

Exam Questions 400-051

CCIE Collaboration

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NEW QUESTION 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
```

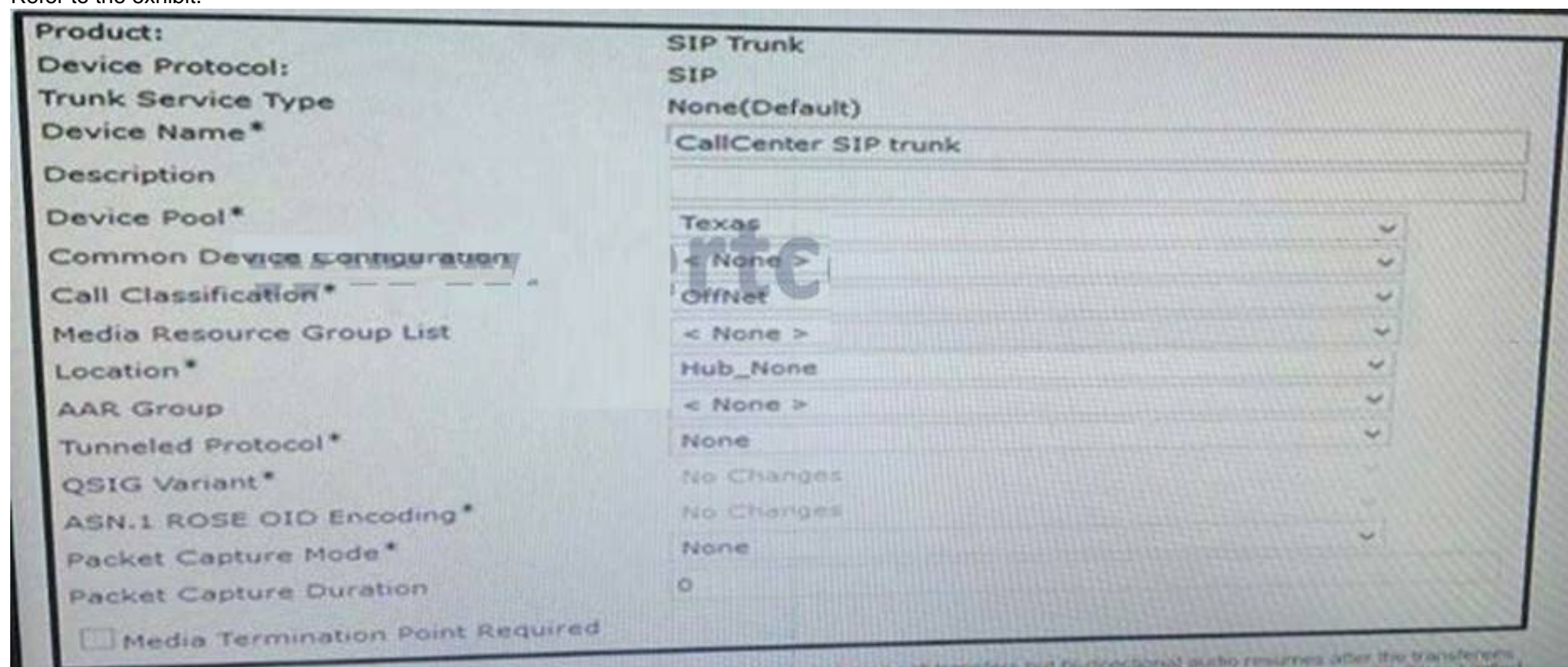
A Jabber for Windows application fails to connect to the voicemail server. Which two options cause this problem? (Choose two)

- 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: CE

NEW QUESTION 2

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: AB

NEW QUESTION 3

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

NEW QUESTION 4

ACisco 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: BCDG

NEW QUESTION 5

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

NEW QUESTION 6

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: AB

NEW QUESTION 7

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. urlinformation http://10.10.1.1/information/info.html
- B. urlservices http://10.10.1.1/voiceview/common/login.do
- C. urlauthentication http://10.10.1.1/CCMCIP/authenticate.asp
- D. url messages http://10.10.1.1/messages/common/login.do
- E. urlservices http://10.10.1.1/CMEUser/123456/urlsupport.html

Answer: BC

NEW QUESTION 8

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

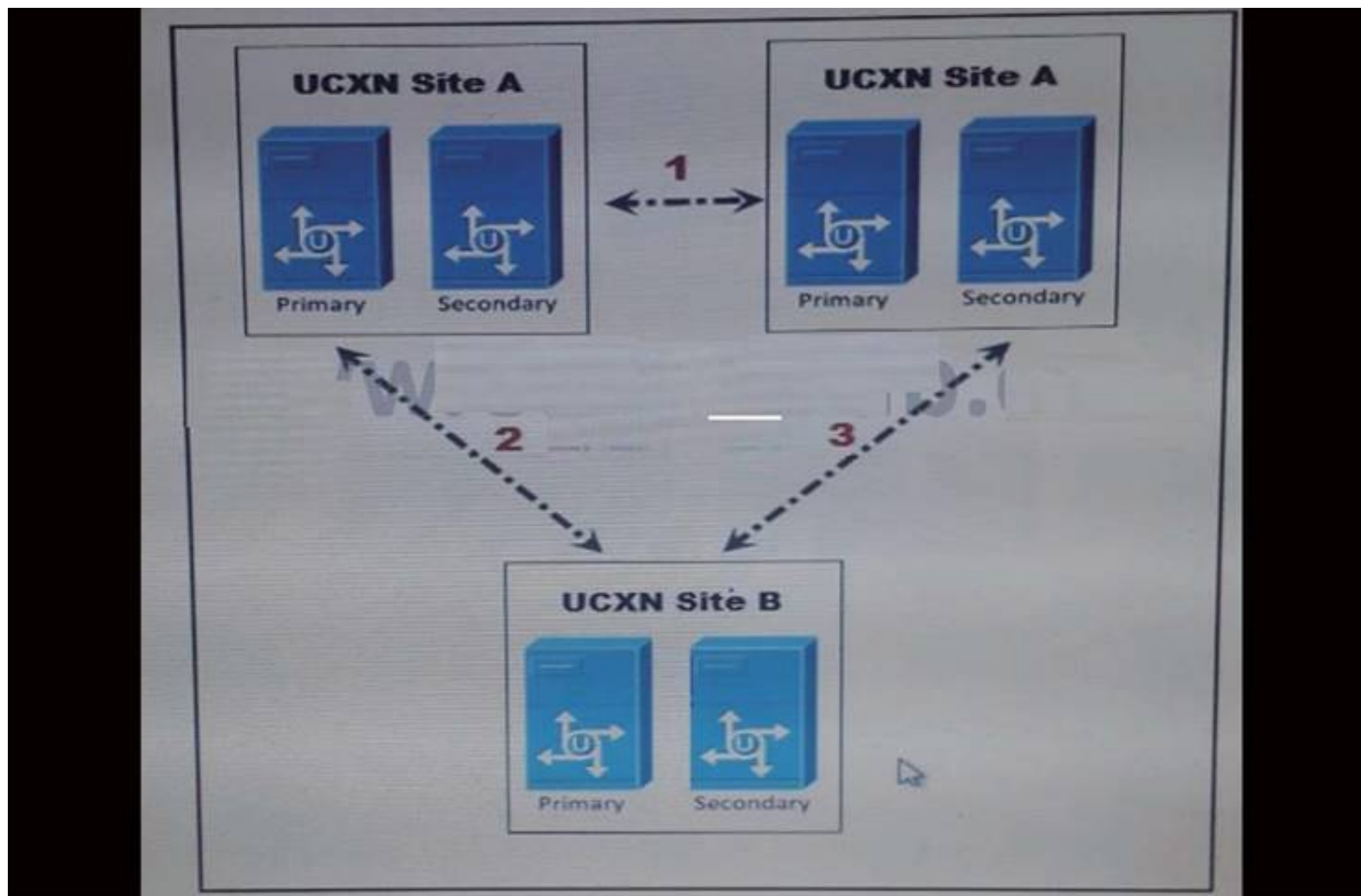
Which two sets of commands satisfy these requirements? (choose two)

- A. FastEthernet 0/1 mls qos trust cos
- B. FastEthernet 0/1switchport priority extend cos 0
- C. FastEthernet 0/1switchport priority extend trust
- D. FastEthernet 0/1 mls qos cos override
- E. FastEthernet 0/1 mls qos cos 0
- F. FastEthernet 0/1mls qos trust device cisco-phone

Answer: AB

NEW QUESTION 9

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 SMTP2 - HTTP/HTTPS, SMTP3 None
- B. 1 HTTP/HTTPS, SMTP 2 SMTP3 None
- C. 1 - HTTP/HTTPS, SMTP 2 - HTTP/HTTPS, SMTP3 - HTTP/HTTPS, SMTP
- D. 1 SMTP1 SMTP1 SMTP

Answer: A

NEW QUESTION 10

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

NEW QUESTION 10

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

NEW QUESTION 13

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: AC

NEW QUESTION 14

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: AB

NEW QUESTION 17

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

NEW QUESTION 19

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: AB

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#

NEW QUESTION 23

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: DEF

NEW QUESTION 27

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: AE

NEW QUESTION 32

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: AF

NEW QUESTION 34

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: DEF

NEW QUESTION 38

Refer to the exhibit.



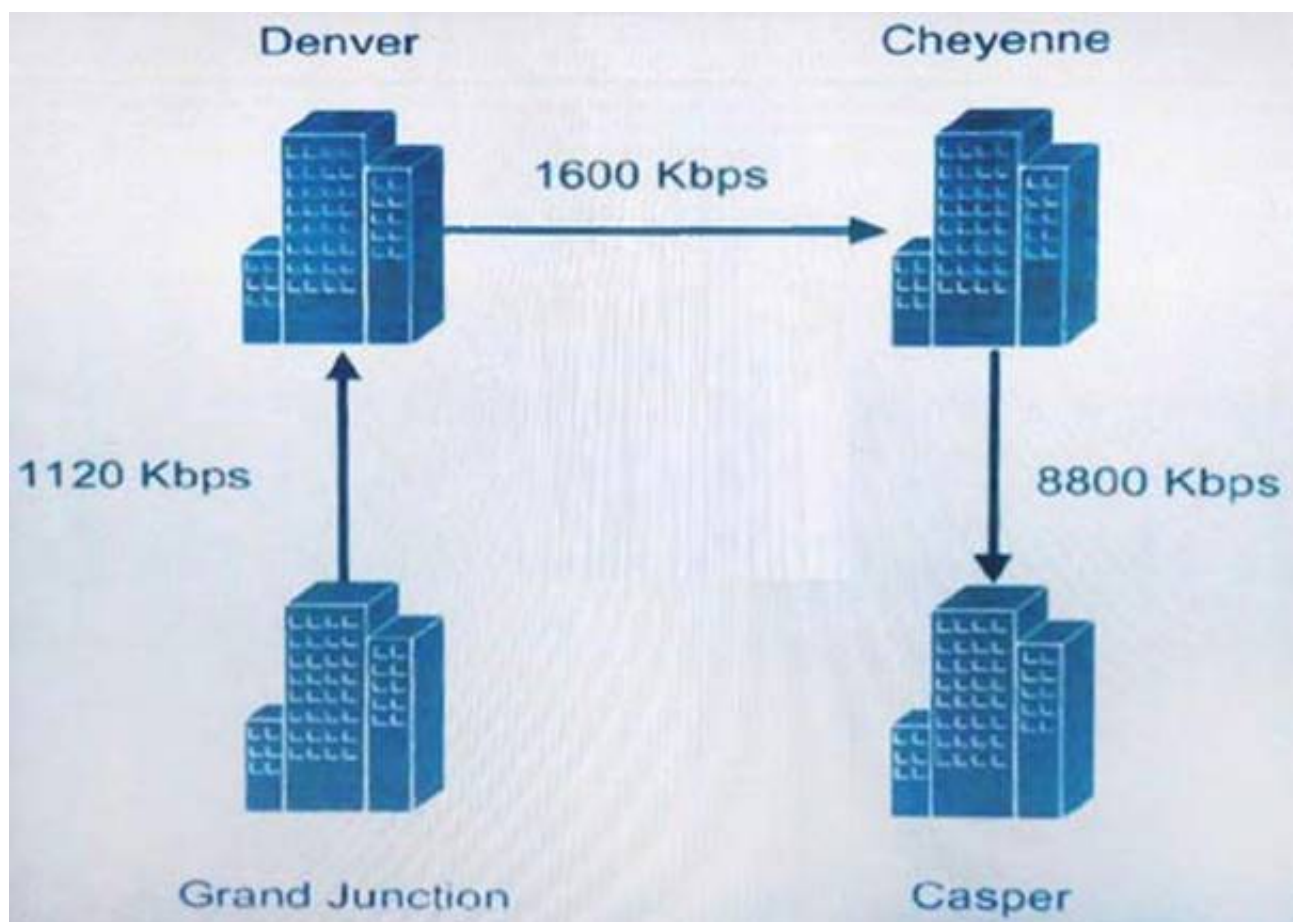
The SIP service Provider does not support re-invites unless media changes. Which two commands are needed on the CUBE to standby the SIP service provider requirements? (choose two)

- 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: DF

NEW QUESTION 43

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: CE

NEW QUESTION 48

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

NEW QUESTION 52

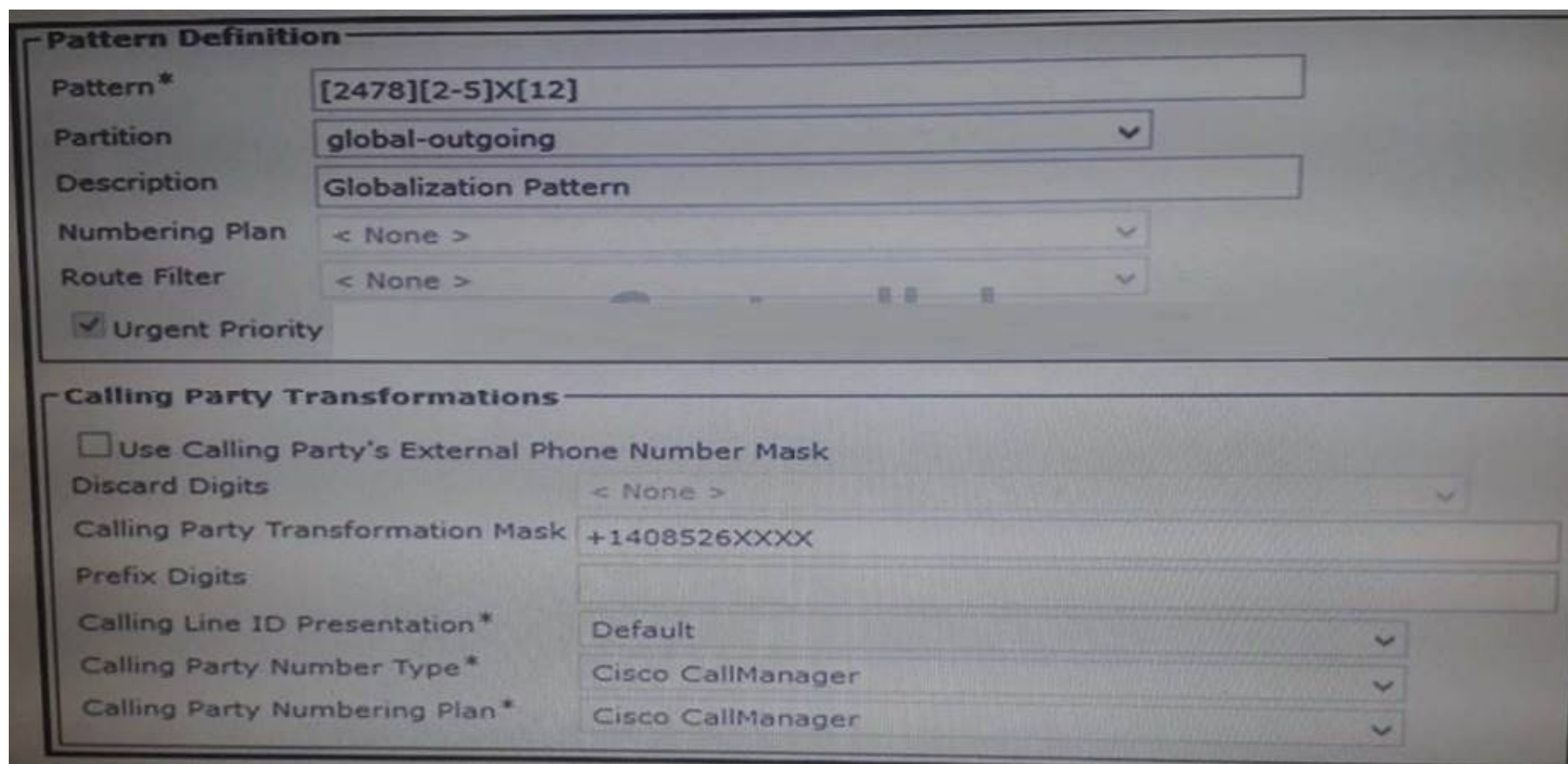
What is the minimum number of TCP sessions needed to complete a H323 call between two H323 gateways using slow start?

- A. 3
- B. 2
- C. 4
- D. 1

Answer: C

NEW QUESTION 56

Refer to the Exhibit,



Pattern Definition

Pattern* [2478][2-5]X[12]

Partition global-outgoing

Description Globalization Pattern

Numbering Plan < None >

Route Filter < None >

☒ Urgent Priority

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Discard Digits < None >

Calling Party Transformation Mask +1408526XXXX

Prefix Digits

Calling Line ID Presentation* Default

