com was defined as a directory URI for that end user.

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

A. bob@cisco.com

B. bobby@cisco.com

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.

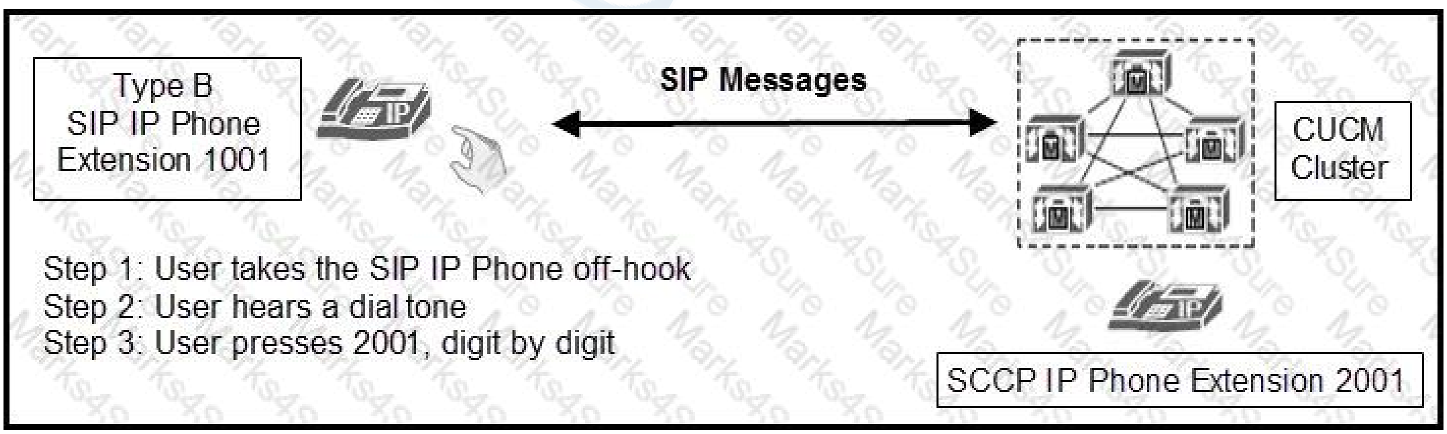
Answer: B

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BK_CD2F83FA_00

Question #:550 - (Exam Topic 4)

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.

Answer: E

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 #:551 - (Exam Topic 4)

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

Answer: A

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 #:552 - (Exam Topic 4)

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

Answer: D

Explanation

408 Request Timeout

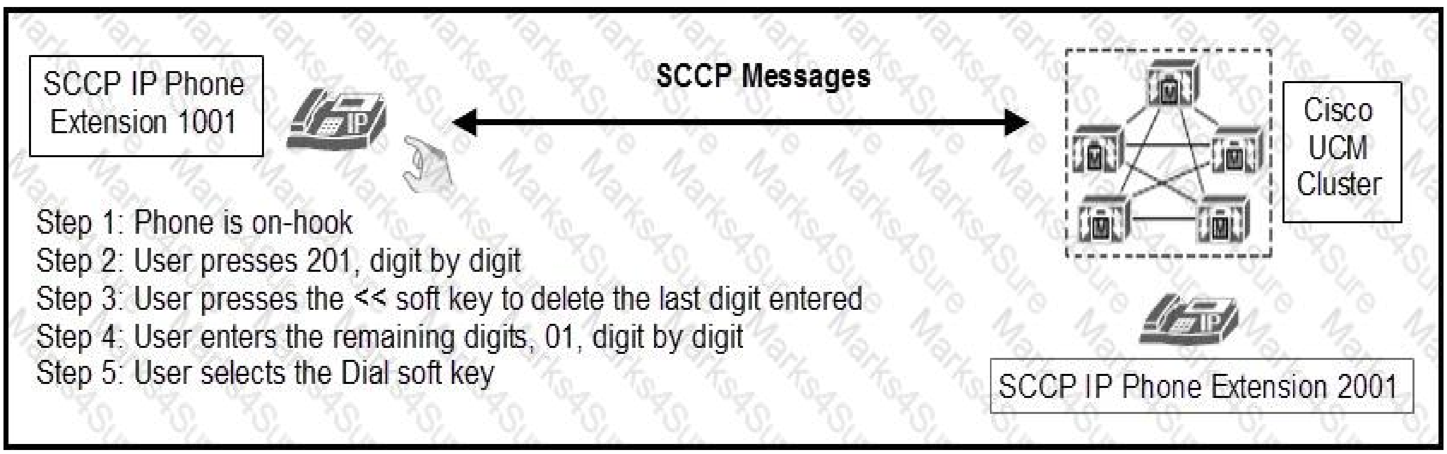
Couldn't find the user in time. The server could not produce a response within a suitable amount of time, for example, if it could not determine the location of the user in time. The client MAY repeat the request without modifications at any later time

Reference:

http://en.wikipedia.org/wiki/List_of_SIP_response_codes

Question #:553 - (Exam Topic 4)

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.

Answer: B

Explanation

Because sccp phones send digits DIGIT-by-DIGIT i.e. it sends each digit in real time.

Link:-https://supportforums.cisco.com/document/87236/working-concept-sccp-sip-phones-and-dial-rules

Question #:554 - (Exam Topic 4)

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

Answer: D

Explanation

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.

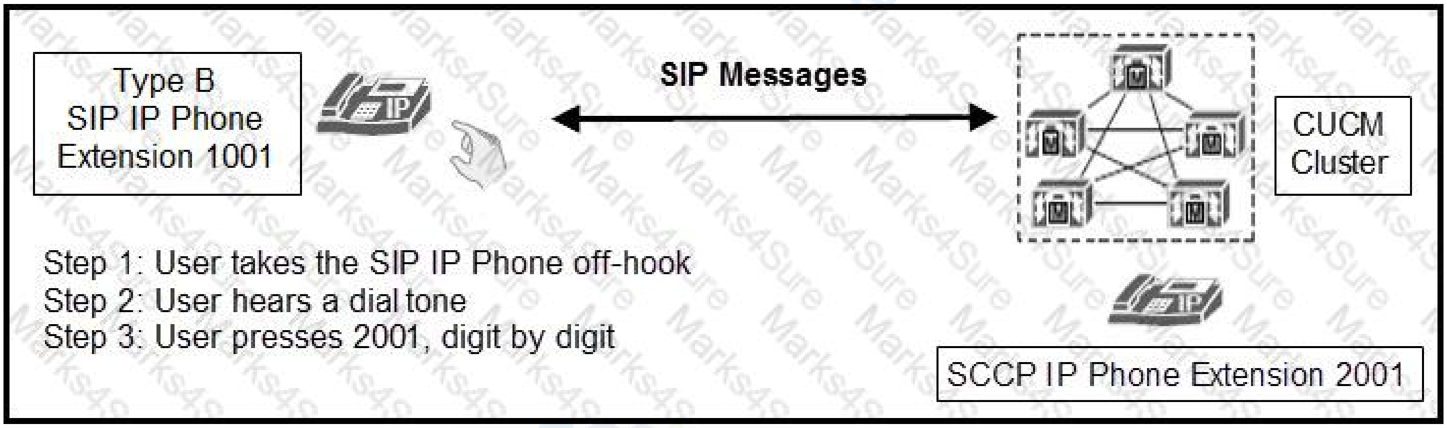
Cisco Unified Communications Manager uses the x-cisco-number tag in the SIP identity headers to communicate a blended address. When both a directory URI and directory number are available for the sending phone and blended addressing is enabled, Cisco Unified Communications Manager uses the directory 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_0

Question #:555 - (Exam Topic 4)

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.

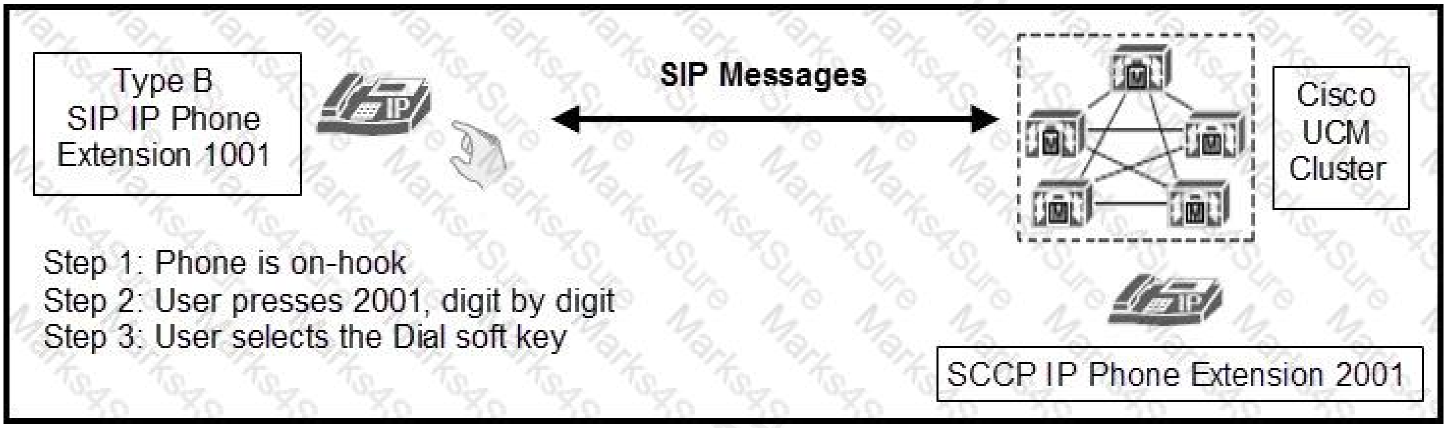
Answer: A

Explanation

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 #:556 - (Exam Topic 4)

Refer to the exhibit.



A user is going through a series of dialing steps on a SIP Type B 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 associated with a SIP dial rule with a pattern value of "2001".

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. 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.

