

400-051 Dumps

CCIE Collaboration

<https://www.certleader.com/400-051-dumps.html>



NEW QUESTION 1

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

- A. Using Only Virtual Machines
- B. Using Micro Services Application
- C. Using Virtual machines in containers
- D. Using Only containers
- E. Using containers in virtual machines

Answer: D

NEW QUESTION 2

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

- A. The hunt member remains logged in if Automatically Logout Hunt Member on No Answer is not selected in the Hunt Pilot configuration page.
- B. The hunt member is logged off if Automatically Logout Hunt Member on No Answer is selected on the Line Group Configuration page.
- C. The hunt member is logged off automatically and must manually reset the phone to log back in.
- D. The hunt member remains logged in if Automatically Logout Hunt Member on No Answer is not selected in Cisco Unified Communications Manager Service Parameters.
- E. The hunt member is logged off automatically and guest press HLOG to log back in.

Answer: B

NEW QUESTION 3

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 4

ACollaboration Engineer implemented Cisco EMCC between Cisco Unified CM clusters. The administrator has configured the bulk certificate management and exported the certificates to the SFTP server. After importing the certificates into each of the clusters, the administrator tested Cisco EMCC on a phone, but

received "Login is unavailable (208)". Which two steps resolve this error? (Choose two)

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

Answer: AE

NEW QUESTION 5

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 6

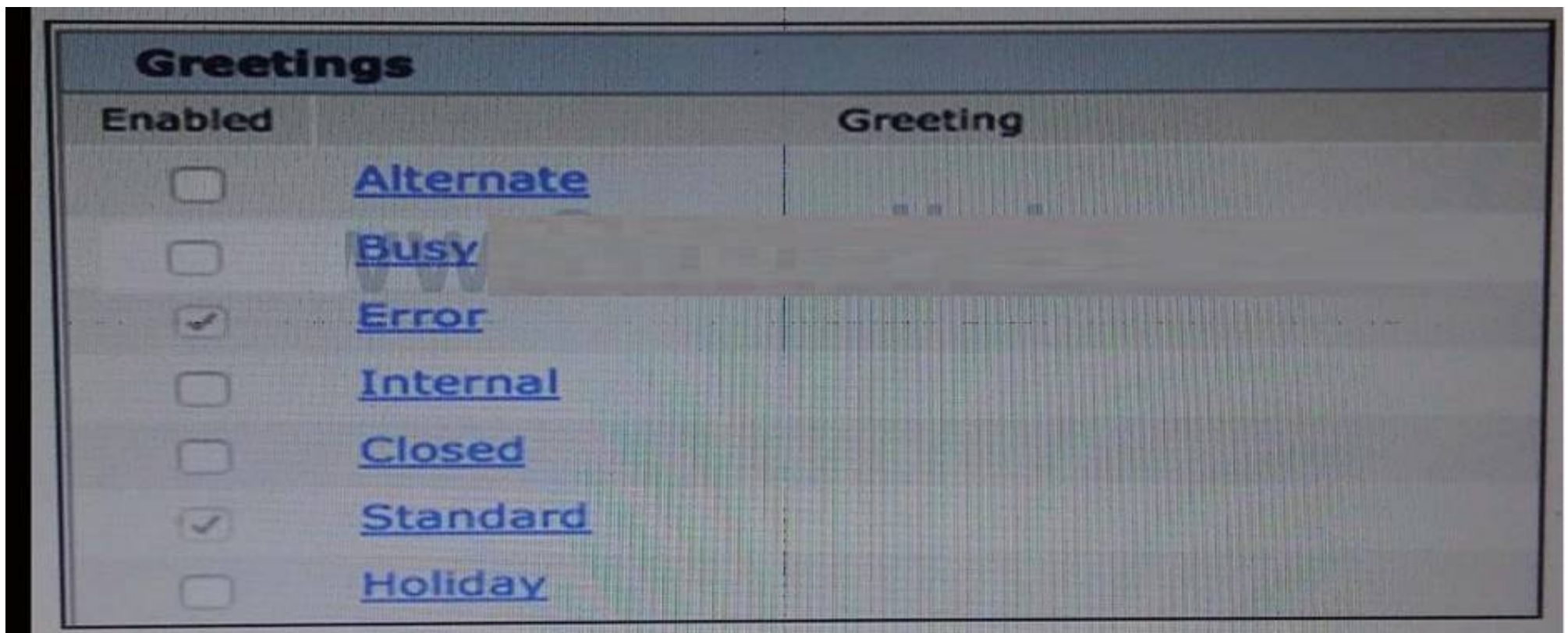
A collaboration engineer has been asked to implement secure real-time protocol between a Cisco Unified CM and its SIP gateway. Which option is a consideration for this implementation?

- 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 7

Exhibit:



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

- A. Enable standard, alternate and error greetings
- B. Enable holiday, closed and standard greetings
- C. Enable holiday, closed, error and standard greetings
- D. Enable standard and error greetings
- E. Enable holiday, closed, standard, alternate and error greetings

Answer: C

NEW QUESTION 8

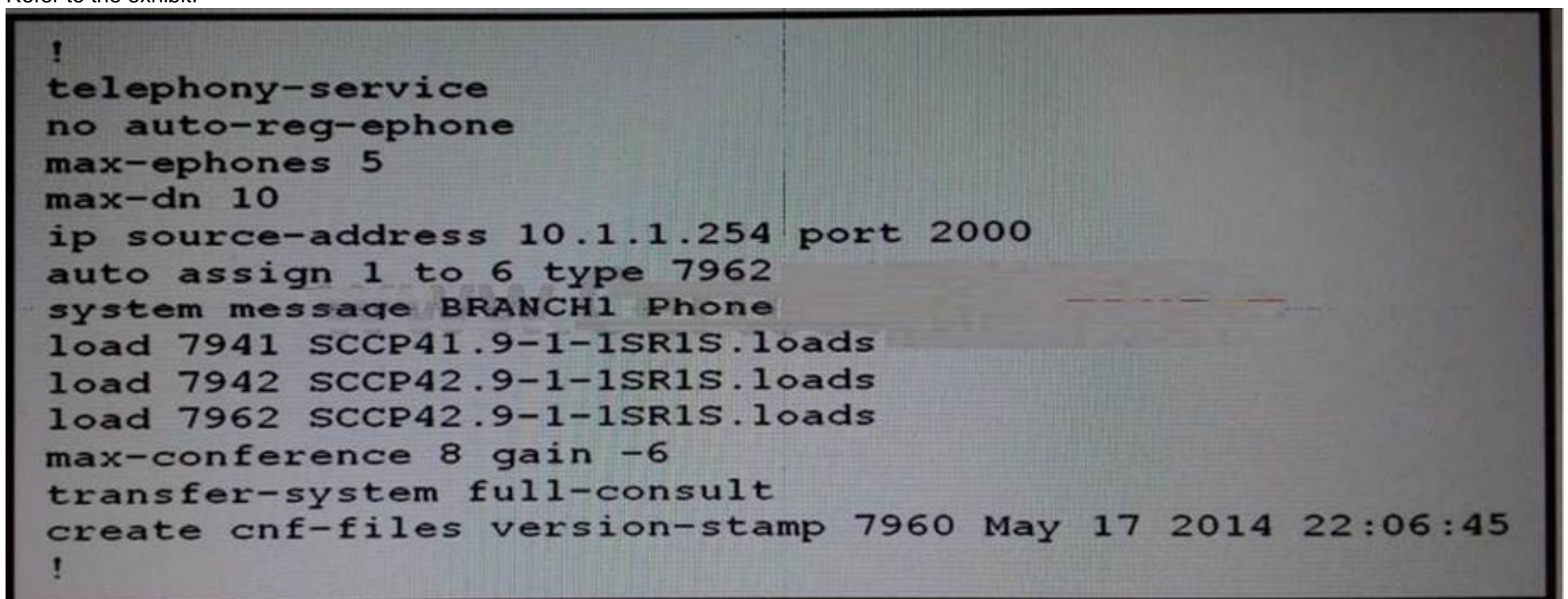
Which two settings must be the same between the backup source and restore target with DRS in Cisco Unified Communications Manager? (Choose two.)

- A. Server Hostname
- B. Server IP Address
- C. Cluster Security Password
- D. NTP Servers
- E. Domain Name
- F. Certificate Information

Answer: AB

NEW QUESTION 9

Refer to the exhibit.



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 10

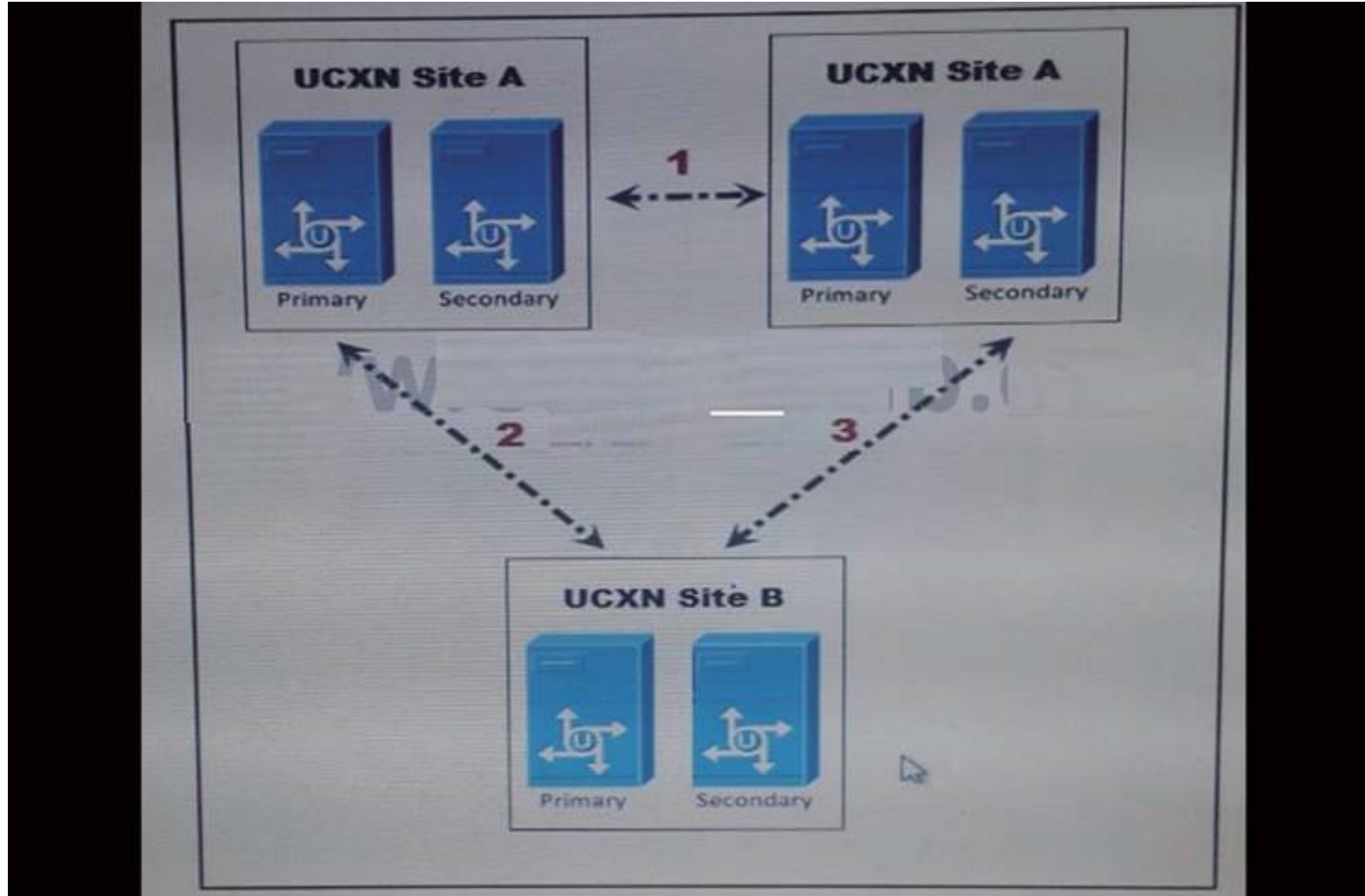
A SIP carried delivers DID to a Cisco Unified Border Element in the form of +155567810XX, where the last two digits could be anything from 00 to 99. To match the internal dial plan, that number must be changed to 6785XXX, where the last two digits should be retained. Which two translation profiles create the required outcome? (Choose two)

- A. rule 1 /555\(.*)\.*\(.*)/ ^150\2/
- B. rule 1 /+ 1555\(...)\.(\...)\\$/ ^15\2/
- C. rule 1 /^+ 1555\678\10\(\...)\\$/ ^150\2/
- D. rule 1 /^15+678\(\... \.)/678\1/
- E. rule 1 /.15+678?10?\(\...)/ /67850\1/

Answer: CE

NEW QUESTION 10

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 11

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 13

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

- A. Instant message and chat for Jabber using WebEx messenger service
- B. Softphone mode for Jabber using Cisco Unified Communications Manager
- C. Visual voicemail for Jabber on Cisco unity
- D. Audio and video conference for Jabber using WebEx meeting center

Answer: B

NEW QUESTION 15

Refer to the exhibit.

```
Jun  1 17:59:16.839: ISDN Se0/0/0:15 Q931: RX <- SETUP pd = 8  callref = 0x17FF
    Bearer Capability i = 0x8090A3
        Standard = CCITT
        Transfer Capability = Speech
        Transfer Mode = Circuit
        Transfer Rate = 64 kbit/s
    Channel ID i = 0xA98392
        Exclusive, Channel 18
    Progress Ind i = 0x8283 - Origination address is non-ISDN
    Calling Party Number i = 0x2181, '2014582589'
        Plan:ISDN, Type:National
    Called Party Number i = 0xC1, '6727498'
        Plan:ISDN, Type:Subscriber(local)
    Sending Complete
Jun  1 17:59:16.847: ISDN Se0/0/0:15 Q931: TX -> CALL_PROC pd = 8  callref = 0x97FF
    Channel ID i = 0xA98392
        Exclusive, Channel 18
PSTN_VG01#
```

A PSTN caller initiates an inbound call. Which two dial peers can be selected as inbound dial peers? (Choose two.)

- A. dial-peer voice 100 potsanswer-address [2-9]..[2-9]...\$ voice-port 0/0/0:23
- B. dial-peer voice 200 potsdestination-pattern [2-9]..[2-9]..[2-9]...\$ voice-port 0/1/0:15
- C. dial-peer voice 300 potsincoming called-number 704[2-9]...\$ voice-port 0/1/0:15
- D. dial-peer voice 400 pots answer-address 672[2-9]...\$ voice-port 0/0/0:15
- E. dial-peer voice 500 potsincoming called-number 6..[2345689]...\$ voice-port 0/1/0:15

Answer: BE**NEW QUESTION 17**

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

- A. Box-To-Box High Availability Support feature is not supported.
- B. Configure LATryl under a voice class or dial peer is not supported.
- C. SIP UPDATE message outlined in RFC3311 is not supported.
- D. Codec transcoding between LATM and other codecs is not supported
- E. Dual tone multi-frequency interworking with LATM codec is not supported.
- F. Basic calls using flow-around or flow-through is not supported.

Answer: DE**NEW QUESTION 18**

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 22**

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 25

Refer to the exhibit.

Cluster Detailed View from PUB (3 Servers):

| SERVER-NAME | IP ADDRESS | PING (ms) | RPC? | REPLICATION STATUS | REPL. QUEUE | DBVERS | RPL. LOOP? | REPL. REPLICATION SETUP (RTMT): & details |
|-------------|---------------|-----------|-----------|--------------------|-------------|--------|------------|---|
| CUCMPUB | 172.16.100.50 | 0.033 | Connected | 0 | Match | Yes | (3) | PUB Setup Completed |
| CUCNSub1 | 172.16.100.51 | 0.855 | Connected | 0 | Match | Yes | (4) | Setup Failed |
| CUCNSub2 | 172.16.100.52 | 1.025 | Connected | 0 | Match | Yes | (4) | Setup Failed |
| CUCNSub3 | 172.16.100.53 | 3.250 | Connected | 0 | Match | Yes | (4) | Setup Failed |

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

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

- A. run the command `utils dbreplication dropadmindb` on all subscribers
- B. run the command `utils dbreplication repairable all` from the publisher
- C. run the command `utils dbreplication stop` on the publisher
- D. run the command `utils dbreplication reset all` from the publisher
- E. run the command `utils dbreplication repair all` from the publisher
- F. run the command `utils dbreplication stop` on all subscribers

Answer: CDF

NEW QUESTION 26

Refer to the exhibit.



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

- A. unencrypted call signaling and media
- B. encrypted call signaling but unencrypted call media
- C. encrypted call media but unencrypted call signaling
- D. encrypted call signaling and media
- E. Not enough information provided to answer this question.

Answer: D

NEW QUESTION 29

Refer to the exhibit.



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

- A. The Cisco Unified Mobility Access Voice directory number is not shown in the exhibit
- B. 4155551234
- C. 5151234
- D. 5251234

Answer: B

NEW QUESTION 30

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 31

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

- A. Called party transformation at route list level is not shown on the display of the calling phone
- B. Called party transformations at route list level is reflected on the display of the calling phone
- C. Digit manipulation occur once per route list
- D. Only called transformation is available at route list level
- E. Called party transformations at route list level is replaced by called party transformations at route pattern level

Answer: E

NEW QUESTION 36

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 40

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 42

Refer to the exhibit.