Calling Party Number Type* Cisco CallManager

Calling Party Numbering Plan* Cisco CallManager

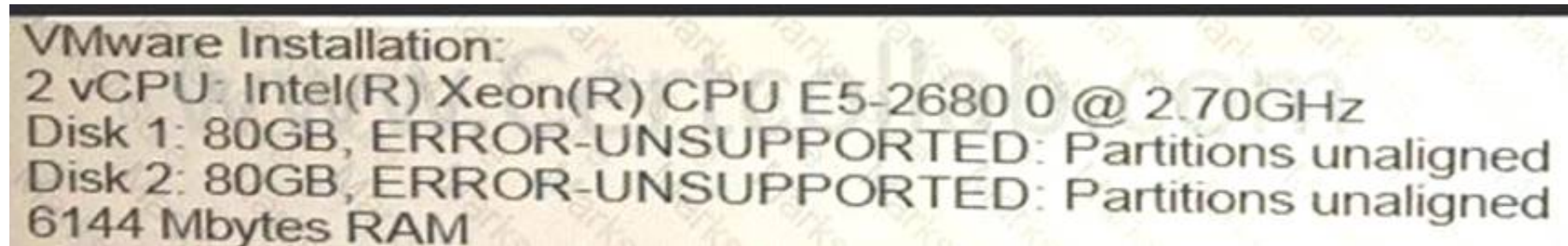
Which three cisco Unified CM internal extension match the globalization pattern shown and provide a globalized calling party number? (Choose three)

- A. 2401
- B. 2671
- C. 3392
- D. 4202
- E. 7352
- F. 8253

Answer: ADE

NEW QUESTION 57

Refer to the exhibit.



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: BEF

NEW QUESTION 62

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

- A. Cluster
- B. Distributed
- C. Centralized
- D. Grid

Answer: B

NEW QUESTION 66

Refer to the exhibit.

Toll Free Numbers: 1855, 1866, 1877, 1888					Gateway Name			Area Code	Max. No. of Ports
Condition	No. of Digits	Pattern	Call Type		Gateway Name			Area Code	Max. No. of Ports
1	5	1	On Net		MGLocal			914	1
2	7	1	Local		DOISUB001-6254e07W			625,914	24
3	10	T1	Others						
4	10	G1	Local						
5	10	1	Long Distance						
6	11	T1	Others						
7	11	MG1	Local						
8	11	MG1	Local						
9	11	1	Long Distance						
10	3	0111	International						

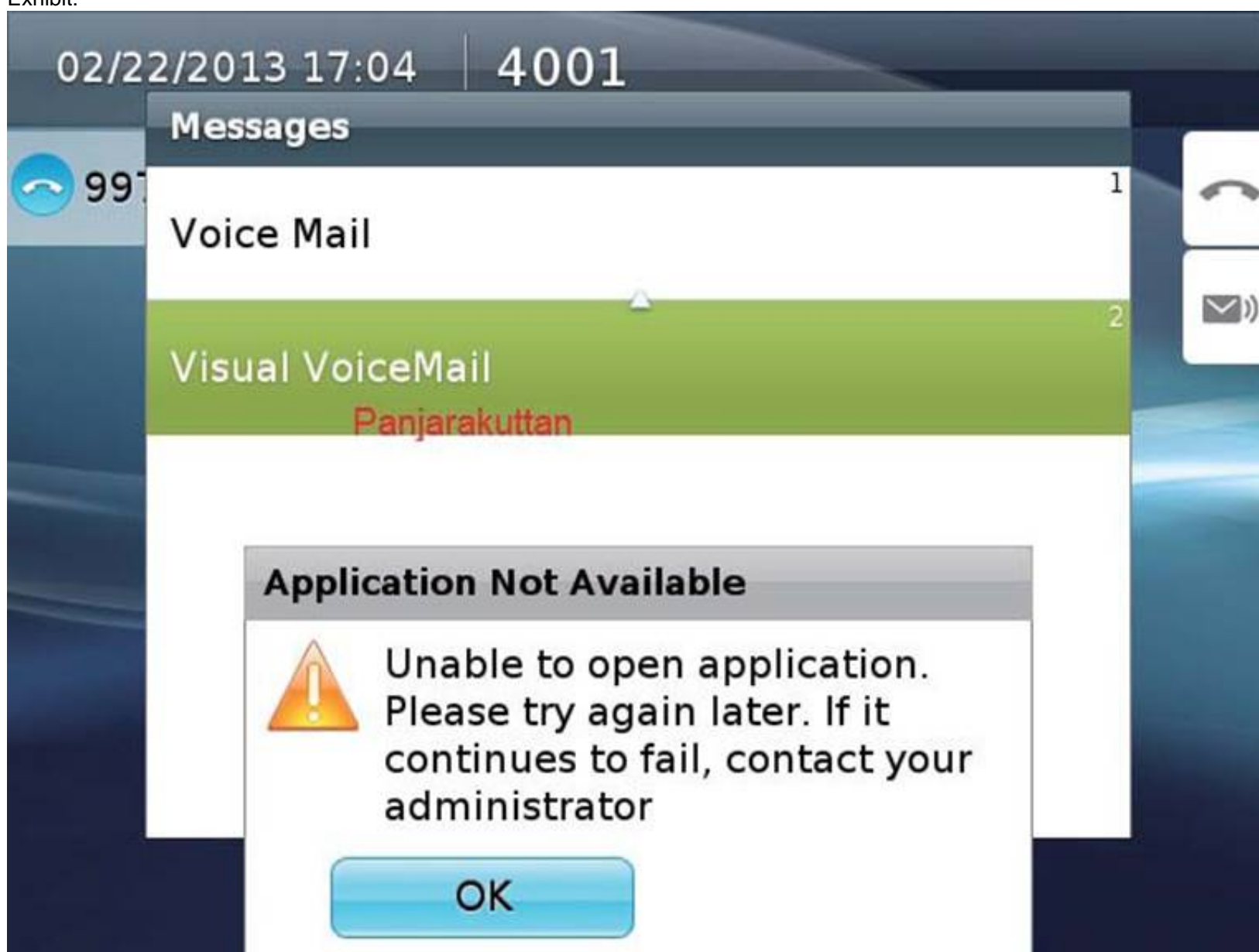
A customer is configuring CAR costing for call. When the customer runs the costing reports calls are not being tagged correctly. Which two changes allow proper costing to be determined for these calls? (Choose two)

- 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: AB

NEW QUESTION 67

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).

NEW QUESTION 72

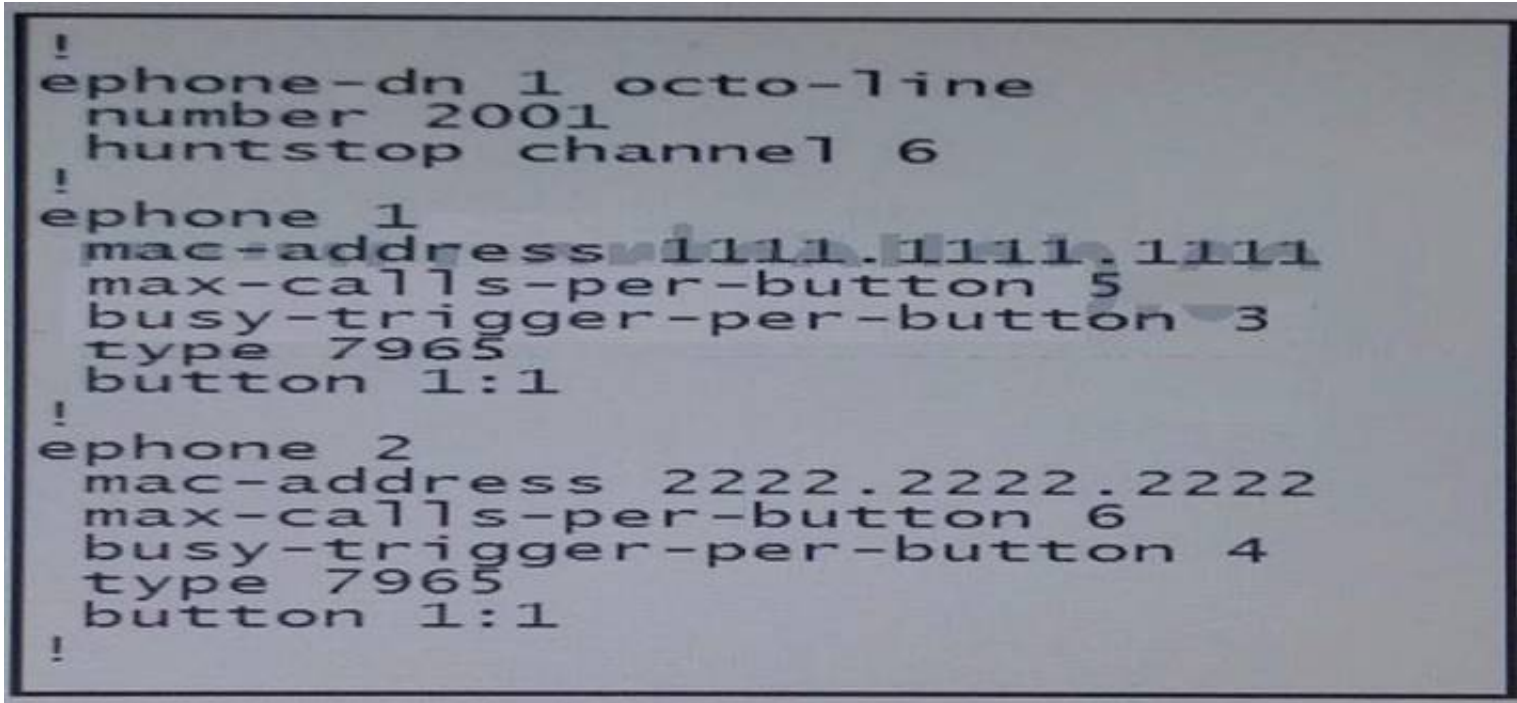
A company is closing two offices and are transferring employees to a new location over a 30 day period. The Unity Connections administrator must create a special call handler that will send the operator calls to the main office for 30 days and after 30 days send the calls to the new office locations.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

NEW QUESTION 74

Exhibit:



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

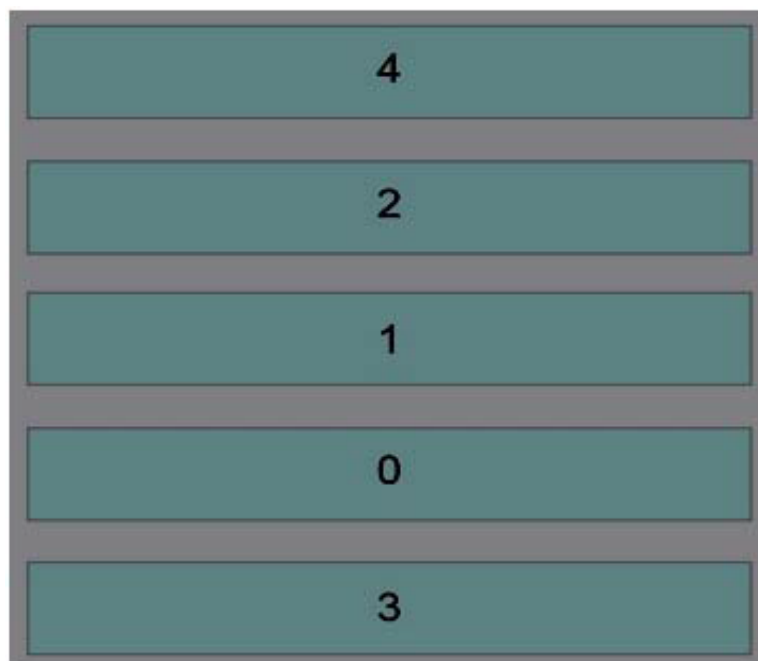
NEW QUESTION 76

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

0	Replication setup did not succeed
1	Logical connections successful and all tables match between servers
2	Incorrect replicate counts
3	Replication did not start or is still initializing
4	Logical connections established but tables did not match

Answer:

Explanation:



NEW QUESTION 78

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: CE

NEW QUESTION 83

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

NEW QUESTION 86

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: AD

NEW QUESTION 90

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: AB

NEW QUESTION 94

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_rcvdata: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

NEW QUESTION 97

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: AD

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

NEW QUESTION 100

In OpenStack, while project stores and retrieves arbitrary unstructured data objects?

- A. Nova
- B. Swift
- C. Cinder
- D. Keystone

Answer: D

NEW QUESTION 104

ACisco 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. ACCD 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: AF

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.

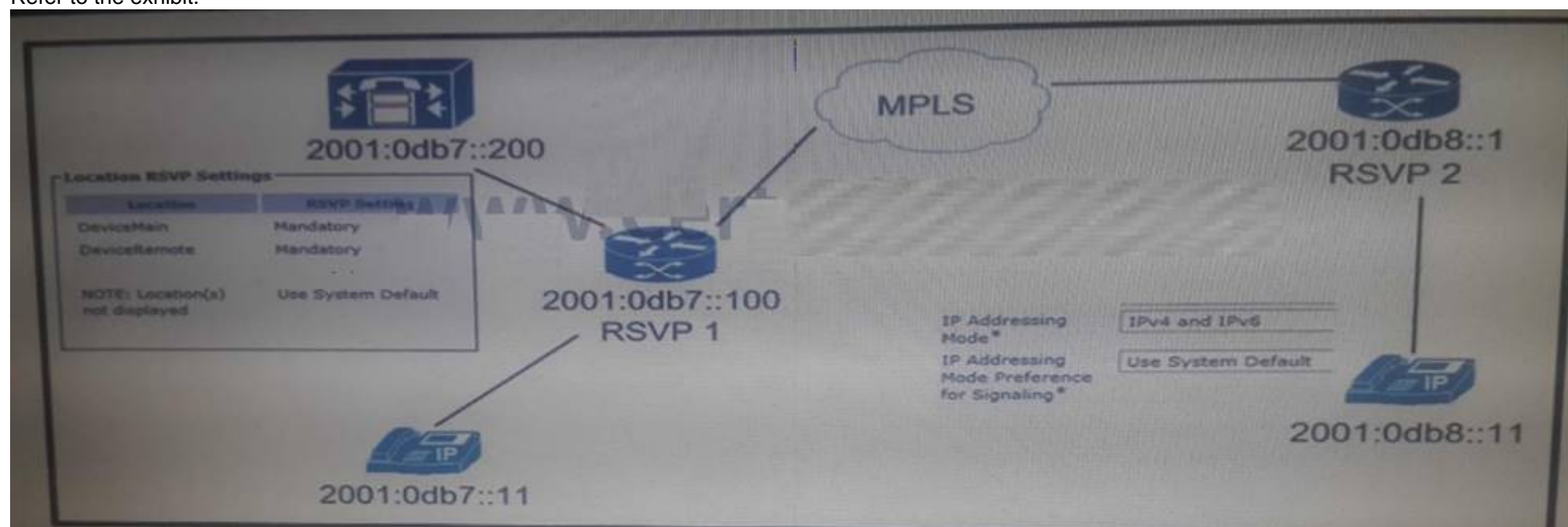
The local Cisco Unified Communications Manager cluster uses SAF-enabled trunks that are assigned to the CCD requesting service to route outbound calls to remote call-control entities that use the SAF network.

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 DN; 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.

NEW QUESTION 106

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

NEW QUESTION 109