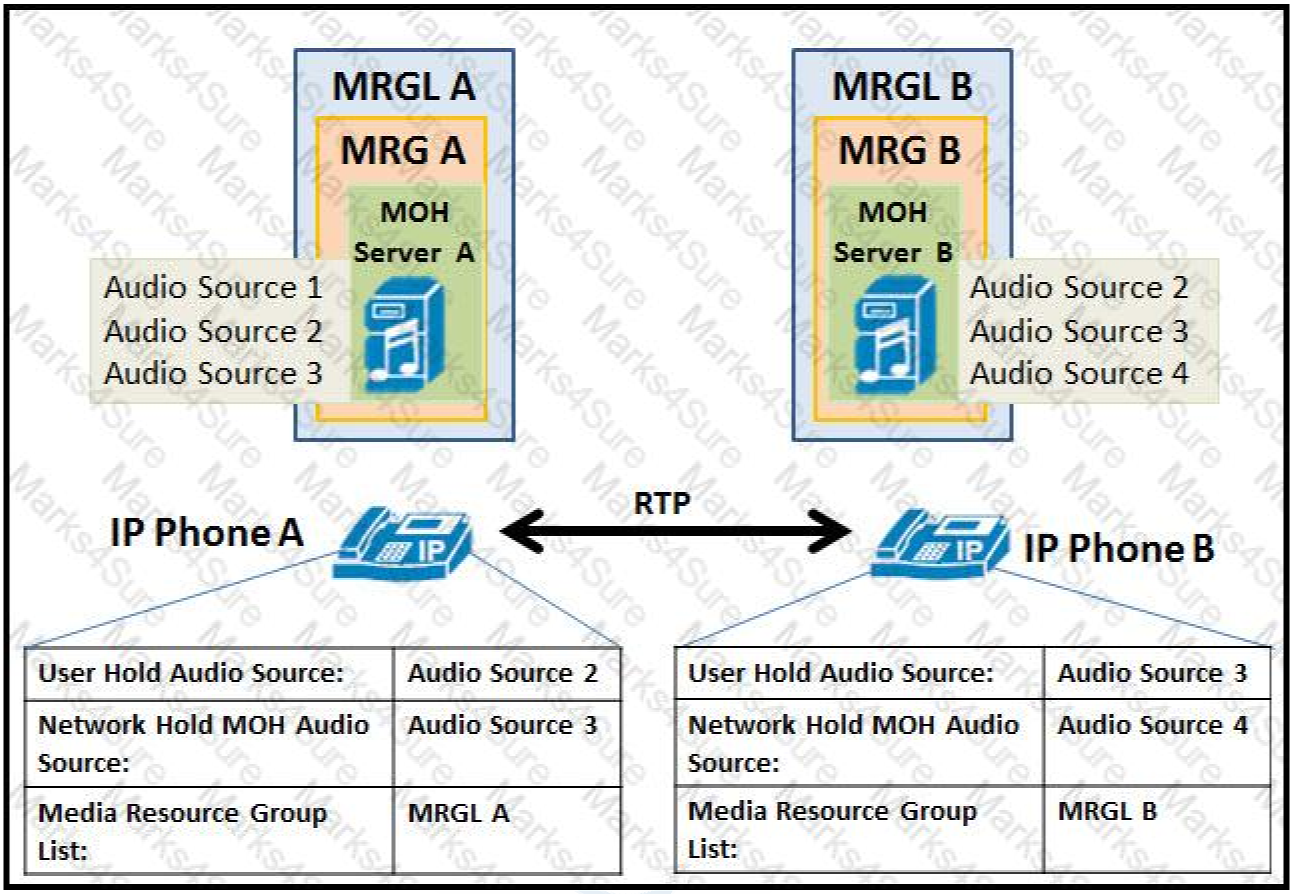
E. The SIP IP phone waits for the inter-digit timer expiry, and then sends all digits to the Cisco Unified Communications Manager in a SIP INVITE message.

Answer: C

Reference: https://supportforums.cisco.com/document/87236/working-concept-sccp-sip-phones-and-dial-rules.

Question #:557 - (Exam Topic 4)

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.

Answer: C

Explanation

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

Question #:558 - (Exam Topic 4)

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

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.

Answer: B E

Explanation

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.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/bat/9_1_1/CUCM_BK_C271A69D_00_cucm-bulk

Question #:559 - (Exam Topic 4)

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 inserts a 500-millisecond pause between digits.

B. It causes the phone to generate a secondary dial tone.

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.

Answer: B

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 insert a 2 second delay Commas may be duplicated to create longer delays.

Question #:560 - (Exam Topic 4)

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

Answer: D

Explanation

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 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.

Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/6_1_1/ccmfeat/cmfsgd611/fsdnd.html

Question #:561 - (Exam Topic 4)

Which two statements about Cisco Unified Communications Manager mixed-mode clusters are true? (Choose two.)

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.

Answer: A E

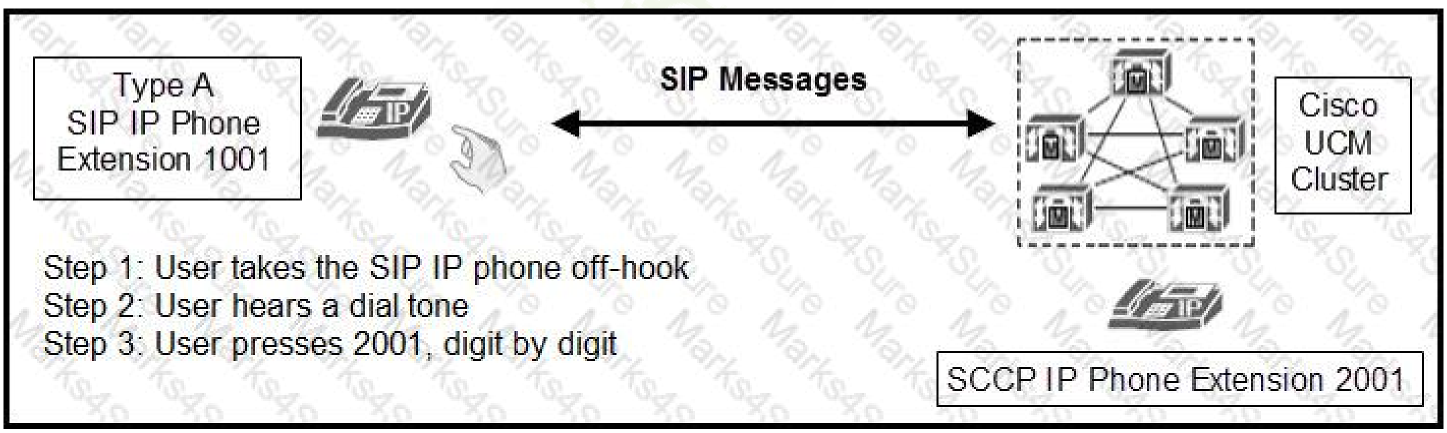
Explanation

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

Reference:http://www.cisco.com/c/en/us/td/docs/voice
_ip_comm/cucm/security/7_0_1/secugd/sec701-cm/secuauth.html#wp1037433

Question #:562 - (Exam Topic 4)

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.

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.

Answer: C

Explanation

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.

Reference: https://supportforums.cisco.com/document/87236/working-concept-sccp-sip-phones-and-dial-rules.

Question #:563 - (Exam Topic 4)

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

Answer: A

Reference:

http://www.techtronicssolution.com/blog/?p=1201

Question #:564 - (Exam Topic 4)

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.

Answer: D

Explanation

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 #:565 - (Exam Topic 4)

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 matches one or more digits from 0 to 9.

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 digits from 0 to 9, or the asterisk (*) or pound (#) symbols.

Answer: D

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 #:566 - (Exam Topic 4)

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

Answer: A B

Explanation

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 #:567 - (Exam Topic 4)

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

Answer: C

Explanation

Cisco Unified Communications Manager supports dialing using directory URIs for call addressing. Directory URIs look like email addresses and follow the username@host 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 #:568 - (Exam Topic 4)

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

Answer: A

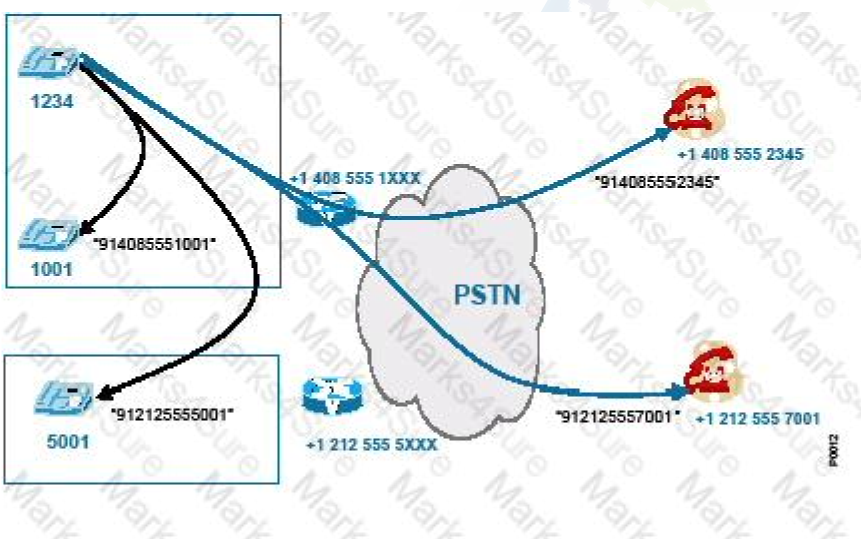
Explanation

- **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.

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09/dialplan.html

Question #:569 - (Exam Topic 4)

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.

Answer: D

Explanation

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.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/bat/9_1_1/CUCM_BK_C271A69D_00_cucm-bulk

Question #:570 - (Exam Topic 4)

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

Answer: A

Explanation

When considering a megacluster deployment, the primary areas impacting capacity are as follows:

•The megacluster 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.

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_5_1/ccmfeat/fsgd-851-cm/fsmoh.html#wp

Question #:571 - (Exam Topic 4)

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.

Answer: B C E

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. SeeLBM 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 onCall 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.xEnhanced 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 #:572 - (Exam Topic 4)

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.

Answer: E

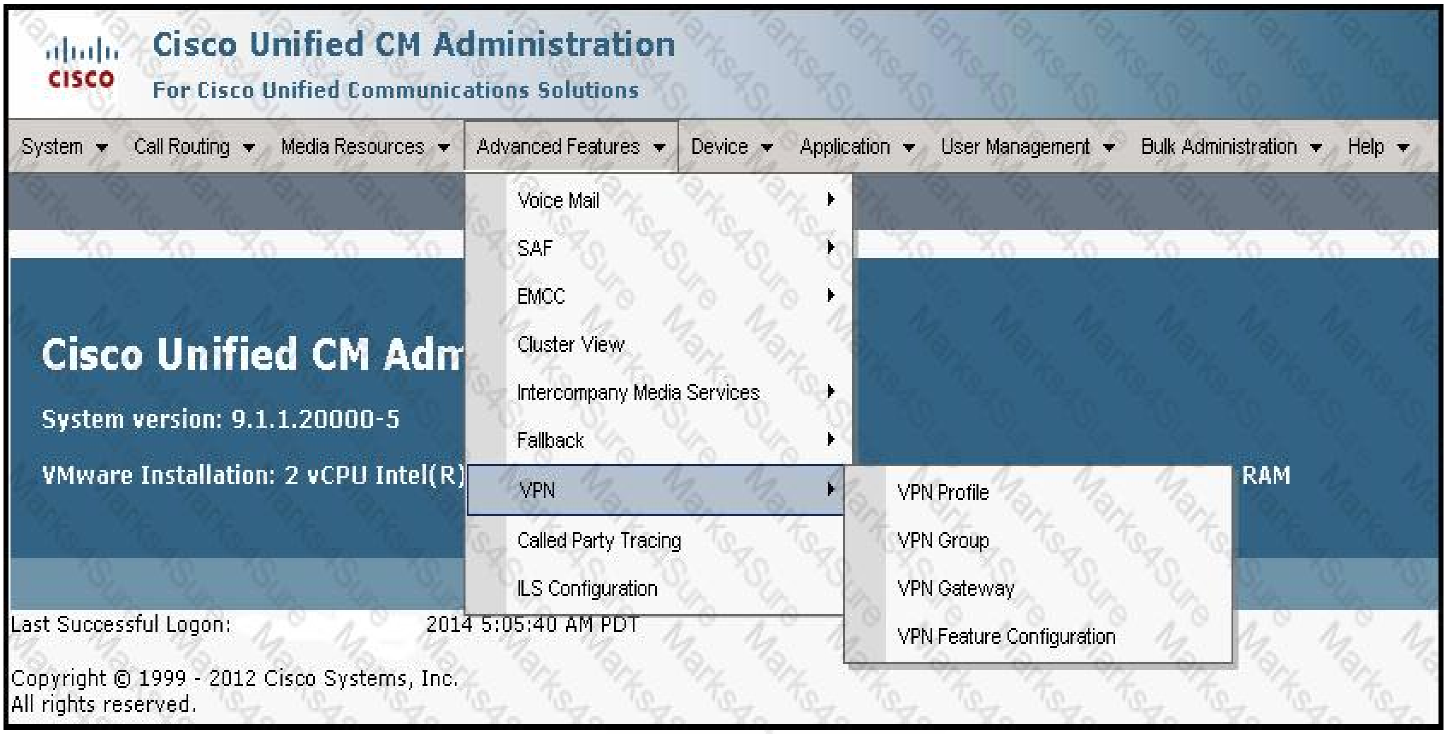
Explanation

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.

