350-801 Dumps

Implementing and Operating Cisco Collaboration Core Technologies

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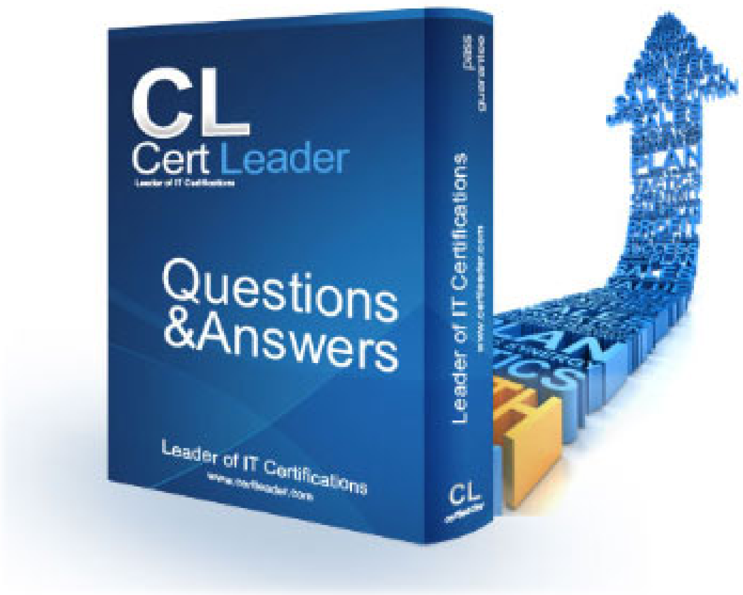
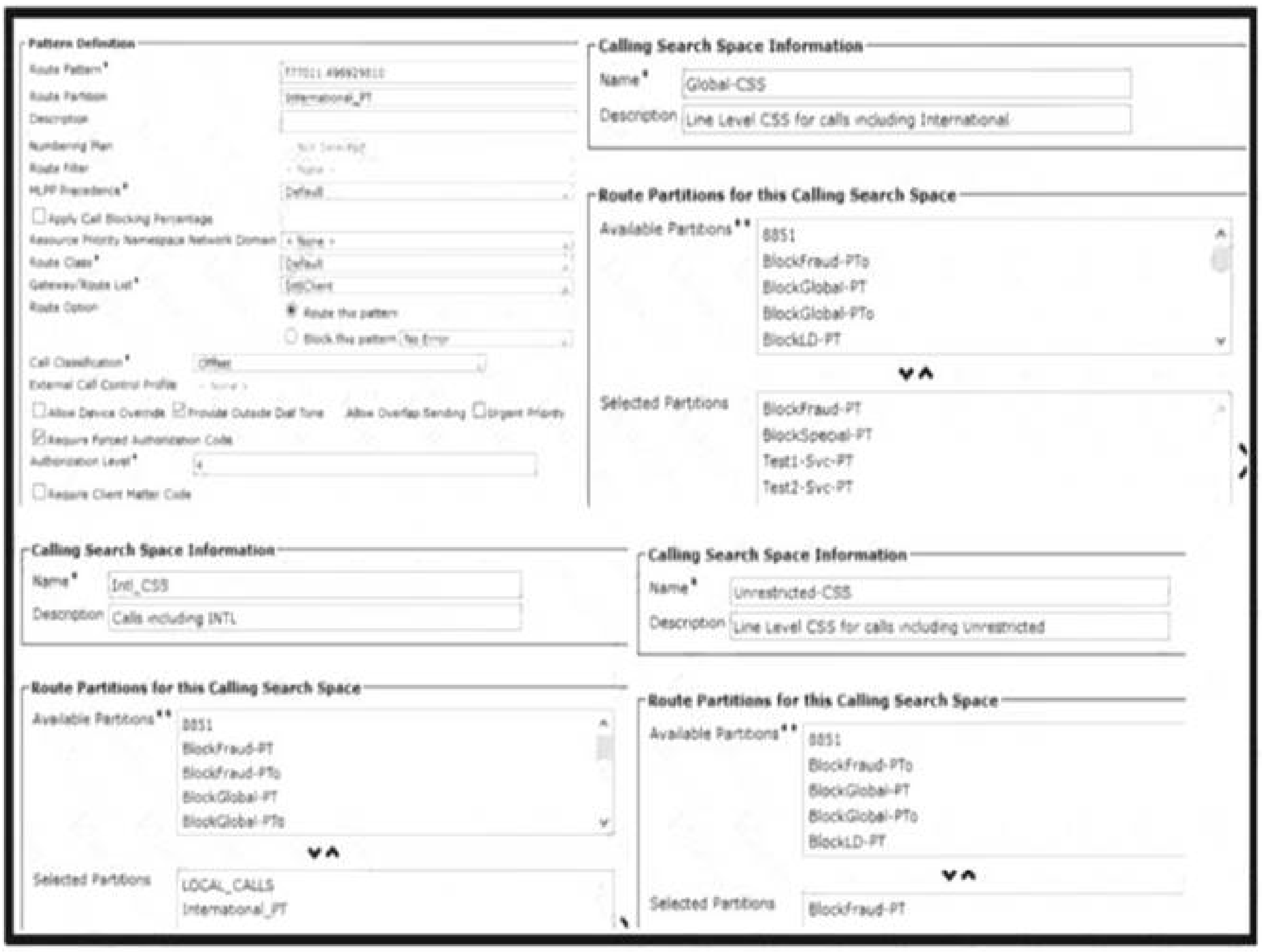