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 conform-action transmit exceed-action drop
- C. police 70000000 13125000 conform-action transmit exceed-action drop
- D. traffic-shape rate 70000000 8750000 8750000

Answer: D

NEW QUESTION 111

ACisco 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: ACE

NEW QUESTION 116

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: BCD

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.

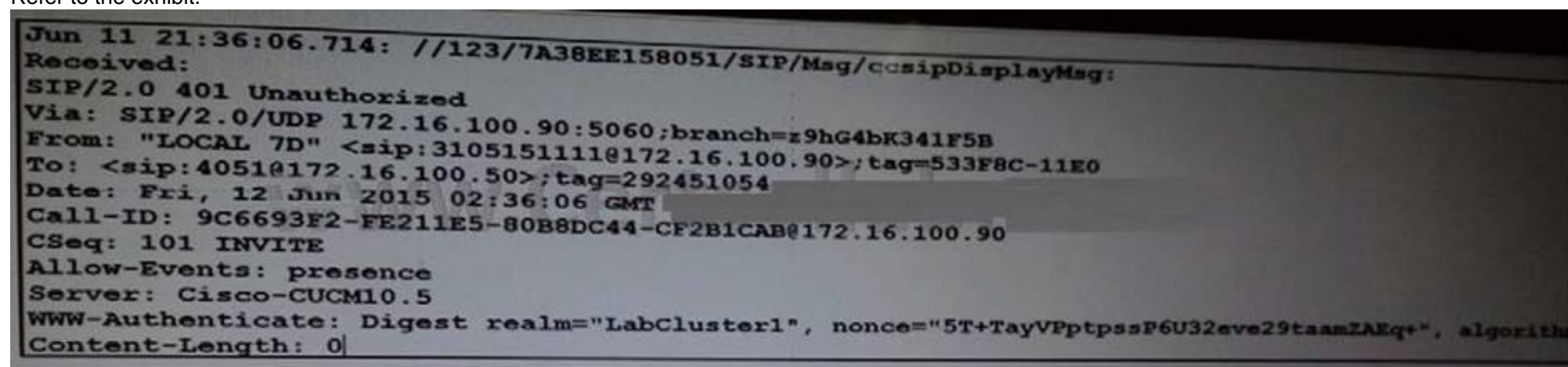
This CSS must include the partitions of the devices you are trying to receive status on.

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>

NEW QUESTION 117

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-uaRegistrar ipv4:172.16.100.90 expires 3600 Authentication cisco realm MD5 LabCluster1
- B. sip-uaAuthentication username cisco password 7 cisco realm LabCluster1
- C. Voice service voip SipAuthenticate digest MD5 LabCluster1
- D. Voice service voip SipAuthenticate digest LabCluster1 MD5

Answer: B

NEW QUESTION 122

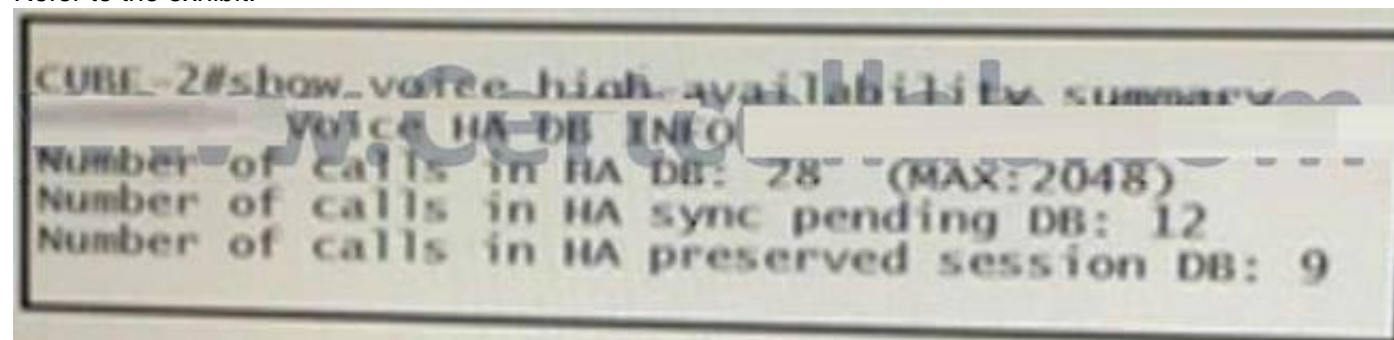
Which four requirements are mandatory to enable a mixed mode Cisco Unified CM cluster? (Choose four.)

- 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: ABDF

NEW QUESTION 126

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 HADB" + "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.

NEW QUESTION 131

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

NEW QUESTION 136

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

- A. Domain 1 only
- B. Domain 4 only
- C. Domain 2 only
- D. Domain1 and Domain4

Answer:

NEW QUESTION 140

ACisco 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: BD

NEW QUESTION 141

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 Dialed Extensions and configure complete extension number on the Edit Caller Input page for the selected key.

Answer: D

NEW QUESTION 146

XMPP Protocol FEATURES(stream) []

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

REQUIRED

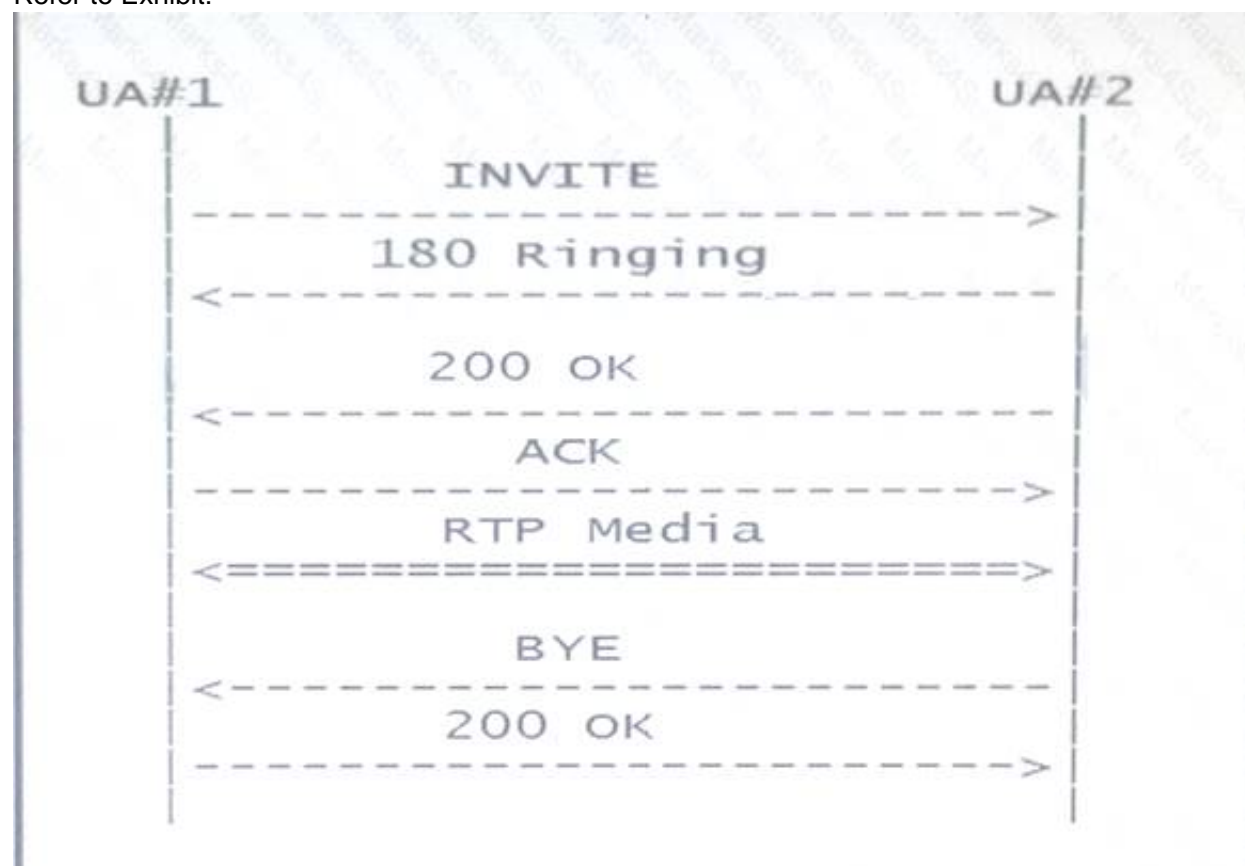
Refer to the exhibit. Which message is used to negotiate the TLS requirement while federating with an external domain?

- A. xmpp-server message
- B. FEATURES message
- C. server hello
- D. STARTTLS message
- E. client hell

Answer: D

NEW QUESTION 147

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

NEW QUESTION 150

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

NEW QUESTION 151

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 dialling 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: BCE

NEW QUESTION 152

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: EF

NEW QUESTION 156

ACisco 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.
- D. The user controls the destination of that call handler by logging into the call handler via greetings administrator.
- E. The administrator informs the user that this feature is a built-in option to the user Cisco PCA page under caller input.
- F. The administrator informs the user that this feature is a built-in option for the user in the TUI under personal settings.

Answer: A

NEW QUESTION 161

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

- A. ADIO is an ICMPv6 RPL control message whose main function is to perform DODAG secure message counters.
- B. ADIO is an ICMPv4 RPL control message whose main function is to perform DODAG discovery, formation, and maintenance.
- C. ADIO is an ICMPv6 RPL control message whose main function is to perform DODAG discovery, formation, and maintenance.
- D. ADIO is an ICMPv4 RPL control message whose main function is to propagate destination information in a RPL network.

Answer: C

NEW QUESTION 162

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

NEW QUESTION 165

Refer to the exhibit.