Reference: http://www.cisco.com/c/en/us/support/
docs/voice-unified-communications/unified-communications-manager-callmanager/81541-fac-config-ex.html

Question #:573 - (Exam Topic 4)

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

Answer: A D

Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/security/8_5_1/secugd/sec-851-cm/secvpfet.html

Question #:574 - (Exam Topic 4)

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

Answer: B

Explanation

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 get used, and only the first two IP addresses that the DHCP server provides get accepted.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_0_1/ccmsys/CUCM_BK_CD2F83FA_00_

Question #:575 - (Exam Topic 4)

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

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

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 Forward Hunt Busy settings on the Line Group members.

D. Calls are handled according to the Hunt Options 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.

Answer: E

Explanation

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 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

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" settings) 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

Topic 5, Cisco IOS UC Applications and Features

Question #:576 - (Exam Topic 5)

Refer to the exhibit.

```
voice class codec 1
 codec preference 1 g729r8
 codec preference 2 g711ulaw

!
voice class sip-profiles 102
 request INVITE sip-header Allow-Header modify ", UPDATE" ""
 request REINVITE sip-header Allow-Header modify ", UPDATE" ""
 response 180 sip-header Allow-Header modify ", UPDATE" ""
 response 200 sip-header Allow-Header modify ", UPDATE" ""
 request INVITE sip-header Diversion remove
 request ANY sip-header Diversion remove
!
dial-peer voice 7 voip
 translation-profile outgoing CALLED_DIGIT_STRIP
 destination-pattern 6911
 modem passthrough nse codec g711ulaw
 session protocol sipv2
 session target ipv4:10.0.0.1
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 voice-class sip early-offer forced
 voice-class sip profiles 102
 dtmf-relay rtp-nte
 ip qos dscp af41 signaling
 no vad
```

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

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.

Answer: B E

Explanation

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.

Question #:577 - (Exam Topic 5)

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 check-pointed 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.

F. The SIP media signal must be bound to the loopback interface.

Answer: B D

Explanation

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 Fast Ethernet (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 a dedicated FE link between the CSSs..

•Require the connection of a Layer 2 device between the redundant CSS peers..

Question #:578 - (Exam Topic 5)

Refer to the exhibit.

```
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
```

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

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 #:579 - (Exam Topic 5)

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

Answer: E

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 #:580 - (Exam Topic 5)

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 be down.

C. Its admin and operational state will remain up.

D. Its admin state will be up but operational state will be "busy-out".

E. Its admin and operational state will be "busy-out".

Answer: A

Explanation

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,Icpif = 30, VAD = disabled, Poor QOV Trap = disabled,

Connect Time = 0, Charged Units = 0

Successful Calls = 0, Failed Calls = 0

Accepted Calls = 0, Refused 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 dialplan number** command. The following example displays the VoIP dial peer associated with the destination pattern 51234:.

router# **show dialplan 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, Icpif = 30,

VAD = enabled, Poor QOV Trap = disabled

Connect Time = 0, Charged Units = 0

Successful Calls = 0, Failed Calls = 0

Accepted Calls = 0, Refused 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 #:581 - (Exam Topic 5)

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

Answer: C

Explanation

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 #:582 - (Exam Topic 5)

Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 call-forward b2bua busy 2100
 call-forward b2bua noan 2200 timeout 20
 shared-line max-calls 5
 huntstop channel 4
!
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 3
 id mac 2222.2222.2222
 type 7965
 number 1 dn 1
!
```

IP phone 1 has the MAC address 11111.1111.1111, 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.

Answer: B

Explanation

In shared line configuration phone share the same line so it is possible for any phone to answer the call.

Question #:583 - (Exam Topic 5)

Refer to the exhibit.

```
CUBE-2#show voice high-availability summary
======== Voice HA DB INFO ========
Number of calls in HA DB: 28  (MAX:2048)
Number of calls in HA sync pending DB: 12
Number of calls in HA preserved session DB: 9
```

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

Answer: D

Explanation

Total no of calls =28+12=40.

So, native calls are =40-9=31.

Reference:
http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-border-element/112095-cube
ig-00.html.

Question #:584 - (Exam Topic 5)

Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 call-forward b2bua busy 2100
 shared-line
 huntstop channel 6
!
voice register pool 1
 busy-trigger-per-button 4
 id mac 1111.1111.1111
 type 7965
 number 1 dn 1
!
voice register pool 2
 busy-trigger-per-button 5
 id mac 2222.2222.2222
 type 7965
 number 1 dn 1
!
```

How many simultaneous inbound calls can be handled by these two IP phones?

A. 2

B. 4

C. 6

D. 9

E. 10

Answer: A

Explanation

The line is configured as shared line so it will support maximum two calls at a time.

Question #:585 - (Exam Topic 5)

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 arrived 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.

Answer: E

Explanation

- 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, 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.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmesnr.html

Question #:586 - (Exam Topic 5)

Refer to the exhibit.

```
SIP-Gateway#show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED

SIP User Agent for TLS over TCP : ENABLED
SIP User Agent bind status(signaling): ENABLED  10.1.1.1
SIP User Agent bind status(media): ENABLED  10.1.1.1
SIP early-media for 180 responses with SDP: DISABLED
SIP max-forwards : 70
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED
Reason Header will override Response/Request Codes: DISABLED
Out-of-dialog Refer: DISABLED
Presence support is DISABLED
protocol mode is ipv4

SDP application configuration:
 Version line (v=) required
 Owner line (o=) required
 Timespec line (t=) required
 Media supported: audio video image
 Network types supported: IN
 Address types supported: IP4 IP6
 Transport types supported: RTP/AVP udptl
```

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.

Answer: B

Question #:587 - (Exam Topic 5)

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

Answer: C

Explanation

- Router(config)#**crypto pki trustpoint CAPF**

enrollment terminal

authorization username subjectname commonname

revocation-check none

Router(config)#**crypto pki authenticate CAPF**

Router(config)#

Things to Note:

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

Reference:

http://www.cisco.com/c/en/us/support/docs/ios-nx-os-software/authentication-authorization-accounting-aaa/1163

Question #:588 - (Exam Topic 5)

Refer to the exhibit.

```
Branch-Router#debug ip dhcp packet

*Aug 15 22:13:23.924: DHCPD: Finding a relay for client 01ec.4476.1e3e.7d on interface Vlan101.
*Aug 15 22:13:23.924: DHCPD: setting giaddr to 10.101.15.1.
*Aug 15 22:13:23.924: DHCPD: BOOTREQUEST from 01ec.4476.1e3e.7d forwarded to 10.100.1.1.
*Aug 15 22:13:23.940: DHCPD: forwarding BOOTREPLY to client ec44.761e.3e7d.
*Aug 15 22:13:23.940: DHCPD: broadcasting BOOTREPLY to client ec44.761e.3e7d.

Branch-Router#
```

The exhibit shows the Cisco IOS CLI output of debug ipdhcp 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 two statements about the network components are true? (Choose two.)

A. The MAC address of the IP phone is 01ec44761e3e.

B. The IP address of the DHCP server is 10.101.15.1.

C. The MAC address of the VLAN 101 interface is ec44761e3e7d.

D. The IP address of the VLAN 101 interface is 10.101.15.1.

E. There is IP connectivity between the VLAN 101 interface of the branch router and the ip-helper address that is configured on this interface.

F. There is IP connectivity between the IP phone and the ip-helper address on the VLAN 101 interface.

Answer: D E

Explanation

As we can see from the logs given first line relate that dhcp request is being relayed. So it clarifies there must be ip helper address commend given by the admin on interface vlan 101. Now we can see from the second line that giaddress is set as source address of vlan 101 by the dhcp as 10.101.15.1 to unicast the dhcp request

Question #:589 - (Exam Topic 5)

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.

Answer: C

Explanation

Table 5-2 Trigger Pattern Wildcards and Special Characters

Character

Description

Examples

X

The X wildcard matches any single digit in the range 0 through 9.

The trigger pattern 9XXX matches all numbers in the range 9000 through 9999.

!

The exclamation point (!) wildcard matches one or more digits in the range 0 through 9.

The trigger pattern 91! matches all numbers in the range 910 through 91999999999999999999999999999999.

?

The question mark (?) wildcard matches zero or more occurrences of the preceding digit or wildcard value.

The trigger pattern 91X? matches all numbers in the range 91 through 91999999999999999999999999999999.

+

The plus sign (+) wildcard matches one or more occurrences of the preceding digit or wildcard value.

The trigger pattern 91X+ matches all numbers in the range 910 through 91999999999999999999999999999999.

[ ]

The square bracket ([ ]) characters enclose a range of values.

The trigger pattern 813510[012345] matches all numbers in the range 8135100 through 8135105.

-

The hyphen (-) character, used with the square brackets, denotes a range of values.

The trigger pattern 813510[0-5] matches all numbers in the range 8135100 through 8135105.

^

The circumflex (^) character, used with the square brackets, negates a range of values. Ensure that it is the first character following the opening bracket ([).

Each trigger pattern can have only one ^ character.

The trigger pattern 813510[^0-5] matches all numbers in the range 8135106

Question #:590 - (Exam Topic 5)

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

A. sequential

B. peer

C. longest idle

D. parallel

E. overlay

Answer: D

Explanation

Parallel Hunt-Group, allows a user to dial a pilot number that rings 2-10 different extensions simultaneously. The first extension to answer gets connected to the caller while all other extensions will stop ringing. A timeout value can be set whereas if none of the extensions answer before the timer expires, all the extensions will stop ringing and one final destination number will ring indefinitely instead. The final number could be another voice hunt-group pilot number or mailbox.

The following features are supported for Voice Hunt-Group:

- Calls can be forwarded to Voice Hunt-Group
- Calls can be transferred to Voice Hunt-Group
- Member of Voice Hunt-Group can be SCCP, ds0-group, pri-group, FXS or SIP phone/trunk
- Max member of Voice Hunt-Group will be 32

Question #:591 - (Exam Topic 5)

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

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:

Command

Purpose

Step 1

Router(config)# **caller-id enable**

Enables caller ID. This command applies to FXS voice ports that send caller-ID information and to FXO ports that receive it. By default caller ID is disabled.

Note If the **station-id** or a **caller-id alerting** command is configured on the voice port, these automatically enable caller ID, and the **caller-id enable** command is not necessary..

Step 2

Router(config-voiceport)# **station-id name***name*

Configures the station name on FXS voice ports connected to user telephone sets. This sets the caller-ID information for on-net calls originated by the FXS port. You can also configure the station name on an FXO port of a router for which incoming Caller ID from the PSTN subscriber line is expected. In this case, if no caller-ID information is included on the incoming PSTN call, the call recipient receives the information configured on the FXO port instead. If the PSTN subscriber line does provide caller-ID information, this information is used and the configured station name is ignored.

The *name* argument is a character string of 1 to 15 characters identifying the station.