NEW QUESTION 1
What happens when a Cisco IP phone loses connectivity to the duster during an active call?

A. The call continues to be active, but features like transfer or hold do not work.
B. The call continues and all features work.
C. The call drops immediately.
D. The call drops after missing two keepalives from Cisco UCM.

Answer: D

NEW QUESTION 2
Refer to the exhibit.



How must the +E.164 translation pattern be configured to reach international number 496929810?

Pattern= \+.496929810, CSS=Unrestricted-CSS, PreDot, Prefix=777011

Pattern= \+.777011496929810, CSS=Intl_CSS

Pattern= \+.011496929810, CSS=Global-CSS, PreDot, Prefix=777

Pattern= \+.496929810, CSS=Intl_CSS, PreDot, Prefix=777011

A. Option A
B. Option B
C. Option C
D. Option D

Answer: C

NEW QUESTION 3
How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

A. Change the service parameter Apply Transformations On Remote Number to True.
B. Configure a translation pattern that has a Calling Party Transform Mask of XXXXXXXXXX.
C. On the inbound SIP trunk, change Significant Digits to 10.
D. Configure a calling party transformation pattern that keeps only the last 10 digits.

Answer: A

NEW QUESTION 4
What is an advantage of using Cisco Webex Control HuB?

A. enables the provisioning, administration, and management of Webex services and Webex Hybrid Services

B. brings Video, audio, and web communication together to meet the collaboration needs of the modern workplace
C. provides streamlined communication and collaboration for a hybrid workforce
D. offers easy contact management, centralized administration, and centralized configuration management

Answer: A

Explanation:
Cisco Webex Control Hub is a cloud-based management platform that enables you to provision, administer, and manage Webex services and Webex Hybrid Services. It provides a single pane of glass for managing all of your Webex services, including Webex Meetings, Webex Teams, and Webex Calling.
Webex Control Hub offers a number of features and benefits, including:
- A single pane of glass for managing all of your Webex services
- Centralized user management
- Simplified provisioning and administration
- Real-time analytics and reporting
- Enhanced security and compliance

Webex Control Hub is a powerful tool that can help you manage your Webex services more effectively. It is easy to use and provides a number of features and benefits that can help you improve your productivity and efficiency.

NEW QUESTION 5
Refer to the exhibit.

```
Sent:
INVITE sip:2004@192.168.100.100:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.100.200:5060;branch=z9hG4bKF1FED
From: "7000" <sip:7000@192.168.100.200>;tag=43CDE-1A22
To: <sip:2004@192.168.100.100>
Call-ID: 26BCA00-4C4E11EA-80169514-B1C46126@192.168.100.200
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE:  1800
User-Agent: Cisco-SIPGateway/IOS-16.9.5
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Contact: <sip:7000@192.168.100.200:5060>
Expires: 180
Max-Forwards: 68
P-Asserted-Identity: "7000" <sip:7000@192.168.100.200>
Session-Expires:  1800
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsSIP-GW-UserAgent 5871 9974 IN IP4 192.168.100.200
s=SIP Call
c=IN IP4 192.168.100.200
t=0 0
m=audio 8002 RTP/SAVP 0
c=IN IP4 192.168.100.200
a=rtpmap:0 PCMU/8000
a=ptime:20
```

Calls to Cisco Unity Connection are failing across Cisco Unified Border Element when callers try to select a menu prompt Why is this happening and how is it fixed?