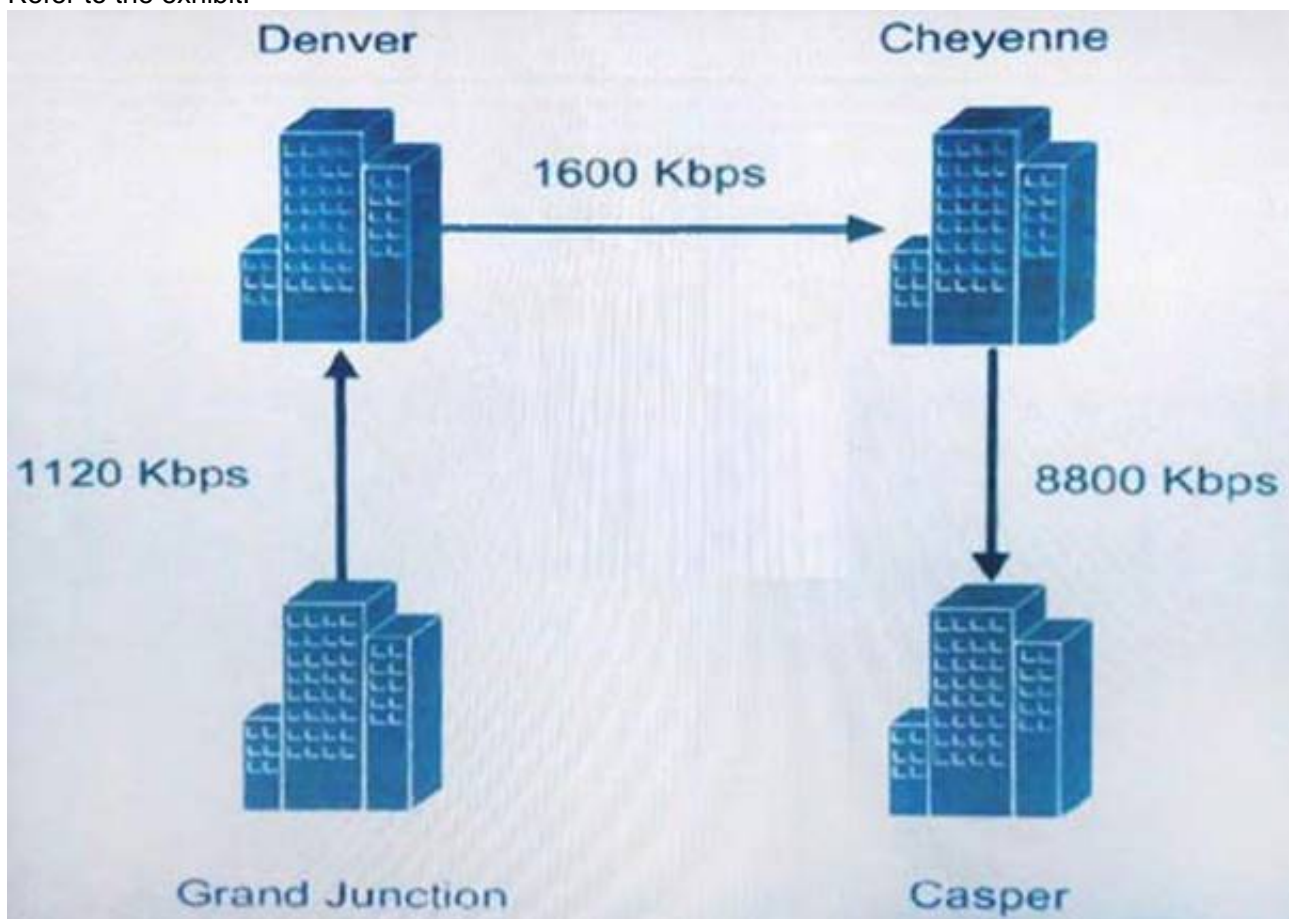
The SIP service Provider does not support re-invites unless media changes. Which two commands are needed on the CUBE to standby the SIP service provider requirements? (choose two)

- 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 47

Refer to the exhibit.



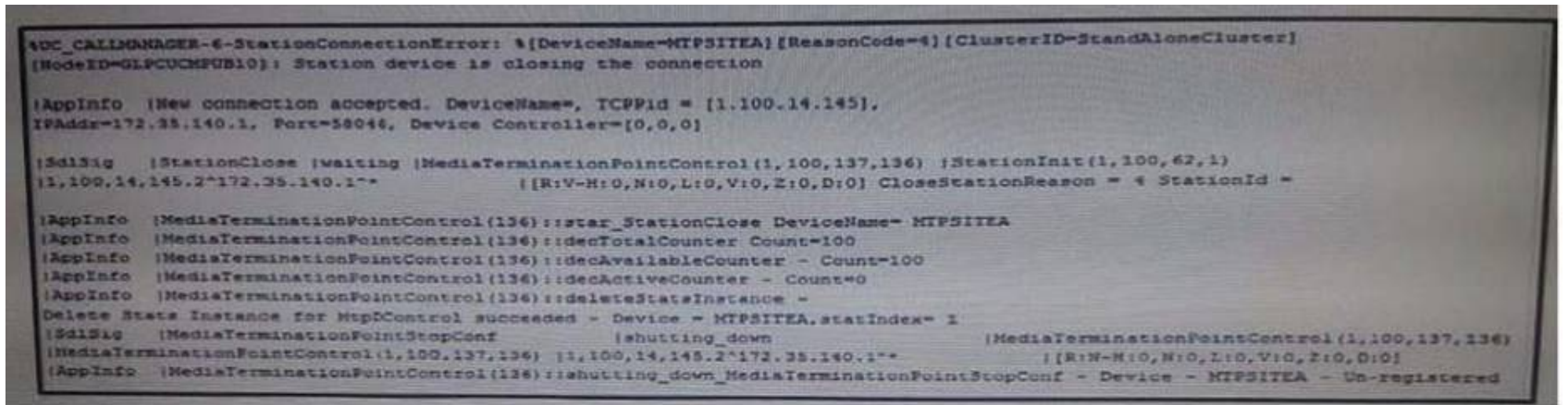
A collaboration engineer configures Cisco Unified CM location using G.711 and iLBC for each site. The bandwidth for each link is shown. Which two options represent the maximum concurrent number of calls supported by grand junction to Casper for each Codec? (Choose two.)

- 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 52

Refer to the exhibit.



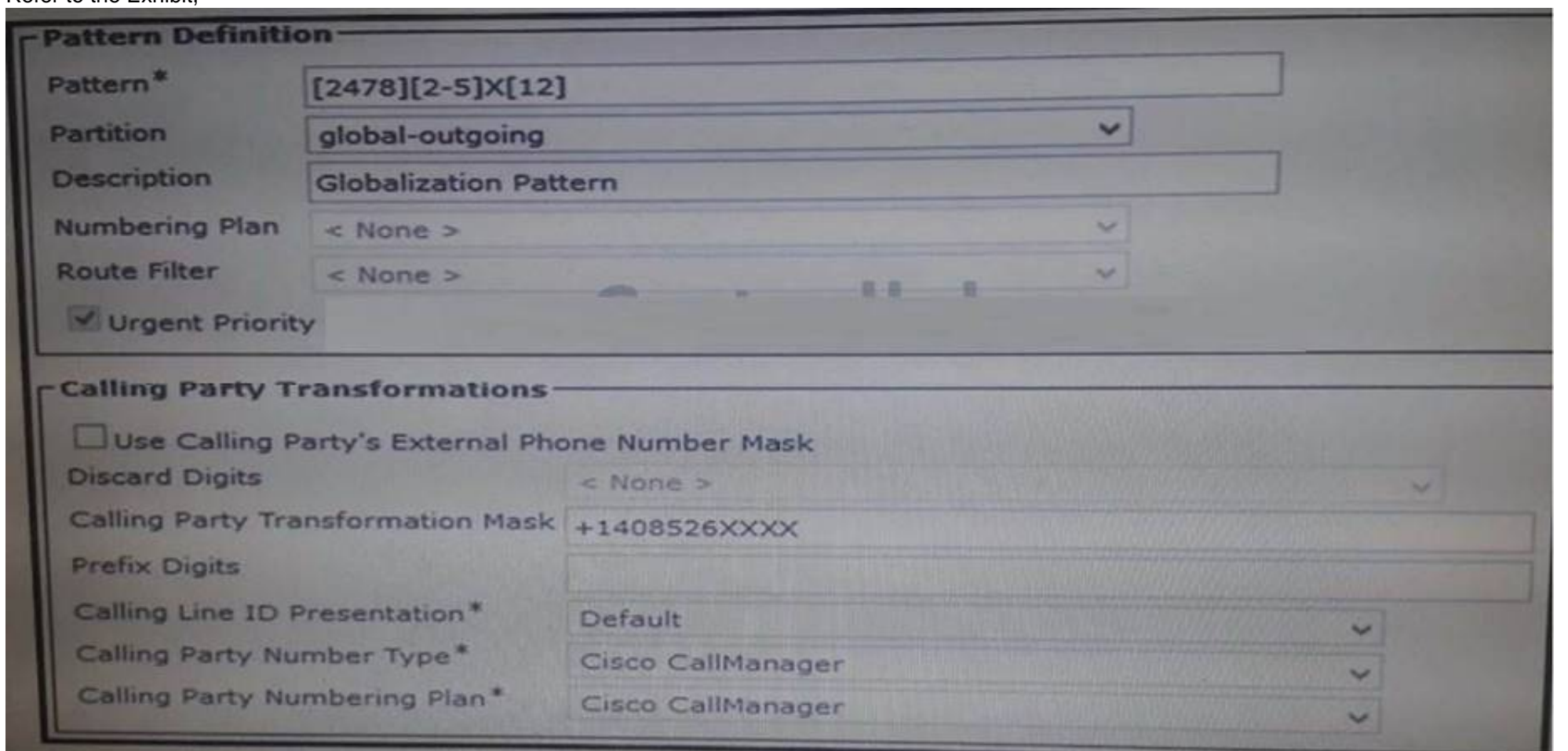
A cisco collaboration engineer discovers that an instance of IOS media termination point (MTP) could not maintain stable registration with CUCM. Call manager traces is showing in the exhibit. What is the reason for the flapping registration?

- A. The CCM version on IOS configuration does not match the CUCM version.
- B. The IOS MTP is experiencing high CPU and is missing its keep-alive.
- C. A Firewall is blocking port 2000 intermittently between IOS Device and CUCM.
- D. Another IOS Media device is attempting to register with the same name.

Answer: D

NEW QUESTION 53

Refer to the Exhibit,



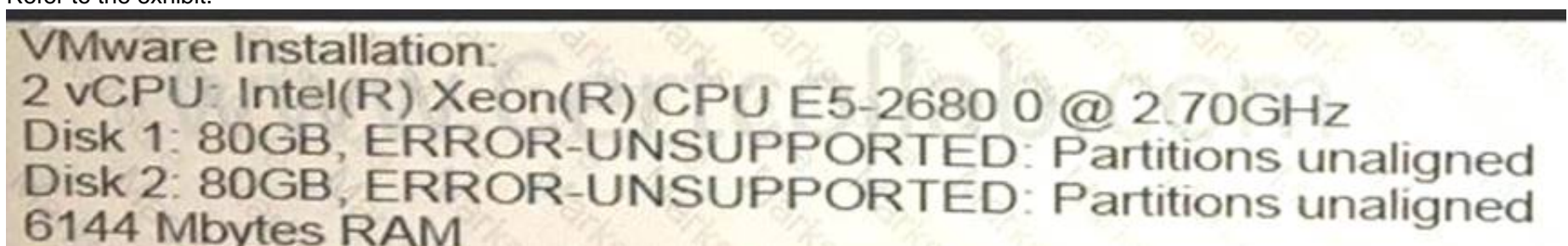
Which three cisco Unified CM internal extension match the globalization pattern shown and provide a globalized calling party number? (Choose three)

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

Answer: ADE

NEW QUESTION 56

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 57

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

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

Answer: B

NEW QUESTION 58

Refer to the exhibit.

| Router#show dial-peer voice sum | | | | | | | | | | | | |
|---------------------------------|------|--------|------|--------|--------------|---------|-----------|---------------|----------|--------|-----------|--|
| dial-peer hunt 0 | | | | | | | | | | | | |
| TAG | TYPE | AD MIN | OPER | PREFIX | DEST-PATTERN | PRE FER | PASS THRU | SESS-TARGET | OUT STAT | PORT | KEEPALIVE | |
| 4300 | voip | up | up | | 4... | 0 | syst | ipv4:10.1.1.4 | | | active | |
| 2300 | voip | up | up | | [2-3]... | 0 | syst | ipv4:10.1.1.3 | | | active | |
| 1111 | voip | up | down | | 1111 | 0 | syst | ipv4:10.1.1.1 | | | busy-out | |
| 20001 | pots | up | up | | 2001\$ | 0 | | | | 50/0/1 | | |
| 20002 | pots | up | up | | 2002\$ | 0 | | | | 50/0/2 | | |

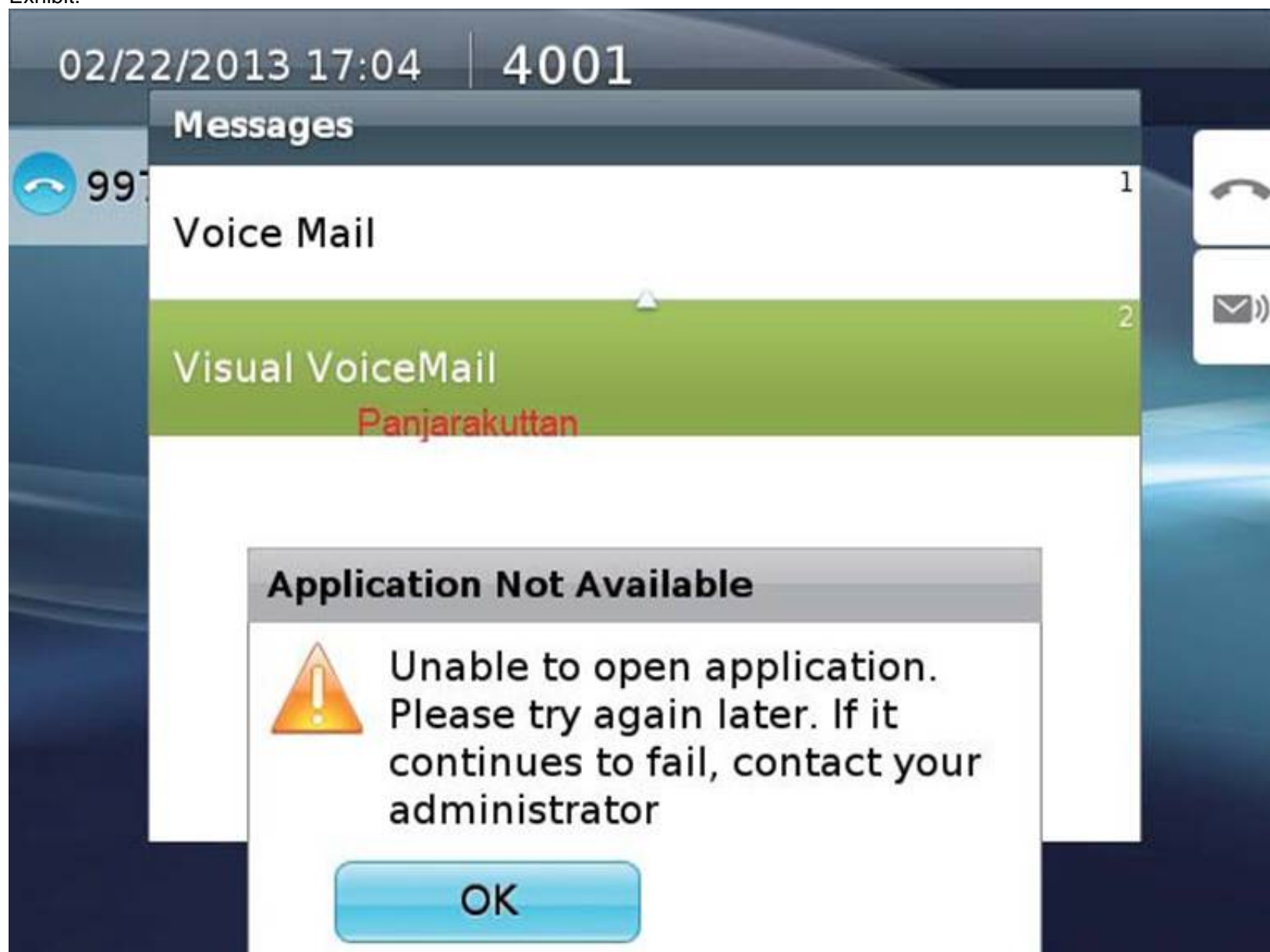
Which out-of-dialog SIP OPTIONS ping response put dial-peer tag 1111 into its current operational state?