Note This command applies only to caller-ID calls, not Automatic Number Identification (ANI) calls. ANI supplies calling number identification only.

Step 3

Router(config-voiceport)# **station-id number***number*

Configure the station number on FXS voice ports connected to user telephone sets. This sets the caller-ID information for on-net calls originated by the FXS port.

You can also configure the station number on an FXO port of a router for which incoming caller ID from the PSTN subscriber line is expected. In this case, if no caller-ID information is included on the incoming PSTN call, the call recipient receives the information configured on the FXO port instead. If the PSTN subscriber line does provide caller-ID information, this information is used and the configured station name is ignored.

If the caller-ID station number is not provided by either the incoming PSTN caller ID or by the station number configuration, the calling number included with the on-net routed call is determined by Cisco IOS software by using a reverse dial-peer search. In this case, the number is obtained by searching for a POTS dial-peer that refers to the voice-port and the destination-pattern number from that dial-peer is used.

Number is a string of 1 to 15 characters identifying the station telephone or extension number.

Reference:

http://www.cisco.com/c/en/us/td/docs/ios/12_2/voice/configuration/guide/fvvfax_c/vvfclid.html

Question #:592 - (Exam Topic 5)

Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 call-forward b2bua busy 2100
 call-forward b2bua noan 2200 timeout 20
 shared-line max-calls 4
 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.

Answer: D

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.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_c1ht.html#wp15

Question #:593 - (Exam Topic 5)

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

Answer: A

Explanation

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.

Reference:
http://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/64282-impedance-choice.htm

Question #:594 - (Exam Topic 5)

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

Answer: A E

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 #:595 - (Exam Topic 5)

Refer to the exhibit.

```
T1-CAS-Gateway(config-controller)#ds0-group 1 time 1-10 type ?
  e&m-delay-dial          E & M Delay Dial
  e&m-fgd                 E & M Type II FGD
  e&m-immediate-start     E & M Immediate Start
  e&m-lmr                 E & M land mobil radio
  e&m-wink-start          E & M Wink Start
  ext-sig                 External Signaling
  fgd-eana                FGD-EANA BOC side
  fgd-os                  FGD-OS BOC side
  fxo-ground-start        FXO Ground Start
  fxo-loop-start          FXO Loop Start
  fxs-ground-start        FXS Ground Start
  fxs-loop-start          FXS Loop Start
  none                    Null Signalling for External Call Control
  <cr>
```

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

A. e&m-wink-start

B. e&m-delay-dial

C. e&m-delay-dial

D. e&m-lmr

E. e&m-fgd

Answer: E

Explanation

Because it can receive ANI information and sends DNIS info. But can’t send ANI.

Reference:

http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/isdn/configuration/15-mt/vi-15-mt-book/vi-imp-t1cas-v

Question #:596 - (Exam Topic 5)

Refer to the exhibit.

```
!
ephone-dn 1
 number 1001
!
ephone-dn 2
 number 1002
!
ephone-dn 3
 number 1003
 ephone-hunt login
!
ephone-dn 4
 number 1004
!
ephone-dn 5
 number 1005
 ephone-hunt login
!
ephone-dn 6
 number 1006
!
ephone-hunt 1 peer
 list 1001,1002,1004,*
 hop 6
 final 1100
!
```

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

Answer: C

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/reference/cme_cr/cme_e1ht.html

Question #:597 - (Exam Topic 5)

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

Answer: D

Explanation

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 as an alternative on the PVDM3

Reference:
http://www.cisco.com/c/en/us/td/docs/routers/access/1900/software/configuration/guide/Software_Configuratior
_config.html

Question #:598 - (Exam Topic 5)

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

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:

dspfarm profile 1 conference video guaranteed-audio codec h264 vga

codec h264 4cif

Reference:

http://www.cisco.com/c/en/us/products/collateral/unified-communications/voice-video-conferencing-isr-routers/qa_c67-649850.html

Question #:599 - (Exam Topic 5)

Refer to the exhibit.

```
Outgoing SIP UDP message to 10.1.1.1:[5060]:

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.1.1.1:5060;branch=z9hG4bK078cE1C7A
From:"Unknown" ;tag=2349872847981
To: "SBC" ;tag=2349872938479
Date: Tue, 11 Dec 2012 15:08:29 GMT
Call-ID: 234098d123147652@20.1.1.1
CSeq: 104 OPTIONS
WWW-Authenticate: Digest realm="StandAloneCluster", nonce="sdflakjdfjklahsfhkhq", algorithm=MD5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Content-Length: 0
```

The exhibit shows an outgoing SIP 401 response message from Cisco Unified Communications Manager to a SIP VoIP service provider gateway. Which action can the Cisco Unified Communications Manager systems administrator use to change the response to "200 OK"?

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.

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_0(

Question #:600 - (Exam Topic 5)

Refer to the exhibit.

```
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
```

Assuming this NFAS-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

Answer: B

Explanation

TEI_ASSIGNED, which indicates that the PRI does not exchange Layer 2 frames with the switch. Use the show controller t1 *x* 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 #:601 - (Exam Topic 5)

Refer to the exhibit.

```
voice translation-rule 2
   rule 1 /^500..$/ /40877\0/
   rule 2 /^5..[0-9]$/ /+1408777\0/
   rule 3 /^500....$/ /1408777..../
voice translation-profile PSTN
   translate calling 2
```

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

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 & \0 means all set of match pattern. Thus, +14087775001.

Question #:602 - (Exam Topic 5)

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. :

Answer: D

Explanation

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 #:603 - (Exam Topic 5)

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

Answer: A B

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:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/8_5_1/ccmcfg/bccm-851-cm/b06siprf.html

Question #:604 - (Exam Topic 5)

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.

Answer: A B

Explanation

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.

Reference: http://www.cisco.com
/c/en/us/td/docs/routers/asr1000/configuration/guide/sbcu/2_xe/sbcu_2_xe_book/sbc_h239.html

Question #:605 - (Exam Topic 5)

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

Answer: B

Explanation

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 #:606 - (Exam Topic 5)

Refer to the exhibit.

```
T1-CAS-Gateway(config-controller)#ds0-group 1 time 1-10 type ?
  e&m-delay-dial         E & M Delay Dial
  e&m-fgd                E & M Type II FGD
  e&m-immediate-start    E & M Immediate Start
  e&m-lmr                E & M land mobil radio
  e&m-wink-start         E & M Wink Start
  ext-sig                External Signaling
  fgd-eana               FGD-EANA BOC side
  fgd-os                 FGD-OS BOC side
  fxo-ground-start       FXO Ground Start
  fxo-loop-start         FXO Loop Start
  fxs-ground-start       FXS Ground Start
  fxs-loop-start         FXS Loop Start
  none                   Null Signalling for External Call Control
  <cr>
```

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

Answer: C

Explanation

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 #:607 - (Exam Topic 5)

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

Answer: C

Explanation

Table 6-2 DSP Resources on Cisco IOS Hardware Platforms with C5510 Chipset

Hardware Module or Chassis

DSP Configuration

Maximum Number of Voice Terminations (Calls) per DSP and per Module

Medium Complexity

(8 calls per DSP)

High Complexity

(6 calls per DSP)

Flex Mode1

(240 MIPS per DSP)

VG-224

Fixed at 4 DSPs

N/A

24 calls per platform

Supported codecs:

•G.711 (a-law, mu-law).

•G.729a.

N/A

NM-HD-1V2

Fixed at 1 DSP

4 calls per NM

4 calls per NM

240 MIPS per NM

NM-HD-2V

Fixed at 1 DSP

8 calls per NM

6 calls per NM

240 MIPS per NM

NM-HD-2VE

Fixed at 3 DSPs

24 calls per NM

18 calls per NM

720 MIPS per NM

NM-HDV2

NM-HDV2-2T1/E1

NM-HDV2-1T1/E1

1 to 4 of:

PVDM2-83(½ DSP)PVDM2-16 (1 DSP)PVDM2-32 (2 DSPs)PVDM2-48 (3 DSPs)PVDM2-64 (4 DSPs)

Calls per PVDM:

48162432

Calls per PVDM:

36121824

MIPS per PVDM:

120240480720960

Reference:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42media.html

Question #:608 - (Exam Topic 5)

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

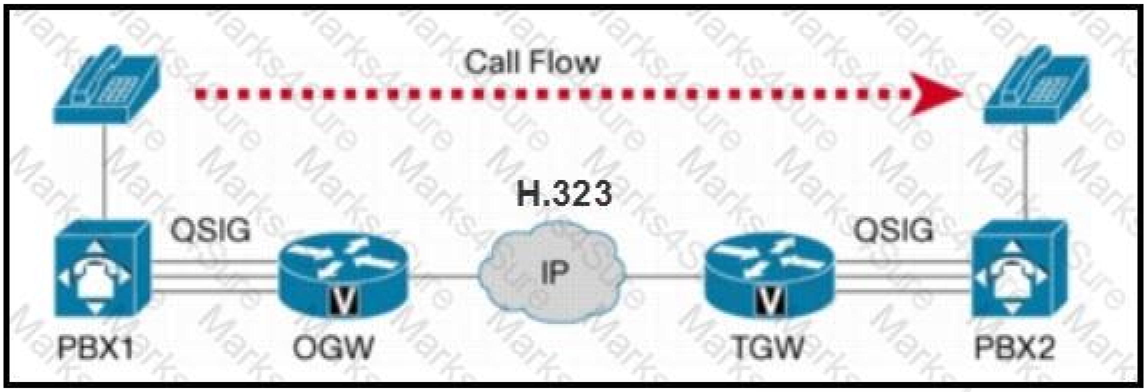
Answer: C

Explanation

Ephone is configured as octo line so maximum call number is 8 and it will be divided between lines.

Question #:609 - (Exam Topic 5)

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

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 ® 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.**

Reference:

http://www.cisco.com/c/en/us/solutions/collateral/enterprise-networks/empowered-branch-solution/white_paper_

Question #:610 - (Exam Topic 5)

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

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.

Reference: http://www.cisco.com/c/en/us/products/coll ateral/unified-communications/voice-video-conferencing-isr-routers/data_sheet_c78-649427.pdf

Question #:611 - (Exam Topic 5)

In Cisco IOS routers, which chipset is the PVDM-12 DSP hardware based on?

A. C542

B. C549

C. C5510

D. C5421

E. C5409

Answer: B

Explanation

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

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/7x/uc7_0/media.html

Question #:612 - (Exam Topic 5)

Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 huntstop channel 10
!
voice register pool 1
 id mac 1111.1111.1111
 type 7965
 number 1 dn 1
!
```

How many calls, inbound and outbound combined, are supported on the IP phone?

A. 1

B. 2

C. 8

D. 12

E. 50

Answer: E

Explanation

Output incomplete to figure out the answer

Question #:613 - (Exam Topic 5)

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

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 #:614 - (Exam Topic 5)

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 inbound calls can be handled simultaneously between ephone 1 and ephone 2 before a user busy tone is returned?

A. 6

B. 7

C. 8

D. 9

E. 11

Answer: A

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.

Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/command/ref erence/cme_cr/cme_e1ht.html

Question #:615 - (Exam Topic 5)

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

Answer: B D

Explanation

- **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.configureterminal

3.cardtypee1*slot*

4.controllere1*slot*/*port*

5.linecode {ami | hdb3}

6.framing {crc4 | no-crc4}

Reference:

http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/interface/c
onfiguration/12-4/ir-12-4-book/ir-12-port-chann-nm.html

Question #:616 - (Exam Topic 5)

Refer to the exhibit.