```
INVITE sip:951241822220172.16.100.90:5060 SIP/2.0
Via: SIP/2.0/TCP 172.16.100.50:5060
From: "Agent A" <sip:51241830010172.16.100.50>
To: <sip:951241822220172.16.100.90>
Date: Tue, 10 Mar 2015 14:25:03 GMT
Call-ID: 3c748f00-4felfebr-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

NEW QUESTION 167

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

NEW QUESTION 171

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

NEW QUESTION 176

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

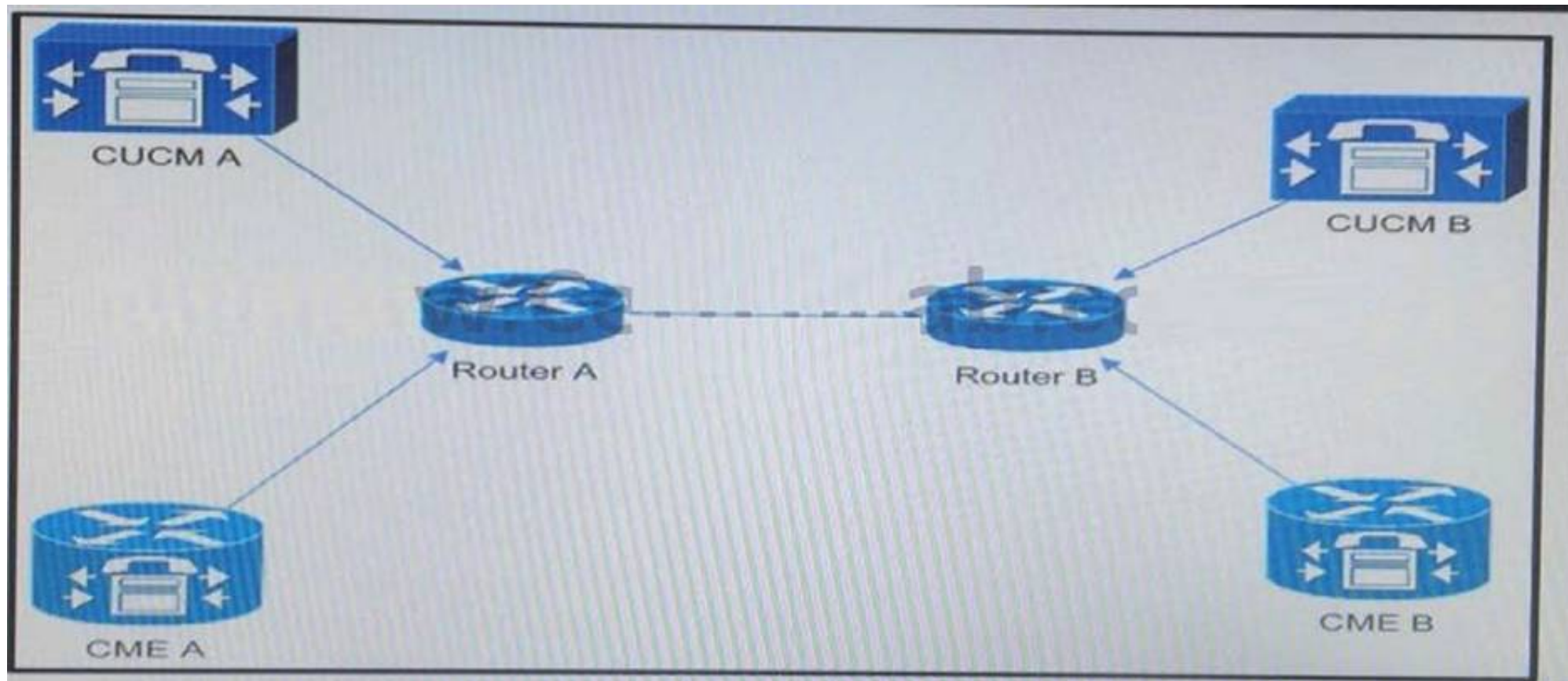
- A. 60 seconds
- B. 1 seconds
- C. 1440 seconds
- D. 600 seconds

E. 3600 seconds

Answer: A

NEW QUESTION 177

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: CF

NEW QUESTION 182

The Cisco Call 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

NEW QUESTION 185

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

NEW QUESTION 190

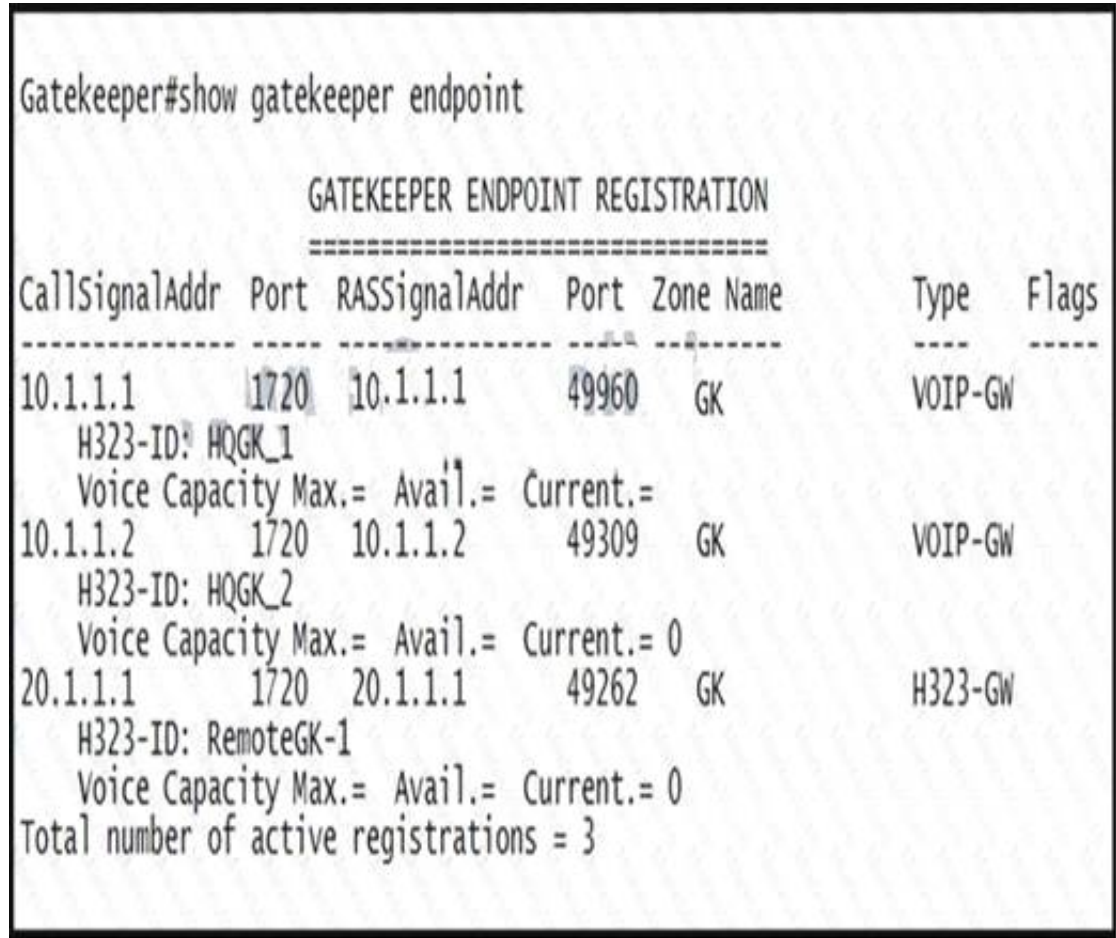
Which two services must be enabled on the routing servers when configuring Partitioned Intradomain Federation? (Choose two.)

- 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: BC

NEW QUESTION 192

Exhibit:



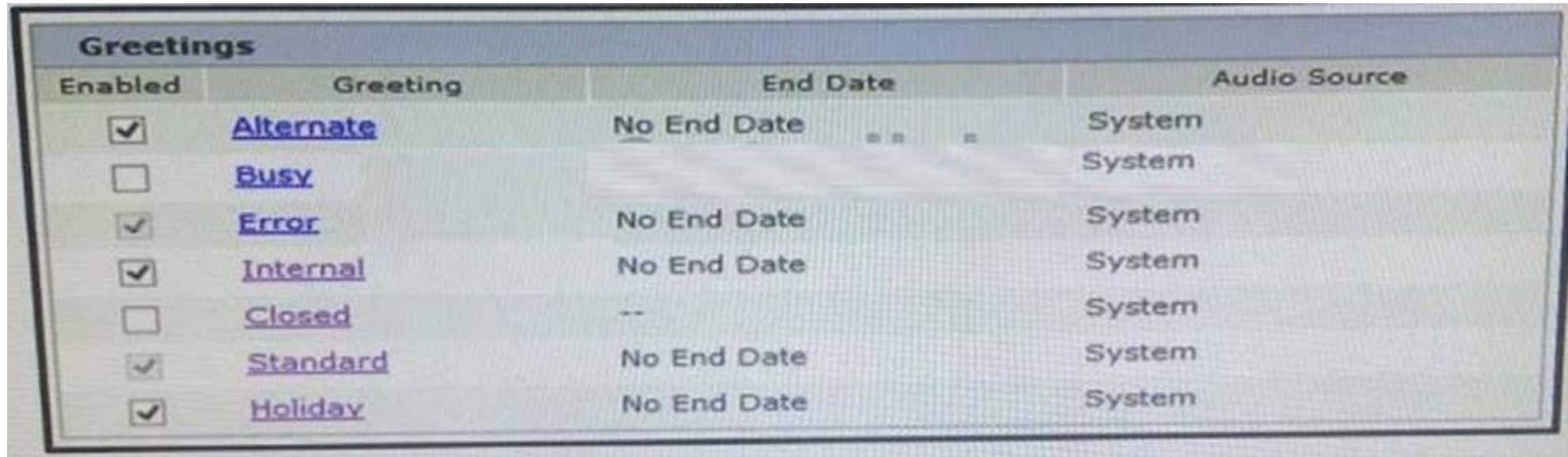
10.1.1.1 and 10.1.1.2 are node IP addresses of a Cisco Unified CM cluster. Which statement describes the correct Cisco Unified CM configurations that produced the output shown in the exhibit?

- 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

NEW QUESTION 195

Refer to the exhibit.



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: ADFG

NEW QUESTION 198

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: AD

NEW QUESTION 201

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: BE

NEW QUESTION 204

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: CF

NEW QUESTION 207

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: AD

NEW QUESTION 212

Refer to the exhibit.



ACUCM 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: ABCD

NEW QUESTION 215

ACUCM 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 MESSAGE 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

NEW QUESTION 216

"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 RCM is stuck in the initializing state
- C. The MAC of the agent phone is not associated with RCM 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: BC

NEW QUESTION 220

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: BC

NEW QUESTION 222

Refer to the exhibit.



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

NEW QUESTION 224

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

NEW QUESTION 226

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: 3c748f00-4felfebf-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=ftm: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: DE