- A. 401 Unauthorized
- B. 505 Version Not Supported
- C. 406 Not Acceptable
- D. 482 Loop Detected
- E. 500 Server Internal Error

Answer: B

NEW QUESTION 61

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 66

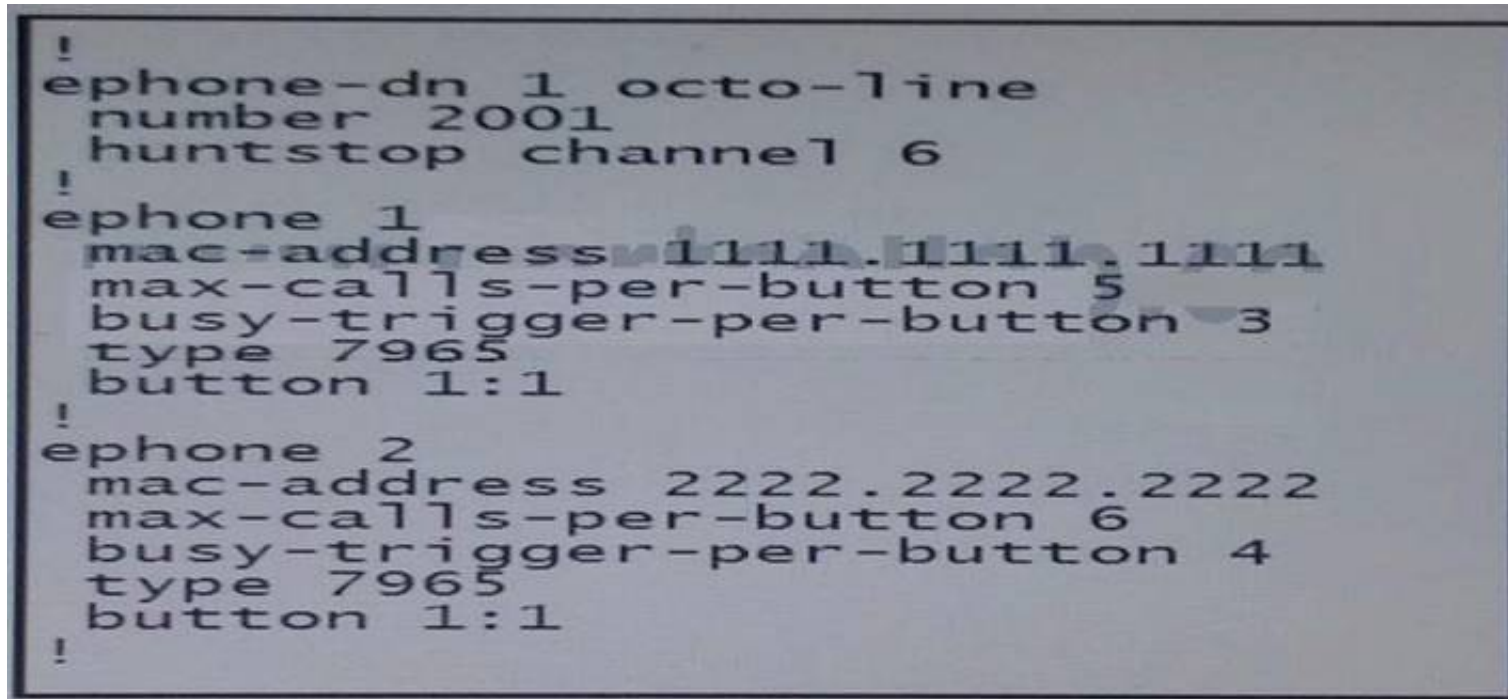
A company is closing two offices and are transferring employees to a new location over a 30 day period. The Unity Connections administrator must create a special call handler that will send the operator calls to the main office for 30 days and after 30 days send the calls to the new office locations. 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 68

Exhibit:



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

How many simultaneous outbound calls are possible with this Cisco Unified Communications Manager Express configuration on these two phones?

- A. 6
- B. 7
- C. 8
- D. 9
- E. 11

Answer: C

NEW QUESTION 69

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:

| |
|---|
| 4 |
| 2 |
| 1 |
| 0 |
| 3 |

NEW QUESTION 70

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 72

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 74

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 77

Which Cisco Unified CM service is responsible for writing Call Management Records into the CDR Analysis and Reporting database?

- A. Cisco CDR Agent
- B. Cisco CAR DB
- C. Cisco CDR Repository Manager
- D. Cisco CAR Scheduler
- E. Cisco Extended Functions

Answer: D

NEW QUESTION 78

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 81

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 82

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

- A. On Hook
- B. Connected
- C. On Hold
- D. Off Hook
- E. Ring Out
- F. Connected Conference

Answer: AD

NEW QUESTION 86

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's; 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 90

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 92

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

- A. Change the service type field
- B. Select the user corporate email address option
- C. Uncheck synchronize connection and exchange mailboxes
- D. Update the Cisco unified Messaging service

Answer: D

NEW QUESTION 94

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

- A. Resume
- B. EndCall
- C. Select
- D. Barge
- E. NewCall
- F. cBarge
- G. Join

Answer: DEF

NEW QUESTION 99

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 sip-header Diversion modify "sip:(.*>)" "123456@company.com>"
- B. voice class sip-profiles 200request INVITE sdp-header Diversion modify "sip:(.*>)" "123456@company.com>" request REINVITE sdp-header Diversion 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 103

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 104

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 107

Refer to the exhibit.

The screenshot displays the 'Forwarded Routing Rule' configuration page for 'Emergency Message'. The 'Status' is set to 'Active'. The 'Language' is set to 'Inherit Language from Caller'. The 'Search Scope' is 'ucxnprod Search Space'. The 'Send Call to' is set to 'Call Handler' and 'Emergency Message'. The 'Routing Rule Conditions' section shows a table with one condition: 'Emergency Message' with status 'Active'. The 'Edit Greeting (Standard)' section shows 'Status' as 'Greeting Enabled with No End Date and Time'. The 'Callers Hear' section shows 'System Default Greeting'. The 'After Greeting' section shows 'Call Action' as 'Route From Next Call Routing Rule' and 'Call Handler' as '3000'. The 'Forwarded Routing Rules in Descending Order of Precedence' table at the bottom lists three rules: 'Emergency Message' (Active), 'Attempt Forward' (Active), and 'Opening Greeting' (Active).

| Display Name | Status | Dialed Number | Calling Number | Forwarding Station | Phone System | Port | Send Call to |
|-------------------|--------|---------------|----------------|--------------------|--------------|------|-----------------------|
| Emergency Message | Active | | | | | | Greeting Conversation |
| Attempt Forward | Active | | | | | | Attempt Forward |
| Opening Greeting | Active | | | | | | Transfer Conversation |

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

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

Answer: B

NEW QUESTION 112

Which two SDP content headers can be found in a SIP INVITE message? (Choose two.)

- A. Expires
- B. Contact
- C. Connection Info
- D. Media Attributes
- E. Allow
- F. CSeq

Answer: CD

NEW QUESTION 116

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 121

Refer to the exhibit.

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

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

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

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

Answer: C

NEW QUESTION 122

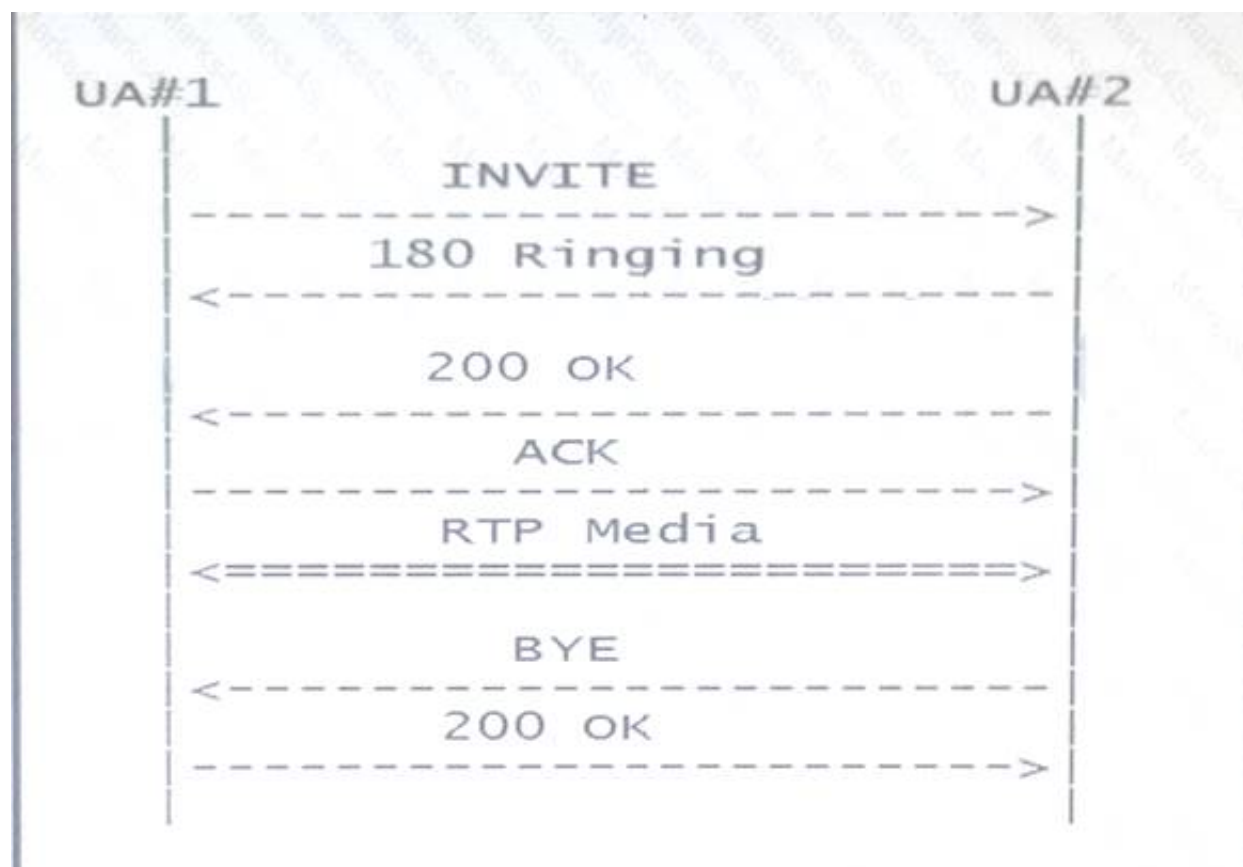
ACisco collaboration engineer is troubleshooting unexpected SIP call disconnects. Which three responses correspond to the 5xx range? (Choose three.)

- A. Temporarily Unavailable
- B. Version not supported
- C. Bad request
- D. Forbidden
- E. Server Timeout
- F. Service Unavailable

Answer: BEF

NEW QUESTION 124

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 125

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

- A. An email received from a colleague.
- B. An email accessed via a web browser.
- C. An email saved on a USB drive.
- D. An email archived locally on a laptop hard drive.
- E. An email sent to a colleague.

Answer: CD

NEW QUESTION 127

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 129

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

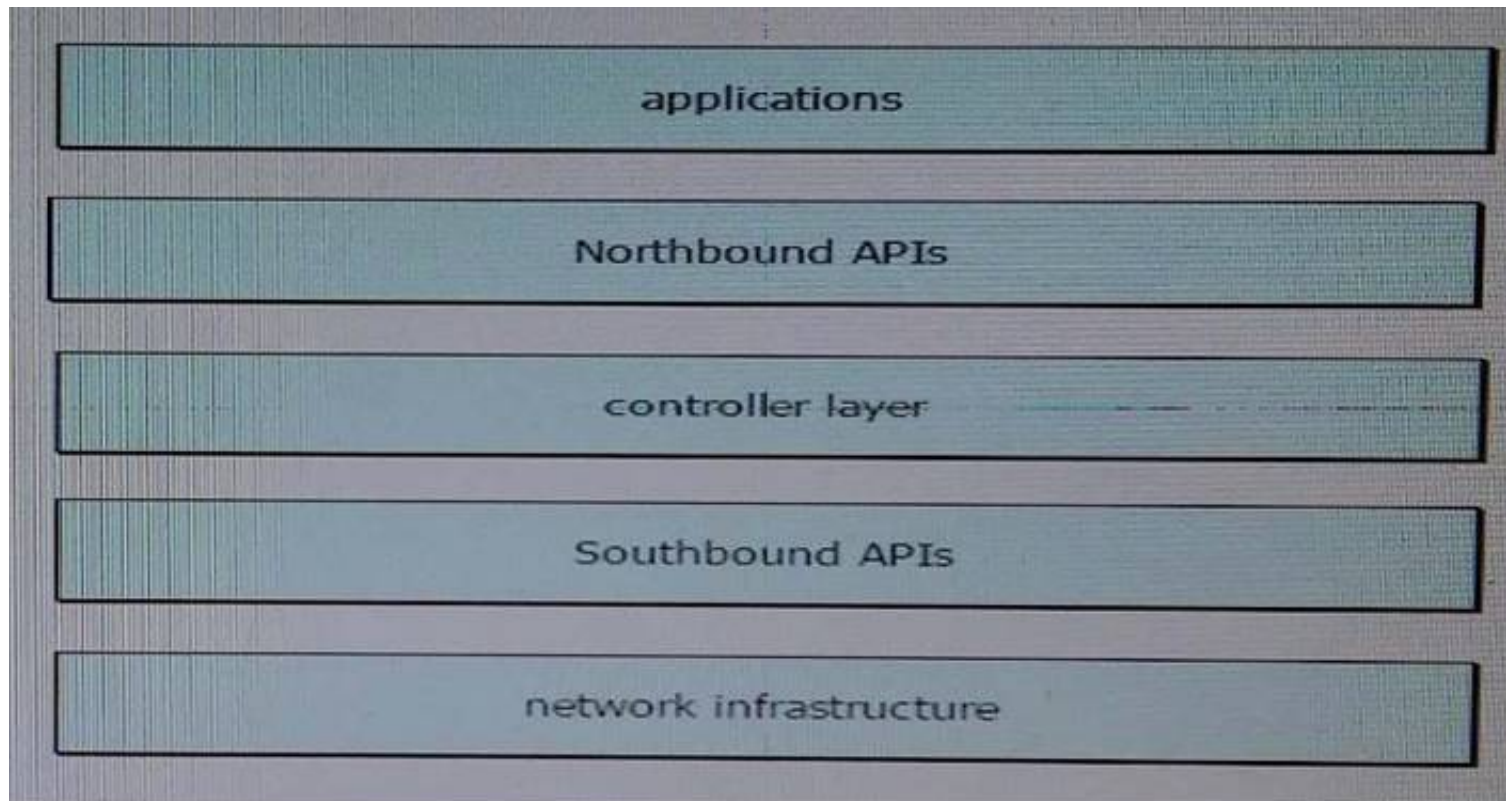
NEW QUESTION 130

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

| | |
|------------------------|---|
| Southbound APIs | 5 |
| Northbound APIs | 4 |
| Applications | 3 |
| Controller Layer | 2 |
| Network infrastructure | 1 |

Answer:

Explanation:



NEW QUESTION 131

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 135

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

- A. NOTIFY
- B. TRYING
- C. PRACK
- D. USE PROXY
- E. UNAUTHORIZED
- F. OPTIONS

Answer: ACF

NEW QUESTION 140

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

- A. The UAC must include an answer in a PRACK
- B. The UAC must include an answer in a PRACKACK
- C. The UAC may include an answer in PRACKACK
- D. The UAC may include an answer in PRACK
- E. The UAC may include an answer in ACK

Answer: B

NEW QUESTION 144

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

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

Answer: AD

NEW QUESTION 145

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 146

Exhibit:



Which two wireless security modes offer these configuration options on a Cisco 9971 IP Phone? (Choose two)

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

Answer: AF

NEW QUESTION 148

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 153

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

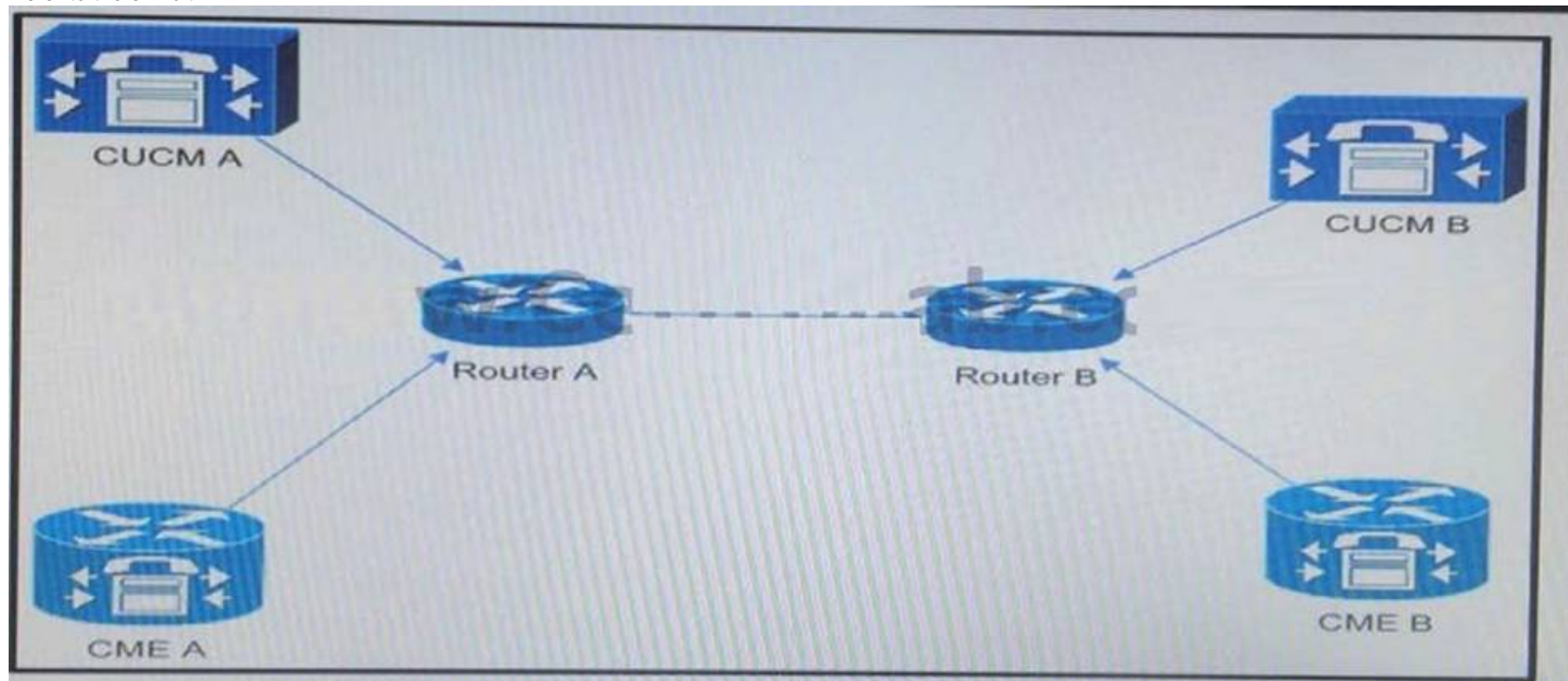
- A. 60 seconds
- B. 1 seconds
- C. 1440 seconds

- D. 600 seconds
- E. 3600 seconds

Answer: A

NEW QUESTION 158

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 162

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 163

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

- A. Microsoft RCC must be enabled
- B. The enable use of Email Address when federating option can be turned on if SIP URI's are different between IM&P and Lync.
- C. You must update the URIs of any users migrated from Lync to IM&P to match the Cisco Unified Presence Server SIP URI format.
- D. Intradomain Federation to Lync is only possible using SIP
- E. A static route must be added to point the local presence domain to the Lync server.
- F. IM&P and Lync should federate to any required remote domains.