A. Cisco Unity Connection is configured on G.729 onl
B. Cisco Unity Connection must be reconfigured to support iLBC.
C. Cisco Unified Border Element is not sending any support for DTM
D. OTMF configuration must be added to the appropriate dial peer.
E. Cisco Unified Border Element is sending the incorrect media IP addres
F. The IP address of the loopback interface must be advertised in the SDP
G. The Cisco Unity Connection Call Handler is configured for a "Release to Switch" transfer type Transfers must be disabled for the Cisco Unity Connection Call Handler

Answer: B

NEW QUESTION 6
Which two protocols can be configured for the Cisco Unity Connection and Cisco UCM integration? (Choose two.)

A. 323
B. SIP
C. SCCP
D. MGCP
E. RTP

Answer: BC

Explanation:
The two protocols that can be configured for the Cisco Unity Connection and Cisco UCM integration are SIP and SCCP. SIP, or Session Initiation Protocol, is a signaling protocol used for initiating, maintaining, and terminating real-time sessions, including voice, video, and messaging applications.
SCCP, or Skinny Client Control Protocol, is a Cisco proprietary signaling protocol used for controlling Cisco IP phones.
H.323 is an older signaling protocol that is no longer widely used. MGCP, or Media Gateway Control Protocol, is a protocol used for controlling media gateways.
RTP, or Real-time Transport Protocol, is a protocol used for transporting real-time data, such as voice and video

NEW QUESTION 7
Which option must be used when configuring the Local Gateway for a Cisco Webex Calling trunk?

A. local authentication
B. certificate-based
C. mutual TLS
D. Auth-based

Answer: B

Explanation:
A certificate-based trunk is a type of trunk that uses certificates to authenticate the connection between Webex Calling and the Local Gateway1. A Local Gateway is a supported session border controller that terminates the trunk on the premises2. A certificate-based trunk requires a certificate authority (CA) to issue and manage the certificates for both Webex Calling and the Local Gateway1.

NEW QUESTION 8
Refer to the exhibit. Which two codec permutations should be transcoded by this dspfarm? (Choose two.)

A. iLBC to G.711ulaw
B. G.728br8 to G.711alaw
C. G.729r8 to G.711ulaw
D. G.722 to G.729r8
E. G.729ar8 to G.711alaw

Answer: CE

NEW QUESTION 9
Which configuration concept allows for high-availability on IM and Presence services in a UC environment?

A. IM and Presence subclusters (configured on Cisco UCM)
B. Presence Redundancy Groups (configured on Cisco Unified IM and Presence)
C. IM and Presence subclusters (configured on Cisco Unified IM and Presence)
D. Presence Redundancy Groups (configured on Cisco UCM)

Answer: D

NEW QUESTION 10
If a phone needs to register with cucm1.cisco.com, which network service assists with the phone registration process?

A. SNMP
B. ICMP
C. SMTP
D. DNS

Answer: D

Explanation:
According to the Cisco Community website1, the phone uses DNS to resolve the hostname of the CUCM server (cucm1.cisco.com) to its IP address. DNS is a network service that translates domain names into IP addresses.

NEW QUESTION 10
Which SNMP service must be activated manually on the Cisco Unified Communications Manager after installation?

A. Cisco CallManager SNMP
B. SNMP Master Agent
C. Connection SNMP Agent
D. Host Resources Agent

Answer: A

NEW QUESTION 11
An administrator must configure the Local Route Group feature on Cisco UCM. Which step will enable this feature?

A. For each route group, check the box for the Local Route Group feature.
B. For each route pattern, select the Local Route Group as the destination.
C. For each device pool, configure a route group to use as a Local Route Group for that device pool
D. For each route list, configure a route group to use as a Local Route Group.