NEW QUESTION 230

Refer to the exhibit.

```

Jul 18 05:00:21.834: //1/xxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent: 1001010.10.10.31:50800;transport=tcp SIP/2.0
NOTIFY sip:1001010.10.10.31:50800;transport=tcp SIP/2.0
via: SIP/2.0/TCP 10.10.10.254:5060;branch=z9hG4bKE698F
From: <sip:1001010.10.10.254>;tag=194EE3C-596
To: <sip:1001010.10.10.31>
Call-ID: D62DE77E-6AAC11E7-81208ED4-64A83368@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:1001010.10.10.254:5060;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

NEW QUESTION 235

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 nodes?

- 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: BFGH

NEW QUESTION 239

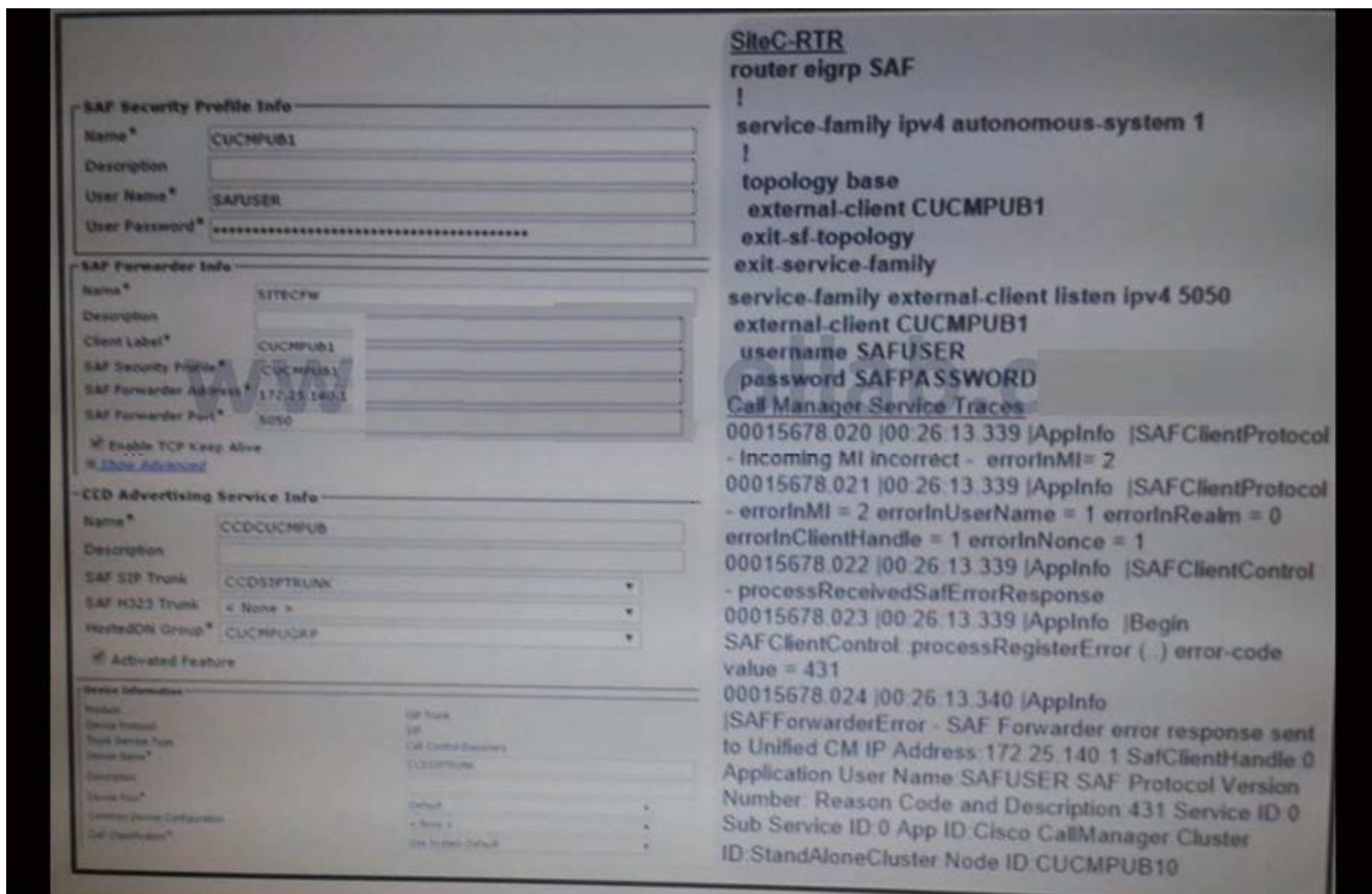
Which Cisco Unified CM service is responsible for writing Call Detail Records into flat files?

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

Answer: A

NEW QUESTION 241

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

NEW QUESTION 245

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: AF

NEW QUESTION 246

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

NEW QUESTION 249

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 200request INVITE sip-header Diversion modify "sip:(.*)>" "123456@company.com>" request REINVITE sipheaderDiversion modify "sip:(.*)>" "123456@company.com>"

- B. voice class sip-profiles 200request INVITE sdp-header Diversion modify "sip:(.*>)" 123456@company.com> request REINVITE sdpheaderDiversion modify "sip:(.*>)" 123456@company.com>"
- C. voice class sip-profiles 200response 200 sdp-header Diversion modify "sip:(.*>)" 123456@company.com>"
- D. voice class sip-profiles 200response 200 sip-header Diversion modify "sip:(.*>)" 123456@company.com>"

Answer: A

NEW QUESTION 253

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: AC

NEW QUESTION 257

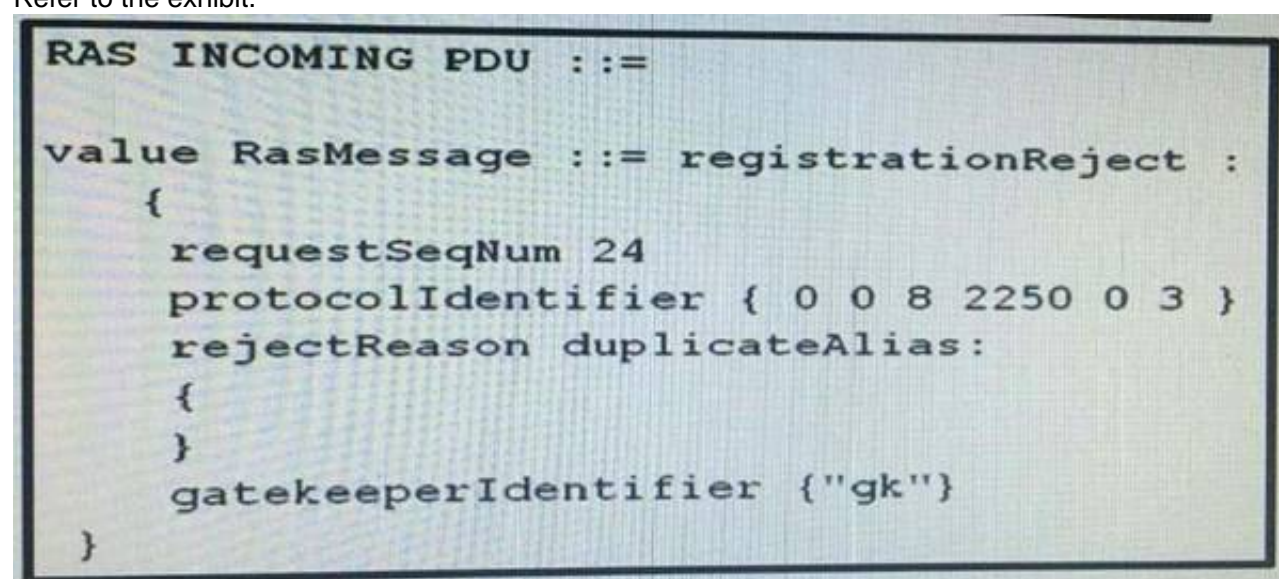
After configuring EM in the CUCM cluster, users are receiving 'Host not found' error message after pressing the Services button. What should be done to fix this problem?

- 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

NEW QUESTION 258

Refer to the exhibit.



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: BE

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>

NEW QUESTION 259

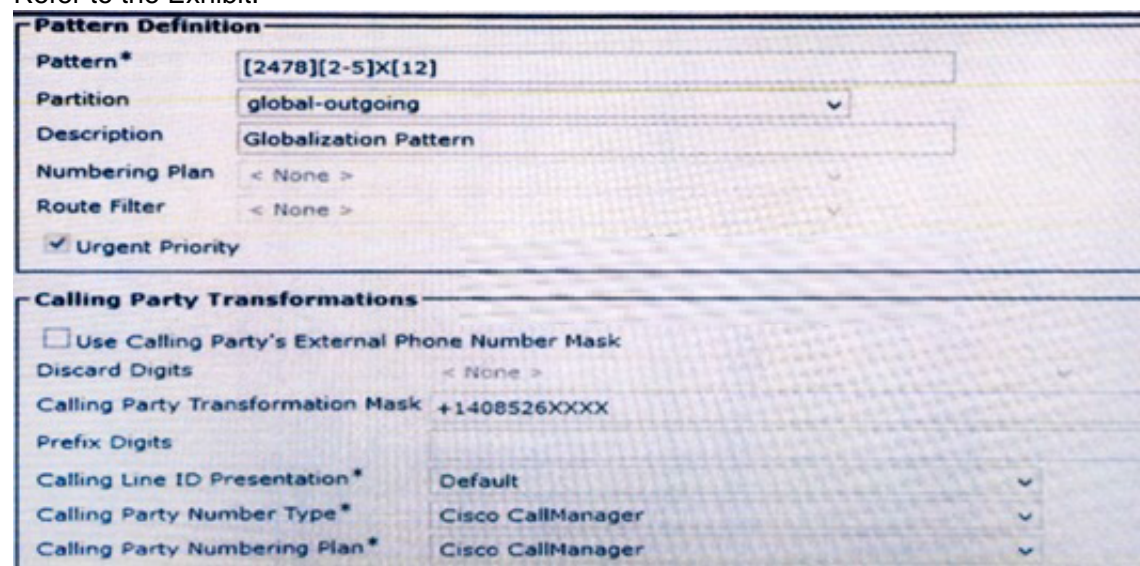
ABC Company has Cisco unified CM version 9.1 cluster with seven nodes. The publisher server suffered a catastrophic hard disk failure without Cisco Disaster Recovery System Backups. Which method restore the Publisher node is valid?

- 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

NEW QUESTION 260

Refer to the Exhibit.



Pattern Definition

Pattern* [2478][2-5]X[12]

Partition global-outgoing

Description Globalization Pattern

Numbering Plan <None>

Route Filter <None>

☒ Urgent Priority

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Discard Digits <None>

Calling Party Transformation Mask +1408526XXXX

Prefix Digits

Calling Line ID Presentation* Default

Calling Party Number Type* Cisco CallManager

Calling Party Numbering Plan* Cisco CallManager

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: ADE

NEW QUESTION 261

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

NEW QUESTION 266

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

NEW QUESTION 268

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)

Order	Blocked	
0		
1	<input checked="" type="checkbox"/>	+
2	<input checked="" type="checkbox"/>	9+
3	<input checked="" type="checkbox"/>	91???????
4	<input checked="" type="checkbox"/>	90???????
5	<input checked="" type="checkbox"/>	91???
6	<input checked="" type="checkbox"/>	900
	<input type="checkbox"/>	*

B)

Order	Blocked	
0		
1	<input checked="" type="checkbox"/>	+
2	<input checked="" type="checkbox"/>	9+
3	<input type="checkbox"/>	91???????
4	<input checked="" type="checkbox"/>	9011???????
5	<input type="checkbox"/>	9?????????
6	<input type="checkbox"/>	9?????????
	<input checked="" type="checkbox"/>	*

C)

Order	Blocked	
0	<input checked="" type="checkbox"/>	9011*
1	<input checked="" type="checkbox"/>	+
2	<input checked="" type="checkbox"/>	9+
3	<input type="checkbox"/>	91???????
4	<input type="checkbox"/>	9?????????
5	<input type="checkbox"/>	9?????????
6	<input checked="" type="checkbox"/>	*

D)

Order	Blocked	
0	<input checked="" type="checkbox"/>	9*
1	<input checked="" type="checkbox"/>	9+
2	<input checked="" type="checkbox"/>	91???????
3	<input type="checkbox"/>	91???????????
4	<input type="checkbox"/>	9?????????
5	<input checked="" type="checkbox"/>	900
6	<input checked="" type="checkbox"/>	*

E)

Order	Blocked	
0		
1	<input checked="" type="checkbox"/>	9=
2	<input checked="" type="checkbox"/>	9+*
3	<input checked="" type="checkbox"/>	9???????*
4	<input type="checkbox"/>	91????????????*
5	<input type="checkbox"/>	9???????*
6	<input checked="" type="checkbox"/>	900
	<input checked="" type="checkbox"/>	=

Order	Blocked	
0		
1	<input checked="" type="checkbox"/>	9+*
2	<input checked="" type="checkbox"/>	9011*
3	<input checked="" type="checkbox"/>	9???*
4	<input checked="" type="checkbox"/>	91*
5	<input type="checkbox"/>	91???*
6	<input type="checkbox"/>	9???????????*
	<input checked="" type="checkbox"/>	=

- A. Exhibit A
- B. Exhibit B
- C. Exhibit C
- D. Exhibit D
- E. Exhibit E
- F. Exhibit F

Answer: BC

NEW QUESTION 271

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: ABD

NEW QUESTION 274

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: AE

NEW QUESTION 277

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 EnabledFlag 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

NEW QUESTION 280

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

NEW QUESTION 284

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.729h, b:8, e:on, gc:1, s:on, t:10, r:g, nt:IN;
  AIM;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, contest, 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 AUPE 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

NEW QUESTION 285

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: AE

NEW QUESTION 287

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: ACE

NEW QUESTION 288

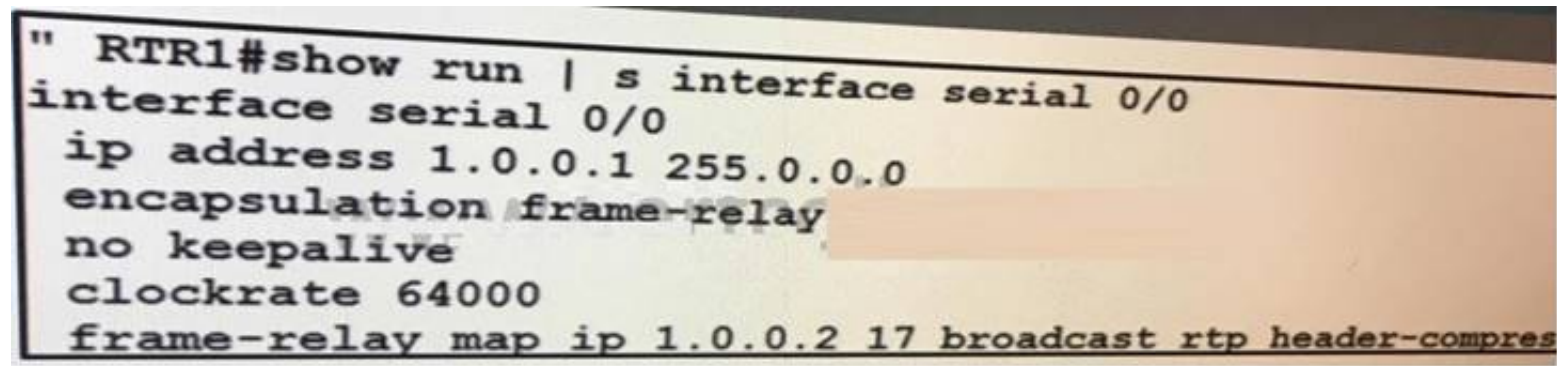
Which MGCP message does a Cisco IOS MGCP gateway send to the backup Cisco Unified CM server when two consecutive keep-alive exchanges failed with the primary Cisco Unified CM server?

- A. AUPE
- B. DLCX
- C. NTFY
- D. RSIP
- E. AUCX

Answer: D

NEW QUESTION 290

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: BC

NEW QUESTION 295

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

NEW QUESTION 296

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: AB

NEW QUESTION 299

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 wilddcarded
- B. advertise callcontrol 1
- C. subscribe callcontrol wilddcarded
- D. register callcontrol wilddcarded
- E. publish callcontrol 1
- F. distribute callcontrol 1

Answer: CE

NEW QUESTION 304

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

NEW QUESTION 306

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

NEW QUESTION 310

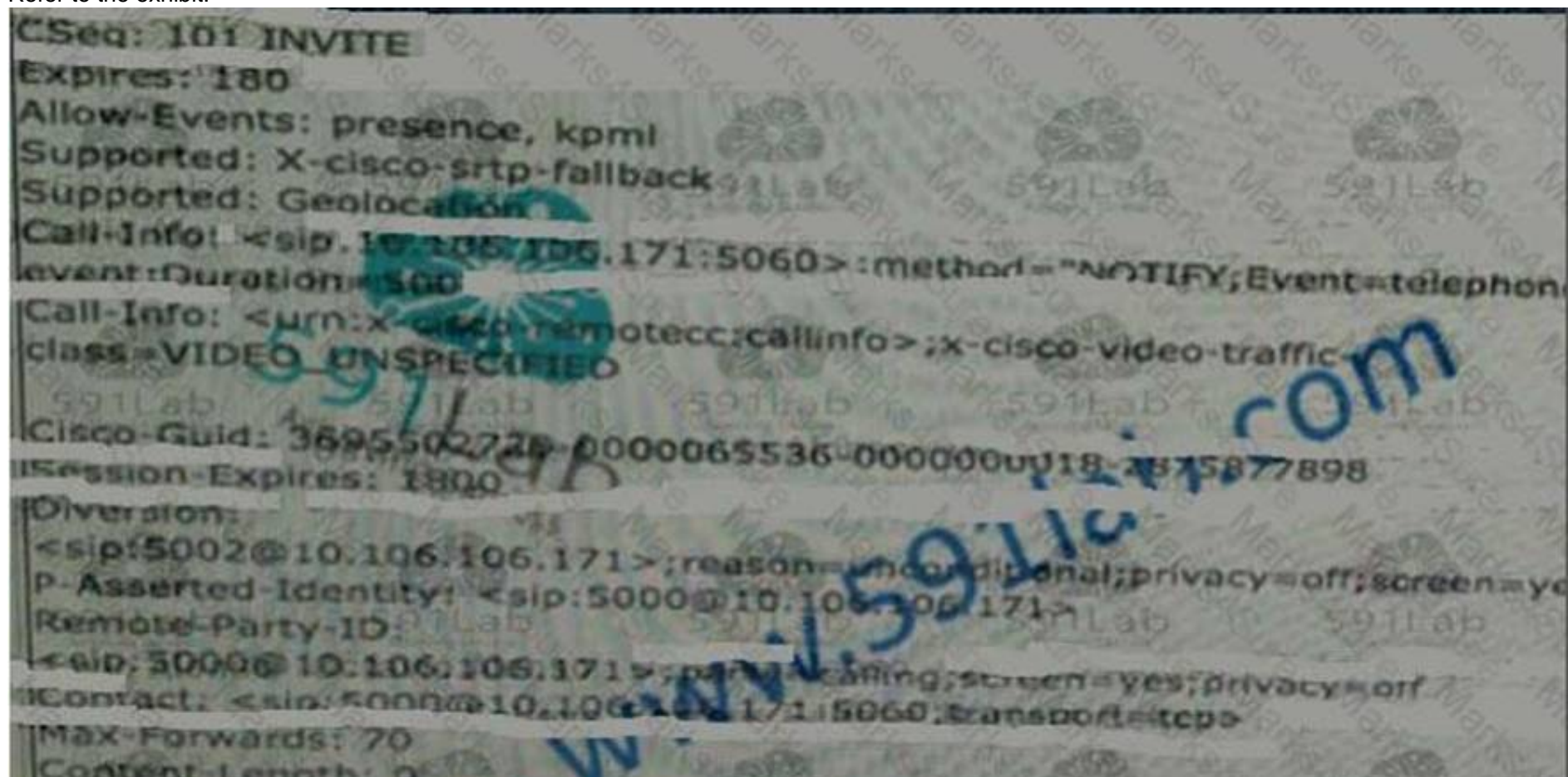
An engineer is planning the voice call bandwidth requirements between two offices. The design requires a capacity of 25 concurrent audio calls using the G.729 codec and IPv6 transport. Which two configurations meet the requirements? (Choose two)

- 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: BD

NEW QUESTION 312

Refer to the exhibit.



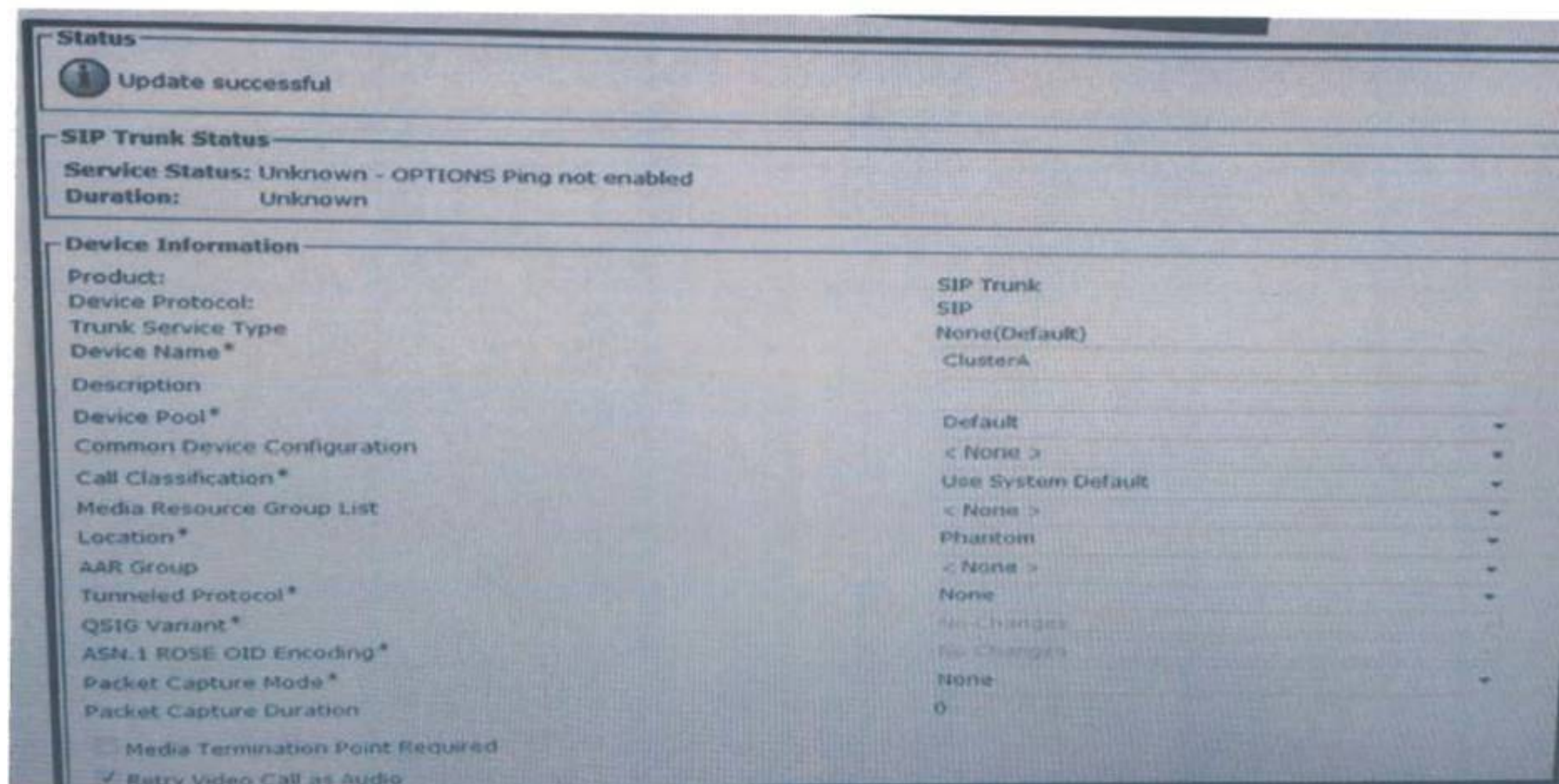
Which Cisco IOS SIP profile is valid for copying value from the "Diversions" header to the "From" header in a SIP INVITE message?

- A. voice class sip-profiles 1request INVITE sip-header Diversion copy "<sip:(.*)@.*" u02 request INVITE sip-header From copy "<sip:(.*)@.*" u01request INVITE sip-header From modify "(.*)<sip:.*(@).*)" "\1<sip:\u01@\2"
- B. voice class sip-profiles 1request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01 request INVITE sip-header From copy "<sip:(.*)@.*" u02request INVITE sip-header From modify "(.*)<sip:.*(@).*)" "\2<sip:\u01@\1"
- C. voice class sip-profiles 1request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01 request INVITE sip-header From copy "<sip:(.*)@.*" u02request INVITE sip-header From modify "(.*)<sip:.*(@).*)" "\1<sip:\u02@\2"
- D. voice class sip-profiles 1request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01 request INVITE sip-header From copy "<sip:(.*)@.*" u02request INVITE sip-header From modify "(.*)<sip:.*(@).*)" "\1<sip:\u01@\1"
- E. voice class sip-profiles 1request INVITE sip-header Diversion copy "<sip:(.*)@.*" u01 request INVITE sip-header From copy "<sip:(.*)@.*" u02request INVITE sip-header From modify "(.*)<sip:.*(@).*)" "\1<sip:\u01@\2"

Answer: E

NEW QUESTION 313

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

NEW QUESTION 318

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.

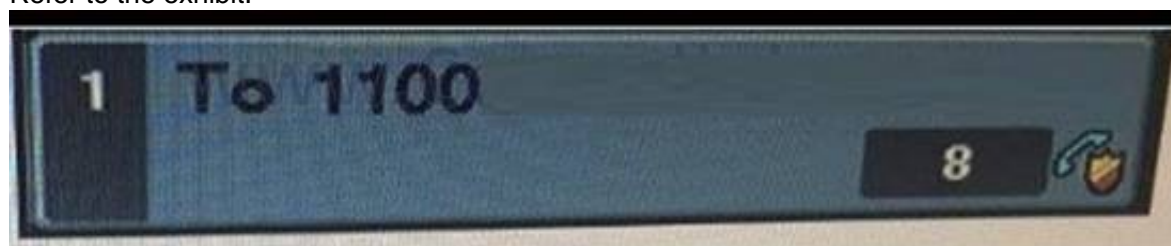
Step 1
Step 2
Step 3
Step 4

Answer:

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

NEW QUESTION 319

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

NEW QUESTION 323

Where can a Cisco Unified CM administrator define Call Detail Records data collection interval?

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

Answer: B

NEW QUESTION 324

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

NEW QUESTION 326

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
```