Answer: CDE

NEW QUESTION 166

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

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

What could resolve this issue? (Choose two)

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

- B. The value of remote destination limits should be change to 2 instead of the default value of 4 under the end user page
- C. The enable mobile voice access option must be selected under the end user page
- D. The value of maximum wait time for desk pickup should be change 20000 instead of the default of 10000, under the end user page
- E. The rerouting calling search space assigned to the user's remote destination profile must have access to international calls

Answer: AE

NEW QUESTION 169

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 171

Exhibit:

```
Gatekeeper#show gatekeeper endpoint

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

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

- A. Device Name on the Cisco Unified CM Gatekeeper configuration page is HQGK.
- B. Device Name on the Cisco Unified CM H.225 Trunk (Gatekeeper Controlled) configuration page is HQGK.
- C. Device Name on the Cisco Unified CM H.225 Trunk (Gatekeeper Controlled) configuration page is HQGK_1, HQGK_2.
- D. Device Name on the Cisco Unified CM Gatekeeper configuration page is HQGK_1, HQGK_2.
- E. Not enough information has been provided to answer this QUESTION NO:.

Answer: B

NEW QUESTION 173

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

- A. Only T.38 and Cisco fax protocol are supported
- B. SIP require the all the time be sent in GMT
- C. Call hold RE-INVITE is not supported
- D. SRTP is supported only in cisco IOS 15.x and higher

Answer: B

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

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

NEW QUESTION 175

Refer to the exhibit.



Which two wireless security modes offer these configuration options on a Cisco 9971 IP Phone? (Choose two)

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

Answer: AF

NEW QUESTION 176

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,

- 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 178

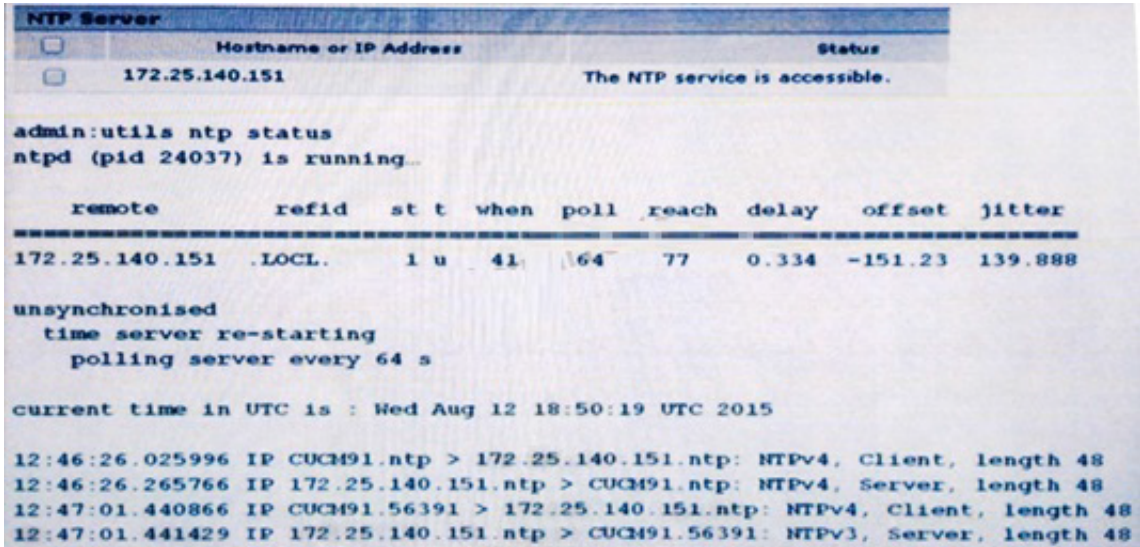
Which two parameters, in the reply of an MGCP gateway to an Audit Endpoint message, indicate to a Cisco Unified CM that it has an active call on an endpoint? (Choose two)

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

Answer: AD

NEW QUESTION 182

Refer to the exhibit.



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

- A. The NTP server used is a Windows based NTP server.
- B. The IOS Command NTP server 172.25.140.151 version 3 is advertising NTPv3.
- C. The NTP server stratum is higher than four.
- D. A firewall is blocking NTP port 123.

Answer: B

NEW QUESTION 184

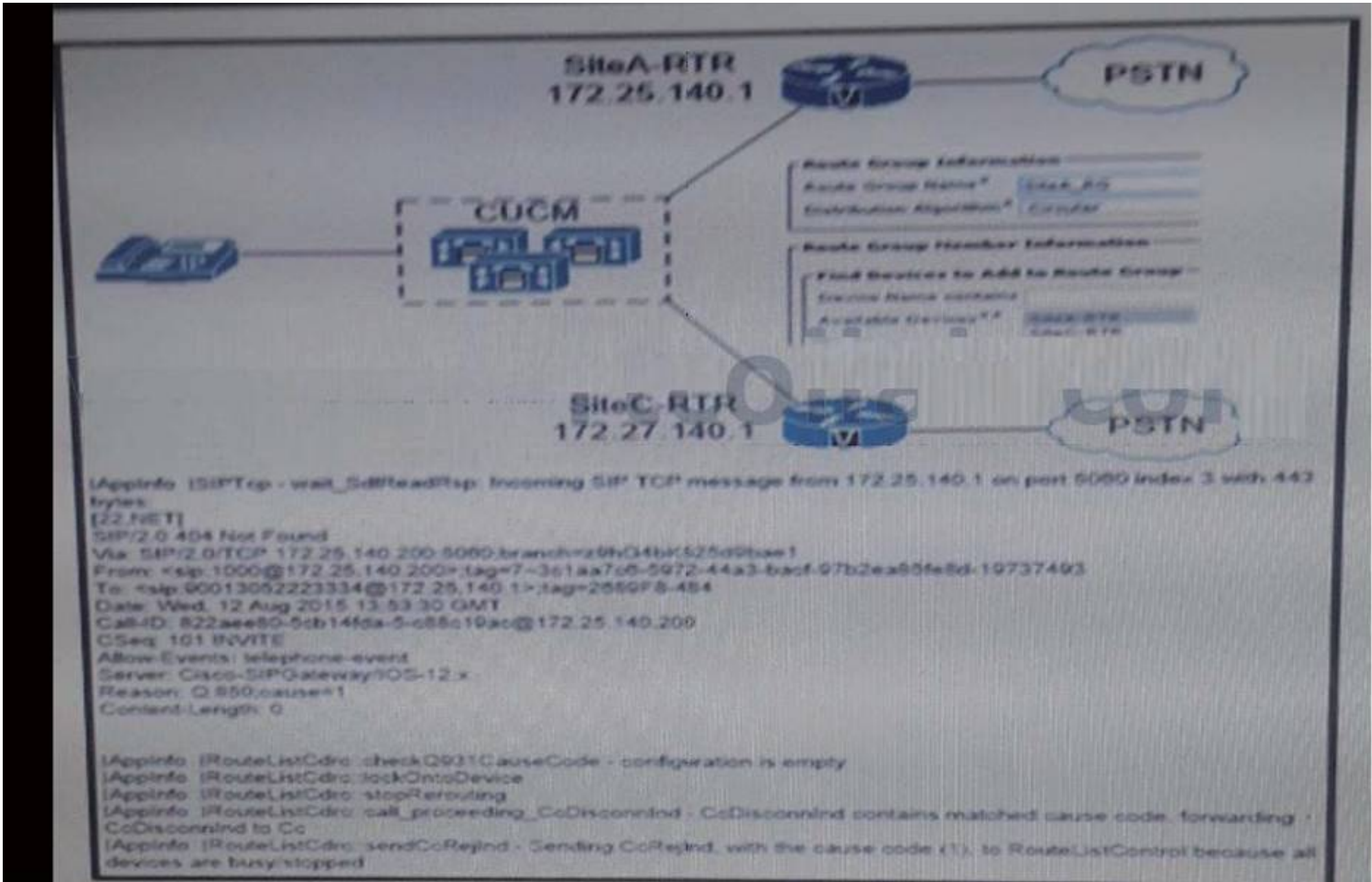
A collaboration engineer is designing a Cisco Unity Connection network for a large client running 10 x. The client has 12 locations, each with their own Cisco Unity Connection cluster. Which two designs are valid? (Choose two)

- 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 189

Refer to the exhibit.



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

is CUCM unable to extend the call setup through SiteC-RTR?

- A. Stop routing on Q.931 Disconnect cause code is set to 27
- B. Stop routing on Unallocated Number Flag is set to true
- C. Stop Routing on User Busy Flag is set to true
- D. Retry count for SIP Invite is set to 1
- E. Retry count for SIP response is set to 1

Answer: B

NEW QUESTION 194

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 195

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

- A. Destination RDP
- B. Source Phone
- C. Source RDP
- D. Destination phone

Answer: B

NEW QUESTION 198

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

- A. The phone is registered to the CUCM cluster but need to update its firmware
- B. The phone is attempting to register to the CUCM cluster for the first time
- C. Auto-registration is disabled on CUCM cluster
- D. The phone has not yet been defined in the CUCM database
- E. The phone is attempting to change from SIP firmware to SCCP firmware
- F. Auto-registration is enabled on CUCM cluster

Answer: BDF

NEW QUESTION 203

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

- A. They cannot be advertised as PSTN fail over number
- B. They can be added into local partition
- C. If the number mask is not configured, the alternate number is invalid
- D. They cannot be added into local partition
- E. They are not eligible to be advertised using Global Dial Plan Replication
- F. If the number mask it is not configured, use DN as alternate number

Answer: BF

NEW QUESTION 207

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 212

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 214

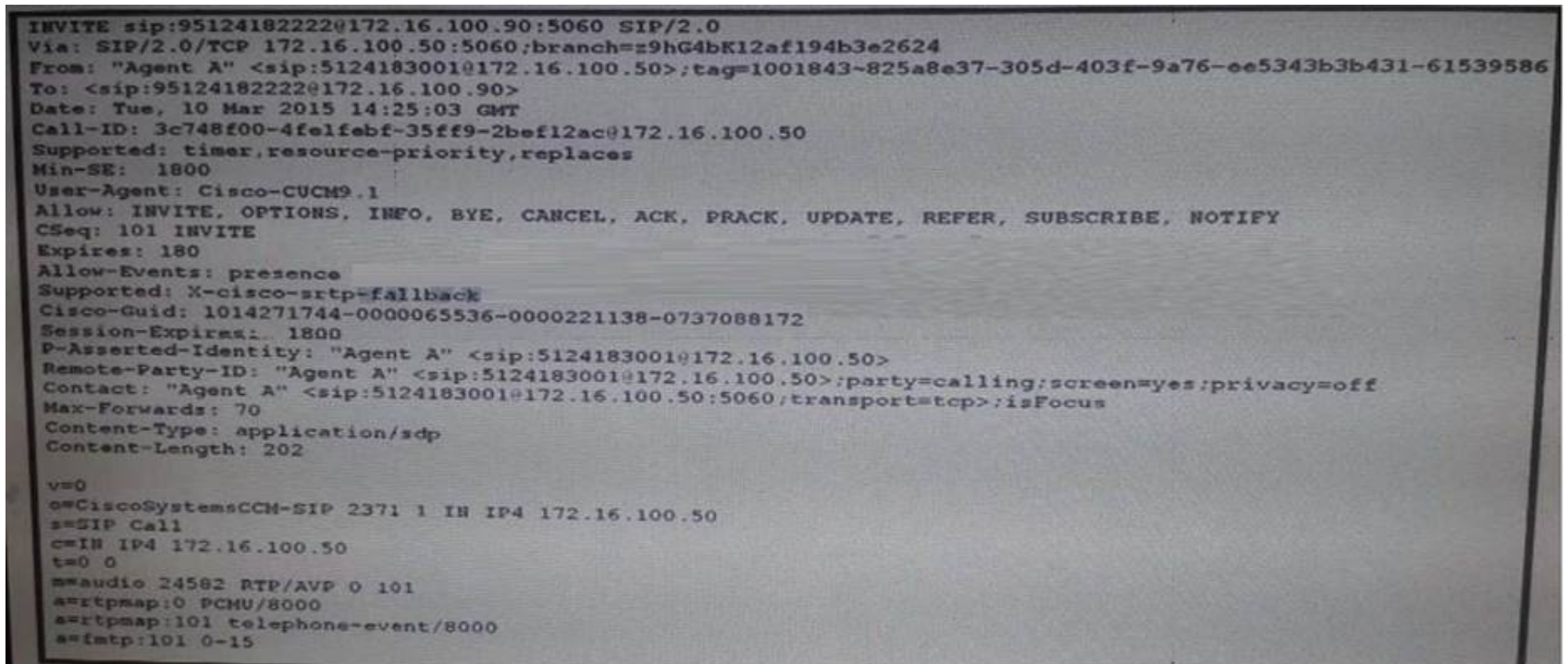
What is the maximum number of subclusters supported on a Cisco IM & Presence

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

Answer: B

NEW QUESTION 216

Refer to the Exhibit.



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 217

Refer to the exhibit.



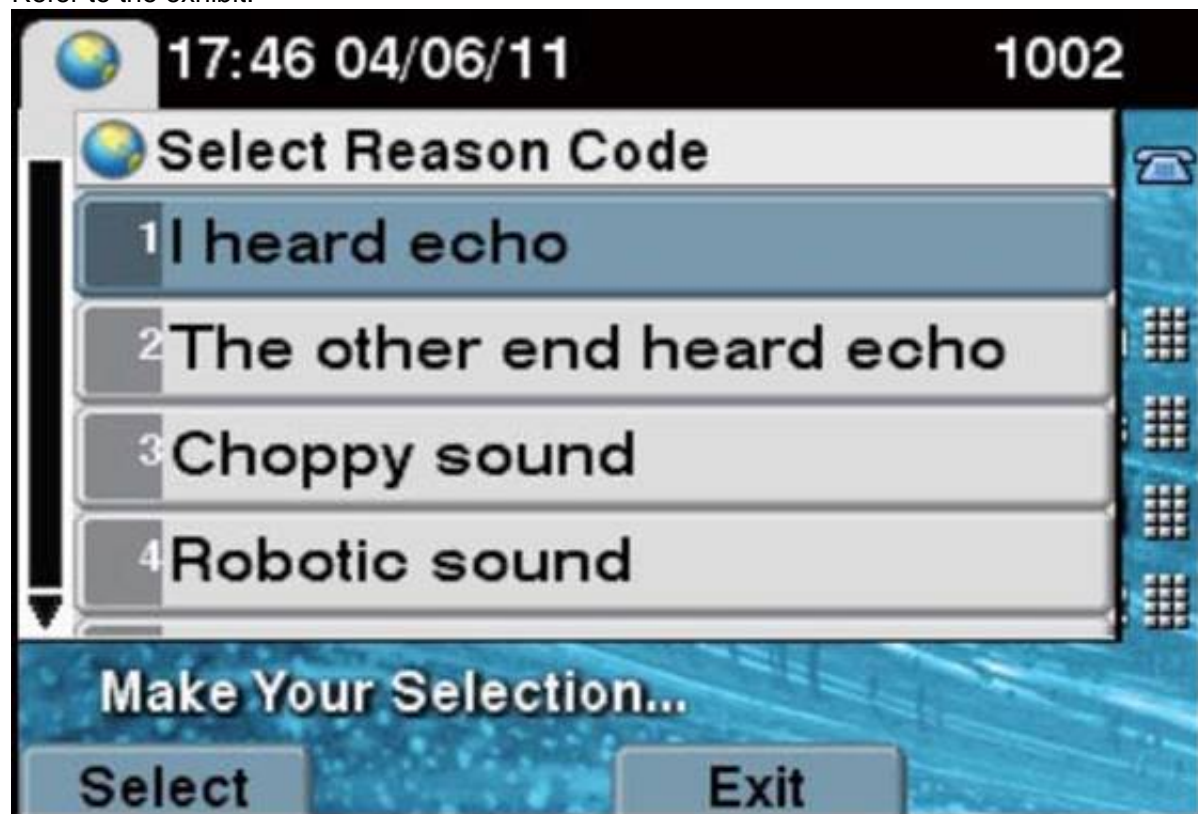
Which description of the event captured in the SIP message on a Cisco Unified Communication Manager Express router with Cisco Unified Express and registered IP phones (SIP and SCCP) is true?

- 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 219

Refer to the exhibit.



Which Cisco Unified CM service interfaces with Cisco IP Phones to allow users to report audio and other general problems on the phones?

- A. Cisco Serviceability Reporter
- B. Cisco Audit Event Service
- C. Cisco CallManager Serviceability
- D. Cisco Extended Functions
- E. Cisco RTMT Reporter Servlet

Answer: D

NEW QUESTION 223

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 227

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

- A. Concatenation of a remote destination profile CSS and line CSS
- B. calling number transformation CSS of the remote destination
- C. remote destination profile rerouting CSS
- D. phone line level CSS of the mobility user
- E. trunk or gateway inbound CSS
- F. mobility user's phone device level CSS

Answer: AD

NEW QUESTION 229

Refer to the exhibit.



| Certificate Name | Certificate Type | .PEM File |
|-------------------|------------------|--|
| tomcat | certs | tomcat.pem |
| ipsec | certs | ipsec.pem |
| tomcat-trust | trust-certs | PUB.pem |
| tomcat-trust | trust-certs | VeriSign Class 3 Secure Server CA - G3.pem |
| tomcat-trust | trust-certs | SUB.pem |
| ipsec-trust | trust-certs | PUB.pem |
| CallManager | certs | CallManager.pem |
| CAPF | certs | CAPF.pem |
| TVS | certs | TVS.pem |
| CallManager-trust | trust-certs | Cisco Manufacturing CA.pem |
| CallManager-trust | trust-certs | CAPF-8b60ebb5.pem |
| CallManager-trust | trust-certs | CAP-RTP-002.pem |
| CallManager-trust | trust-certs | CAP-RTP-001.pem |
| CallManager-trust | trust-certs | SUB.pem |
| CallManager-trust | trust-certs | CAPF-fbb7020d.pem |
| CallManager-trust | trust-certs | Cisco Root CA 2048.pem |
| CAPF-trust | trust-certs | Cisco Manufacturing CA.pem |
| CAPF-trust | trust-certs | CAPF-8b60ebb5.pem |
| CAPF-trust | trust-certs | CAP-RTP-002.pem |
| CAPF-trust | trust-certs | CAP-RTP-001.pem |
| CAPF-trust | trust-certs | Cisco Root CA 2048.pem |

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

- A. PUB.pem tomcat-trust trust-cert
- B. SUB.pem CallManager-trust trust-cert
- C. CAPF.pem CAPF cert
- D. TVS.pem TVS cert
- E. CallManager.pem CallManager cert

Answer: E

NEW QUESTION 230

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

- All IP phones should use audio codec at less than 10K bandwidth and maintain voice quality
- The IP phone audio codec selected should be optimized for speech and music
- The audio codec should be used in outbound SIP early offer signaling

Which configuration best satisfy the requirement?

A)

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

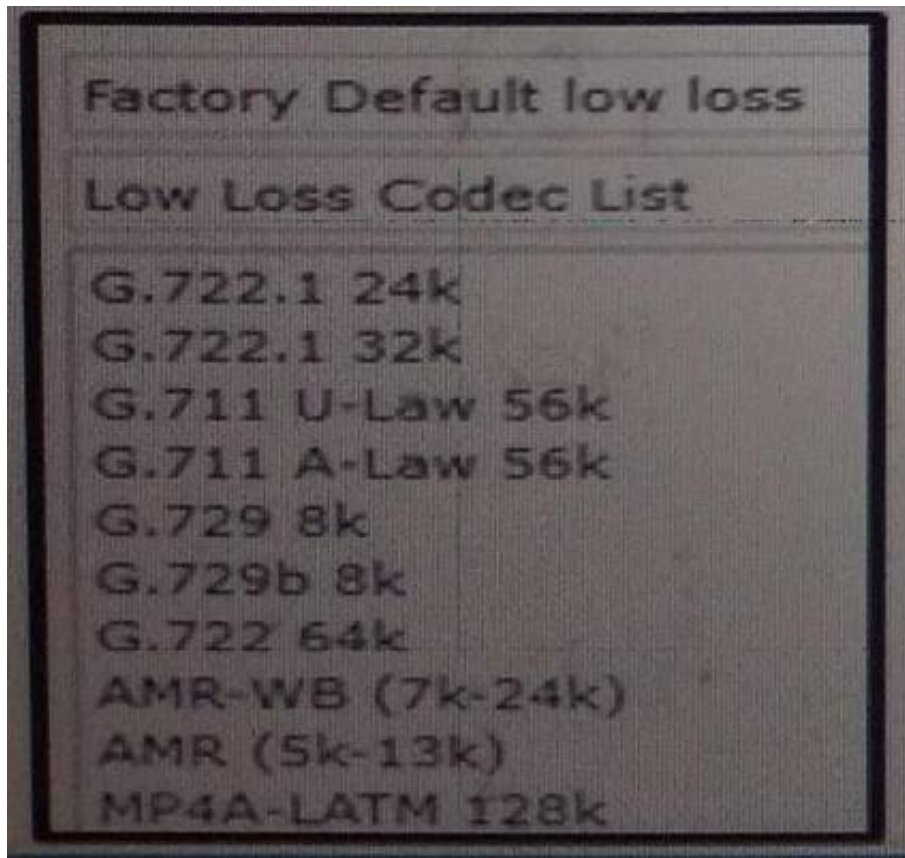
B)

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

C)

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

D)



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

Answer: C

NEW QUESTION 231

Refer to the exhibit.

SAF Security Profile Info

Name* CUCMPUB1

Description

User Name* SAFUSER

User Password* *****

SAF Forwarder Info

Name* SITECFW

Description

Client Label* CUCMPUB1

SAF Security Profile* CUCMPUB1

SAF Forwarder Address* 172.25.140.1

SAF Forwarder Port* 5050

☒ Enable TCP Keep Alive

[Show Advanced](#)

CCD Advertising Service Info

Name* CCDCUCMPUB

Description

SAF SIP Trunk* CCDSIPTRUNK

SAF H323 Trunk* < None >

HostedDN Group* CUCMPUB1

☒ Activated Feature

Media Information

| | |
|------------------------------|------------------------|
| Media | SIP Trunk |
| Device Protocol | SIP |
| Trunk Service Type | Call Control Extension |
| Device Name* | CCDSIPTrunk |
| Description | |
| Device Type* | Default |
| Connect Device Configuration | < None > |
| Call Classification* | Use Trunk-Default |

SiteC-RTR