```
SIP-Gateway#show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED

SIP User Agent for TLS over TCP : ENABLED
SIP User Agent bind status(signaling): ENABLED  10.1.1.1
SIP User Agent bind status(media): ENABLED  10.1.1.1
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 70
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED
Reason Header will override Response/Request Codes: DISABLED
Out-of-dialog Refer: DISABLED
Presence support is DISABLED
protocol mode is ipv4

SDP application configuration:
 Version line (v=) required
 Owner line (o=) required
 Timespec line (t=) required
 Media supported: audio video image
 Network types supported: IN
 Address types supported: IP4 IP6
 Transport types supported: RTP/AVP udptl
```

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.

Answer: B

Explanation

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 an 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.

Reference:

http://www.cisco.com/c/en/us/td/docs/ios/voice/cube/configuration/guide/vb_book/vb_book/vb_1506.html

Question #:617 - (Exam Topic 5)

Refer to the exhibit.

```
!
voice register dn 1
 number 2001
 call-forward b2bua busy 2100
 call-forward b2bua noan 2200 timeout 20
 shared-line max-calls 4
 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.

Answer: C

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 #:618 - (Exam Topic 5)

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
!
```

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

Answer: C

Explanation

As we can see max-calls-per-button set to 5 and 3 calls are active. So, 2 calls remain.

Question #:619 - (Exam Topic 5)

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

Answer: B C

Explanation

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 specify 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 #:620 - (Exam Topic 5)

Refer to the exhibit.

```
%VOICE_IEC-3-GW: Application Framework Core: Internal Error (Toll fraud call rejected):
IEC=1.1.228.3.31.0 on callID 3 GUID=F146D6B0539C11DF800CA596C4C2D7EF
000183: *Jan 3 11:21:31.251: //3/F146D6B0800C/CCAPI/ccCallSetContext:
   Context=0x49EC9978
000184: *Jan 3 11:21:31.251: //3/F146D6B0800C/CCAPI/cc_process_call_setup_ind:
   >>>>CCAPI handed cid 3 with tag 1000 to app "_ManagedAppProcess_TOLLFRAUD_APP"
000185: **Jan 3 11:21:31.251: //3/F146D6B0800C/CCAPI/ccCallDisconnect:
   Cause Value=21, Tag=0x0, Call Entry(Previous Disconnect Cause=0, Disconnect Cause=0)
```

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.

Answer: D

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 #:621 - (Exam Topic 5)

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

Answer: A C

Explanation

Q. What video conferences are supported?

A. Three types of video profiles are supported: homogeneous conferences (video switching), heterogeneous conferences (video mixing), and guaranteed audio conferences (best-effort video).

Reference: http://www.cisco.com/c/en/us/product s/collateral/unified-communications/voice-video-conferencing-isr-routers/qa_c67-649850.html

Question #:622 - (Exam Topic 5)

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

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 #:623 - (Exam Topic 5)

Refer to the exhibit.

```
%Q.931 is backhauled to CCM MANAGER 0x0003 on DSL 0. Layer 3 output may not apply

ISDN Serial0/0/0:23 interface
        dsl 0, interface ISDN Switchtype = primary-ni
        L2 Protocol = Q.921 0x0000  L3 Protocol(s) = CCM MANAGER 0x0003
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = MULTIPLE_FRAME_ESTABLISHED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Active dsl 0 CCBs = 0
    The Free Channel Mask:  0x807FFFFF
    Number of L2 Discards = 0, L2 Session ID = 117
ISDN Serial0/2/1:23 interface
        dsl 1, interface ISDN Switchtype = primary-ni
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State = TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Active dsl 1 CCBs = 0
    The Free Channel Mask:  0x00000000
    Number of L2 Discards = 0, L2 Session ID = 0
Total Allocated ISDN CCBs = 0
```

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.

Answer: B D

Explanation

As you can see the T1 0/0/0:23 interface is active in layer 1,2(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

Question #:624 - (Exam Topic 5)

Refer to the exhibit.

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

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.

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.

Answer: E

Explanation

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 #:625 - (Exam Topic 5)

Refer to the exhibit.

```
!
application
  service app-b-acd-aa
  param voice-mail 2220
  paramspace english index 1
  param max-time-call-retry 40
  param service-name app-b-acd
  param number-of-hunt-grps 1
  param drop-through-option 1
  paramspace english language en
  param handoff-string app-b-acd-aa
  param dial-by-extension-option 3
  param max-time-vm-retry 1
  param aa-pilot 5272000
  paramspace english location flash:
  param queue-overflow-extension 2003
  param second-greeting-time 10
  param drop-through-prompt _bacd_welcome.au
  param call-retry-timer 20
  !
  service app-b-acd
  param queue-len 2
  param aa-hunt1 2100
  param queue-manager-debugs 1
  param number-of-hunt-grps 1
  !
!
ephone-hunt 1 longest-idle
 pilot 2100
 list 2001, 2002
 timeout 10, 10
 final 2120
 statistics collect
!
```

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

Answer: B

Explanation

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

Question #:626 - (Exam Topic 5)

Refer to the exhibit.

```
!
application
  service app-b-acd-aa
  param voice-mail 2220
  paramspace english index 1
  param max-time-call-retry 40
  param service-name app-b-acd
  param number-of-hunt-grps 1
  param drop-through-option 1
  paramspace english language en
  param handoff-string app-b-acd-aa
  param dial-by-extension-option 3
  param max-time-vm-retry 1
  param aa-pilot 5272000
  paramspace english location flash:
  param queue-overflow-extension 2003
  param second-greeting-time 10
  param drop-through-prompt _bacd_welcome.au
  param call-retry-timer 10
  !
  service app-b-acd
  param queue-len 2
  param aa-hunt1 2100
  param queue-manager-debugs 1
  param number-of-hunt-grps 1
  !
!
ephone-hunt 1 longest-idle
 pilot 2100
 list 2001, 2002
 timeout 10, 10
 final 2120
 statistics collect
!
```

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

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?

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.

Answer: B

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.

Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/bacd/configuration/guide/cme40tcl/40bacd.html#wpmkr1105714

Question #:627 - (Exam Topic 5)

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

Answer: C

Explanation

Link:-http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/vcr1/vcr1-cr-book/vcr-c4.html

Question #:628 - (Exam Topic 5)

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.

Answer: A C

Explanation

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 #:629 - (Exam Topic 5)

Assume the IP address of Cisco Unity Express is 10.1.1.1. 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/

Answer: C

Explanation

For

user access cisco unity has predefined url and it is http://10.1.1.1/user

Question #:630 - (Exam Topic 5)

Refer to the exhibit.

```
!
application
  service app-b-acd-aa
  param voice-mail 2220
  paramspace english index 1
  param max-time-call-retry 40
  param service-name app-b-acd
  param number-of-hunt-grps 1
  param drop-through-option 1
  paramspace english language en
  param handoff-string app-b-acd-aa
  param dial-by-extension-option 3
  param max-time-vm-retry 1
  param aa-pilot 5272000
  paramspace english location flash:
  param queue-overflow-extension 2003
  param second-greeting-time 10
  param drop-through-prompt _bacd_welcome.au
  param call-retry-timer 10
  !
  service app-b-acd
  param queue-len 2
  param aa-hunt1 2100
  param queue-manager-debugs 1
  param number-of-hunt-grps 1
  !
!
ephone-hunt 1 longest-idle
 pilot 2100
 list 2001, 2002
 timeout 10, 10
 final 2120
 statistics collect
!
```

Assume the B-ACD configuration on a Cisco IOS Cisco Unified Communications Manager Express router is operational. What will happen to a new call that enters the call queue when there are already two calls in queue?

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.

Answer: C

Explanation

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

Question #:631 - (Exam Topic 5)

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

Answer: B D G

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 end-user experiences. Both homogenous and heterogonous 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.

Reference: http://www.cisco.com/c/en/us/products/collateral/unified-com munications/voice-video-conferencing-isr-routers/data_sheet_c78-649427.html

Question #:632 - (Exam Topic 5)

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

Answer: C

Explanation

Each T1 channel carries a sequence of frames. These frames consist of 192 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.

Reference:

http://www.cisco.com/c/en/us/support/docs/voice/digital-cas/22444-t1-cas-ios.html

Question #:633 - (Exam Topic 5)

Refer to the exhibit.

```
Router#show dial-peer voice sum
dial-peer hunt 0
             AD                                 PRE PASS                    OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN    FER THRU SESS-TARGET      STAT PORT     KEEPALIVE

4300   voip  up   up             4...              0 syst ipv4:10.1.1.4                   active
2300   voip  up   up             [2-3]...          0 syst ipv4:10.1.1.3                   active
1111   voip  up   down           1111              0 syst ipv4:10.1.1.1                   busy-out
20001  pots  up   up             2001$             0                           50/0/1
20002  pots  up   up             2002$             0                           50/0/2
```

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

D. 486 Busy Here

E. 503 Service Unavailable

Answer: E

Explanation

SIP 503 Service Unavailable is commonly seen in a VoIP network when a SIP device (such as a SIP server) is knowingly unable to process a call. Typically when this happens the endpoint that originated the Invite will try the next available host it receives in the SIP Contact header.

Question #:634 - (Exam Topic 5)

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

Answer: D

Explanation

- **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.

•QSIG FACILITY/NOTIFY/INFO.

<—>

SIP INFO

•QSIG SETUP.

<—>

SIP INVITE

•QSIG ALERTING.

<—>

SIP 180 RINGING

•QSIG PROGRESS.

<—>

SIP 183 PROGRESS

•QSIG CONNECT.

<—>

SIP 200 OK

•QSIG DISCONNECT.

<—>

SIP BYE/CANCEL/4*xx*—6*xx* Response

Reference:

http://www.cisco.com/c/en/us/td/docs/ios/voice/sip/configuration/guide/15_0/sip_15_0_book/tunneling_qsig.htm

Question #:635 - (Exam Topic 5)

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.

Answer: E

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 #:636 - (Exam Topic 5)

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

Answer: A

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.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmesnr.html

Question #:637 - (Exam Topic 5)

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.

Answer: A C

Explanation

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.

Reference:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/command/reference/cme_v1ht.html

Question #:638 - (Exam Topic 5)

Refer to the exhibit.

```
!
application
  service app-b-acd-aa
  param voice-mail 2220
  paramspace english index 1
  param max-time-call-retry 40
  param service-name app-b-acd
  param number-of-hunt-grps 1
  param drop-through-option 1
  paramspace english language en
  param handoff-string app-b-acd-aa
  param dial-by-extension-option 3
  param max-time-vm-retry 1
  param aa-pilot 5272000
  paramspace english location flash:
  param queue-overflow-extension 2003
  param second-greeting-time 10
  param drop-through-prompt _bacd_welcome.au
  param call-retry-timer 10
  !
  service app-b-acd
  param queue-len 2
  param aa-hunt1 2100
  param queue-manager-debugs 1
  param number-of-hunt-grps 1
  !
!
ephone-hunt 1 longest-idle
 pilot 2100
 list 2001, 2002
 timeout 10, 10
 final 2120
 statistics collect
!
```

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.

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 #:639 - (Exam Topic 5)

Refer to the exhibit.