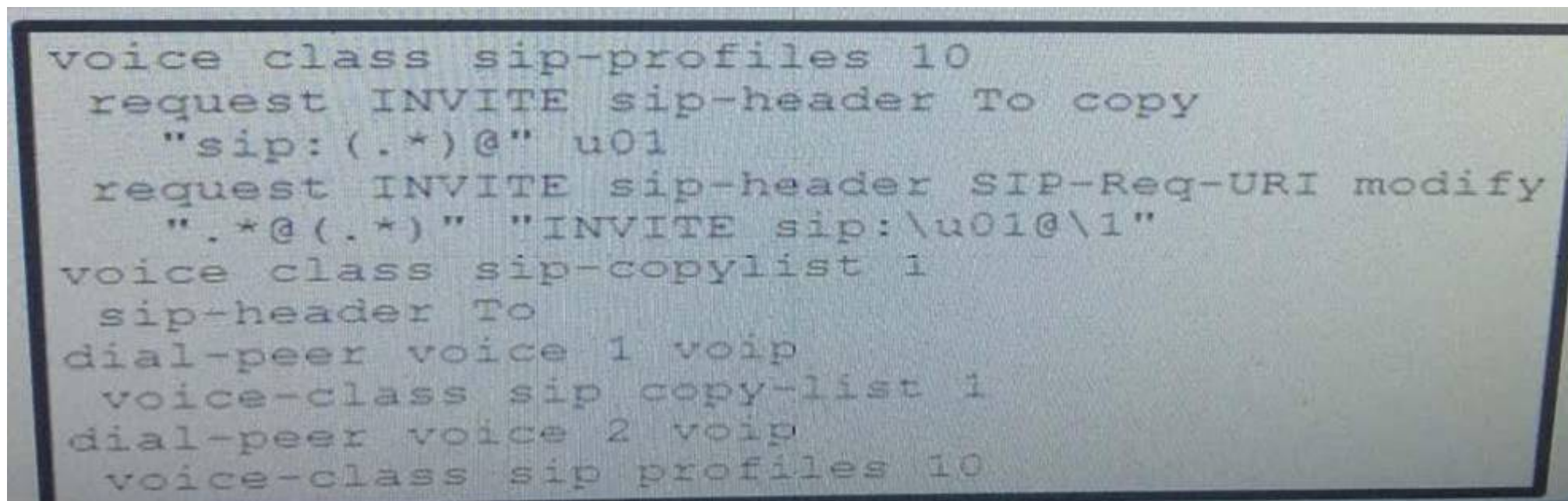
B)

```
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)



- A. Exhibit A
- B. Exhibit B
- C. Exhibit C
- D. Exhibit D

Answer: A

NEW QUESTION 327

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: e1817d00-4fffd18b-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 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: BDF

NEW QUESTION 328

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

NEW QUESTION 332

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="sdf1akjdfjklahsfhkhq", algorithm=MD5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Content-Length: 0
```

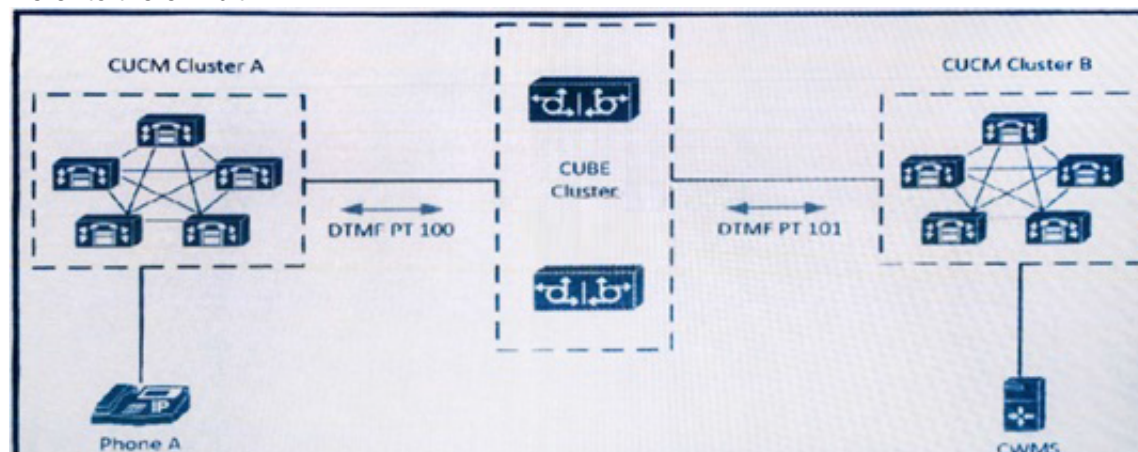
The exhibit shows an outgoing SIP 401 response message from Cisco Unified Communications Manager to a SIP VoIP service provider gateway. Which action can the Cisco Unified Communication Manager Systems administrator use to change the response to "200 OK"?

- 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

NEW QUESTION 336

Refer to the exhibit.



ACUBE 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 under global 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.

NEW QUESTION 338

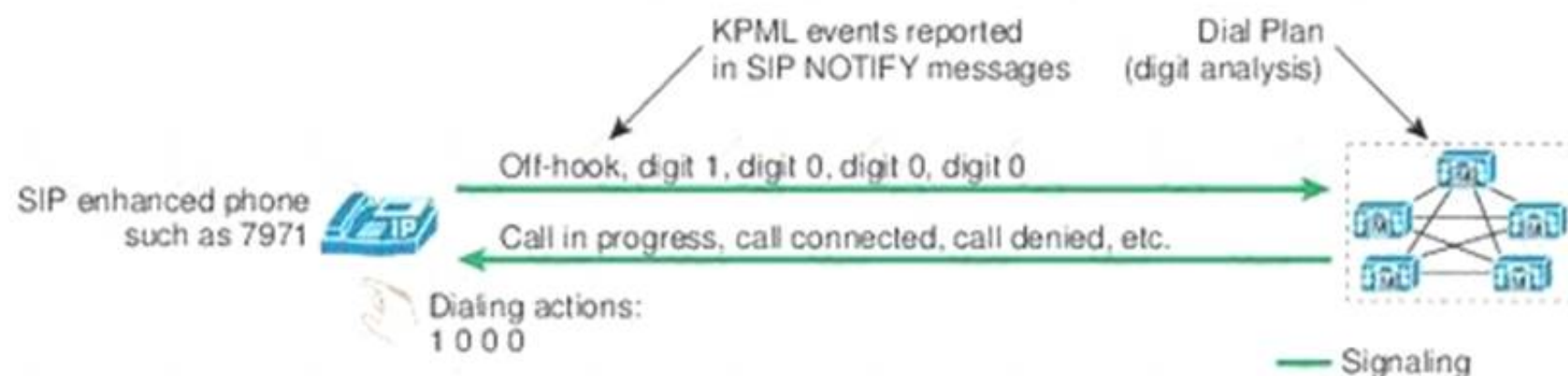
ACUCM engineer has deployed Type B SIP Phones on a remote site and no SIP dial rules were deployed for these phones. How Will CUCM receive the DTMF after the phone goes off-hook and the button are pressed?