```

router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
external-client CUCMPUB1
exit-sf-topology
exit-service-family
service-family external-client listen ipv4 5050
external-client CUCMPUB1
username SAFUSER
password SAFPASSWORD

```

Call Manager Service Traces

```

00015678.020 |00:26:13.339 |AppInfo |SAFClientProtocol
- Incoming MI incorrect - errorInMI= 2
00015678.021 |00:26:13.339 |AppInfo |SAFClientProtocol
- errorInMI= 2 errorInUserName= 1 errorInRealm= 0
errorInClientHandle= 1 errorInNonce= 1
00015678.022 |00:26:13.339 |AppInfo |SAFClientControl
- processReceivedSafErrorResponse
00015678.023 |00:26:13.339 |AppInfo |Begin
SAFClientControl: processRegisterError (.) error-code
value= 431
00015678.024 |00:26:13.340 |AppInfo
|SAFForwarderError - SAF Forwarder error response sent
to Unified CM IP Address: 172.25.140.1 SafClientHandle 0
Application User Name: SAFUSER SAF Protocol Version
Number: Reason Code and Description 431 Service ID 0
Sub Service ID: 0 App ID: Cisco CallManager Cluster
ID: StandAloneCluster Node ID: CUCMPUB10

```

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 233

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 238

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 241

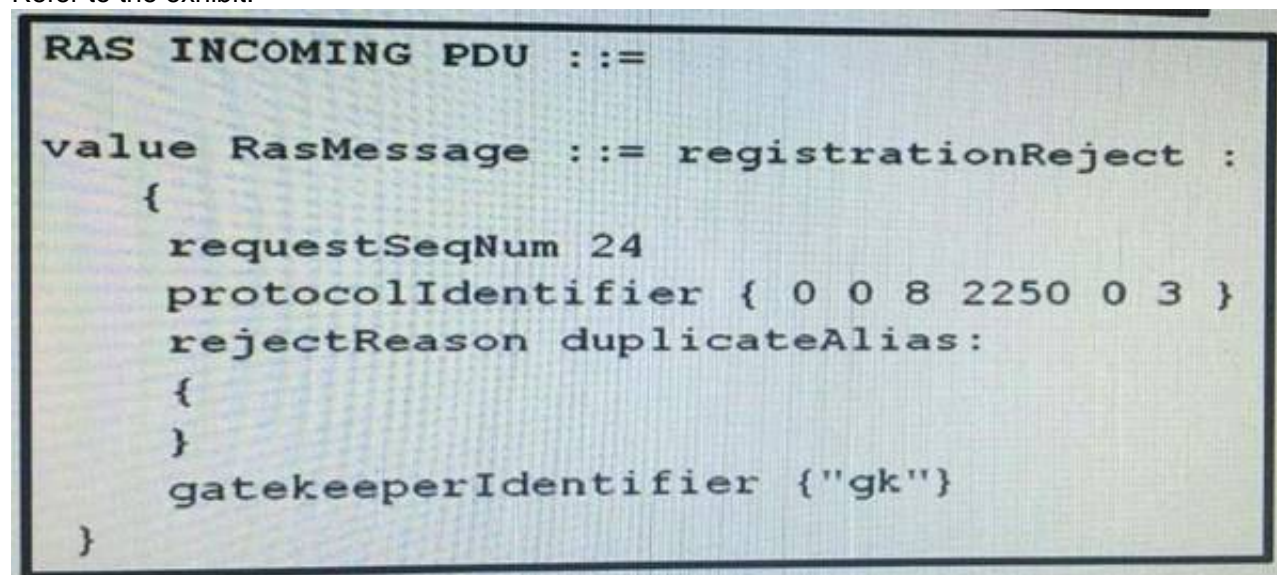
After configuring EM in the CUCM cluster, users are receiving 'Host not found' error message after pressing the Services button. What should be done to fix this problem?

- 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 244

Refer to the exhibit.



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

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

- A. The same zone prefix is configured in two different gatekeepers
- B. The same H323-ID is configured in two different gateways
- C. The same gw-type-prefix is configured in two different zone subnets IDs
- D. The same zone subnet ID is configured in two different gatekeepers
- E. The same E164-ID is configured in two different gateways

Answer: 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 245

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 248

A user is troubleshooting an FXO line on a Cisco IOS router that remains connected even after the call ends. Which three disconnect methods can be configured to fix this problem? (Choose three)

- A. Loop Current Feed Open Signalling Disconnect
- B. Hookflash Duration Signalling Disconnect
- C. CP-TONE Dual Supervisory Disconnect
- D. Power Denial-based Supervisory Disconnect
- E. Ground-start Signalling Disconnect
- F. Tone-based Supervisory Disconnect

Answer: DEF

NEW QUESTION 251

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 253

Refer to the exhibit.

```
router eigrp 1
|
  service-family ipv4 autonomous-system 1
    sf-interface loop0
      no split-horizon
    exit-sf-interface
    topology base
    exit-sf-topology
  exit-service-family
|

client username cmcluster passwrod 0 password12345
domain 1 default

voice service saf
profile trunk-toute 1
  session protocol h323 interface Gig0/0 transport tcp port 1720
|
profile dn-block 1
pattern 1 type extension 3xxx

profile callcontrol 1
dn-service
trunk-route 1
dn-block 1
|
channel 1 vrouter 1 asystem 1
subscribe callcontrol wilcarded
publish callcontrol 1
|

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

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

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

Answer: B

NEW QUESTION 255

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 258

Refer to the exhibit.



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

- A. The token is locked for five days, after which the retry counter resets.
- B. The token is locked until unlocked by Cisco TAC.
- C. The token is locked until Cisco CTL Client is uninstalled and reinstalled on the client PC.
- D. The token cannot be used on the same client PC again
- E. It can be used with another Cisco CTL Client on a different PC.
- F. The token is locked forever.

Answer: E

NEW QUESTION 262

Assume that your customer domain is customer.com and the Cisco Unified Communication Manager IM & Presences environment is 8.X
Which internal DNS SRV record(s) is needed on the DNS server to facilitate service discovery so the Cisco Jabber Client can automatically find the appropriate servers to connect?

- A. _cisco-uds_tcp customer.com
- B. _collab-edge_tls customer.com
- C. _cuplogin_tcp customer.com
- D. _cisco-uds_tls customer.com
- E. _collab-edge_tcp customer.com

Answer: C

NEW QUESTION 266

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

F)

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

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

- A. storage arrays such as those from EMC or NetApp, if applicable
- B. configuration of virtual-to-physical network interface mapping
- C. step-by-step procedures for hardware BIOS, firmware, drivers, and RAID setup
- D. server model and local components (CPU, RAM, adapters, local storage) at the part number level
- E. configuration settings and patch recommendations for VMware software

Answer: D

NEW QUESTION 275

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 276



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

- A. A "487 Request Terminated" is sent by Alice
- B. An ACK sent by Alice
- C. A "199 Early Dialog Terminated" is sent by Alice
- D. A "202 Accepted" is sent by Alice
- E. A "200 OK" is sent by Alice

Answer: E

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 278

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

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

Answer: CF

NEW QUESTION 281

In Cisco Unity Connection, which three configuration dialog boxes can a user assign a search space? (Choose three.)

- A. Routing Rule
- B. Call Handler
- C. Interview Handler
- D. Contacts
- E. Users
- F. Port
- G. Phone System

Answer: ABE

NEW QUESTION 282

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 283

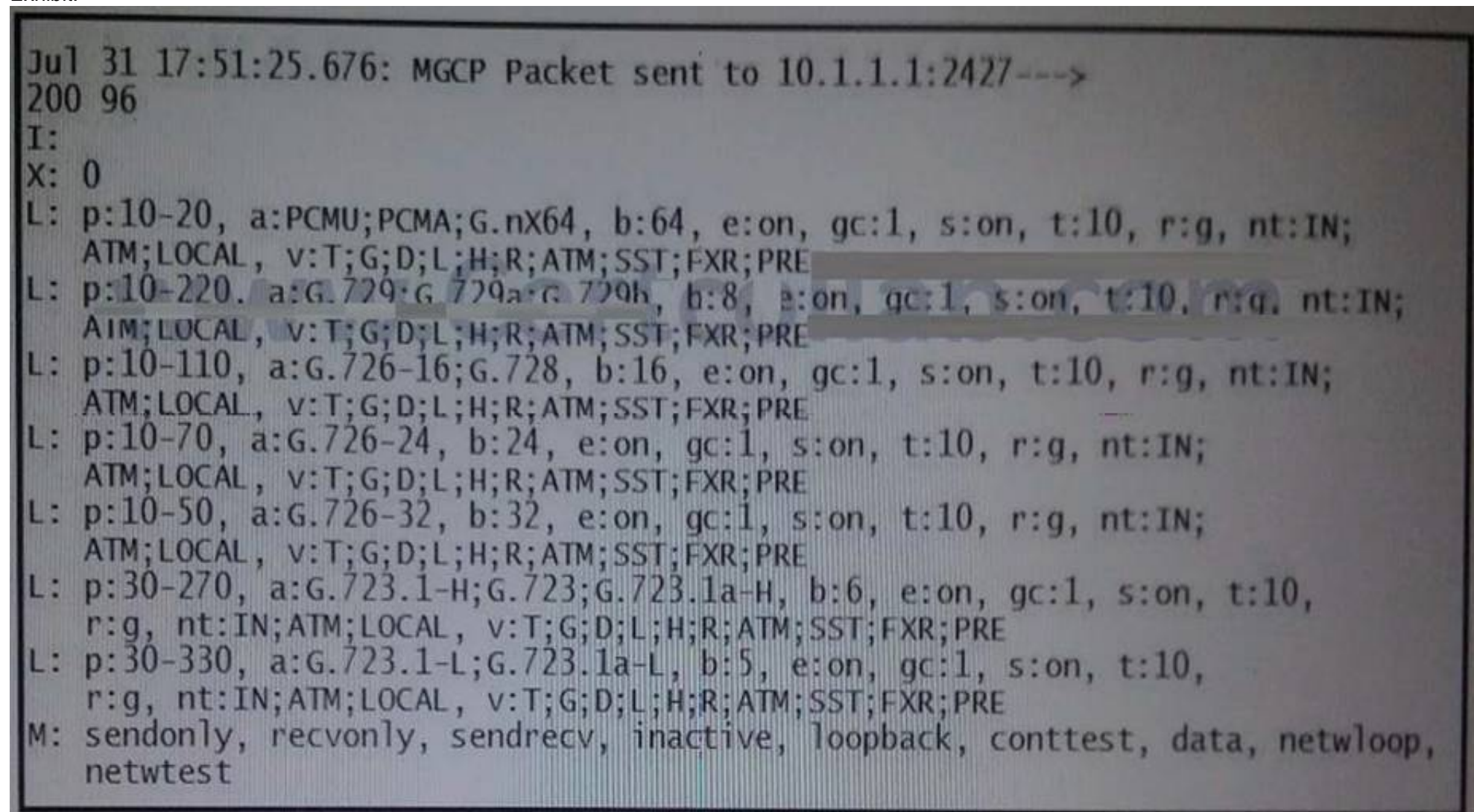
Where can a Cisco Unified CM administrator define Billing Application Server(s) for Call Detail Records?