```
Branch-Router#debug ip dhcp packet

*Aug 15 22:13:23.924: DHCPD: Finding a relay for client 01ec.4476.1e3e.7d on interface Vlan101.
*Aug 15 22:13:23.924: DHCPD: setting giaddr to 10.101.15.1.
*Aug 15 22:13:23.924: DHCPD: BOOTREQUEST from 01ec.4476.1e3e.7d forwarded to 10.100.1.1.
*Aug 15 22:13:23.940: DHCPD: forwarding BOOTREPLY to client ec44.761e.3e7d.
*Aug 15 22:13:23.940: DHCPD: broadcasting BOOTREPLY to client ec44.761e.3e7d.

Branch-Router#
```

The exhibit shows the Cisco IOS CLI output of debug ipdhcp 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.

Answer: D F

Explanation

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.

Topic 6, Quality of Service and Security in Cisco Collaboration Solutions

Question #:640 - (Exam Topic 6)

To which QoS tool category does compressed RTP belong?

A. classification

B. marking

C. link efficiency

D. queuing

E. prioritization

Answer: C

Explanation

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 #:641 - (Exam Topic 6)

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

Answer: D

Explanation

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 #:642 - (Exam Topic 6)

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

Answer: D

Explanation

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 sends packets 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 #:643 - (Exam Topic 6)

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. 274.7 kb/s

E. 238.4 kb/s

Answer: A

Reference:

http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a0080094ae2.shtml

Question #:644 - (Exam Topic 6)

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.

Answer: C E

Explanation

When using TLS 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 #:645 - (Exam Topic 6)

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

Answer: D

Reference:

http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a0080094ae2.shtml

Question #:646 - (Exam Topic 6)

Refer to the exhibit.

```
!
class-map match-any signal
 match ip dscp cs3
class-map match-any rtp
 match ip dscp ef
!
policy-map VoIP
 class rtp
    bandwidth percent 33
    compress header ip rtp
 class signal
    bandwidth percent 5
 class class-default
    fair-queue
!
interface serial 0/1/0
 service-policy output VoIP
!
```

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

Answer: B

Explanation

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

Question #:647 - (Exam Topic 6)

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

Answer: B

Explanation

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

Question #:648 - (Exam Topic 6)

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

Answer: C

Explanation

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 #:649 - (Exam Topic 6)

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

Answer: C

Explanation

Enterprise Medianet QoS Recommendations

Application Class	Per-Hop Behavior	Admission Control	Queuing and Dropping	Media Application Examples
VoIP Telephony	EF	Required	Priority Queue (PQ)	Cisco IP Phones (G.711, G.729)
Broadcast Video	CS5	Required	(Optional) PQ	Cisco IP Video Surveillance/Cisco Enterprise TV
Real-Time Interactive	CS4	Required	(Optional) PQ	Cisco TelePresence
Multimedia Conferencing	AF4	Required	BW Queue + DSCP WRED	Cisco Unified Personal Communicator
Multimedia Streaming	AF3	Recommended	BW Queue + DSCP WRED	Cisco Digital Media System (VoDs)
Network Control	CS6		BW Queue	EIGRP, OSPF, BGP, HSRP, IKE
Signaling	CS3		BW Queue	SCCP, SIP, H.323
Ops/Admin/Mgmt (OAM)	CS2		BW Queue	SNMP, SSH, Syslog
Transactional Data	AF2		BW Queue + DSCP WRED	Cisco WebEx/MeetingPlace/ERP Apps
Bulk Data	AF1		BW Queue + DSCP WRED	E-mail, FTP, Backup Apps, Content Distribution
Best Effort	DF		Default Queue + RED	Default Class
Scavenger	CS1		Min BW Queue	YouTube, iTunes, BitTorrent, Xbox Live

Reference:

http://www.cisco.com/c/en/us/td/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND_40/QoSIntro_40.html

Question #:650 - (Exam Topic 6)

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.

Answer: B

Explanation

TLS Proxy is typically deployed in front of Cisco Unified Communications Manager and other unified communications application servers that utilize media encryption.TLS Proxy is not designed to provide remote-access encryption services for remote phones or client endpoints. Other solutions such as Cisco ASA Phone Proxy or IP Security/Secure Sockets Layer (IPsec/SSL) VPN services are more appropriate. TLS Proxy is not designed to provide a secure campus soft phone solution where the requirement is to provide secure data to phone VLAN traversal or for proxying connections to Cisco Unified Communications Manager.

Question #:651 - (Exam Topic 6)

Refer to the exhibit.

```
!
class-map match-any signal
 match ip dscp cs3
class-map match-any rtp
 match ip dscp ef
!
policy-map VoIP
 class rtp
    bandwidth percent 33
    compress header ip rtp
 class signal
    bandwidth percent 5
 class class-default
    fair-queue
!
interface serial 0/1/0
 service-policy output VoIP
!
```

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

Answer: E

Explanation

Since the use of the “priority” keyword was not used in this example 0% is the correct answer.

Question #:652 - (Exam Topic 6)

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.

Answer: D

Explanation

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 #:653 - (Exam Topic 6)

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

Answer: D

Explanation

By default, LSC certificates are not installed on Cisco IP phones. Cisco IP phones that are required to use LSC certificates must be provisioned to allow TLS transactions before deployment in the field. LSC certificates can be provisioned to the Cisco IP phones through the Certificate Authority Proxy Function (CAPF) process. This process is completed using TLS and USB tokens coupled with the CTL client. Moreover, the Cisco ASA Phone Proxy feature can serve LSC certificates to the Cisco IP phones. Cisco IP phones will only work with the Cisco ASA Phone Proxy and will not establish secure connectivity with the Cisco Unified Communications Manager.

Question #:654 - (Exam Topic 6)

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

Answer: B

Explanation

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 #:655 - (Exam Topic 6)

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

Answer: A

Explanation

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 #:656 - (Exam Topic 6)

Which two statements describe security services that are provided by the Phone Proxy function on a Cisco ASA appliance? (Choose two.)

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.

Answer: D E

Explanation

TLS Proxy is typically deployed in front of Cisco Unified Communications Manager and other unified communications application servers that utilize media encryption. TLS Proxy is not designed to provide remote-access encryption services for remote phones or client endpoints. Other solutions such as Cisco ASA Phone Proxy or IP Security/Secure Sockets Layer (IPsec/SSL) VPN services are more appropriate. TLS Proxy is not designed to provide a secure campus soft phone solution where the requirement is to provide secure data to phone VLAN traversal or for proxying connections to Cisco Unified Communications Manager.

Question #:657 - (Exam Topic 6)

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.

Answer: C

Explanation

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 number 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 ASAs 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 Designs (SRND) page 4-34

Question #:658 - (Exam Topic 6)

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

Answer: C

Explanation

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 304 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.

Topic 7, Cisco Unity Connection

Question #:659 - (Exam Topic 7)

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.

Answer: B

Explanation

Default call handler is selected when we don’t assign any call handler to user and with this default call handler no specific user assigned so it don’t go to any specific mail box and goes to It will be sent to the Undeliverable Messages distribution list

Question #:660 - (Exam Topic 7)

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.

Answer: B

Explanation

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 secresmapclient.

Question #:661 - (Exam Topic 7)

Refer to the exhibit.