Answer: C

Explanation:
The Local Route Group feature allows you to use a route group as the destination for calls that are placed from a device pool. The route group that you use as the destination for calls from a device pool is called the Local Route Group for that device pool.
To configure the Local Route Group feature, you must first create a route group. You can then configure the Local Route Group feature for a device pool by selecting the route group that you want to use as the Local Route Group for that device pool.

NEW QUESTION 12
An engineer must configure codec on a Cisco Unified Border Element to prefer the G.711 ulaw and use G.711 codec as the next The engineer logs In to the CUBE, enters the dial.peer configuration level, and runs the voice-class codec 100 command. Which set of commands completes the configuration?

A. voice class codec 100 codec g7llalaw preference 1 codec a7Hulaw preference 2
B. voice class codec 11j codec <?7iiulaw preferred codec g7iialaw
C. vice class codec 100 codec preference 1 g7llulaw codec preference 2 o711alaw
D. voice class codec ::: codec g711ulaw g711alaw

Answer: C

Explanation:
The following commands are used to configure the codec on a Cisco Unified Border Element to prefer the G.711 ulaw and use G.711 alaw as the next codec:
Code snippet
voice class codec 100
codec preference 1 g711ulaw codec preference 2 g711alaw
The voice class codec 100 command creates a new voice class with the ID of 100. The codec preference 1 g711ulaw command sets the preference for the G.711 ulaw codec to 1. The codec preference 2 g711alaw command sets the preference for the G.711 alaw codec to 2.

NEW QUESTION 17
Which type of message must an administrator configure in the SIP Trunk Security Profile for a Message Waiting Indicator light to work with a SIP integration between Cisco UCM and Cisco Unity Connection?

A. Unsolicited NOTIFY
B. 200 ok
C. SIP Register
D. TCP port 5060

Answer: A

NEW QUESTION 18
Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

A. SSO
B. H.323 registration proxy to Cisco Unified Communications Manager
C. MGCP gateway registration
D. SRST

Answer: A

NEW QUESTION 19
Which type of input is required when configuring a third-party SIP phone?

A. digest user
B. manufacturer
C. serial number350-801 2023-4
D. authorization code

Answer: A

NEW QUESTION 24
On which protocol and port combination does Cisco Prime Collaboration receive notifications (Traps and InformRequests) from several network devices in the Collaboration infrastructure for which it has requested notifications?