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

Answer: A

NEW QUESTION 287

Exhibit:



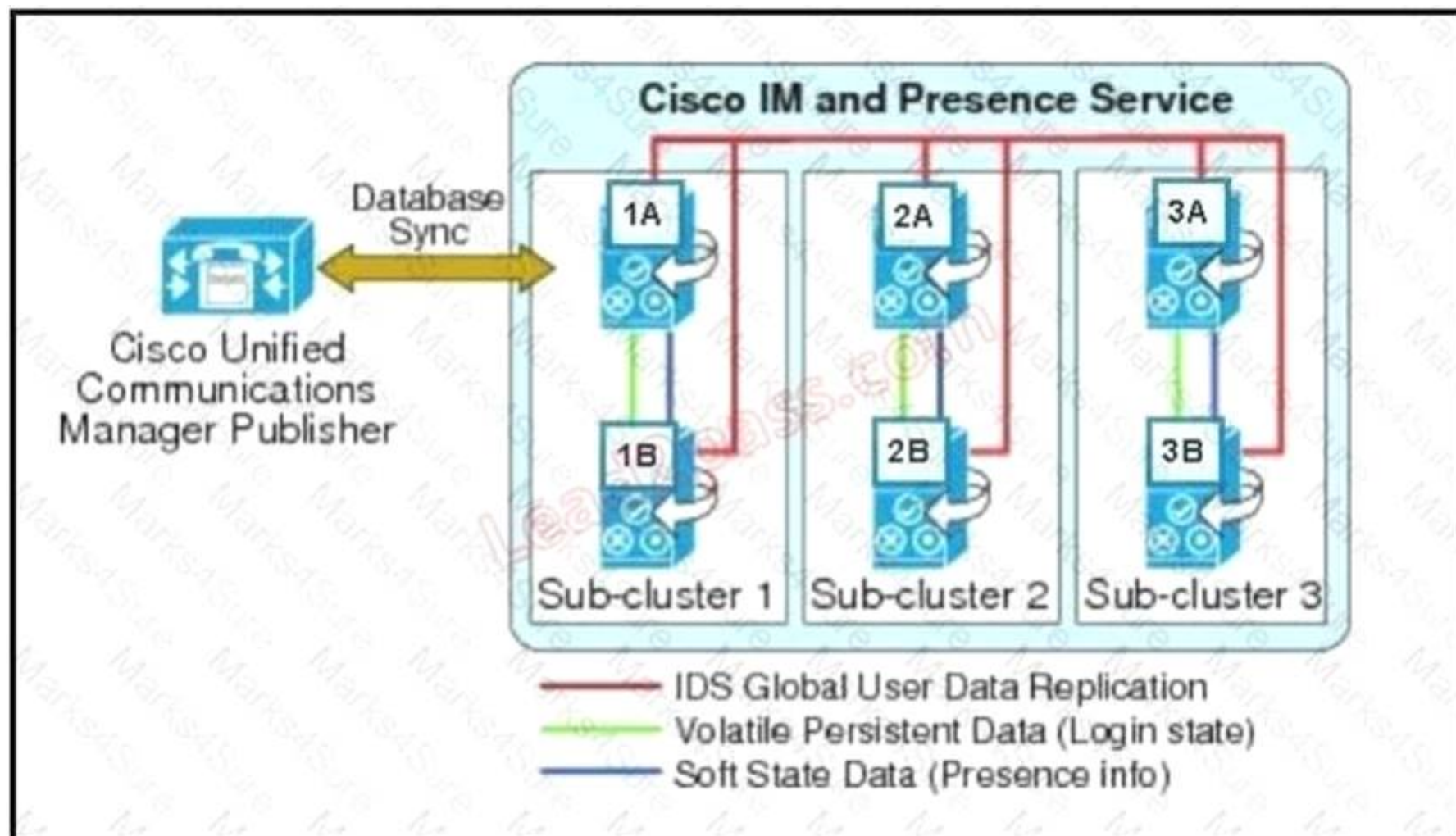
You received this debug output to troubleshoot a Cisco IOS MGCP gateway problem at a customer site. What is the purpose of this message?

- A. The MGCP gateway uses this message to respond to an RQNT message from Cisco Unified Communications Manager.
- B. The MGCP gateway uses this message to respond to an AUCX message from Cisco Unified Communications Manager.
- C. The MGCP gateway uses this message to respond to an AUEP message from Cisco Unified Communications Manager.
- D. The MGCP gateway uses this message to respond to a DLCX message from Cisco Unified Communications Manager.
- E. The MGCP gateway uses this message to respond to an NTFY message from Cisco Unified

Answer: C

NEW QUESTION 288

Refer to the exhibit.



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

- A. baic
- B. hybrid
- C. high availability
- D. mixed

Answer: C

NEW QUESTION 292

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

- A. Forbidden
- B. Unauthorized
- C. Request timeout
- D. Service unavailable
- E. Bad gateway
- F. Internal server error

Answer: DEF

NEW QUESTION 295

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 299

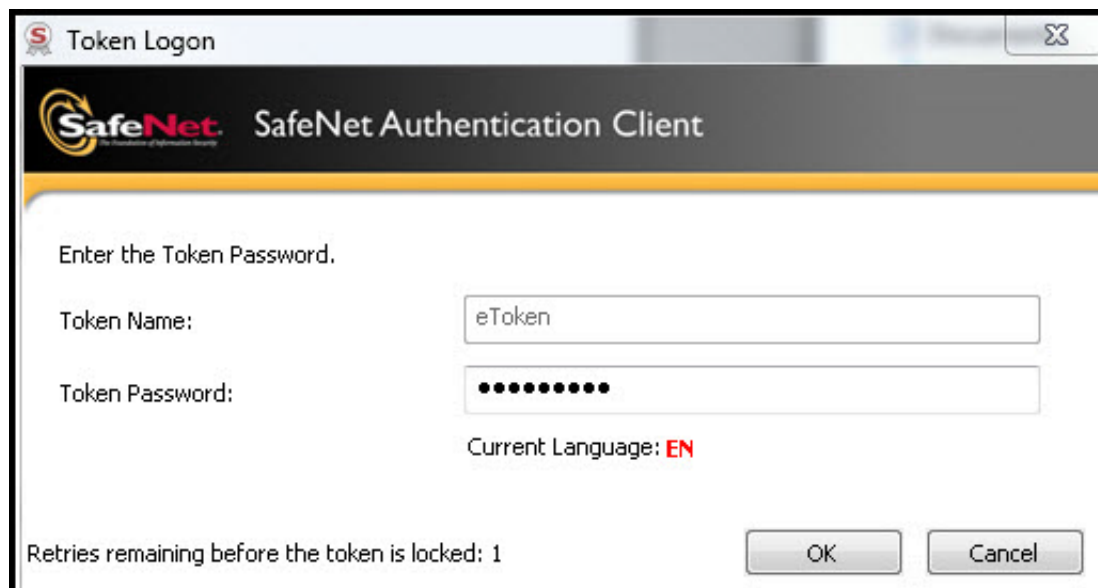
Which two SCCP call states support the CallBack softkey? (Choose two)

- A. On Hook
- B. Remote In Use
- C. Connected Transfer
- D. Ring In
- E. Off Hook
- F. Connected Conference

Answer: AC

NEW QUESTION 303

Refer to the exhibit.



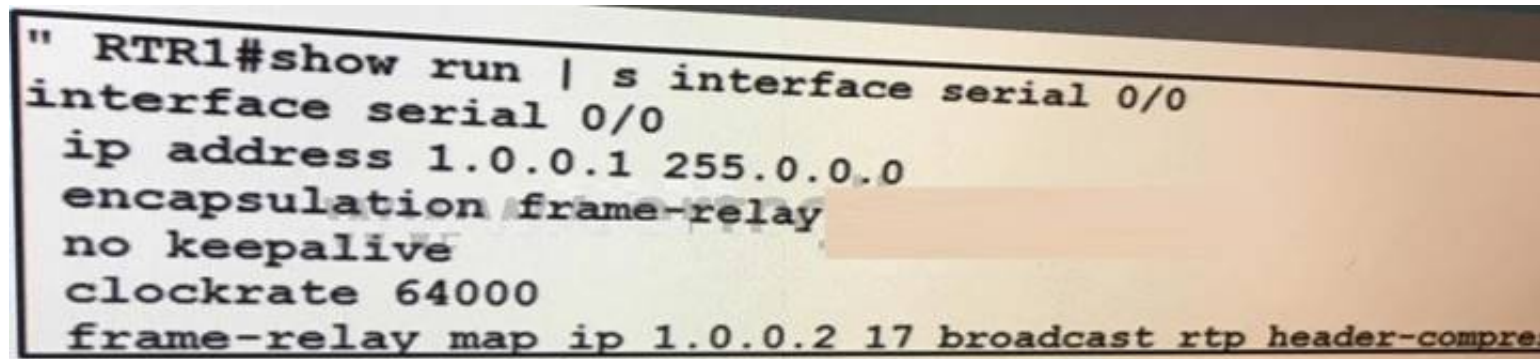
What happens to the USB e-token after the administrator fails to enter the correct password at the next attempt?

- A. The token is locked for five days, after which the retry counter resets.
- B. The token is locked until unlocked by Cisco TAC.
- C. The token is locked until Cisco CTL Client is uninstalled and reinstalled on the client PC.
- D. The token cannot be used on the same client PC again
- E. It can be used with another Cisco CTL Client on a different PC.
- F. The token is locked forever.

Answer: E

NEW QUESTION 304

Refer to the exhibit.



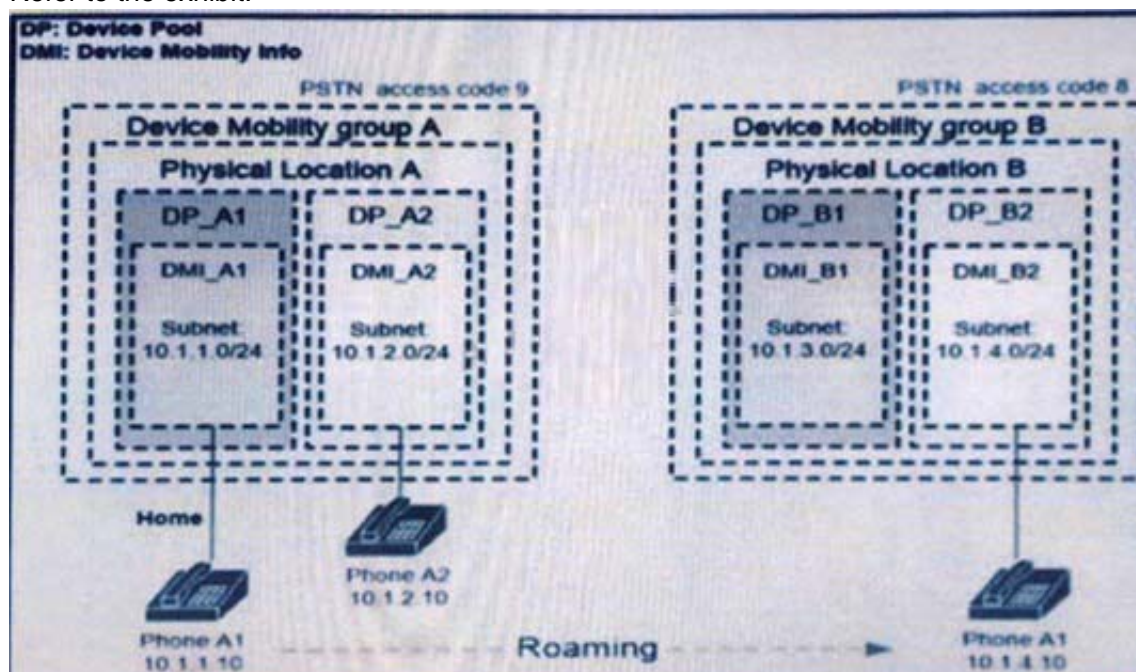
An engineer is upgrading existing frame-relay network to MPLS using Ethernet. The amount of bandwidth from service provider will remain the same. What two issues the engineer must consider when changing from frame-relay to Ethernet for voice connectivity? (Choose two)

- 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 307

Refer to the exhibit.



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

- A. Phone A PSTN calls preserve home location dialing behavior.
- B. Phone A inherits CSS from roaming device pool DP_B2.
- C. Phone A PSTN calls Adopts roaming location dialing behavior.
- D. Phone A retains CSS from Home device pool DP_A1.
- E. Phone A retains media Resource group list from home device pool DP_A1.
- F. Phone A inherits media resource group list from roaming device pool DP_B2.

Answer: BCE

NEW QUESTION 310

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

- A. It provides internetworking to ensure that extended IP Phone traffic is encrypted as long as the Cisco Unified Communications Manager is in secure mode.
- B. It manipulates the call signalling to ensure that all media is routed via the adaptive security appliance.
- C. It requires a remote routing device with an IPsec VPN tunnel.
- D. It intercepts and authenticates soft clients before they reach Cisco Unified Communications Manager clusters.
- E. It supports encrypted TFTP operation of IP phone configuration files.

Answer: BD

NEW QUESTION 312

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 315

Refer to the exhibit.


```

interface GigabitEthernet1/0/15
 description Phone and PC port
 srr-queue bandwidth share 1 10 30 100
 srr-queue bandwidth shape 1 0 0 0
 queue-set 2
 mls qos trust device cisco-phone
 mls qos trust cos

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

```

In a Cisco Unified CM environment with default QoS configuration in the cluster, IP phone users report voice quality issues when they are downloading large files to their PC. Which two configuration changes solve this problem? (Choose two)

- A. The srr-queue bandwidth share command must be changed to increase the weight of queue 1.
- B. The global configuration of threshold 3 of queue 4 must be changed to mls qos srr-queue cos-map queue 4 threshold 3 0 5.
- C. The srr-queue bandwidth shape command must be changed to increase the weight of queue 1.
- D. The srr-queue bandwidth shape command must be removed from the interface configuration.
- E. The priority-queue out command is missing from the interface configuration.

Answer: CE

NEW QUESTION 317

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 320

Refer to the exhibit.

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

- A. The IP phone displays the message 'registration rejected' because it cannot support IPv6
B. The IP phone invokes MTP to translate from ipv4 to IPv6
C. The IP phone ignores the settings and registers with the ipv4 address
D. The IP phone uses SLAAC to acquire an IPv6 address and override the limitation

Answer: A

NEW QUESTION 324

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 325

An Engineer is troubleshooting this situation

- MPLS connection between site A and site B is flapping in intervals of 30 seconds at least twice an hour
- Phones at site B failover to their survivable remote site, Telephony configured gateway.
- Phones fail back to their primary CUCM server as soon as the connection is reestablished.
- Phones failover again to SRST.

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

- A. Keepalive timeout timer
B. Connection Monitor Duration
C. Station Keepalive interval
D. Maximum phone fallback queue depth
E. T302 Timer

Answer: B

NEW QUESTION 327

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

- A. Add "appadmin" application user in Cisco UCM and IM & Presence
B. Add "appadmin" application user in Cisco IM & Presence
C. Add "appaduser" application user in Cisco UCM
D. Add "appadmin" application user in Cisco UCM
E. Add "appaduser" application user in Cisco UCM and IM & Presence

Answer: D

NEW QUESTION 329

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

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

Answer: ABC

NEW QUESTION 331

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 332

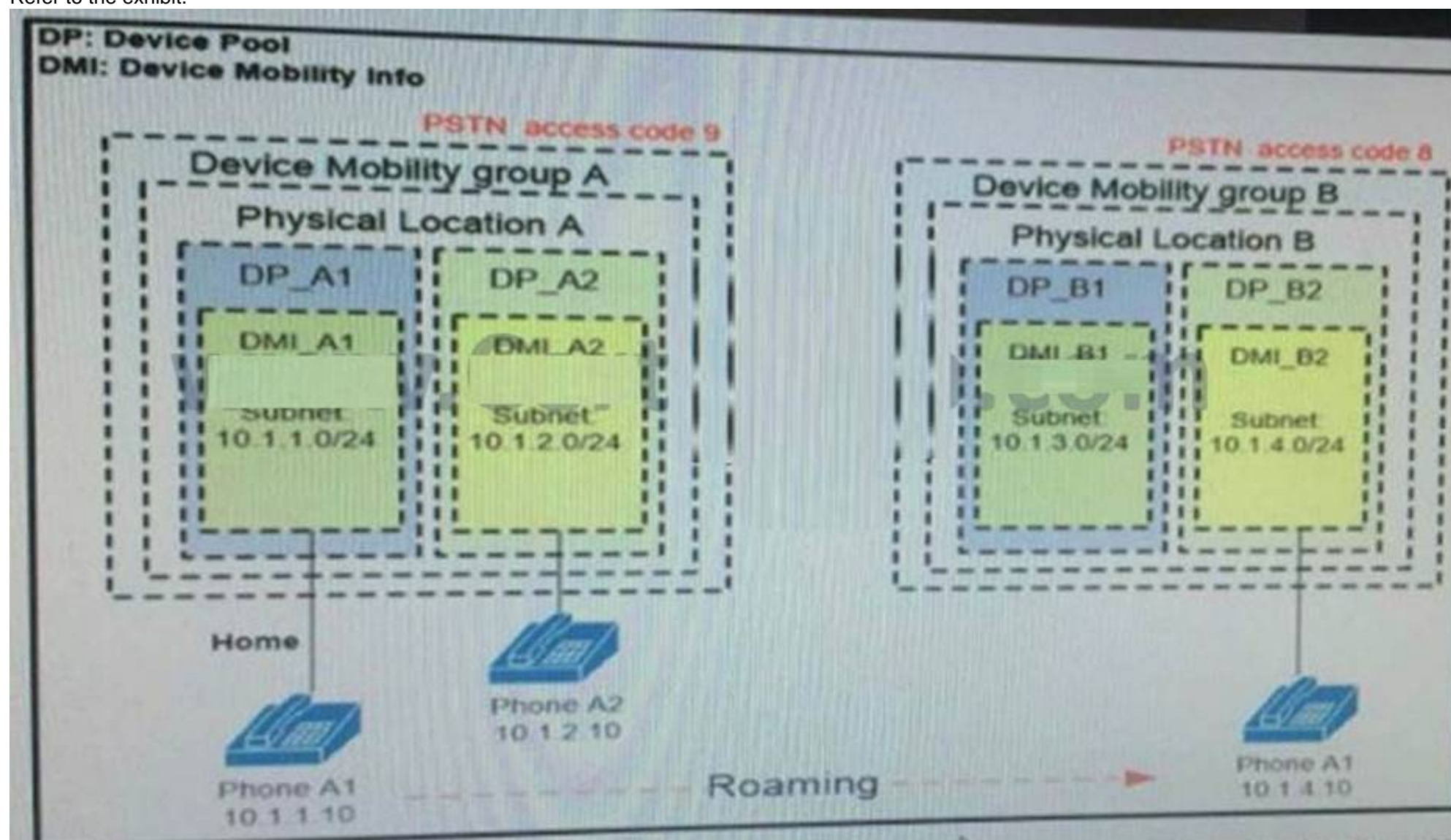
Assume 6 bytes for the Layer 2 header, 1 byte for the end-of-frame flag, and a 40-millisecond voice payload, how much bandwidth should be allocated to the strict priority queue for five VoIP calls that use a G.729 codec over a multilink PPP link?