- 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 will 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.

User Input and Feedback for Type-B SIP Phones with No Dial Rules Configured



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>

NEW QUESTION 343

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: AC

NEW QUESTION 345

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: DE

NEW QUESTION 348

ACisco 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: ACE

NEW QUESTION 351

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

NEW QUESTION 355

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: ACD

NEW QUESTION 356

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: AC

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

NEW QUESTION 359

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: AF

NEW QUESTION 363

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

NEW QUESTION 368

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 infrastructure
- 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

NEW QUESTION 373

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

NEW QUESTION 378

ACisco 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 Cfgdall 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: BC

NEW QUESTION 383

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: ACDF

NEW QUESTION 386

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

NEW QUESTION 388

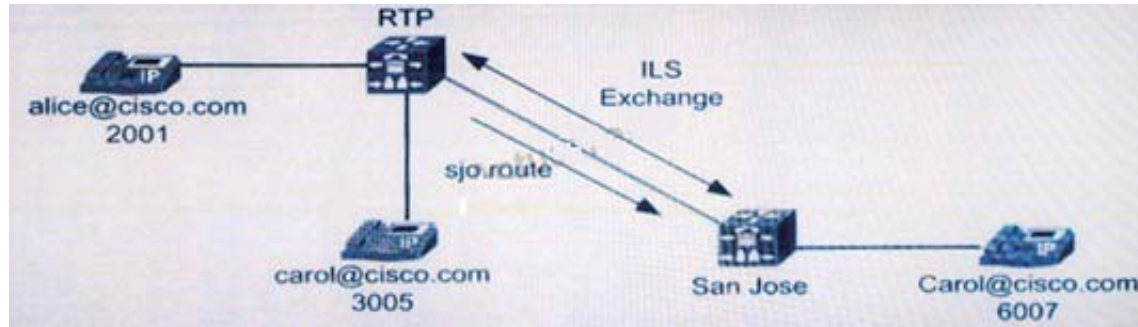
A collaboration engineer is designing a phone VPN infrastructure and the company security team requires Active Directory for authentication. Which two phone VPN configurations meet this requirement? (Choose two)

- 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: CE

NEW QUESTION 390

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: BDE

NEW QUESTION 391

Refer to the exhibit.

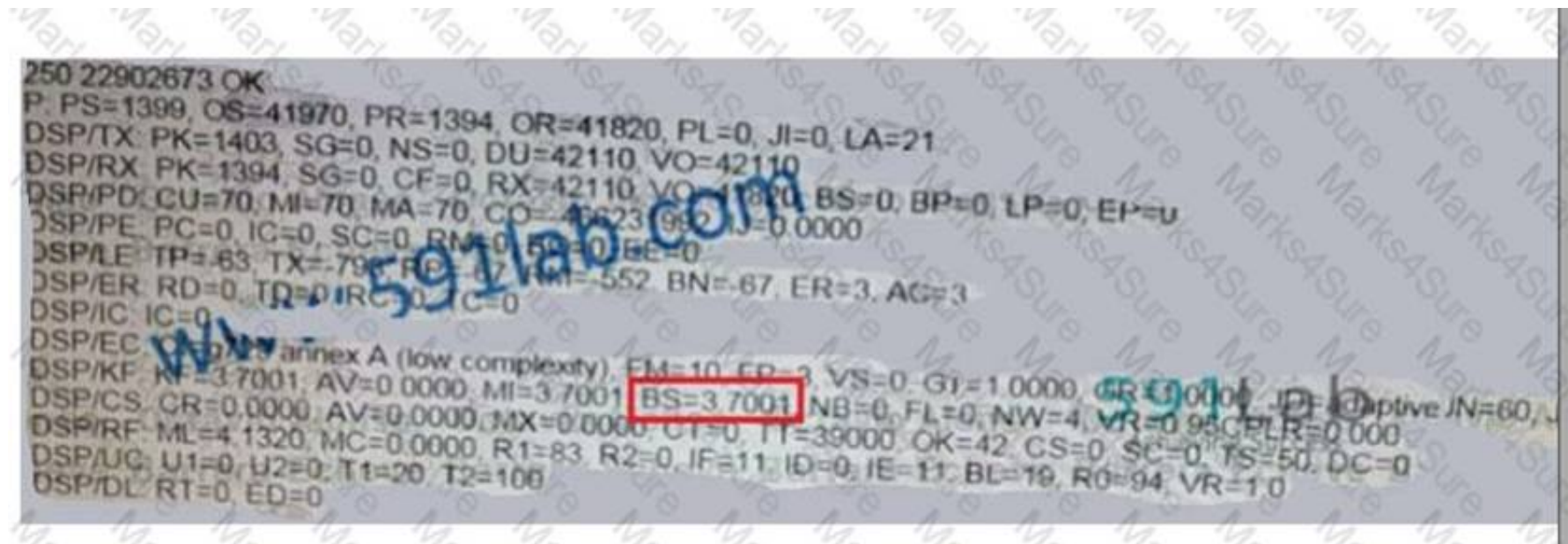


A reporting specialist found that calls answered by the phones are not being recorded. Which two configuration changes can resolve this issue? (Choose two)

- 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: AB

NEW QUESTION 392



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

NEW QUESTION 395

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.

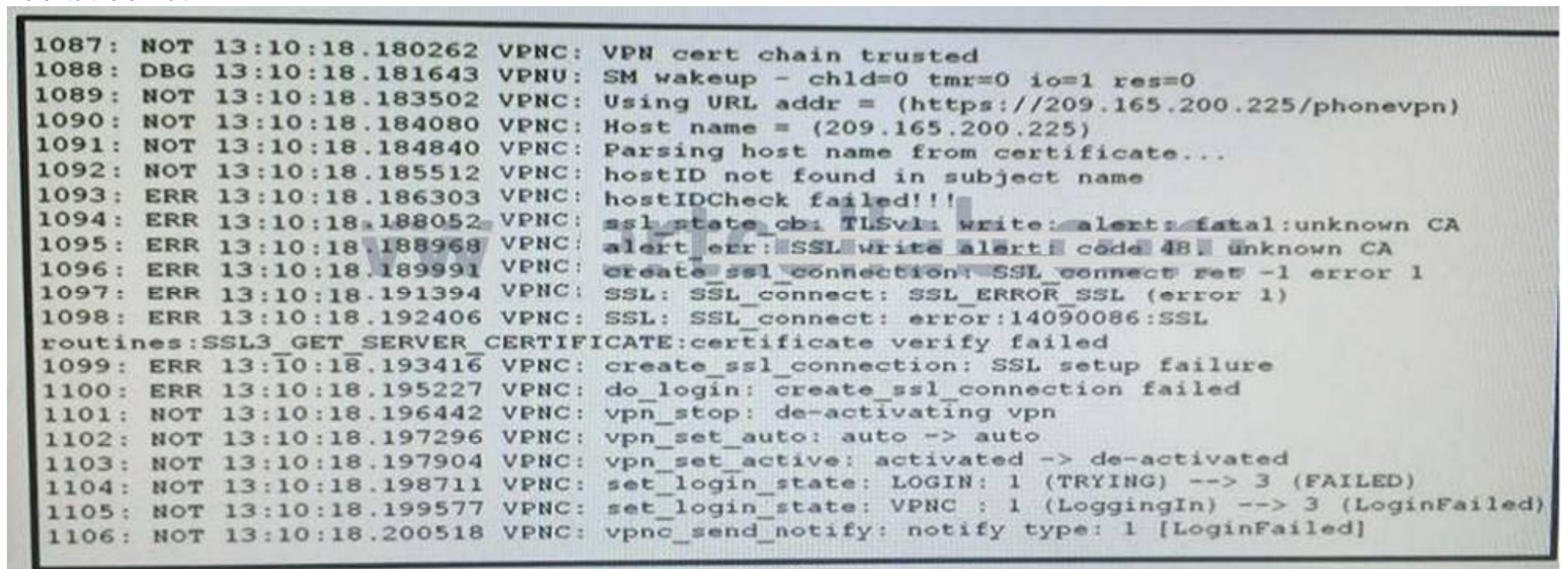
Which setting resolves this issue and always end-user webpage access?

- 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

NEW QUESTION 400

Refer to the exhibit.



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: BE

NEW QUESTION 403

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

NEW QUESTION 408

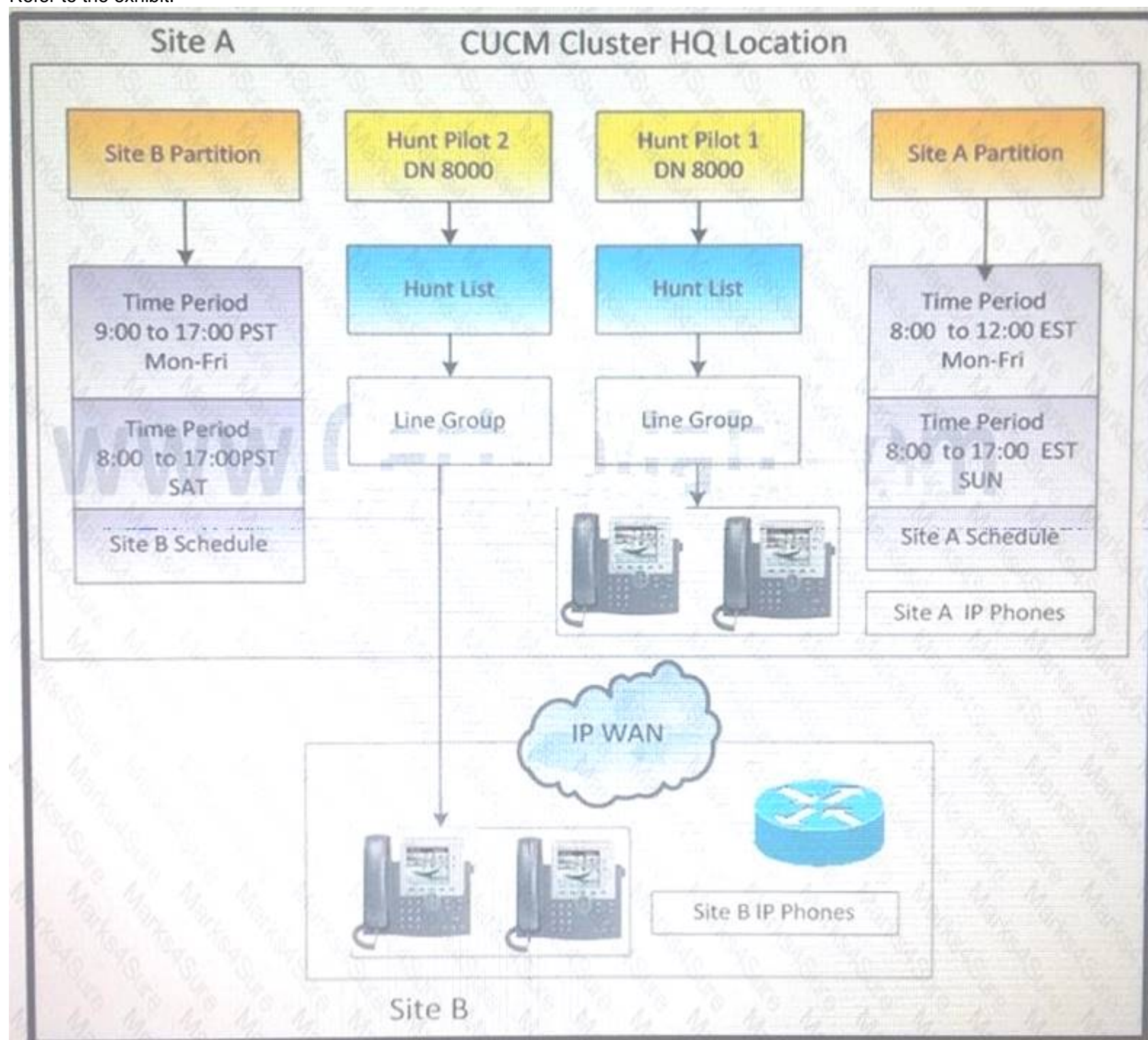
An engineer notices that two Cisco utility Connection servers in a cluster are in split-brain mode. The engineer corrects a network issue that allows the two servers to communicate again. Which two statements describe negative effects of this event? (Choose two)

- 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 can become out of sync after the split-brain recover
- C. Forcing the administrator to run an MWI Synchronization.
- D. The replication between the nodes becomes defunct, requiring the administrator to run utils cuc cluster activate to re-establish intracluster.
- E. A message left on the subscriber server during the outage may be lost during the cluster recovery.
- F. The replication between the nodes becomes defunct, requiring the administrator to run utils cuc cluster renegotiate to re-establish intracluster communication.
- G. The Unity Connection Database can become corrupted, causing the need to reinstall the subscriber server.