```
Jan 29 17:32:01.723: CRYPTO_PKI: (A0076) Starting CRL revocation check
Jan 29 17:32:01.723: CRYPTO_PKI: Matching CRL not found
Jan 29 17:32:01.723: CRYPTO_PKI: (A0076) CDP does not exist. Use SCEP to
query CRL.
Jan 29 17:32:01.723: CRYPTO_PKI: pki request queued properly
Jan 29 17:32:01.723: CRYPTO_PKI: Revocation check is complete, 0
Jan 29 17:32:01.723: CRYPTO_PKI: Revocation status = 3
Jan 29 17:32:01.723: CRYPTO_PKI: status = 0: poll CRL
Jan 29 17:32:01.723: CRYPTO_PKI: Remove session revocation service providers
CRYPTO_PKI: Bypassing SCEP capabilies request 0
Jan 29 17:32:01.723: CRYPTO_PKI: status = 0: failed to create GetCRL
Jan 29 17:32:01.723: CRYPTO_PKI: enrollment url not configured
Jan 29 17:32:01.723: CRYPTO_PKI: transaction GetCRL completed
Jan 29 17:32:01.723: CRYPTO_PKI: status = 106: Blocking chain verification
callback received status
Jan 29 17:32:01.723: CRYPTO_PKI: (A0076) Certificate validation failed
```

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.

Answer: C

Explanation

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. CAs 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 #:662 - (Exam Topic 7)

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.

Answer: E

Explanation

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 #:663 - (Exam Topic 7)

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

Answer: A B E

Explanation

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 #:664 - (Exam Topic 7)

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

Answer: C D

Explanation

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.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/unity_exp/administrator/AA_and_VM/guide/vmadmin_b

Question #:665 - (Exam Topic 7)

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.

Answer: B

Explanation

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 #:666 - (Exam Topic 7)

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

A. holiday

B. closed

C. internal

D. busy

E. alternate

Answer: E

Explanation

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 #:667 - (Exam Topic 7)

Which three Cisco Unity Connection call handler greetings can be overridden by the internal greeting? (Choose three.)

A. holiday

B. alternate

C. error

D. busy

E. closed

F. standard

Answer: A E F

Explanation

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 #:668 - (Exam Topic 7)

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

Answer: A

Explanation

As soon as unity finds the unexpected behavior it prompts the error message to the user.

Question #:669 - (Exam Topic 7)

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.

Answer: C

Explanation

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 #:670 - (Exam Topic 7)

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.

Answer: A

Explanation

When installed on the same server there is no way to create trunk that is why sccp is the only way Cisco Unity Connection and Cisco Unified Communications Manager are installed on the same server.

Question #:671 - (Exam Topic 7)

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.

Answer: D

Explanation

Voice messages queue for delivery in the Voice Outbox folder that is why it shows in Deleted Items folder.

Question #:672 - (Exam Topic 7)

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

Answer: A B

Explanation

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.

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/8x/administration/guide/8xcucsagx/8xcucsag2

Question #:673 - (Exam Topic 7)

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

Answer: A D

Explanation

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 phonenumber which is dialed to invoke the desired application..

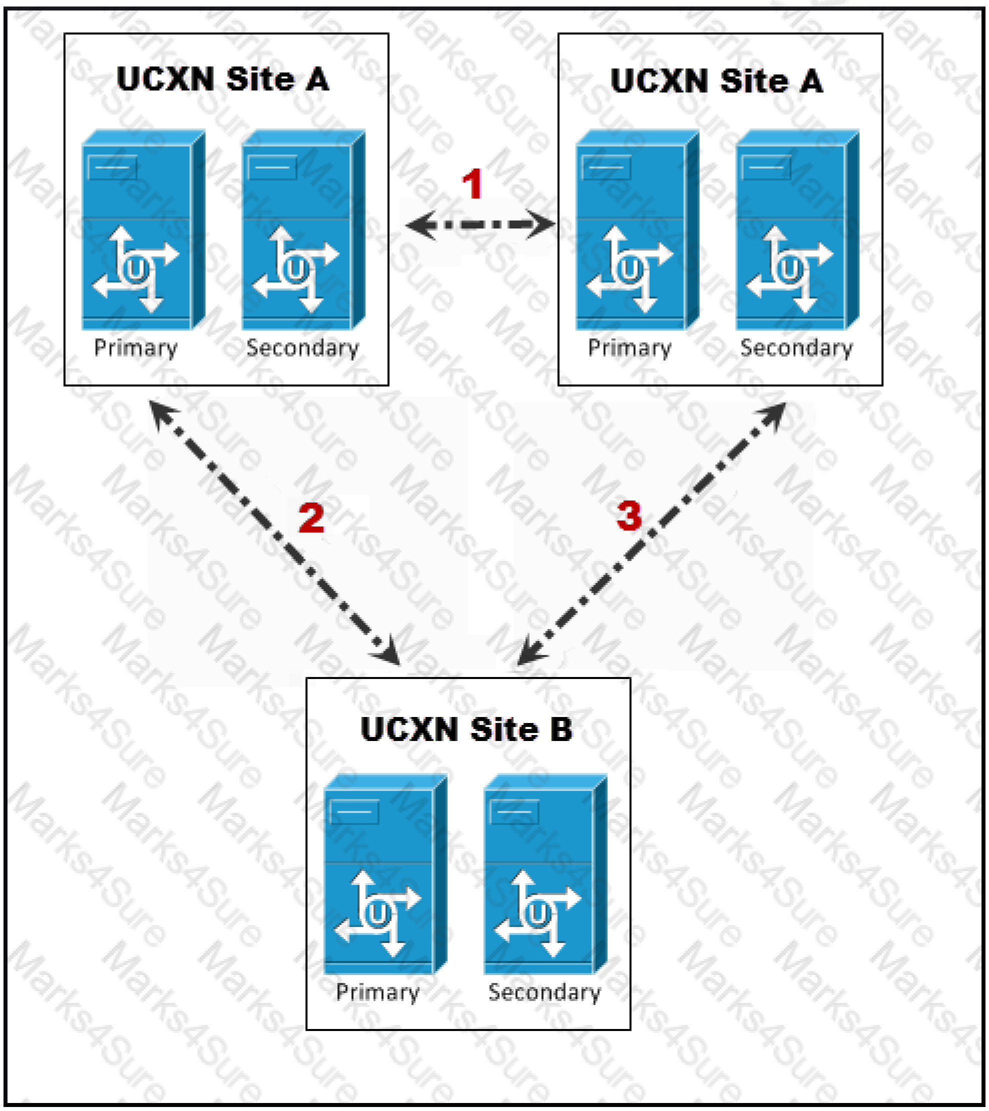
•JTAPI triggers — Use this type of trigger to invoke applications in Cisco Unified Communications Manager mode. This type of trigger is identified by the phonenumber which is dialed to invoke the desired application..

•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 IVR license has been purchased and installed on the system..

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/unity_exp/administrator/AA_and_VM/guide/vmadmin_b .html#wp1126512

Question #:674 - (Exam Topic 7)

Refer to the exhibit.



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

Answer: A

Explanation

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.

Reference:
http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/8x/networking/guide/8xcucnetx/8xcucnet010.html

Question #:675 - (Exam Topic 7)

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.

Answer: D

Explanation

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 #:676 - (Exam Topic 7)

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.

Answer: A B

Explanation

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, 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.

Reference: http://www.cisco.com/
c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmesnr.html#pgfId-1012065

Question #:677 - (Exam Topic 7)

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

Answer: A

Explanation

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.

Topic 8, Cisco Unified Contact Center Express

Question #:678 - (Exam Topic 8)

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

Answer: A

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).

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cust_contact/contact_center/crs/express_9_02/design/gui

Question #:679 - (Exam Topic 8)

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.

Answer: A C

Explanation

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

- Choose **Device**>**Phone** in Unified Communications Manager Administration. The Find and List Phones window is displayed.

Enter search criteria to locate a specific phone and click **Find**. A list of phones that match the search criteria is displayed. Click the device name of the phone to which you want to add a directory number. The Phone Configuration window is displayed.

In the Unified Communications Manager Administration Phone Configuration web page, select the required Association Information (on the left) to get to the Directory Number Configuration web page. On this page, make the following changes:

- In the Multiple Call/Call Waiting Settings section, set the Maximum Number of Calls to 2 (default is 4) for Cisco Unified IP Phones 7900 Series and 3 for Cisco Unified IP Phones 8961, 9951, and 9971.
- In the Multiple Call/Call Waiting Settings section, set the Busy Trigger value to 1 (default is 2).
- In the Call Forward and Call Pickup Settings section, verify that you do not forward any Unified Communications Manager device to the Unified CCX extension of an agent.
- In the Call Forward and Call Pickup Settings section, verify that you do not configure the Unified CCX extension of an agent to forward to a Unified CCX route point.
- 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 Phone 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 Securities 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.
- Do not forward any Unified Communications Manager device to the Unified CCX extension of an agent.
- Do not configure the Unified CCX extension of an agent to forward to a Unified CCX route point.
- Do not use characters other than the numerals 0 to 9 in the Unified CCX extension of an agent.
- Do not configure two lines on an agent phone with the same extension when both lines exist in different partitions.
- Do not assign a Unified CCX extension to multiple devices.
- 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.)
- To use Cisco Unified IP Phones 9900 Series, 8900 Series, and 6900 Series as agent devices, the RmCm application user in Unified Communications Manager needs to have "Allow device with connected transfer/conference" option assigned to itself.

Question #:680 - (Exam Topic 8)

Which two statements describe the remote supervisory monitoring feature in Cisco Unified Contact Center Express? (Choose two.)

A. It is supported on Cisco Unified CCX Enhanced and Premium editions.

B. It does not require 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 supports G.711 and G.729 codecs.

F. It works with SPAN port monitoring only.

Answer: B D

Explanation

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 #:681 - (Exam Topic 8)

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

Answer: B

Explanation

Unified CCX applications might use auxiliary files that interact with callers, 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 #:682 - (Exam Topic 8)

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

D. Recording step