- A. 87 kb/s
- B. 134 kb/s
- C. 102.6 kb/s
- D. 77.6 kb/s
- E. 71.3 kb

Answer: A

NEW QUESTION 334

Refer to the exhibit.



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

- A. Phone A PSTN calls preserve home location dialling behaviour
- B. Phone A inherits CSS from roaming device pool DP_B2
- C. Phone A PSTN calls Adopts roaming location dialling behaviour
- D. Phone A retains CSS from Home device pool DP_A1

- E. Phone A retains media Resource group list from home device pool DP_A1
- F. Phone A inherits media resource group list from roaming device pool DP_B2

Answer: ADE

NEW QUESTION 337

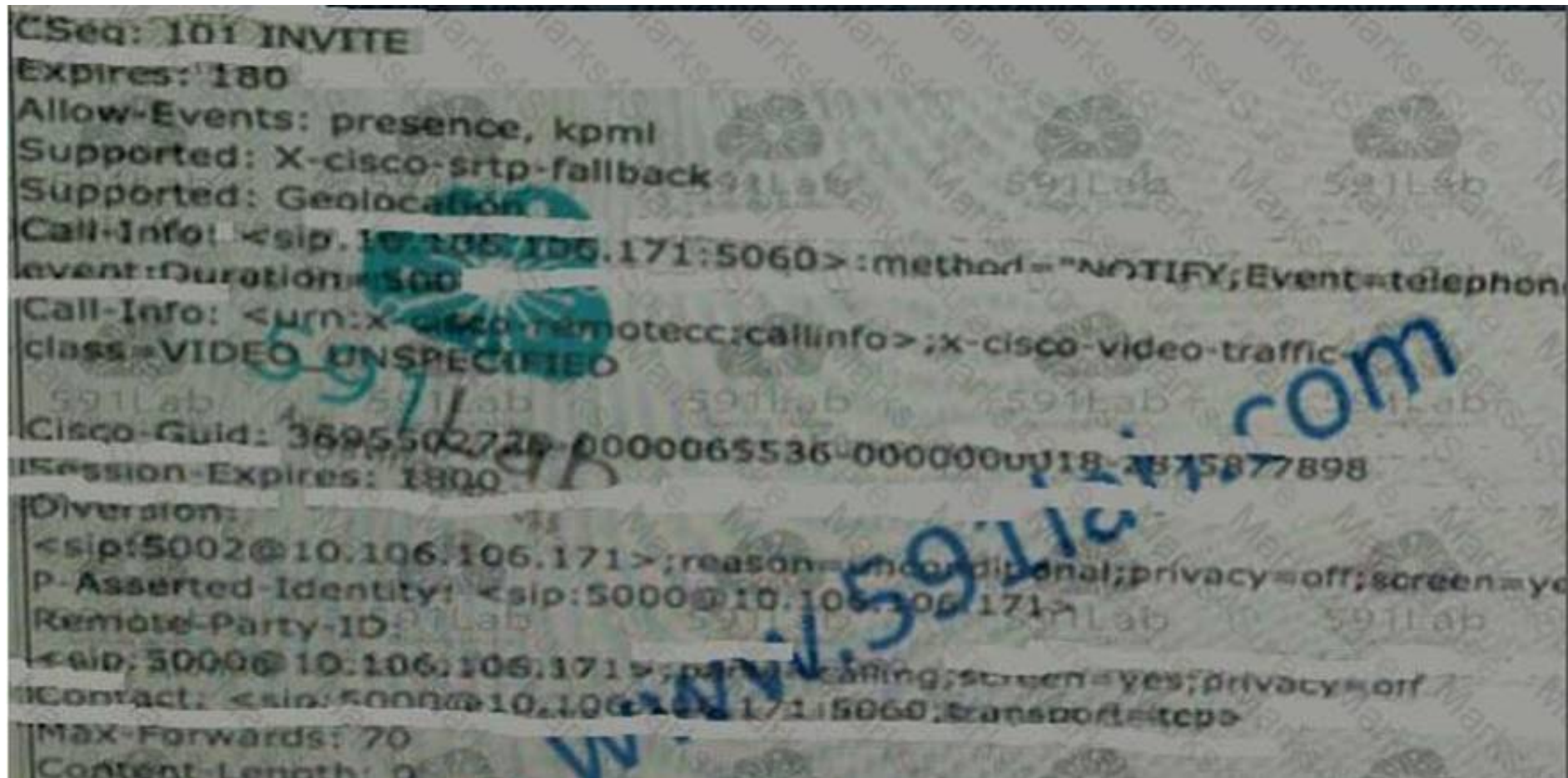
An engineer is planning the voice call bandwidth requirements between two offices. The design requires a capacity of 25 concurrent audio calls using the G.729 codec and IPv6 transport. Which two configurations meet the requirements? (Choose two)

- 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 339

Refer to the exhibit.



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

- A. voice class sip-profiles 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 340

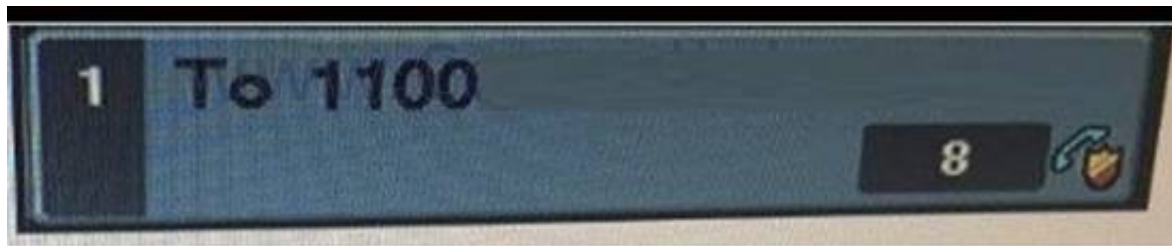
What is a major disadvantage of virtual machines versus containers?

- A. Operational Management
- B. Security
- C. Boot Time
- D. Limited management tools
- E. Vendor lock-in

Answer: C

NEW QUESTION 341

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 346

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 350

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 351

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

- A. Owner user ID is not set on the newly provisioned client.
- B. Incoming calling search space on SIP trunk to IM & P.
- C. Primary DN is not set in end user configuration of the newly provisioned client
- D. Subscriber calling search space is not defined on newly provisioned client.
- E. PC cannot resolve the FQDN of IM & P.
- F. IM & P incoming ACL blocking inbound status.
- G. Subscriber calling search space on SIP trunk to IM & P.

Answer: EFG

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 356

Refer to the exhibit.


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

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

A)

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

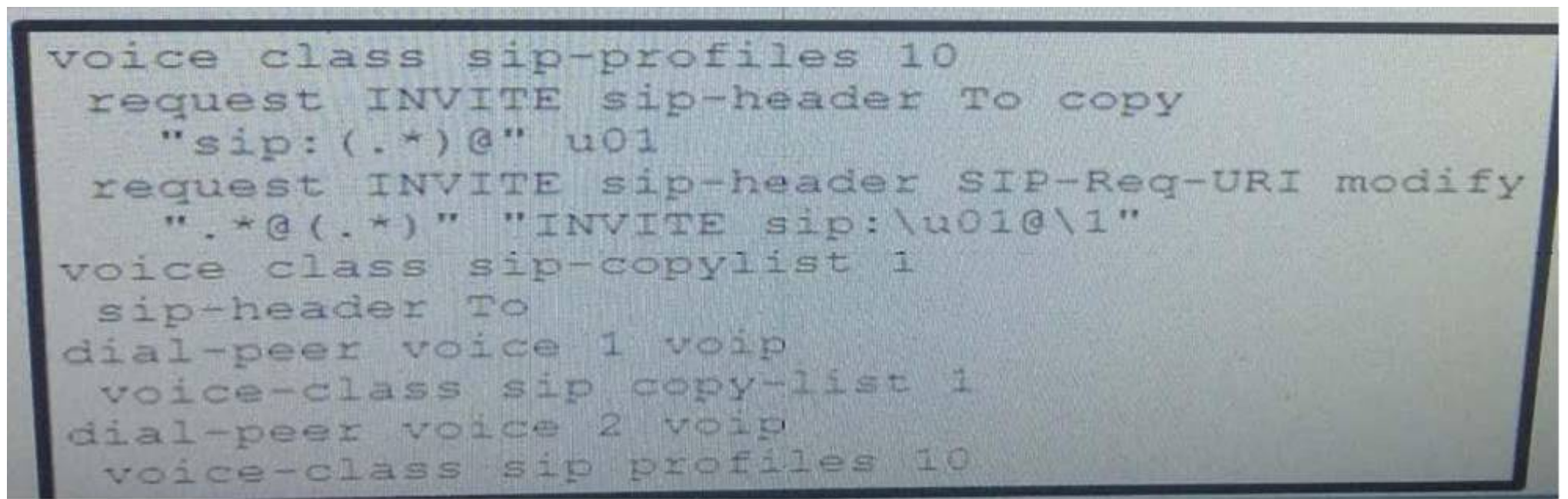
B)

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

C)

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

D)



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

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

Answer: A

NEW QUESTION 357

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 OOBDTMF 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 358

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

- A. SIP IP phones
- B. Media Termination Points
- C. SIP Trunk to Cisco Unity Connection
- D. MGCP gateway with FXO ports
- E. MGCP gateway with FXS ports
- F. MGCP gateway with Q.SIG PRI

Answer: AEF

NEW QUESTION 361

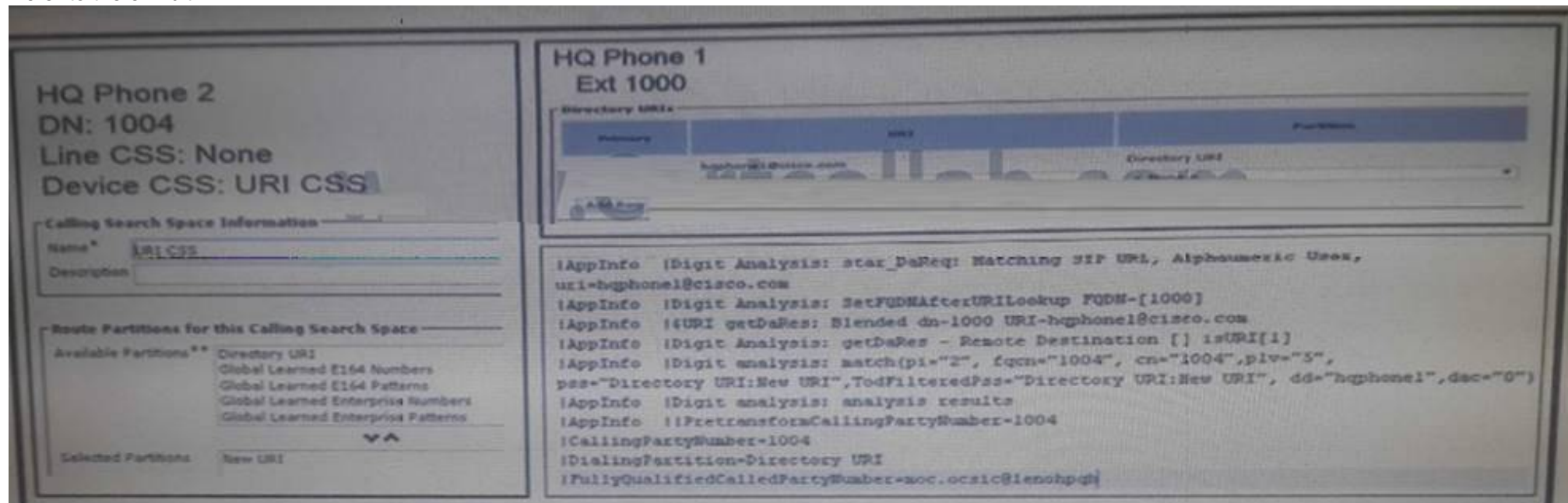
Which route pattern is matched in Cisco Unified Communication Manager Version 11.0 when a user dials 2001?

- A. 200X configured with urgent priority
- B. 20[02-4]1 configured with urgent priority
- C. 200! Configured with urgent priority
- D. 20[*2-4]1 configured with urgent priorit
- E. 20[1-4]1 configured with nonurgent priority

Answer: C

NEW QUESTION 362

Refer to the exhibit.



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

- A. Directory URI Alias partition has been defined in Enterprise parameters
- B. Phone needs to be reset for changes to take effect
- C. Directory URI partition cannot be deleted therefore still will be reachable
- D. CSS Changes failed to be applied after hitting save due to Database replication issue

Answer: A

NEW QUESTION 363

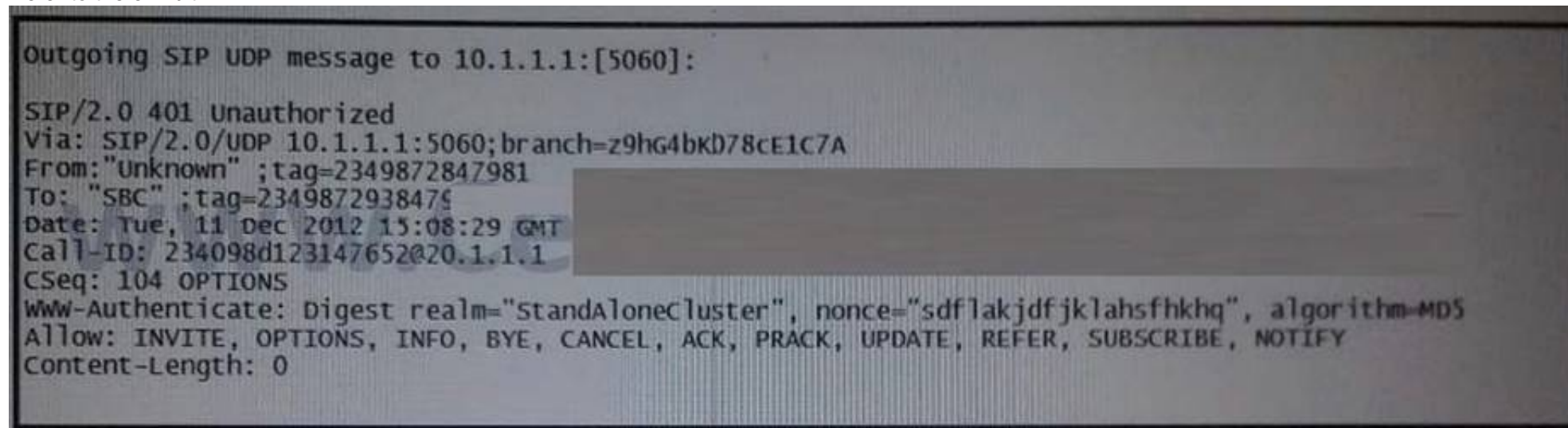
Which two actions does the Cisco Unified IP phone use the initial Trust list to perform? (Choose two.)

- A. Decrypt secure XML files
- B. Encrypt RTP traffic for ip phones that are not registered to the same call manager cluster
- C. Download background image files
- D. Authenticate their configuration file signature
- E. Talk securely to CAPF which is a prerequisite to support configuration files encryption

Answer: DE

NEW QUESTION 368

Refer to the exhibit.



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 372

Refer to the exhibit.


```

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

```

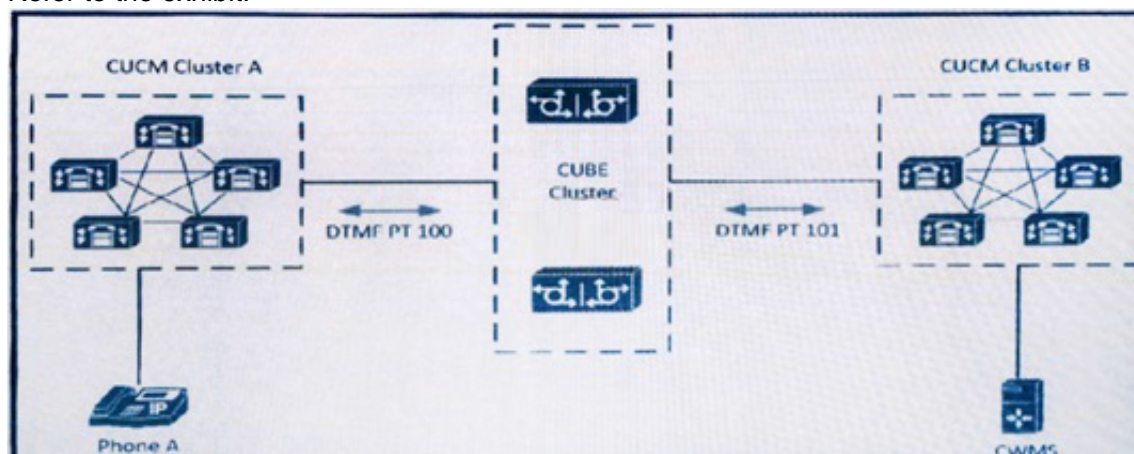
A phone VPN failed to establish a VPN with the Cisco ASA. The support engineer downloaded the console logs and analyzed them. When two steps resolve this issue? (Choose two)

- A. Configure user and password authentication instead of certificate only.
- B. Uncheck the enable Host ID check checkbox under the VPN profile in Cisco Unified CM.
- C. Reset the Cisco Unified CM TFTP service to allow caching of the new certificate.
- D. Delete the current certificate so the phone can download a new one.
- E. Register the phone internally to download the new configuration.