Answer: AC

NEW QUESTION 409

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

NEW QUESTION 414

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: ABC

NEW QUESTION 416

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: ABDH

NEW QUESTION 417

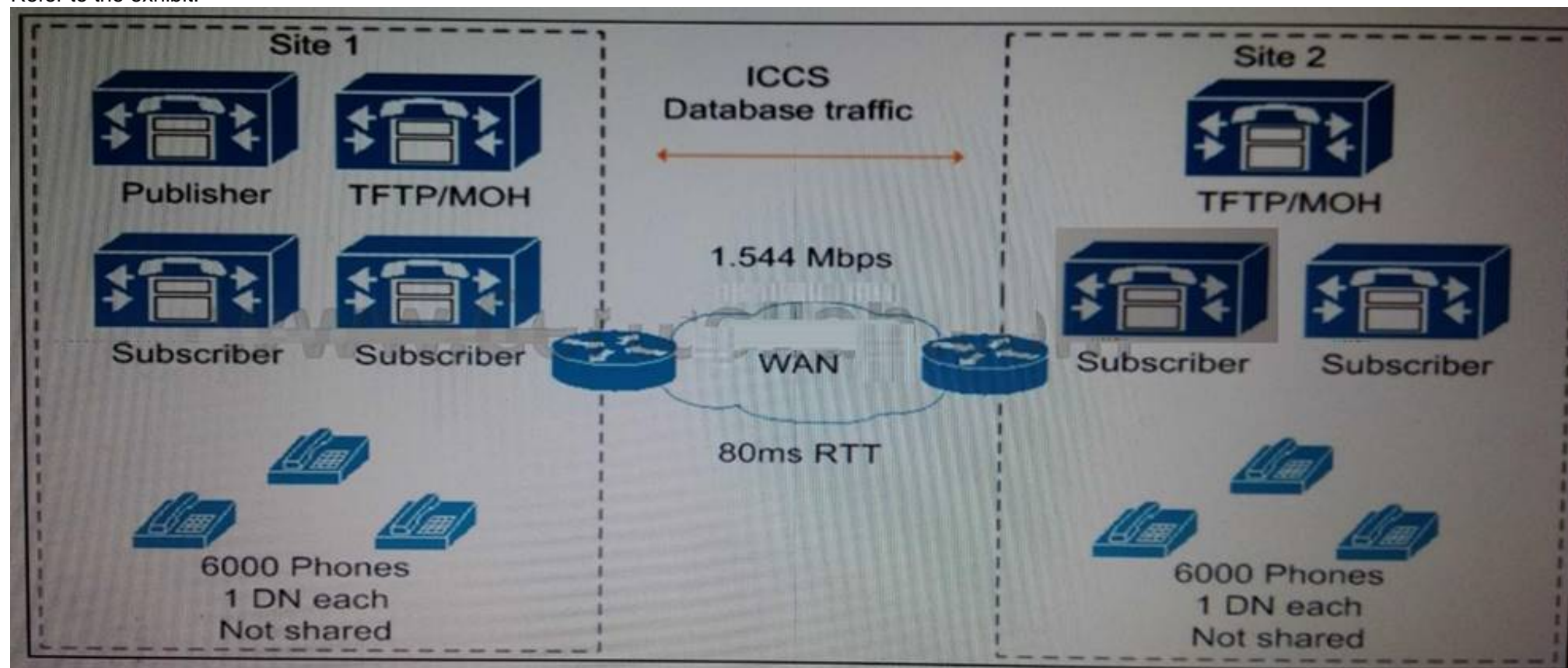
The UCCX consultant is creating a new script must perform an action after the call is terminated between the agent and the caller used to perform this post call action?

- A. The delay step to allow the call to continue after audio call termination.
- B. the OnExceptionClear 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

NEW QUESTION 422

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

NEW QUESTION 426

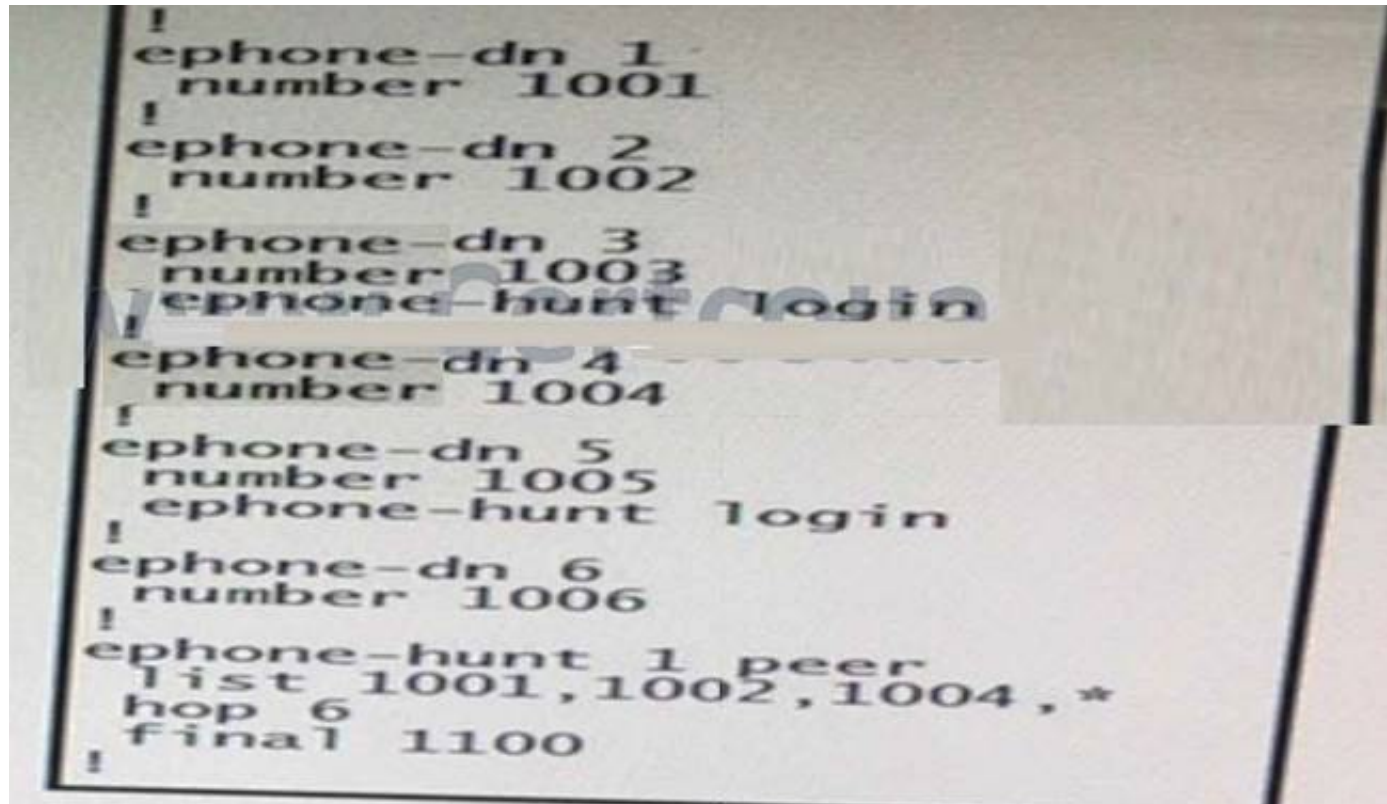
Which directory path on Cisco Unified CM publisher is used to temporarily store the Call Detail Records collected from other nodes until they are processed by the CDR Repository Manager?

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

Answer: B

NEW QUESTION 430

Refer to the exhibit.



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

- A. ephone-dn 1
- B. ephone-dn 2
- C. ephone-dn 3
- D. ephone-dn 4
- E. ephone-dn 6

Answer: C

NEW QUESTION 434

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: AC

NEW QUESTION 439

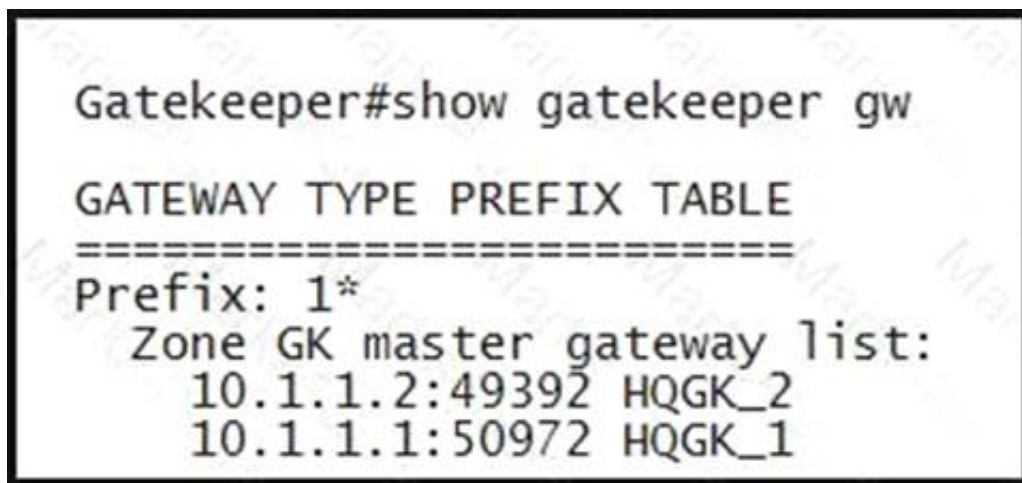
What does a period accomplish when it is used in a SIP Dial Rule pattern that is associated with a Cisco 9971 IP Phone that is registered to Cisco Unified Communications Manager?

- 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

NEW QUESTION 442

Refer to the exhibit.



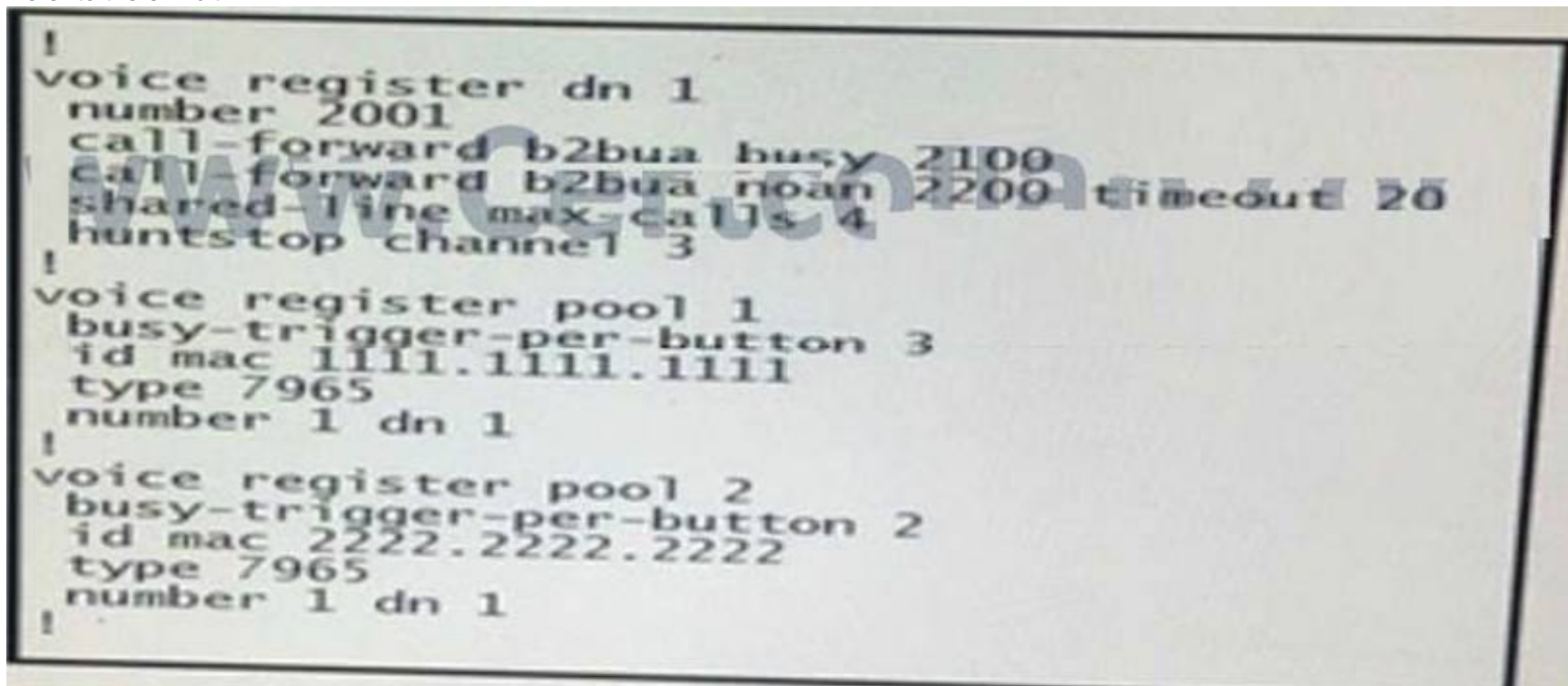
10.1.1.1 and 10.1.1.2 are node IP addresses of a Cisco Unified CM cluster. Which two options are the correct Cisco IOS Gatekeeper configuration that could produce the output shown in the exhibit? (Choose two.)

- 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: BC

NEW QUESTION 444

Refer to the exhibit.



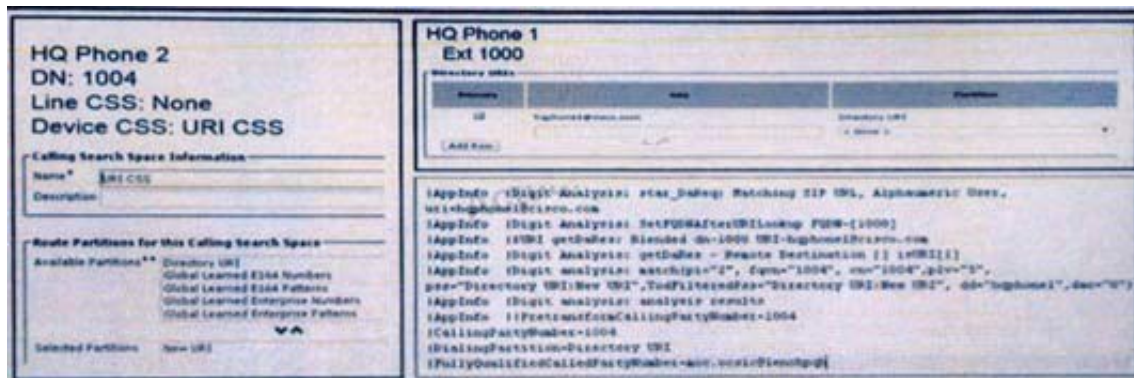
IP phone 1 has the MAC address 1111.1111.1111, while IP phone 2 has the MAC address 2222.2222.2222. The first two incoming calls were answered by IP phone 1, while the third incoming call was answered by IP phone 2. Which option describes what will happen to the fourth incoming call?

- 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

NEW QUESTION 447

Refer to the exhibit.



ACisco 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

NEW QUESTION 449

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.

Device CSS	Priority 1
EMCC CSS	Priority 2
Adjunct CSS	Priority 3
Line CSS	

Answer:

Explanation:

Device CSS	Adjunct CSS
EMCC CSS	Line CSS
Adjunct CSS	EMCC CSS
Line CSS	

NEW QUESTION 450

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: CD

NEW QUESTION 452

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: AEF

NEW QUESTION 456

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 Group1Associate 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: BCF

NEW QUESTION 461

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: BE

NEW QUESTION 462

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>
```

ACisco 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

NEW QUESTION 464

Refer to the exhibit.

Greetings			
Enabled	Greeting	End Date	Audio Source
<input checked="" type="checkbox"/>	Alternate	No End Date	System
<input type="checkbox"/>	Busy	--	System
<input checked="" type="checkbox"/>	Error	No End Date	System
<input checked="" type="checkbox"/>	Internal	No End Date	System
<input type="checkbox"/>	Closed	--	System
<input checked="" type="checkbox"/>	Standard	No End Date	System
<input checked="" type="checkbox"/>	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: CDEF

NEW QUESTION 469

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

NEW QUESTION 473

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

NEW QUESTION 477

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: BDE

NEW QUESTION 480

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