E. Simple Recognition step

Answer: B

Explanation

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 #:683 - (Exam Topic 8)

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. Instruct agents to use the Record button on Cisco IPPA to trigger recording.

B. Instruct supervisors to use the Record button on Cisco Agent Desktop to trigger recording.

C. Instruct supervisors to use the Record button on Cisco Supervisor Desktop to trigger recording.

D. Configure the Cisco Agent Desktop workflow to trigger recording.

E. Recording is not supported on the Cisco Unified CCX Enhanced version. It is supported only on the Premium version.

Answer: D

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 #:684 - (Exam Topic 8)

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 in the RTP payload based on RFC 2833.

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.

Answer: C

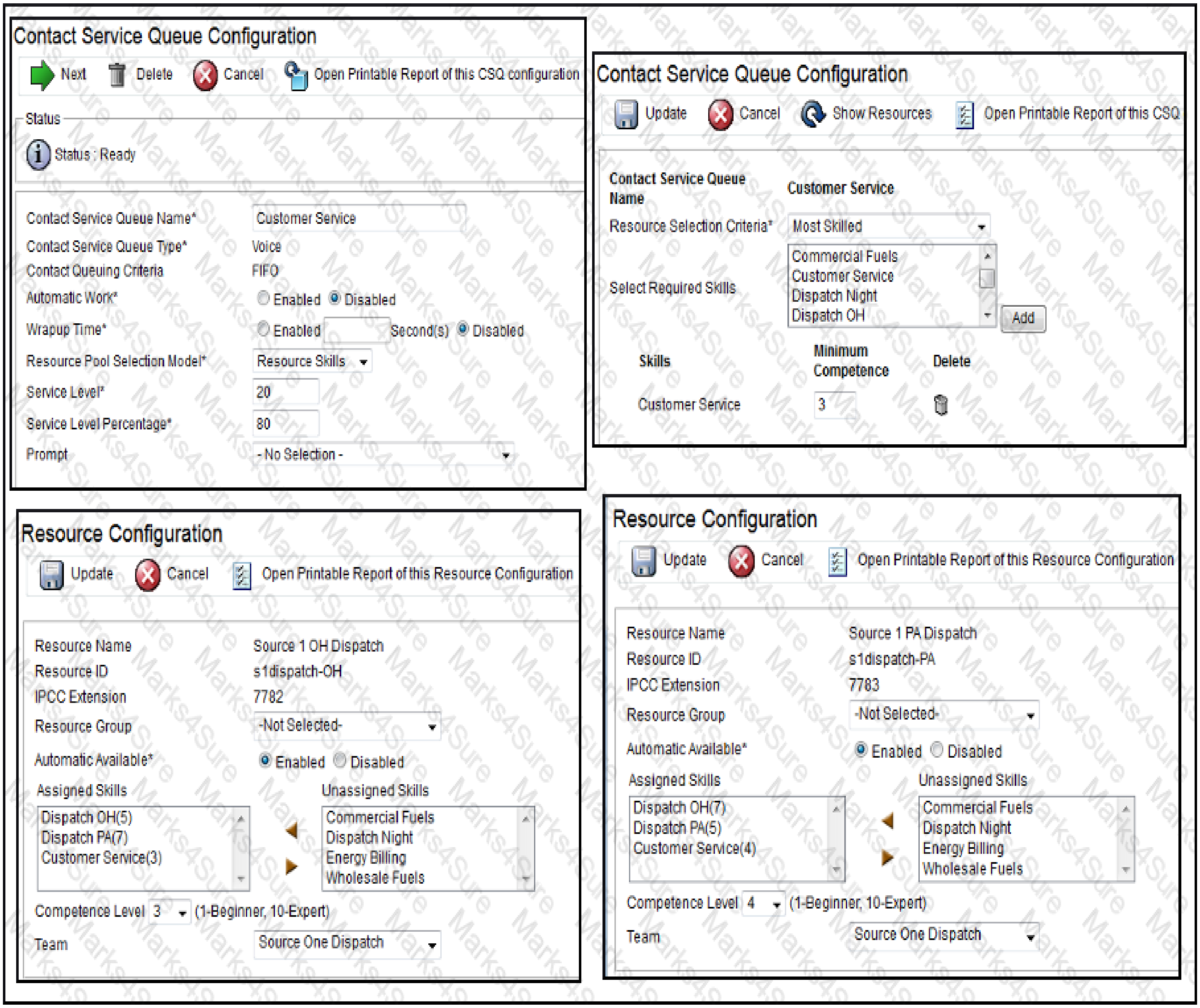
Explanation

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 or are configured to use in-band DTMF, an MTP resource must be invoked by Unified CM to convert the in-band DTMF signaling so that Unified CM can notify Unified CCX of the caller-entered digits. Be sure to enable out-of-band DTMF signaling when configuring voice gateways in order to avoid using the previous MTP resources.

Reference: http://www.cisco.com/en/US/docs/voice_ip_comm/cust_contact/contact_center/crs/express_9_0
2/design/guide/UCCX_BK_C39FDB35_00_cisco-unified-contact-center-express_chapter_010.html

Question #:685 - (Exam Topic 8)

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

Answer: A

Explanation

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 CSQs.

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, 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/cust_contact/contact_center/crs/express_10_0/configurati

Question #:686 - (Exam Topic 8)

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

Answer: D

Explanation

6 RTP streams exist when UCCE agent is engaged in a call when it is being silently monitored.

Reference: http://www.cisco.com/en/US/docs/voice_ip_comm/cust_contact/c
ontact_center/crs/express_9_0/design/UCCX_BK_UD5B347F_00_uccx-solution-reference-network-design_cha

Question #:687 - (Exam Topic 8)

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

Answer: B

Explanation

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 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.

Reference:

http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cust_contact/contact_center/crs/express_9_0/design/UCC

Question #:688 - (Exam Topic 8)

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.

Answer: D

Explanation

There is no mechanism created as of now to record the call so we first span and record it from packet capture or from third party software.

Question #:689 - (Exam Topic 8)

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

Answer: A

Explanation

The Unified CCX Engine enables you to run multiple applications to handle Unified CM Telephony calls or HTTP requests. The Unified CCX 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 Administration web interface: Used to configure system parameters, subsystems, view real-time reports that include total system activity and application statistics, and so on.
- 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.

Topic 9, Cisco Unified IM and Presence

Question #:690 - (Exam Topic 9)

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

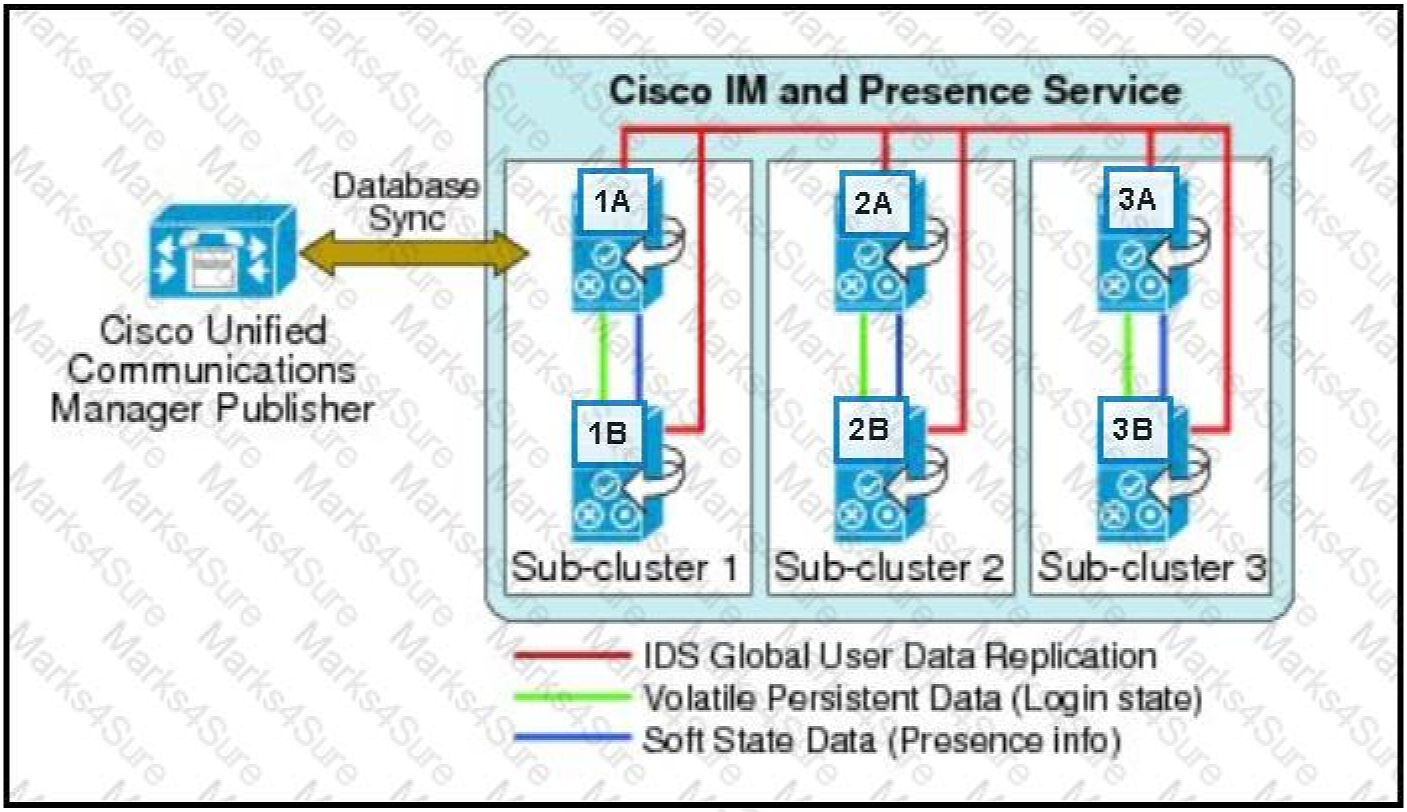
Answer: A B

Explanation

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 #:691 - (Exam Topic 9)

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 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

Answer: C

Explanation

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, as compared to a non-redundant 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 #:692 - (Exam Topic 9)

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.

Answer: B

Explanation

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 sub-cluster 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 #:693 - (Exam Topic 9)

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

Answer: E

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 adhoc (temporary) and permanent chat rooms for the Chat feature. Messages are logged to an external Cisco-supported database.

Question #:694 - (Exam Topic 9)

Which three issues prevent a customer from seeing the presence status of a new contact in their Jabber contact list? (Choose three.)

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.

Answer: B C D

Reference:http://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-presence/97443-cups-cupc-ts.html

Question #:695 - (Exam Topic 9)

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

Answer: C

Explanation

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 #:696 - (Exam Topic 9)

Which three statements about configuring partitioned intradomain federation to Lync are true? (Choose three.)

A. Intradomain federation to Lync is only possible using SIP.

B. IM&P and Lync should federate to any required remote domains.

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.

Answer: A C D

Explanation

Please refer to the link for more information: http://www.cisco
.com/c/en/us/td/docs/voice_ip_comm/cups/8_6/english/integration_notes/Federation/Intradomain_Federation/Pa

Question #:697 - (Exam Topic 9)

Which external database software is required for the Cisco IM and Presence compliance feature?

A. MySQL

B. EnterpriseDB

C. MSSQL

D. SQLite

E. PostgreSQL

Answer: E

Explanation

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..

Reference:
http://www.cisco.com/en/US/docs/voice_ip_comm/cups/8_0/english/install_upgrade/database/guide/Preparing_c
se_setup.html#wp1053954

Question #:698 - (Exam Topic 9)

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 Enable Application Level Authorization.

E. Configure the Outgoing Transport Type to TLS.

Answer: A B

Explanation

Configure SIP Trunk Security Profile for IM and Presence Service

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 CUP 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 to enable these items:

- **Accept Presence Subscription**
- Accept Out-of-Dialog REFER
- Accept Unsolicited Notification
- Accept Replaces 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_C

Question #:699 - (Exam Topic 9)

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

Answer: A D E

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 <new_domain>

This CLI command invokes a reboot of the servers.

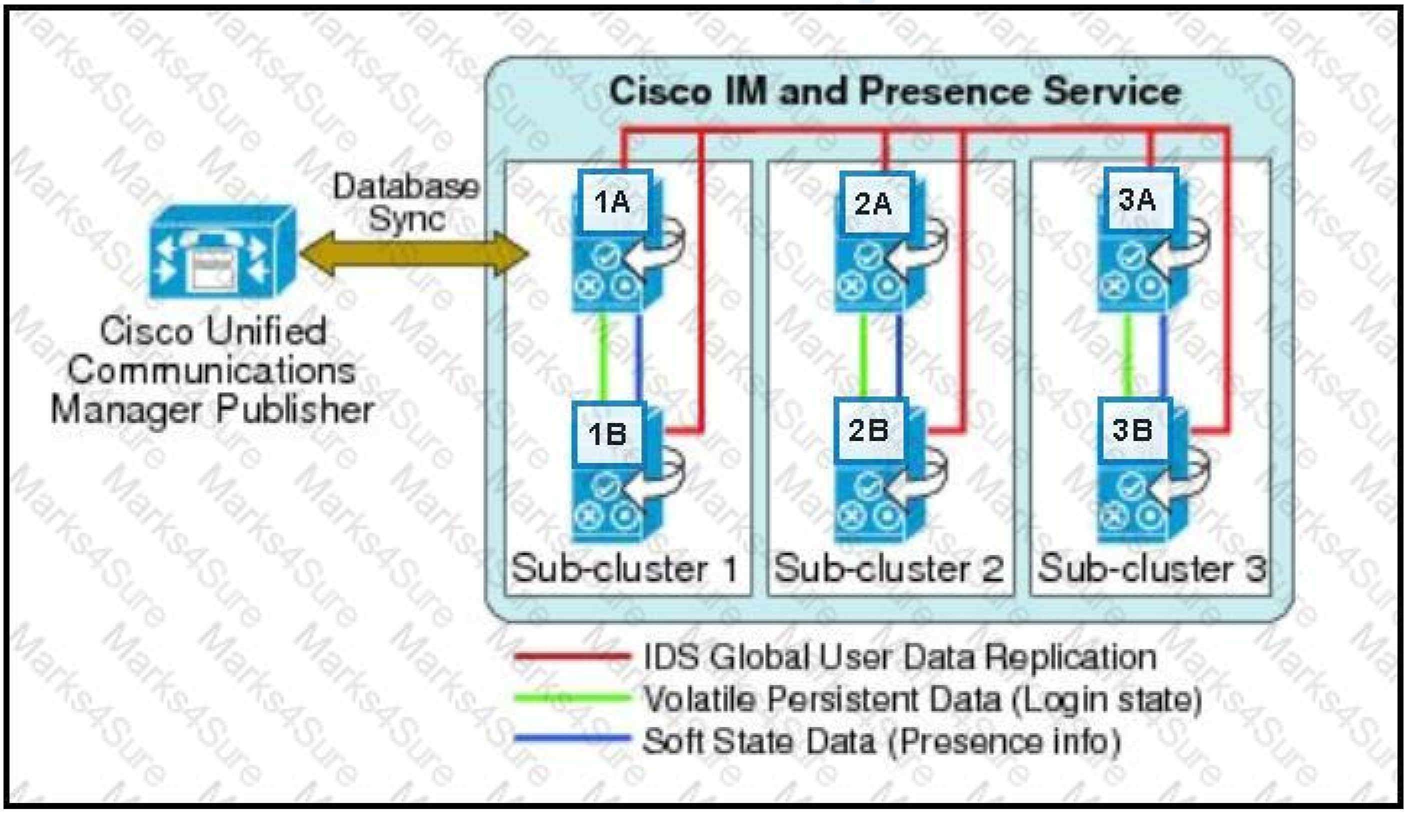
Step 4 On all nodes in the cluster, manually start the Cisco Presence Engine and Cisco XCP Router services after the reboot is complete (if required)..

Step 5 Manually regenerate all certificates on each node in the cluster..

Reference:http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/ip_address_hostname/9_0_

Question #:700 - (Exam Topic 9)

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

Answer: A

Explanation

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 #:701 - (Exam Topic 9)

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

Answer: B

Explanation

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 CUPC to reach the CTI gateway.

Question #:702 - (Exam Topic 9)

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

Answer: C

Explanation

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 #:703 - (Exam Topic 9)

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.

Answer: D

Explanation

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 #:704 - (Exam Topic 9)

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.

Answer: A

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 #:705 - (Exam Topic 9)

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?

A. Switch your Jabber client to use UDS instead of EDI.

B. Switch your Jabber client to use EDI instead of UDS.

C. Update your IM&P server to sync off of the same LDAP directory as your Cisco Unified Communications Manager.

D. Add Jabber to your inbound/outbound firewall rules on your PC.

E. Jabber can only pull directly from LDAP and cannot directly search the Cisco Unified Communications Manager user database.

Answer: A

Explanation

LDAP contact resolution — The client cannot use LDAP for contact resolution when outside of the corporate firewall. Instead, the client must use UDS for contact resolution.

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.

Reference: http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/
jabber/Windows/9_7/JABW_BK_C4C679C9_00_cisco-jabber-for-windows-97/JABW_BK_C4C679C9_00_cis