Answer: B

NEW QUESTION 373

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 374

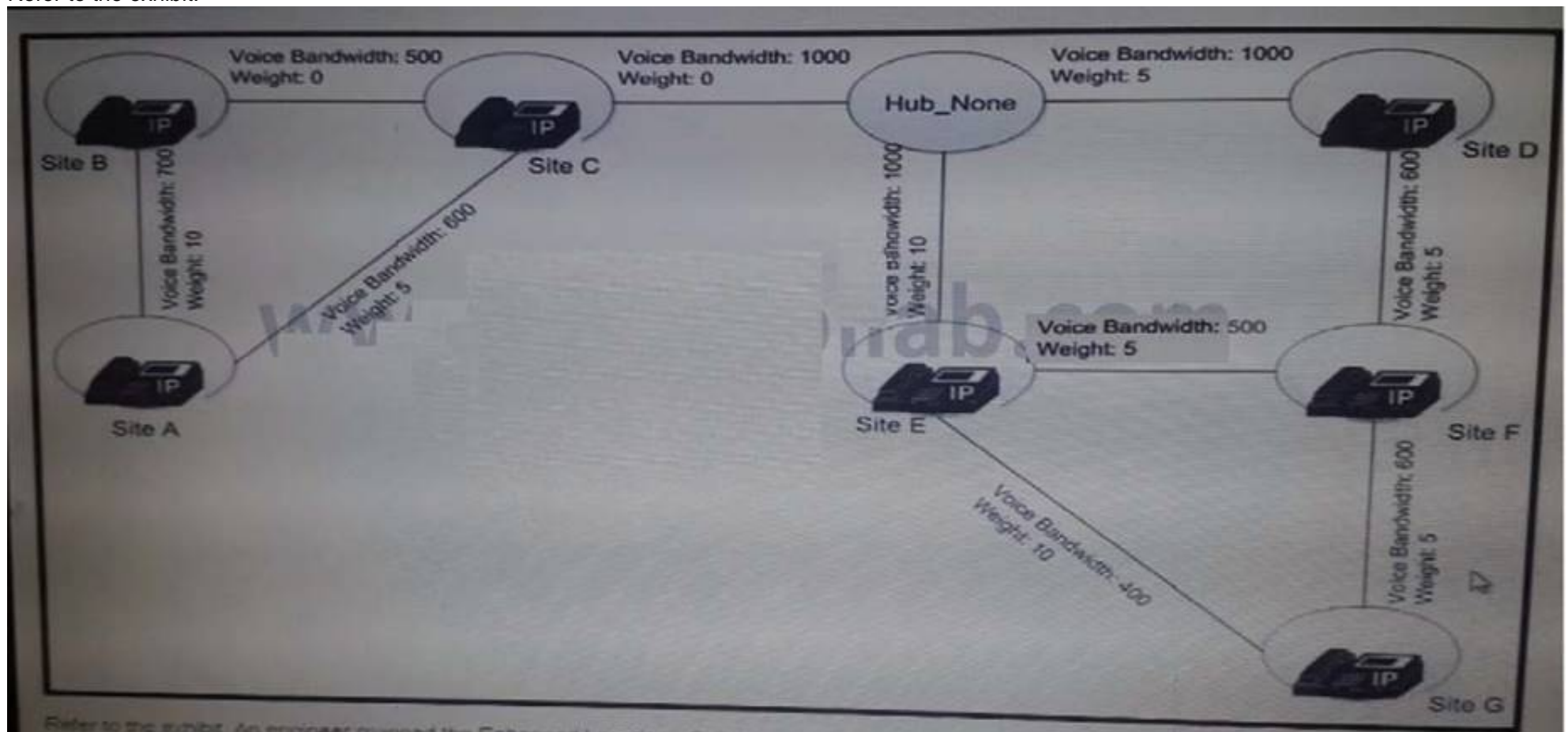
A customer asks you to build a SIP/SIMPLE IM federation between a Cisco Unified Communication Manager IM & Presence environment and a Microsoft Lync 2013 environment. The Lync administrator insists that TLS is used for the federation and that the SSL certificate used by Cisco Unified Communication Manager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

- A. cupxmpp
- B. tomcat
- C. cupxmpps2s
- D. ipsec
- E. cup

Answer: C

NEW QUESTION 379

Refer to the exhibit.



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

- A. 5
- B. 7
- C. 12
- D. 25
- E. 37

Answer: B

NEW QUESTION 383

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 385

Which two fields can be used to uniquely identify the same call in the Call Detail Records and the Call Management Records? (Choose two)

- A. nodeId
- B. globalCallId_callId
- C. callIdentifier
- D. pkid
- E. globalCallId_ClusterId
- F. globalCallId_callManagerId

G. deviceName

Answer: BF

NEW QUESTION 388

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 390

Refer to the exhibit.

```
ephone-dn 3 octo-line
number 1645
label 1645
description John Doe
name John Doe
mobility

!
ephone-template 1
softkeys idle Redial Newcall Mobility 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
```

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. 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 394

ACisco Unity Connection administrator receives a name change request from a voice-mail user, whose Cisco Unity Connection user account was imported from Cisco Unified Communications Manager. What should the administrator do to execute this change?

- A. Change the user data in the Cisco Unity Connection administration page, then use the Synch User page in Cisco Unity Connection administration to push the change to Cisco Unified Communications Manager.
- B. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unity Connection administration to pull the changes from Cisco Unified CM.
- C. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unified CM administration to push the change to Cisco Unity Connection.
- D. Change the user profile from Imported to Local on Cisco Unity Connection Administration, then edit the data locally on Cisco Unity Connection.
- E. Change the user data in Cisco Unity Connection and Cisco Unified Communications Manager separately

Answer: B

NEW QUESTION 398

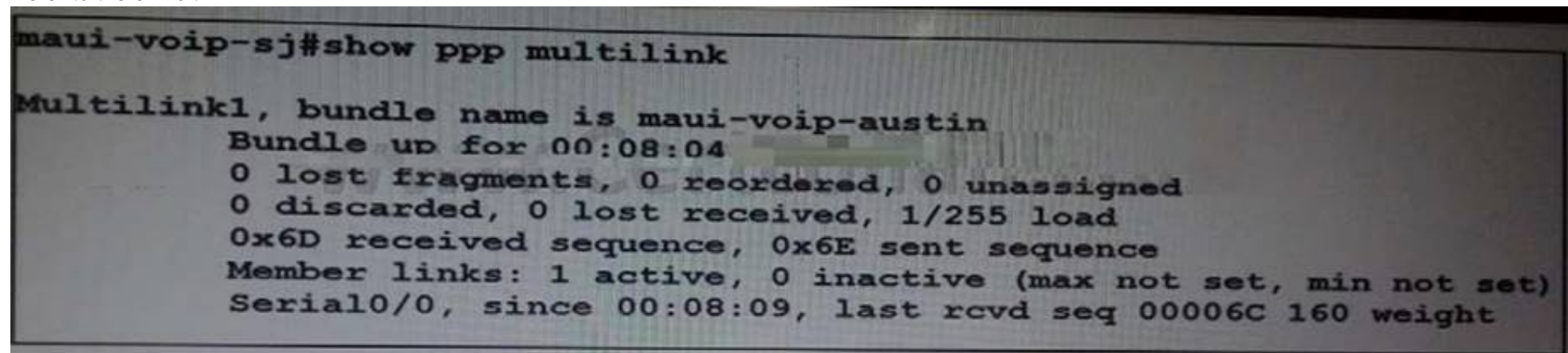
Which three requirements must be met to share Enhanced Location Based Call Admission Control bandwidth usage between clusters? (Choose three.)?

- 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 400

Refer to the exhibit.



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

- A. bandwidth 128
- B. ppp multilink fragment-delay 40
- C. ppp multilink fragment-delay 15
- D. bandwidth 384
- E. ppp multilink fragment-delay 10
- F. bandwidth 512
- G. ppp multilink fragment-delay 20
- H. bandwidth 768

Answer: AE

NEW QUESTION 401

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 402

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 406

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 PostgreSQL is required.
- F. A server running SAP SQL server is required.
- G. A server running Microsoft SQL server is required.
- H. The server must have standard licensing.

Answer: BCDG

NEW QUESTION 411

ACisco Unified Communicator Manager Administrator is working on devices that are roaming within the company using device mobility. Which two configuration settings have priority over the device setting when using the roaming device pool? (choose two)

- A. Physical Location
- B. Device Mobility Calling Search Space
- C. Calling Party Transformation CSS
- D. Adjunct CSS
- E. Media Resource Group List
- F. Region

Answer: EF

NEW QUESTION 413

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 414

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?

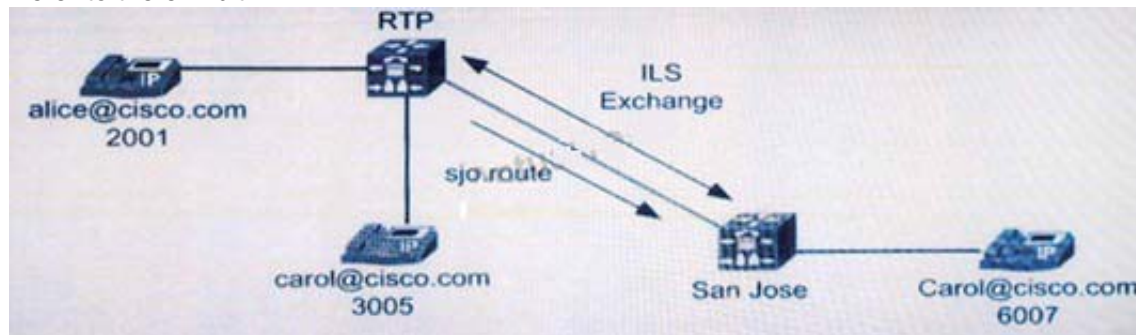
- A. Set CDR Enabled Flag to True.
- B. Set CDR Log Calls with Zero Duration Flag to True.

- 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 415

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 418

Refer to the exhibit.



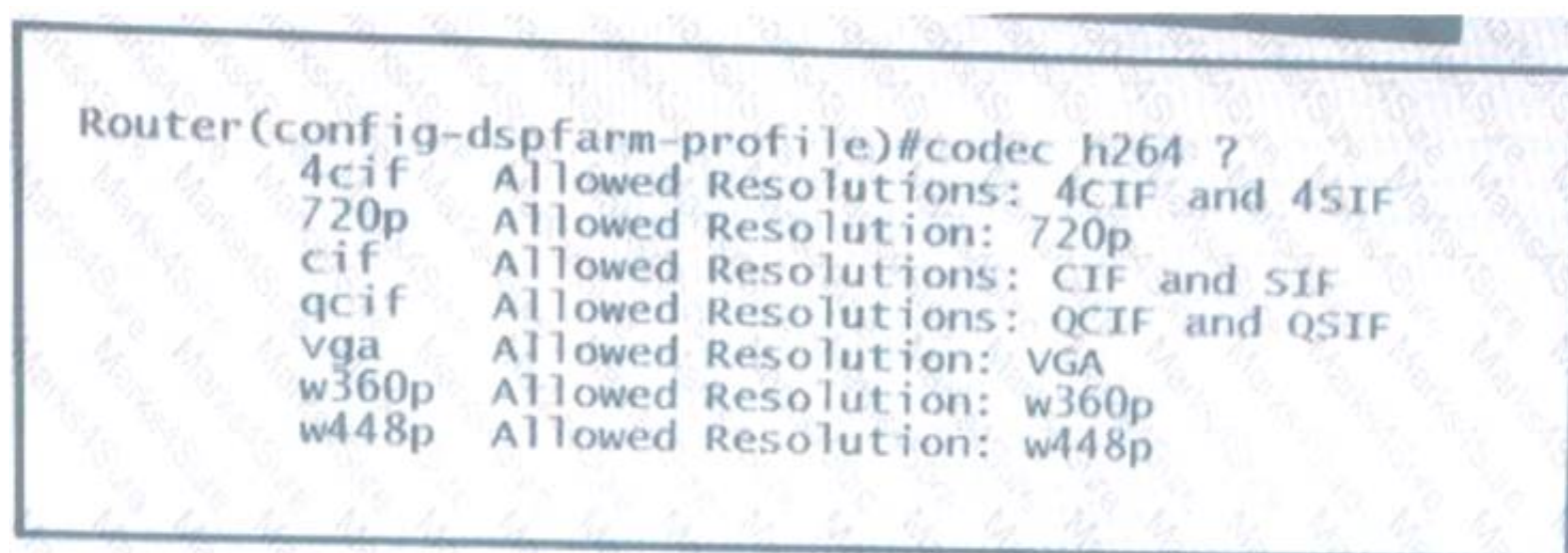
A reporting specialist found that calls answered by the phones are not being recorded. Which two configuration changes can resolve this issue? (Choose two)

- 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 420

Exhibit:



Which Cisco IOS multipoint video conferencing profile is being configured?

- A. Homogeneous
- B. guaranteed-audio
- C. rendezvous
- D. heterogeneous
- E. guaranteed-video

Answer: D

Explanation: Heterogeneous, which means that multiple codecs and multiple bit rates are supported

NEW QUESTION 424

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

- A. Bearer Information
- B. Call ID
- C. Capabilities
- D. Connection ID
- E. Connection Mode
- F. Connection Parameters
- G. Request Identifier
- H. Observed Events

Answer: CDG

NEW QUESTION 428

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

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

Answer: BC

NEW QUESTION 429

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 430

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 432

On which web administration page can you verify database replication health in a two-server Cisco Unified CM cluster running version 9.1?

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

Answer: D

NEW QUESTION 435

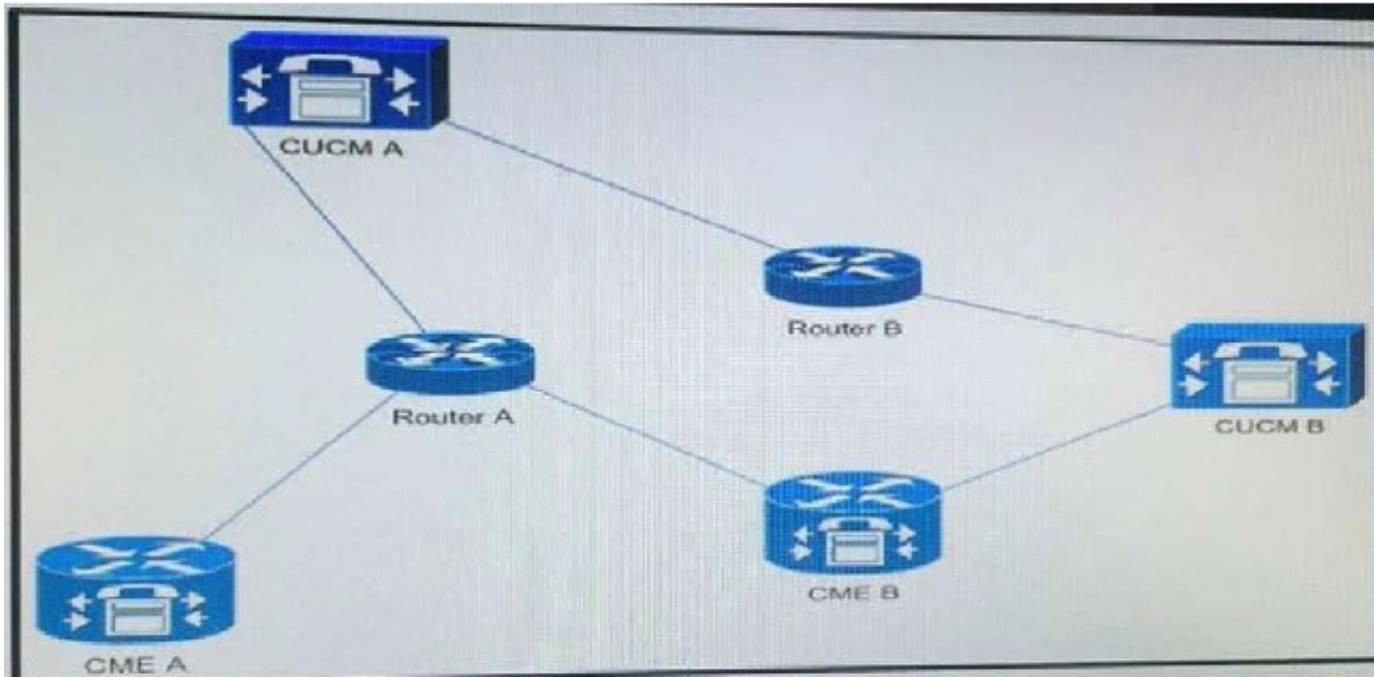
Which directory path on Cisco Unified CM publisher is used to temporarily store the Call Detail Records collected from other nodes until they are processed by the CDR Repository Manager?

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

Answer: B

NEW QUESTION 437

Refer to the exhibit.



A collaboration engineer is configuring dynamic call routing and DN learning between two Cisco UCM and Two Cisco UCM express. What two configuration tasks will support this? (Choose two)

- A. CME B should be configure as service advertisement forwarder only
- B. CME B should be configure as service advertisement client and forwarder
- C. Router A and CME B should be configure to use the same autonomous system number
- D. Router A and CME B should be configure to use the same autonomous system number
- E. Router B and CME B should be configure to use the same autonomous system number
- F. CME B should be configured as service advertisement client only.

Answer: BC

NEW QUESTION 438

.....

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