A. UDP161
B. TCP 161
C. UDP 162
D. TCP 80

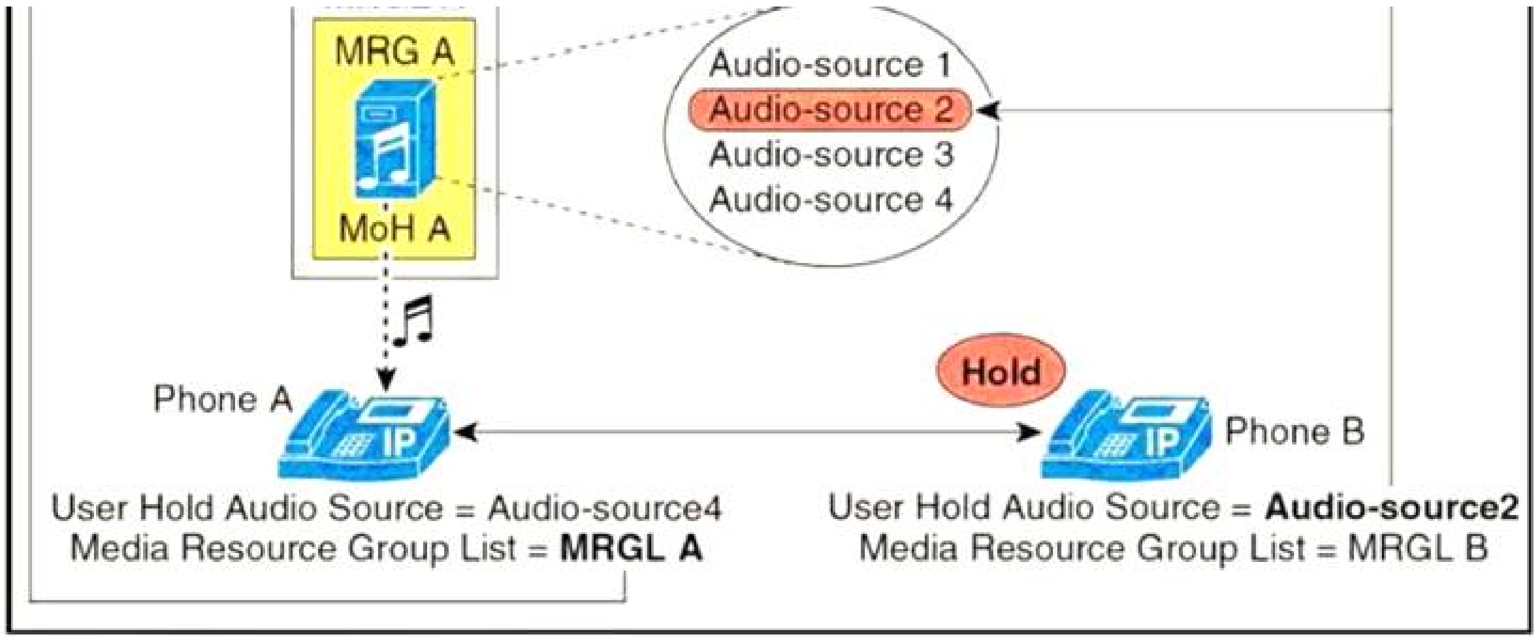
Answer: C

NEW QUESTION 26
What are two features of Cisco Expressway that the customer gets if Expressway-C and Expressway-E are deployed?(Choose two.)

A. highly secure freal-traversal technology to extend organizational reach.
B. additional visibility of the edge traffic in an organization.
C. complete endpoint registration and monitoring capabilities for devices that are local and remote.
D. session-based access to comprehensive collaboration for remote workers, without the need for a separate VPN client.
E. utilization and adoption metrics of all remotely connected devices.

Answer: AD

NEW QUESTION 27
Refer to the exhibit



There is a call flow between Phone A and Phone B Phone B (holder) places Phone A (holder) on hold Which MRGL and Audio Source are played to Phone A?

A. MRGL A and Audio Source 4
B. MRGL B and Audio Source 4
C. MRGL A and Audio Source 2
D. MRGL B and Audio Source 2

Answer: C

NEW QUESTION 31
An engineer with ID012345678 must build an international dial plan in Cisco UCM. Which action is taken when building a variable-length route pattern?

A. configure single route pattern for international calls
B. set up all international route patterns to 0.!
C. reduce the T302 timer to less than 4 seconds
D. create a second route pattern followed by the # wildcard

Answer: D

Explanation:
When building a variable-length route pattern, you need to create a second route pattern followed by the # wildcard. This will allow the user to indicate the end of the number by dialing #. For example, if you want to create a route pattern for international calls, you would create a route pattern like this: 9.011!#
This route pattern will match any number that starts with 9.011, followed by any number of digits, and then ends with #.
The other options are incorrect because:
- Configuring a single route pattern for international calls will not allow the user to indicate the end of the number.
- Reducing the T302 timer to less than 4 seconds will not allow the user to indicate the end of the number.

NEW QUESTION 36
Endpoint A is attempting to call endpoint B. Endpoint A only supports G.711ulaw with a packetization rate of 20 ms, and endpoint B supports packetization rate of 30 ms for G.711ulaw. Which two media resources are allocated to normalize packetization rates through transrating? (Choose two.)

A. software MTP on Cisco IOS Software
B. software MTP on Cisco UCM
C. software transcoder on Cisco UCM
D. hardware transcoder on Cisco IOS Software
E. hardware MTP on Cisco IOS Software

Answer: BE

NEW QUESTION 37
To provide high-quality voice and take advantage of the full voice feature set, which two access layer switches provide support?

A. Use multiple egress queues to provide priority queuing of RTP voice packet streams and the ability to classify or reclassify traffic and establish a network trust boundary.
B. Use 802, 1Q trunking and 802.1p for proper treatment of Layer 2 CoS packet marking on ports with phones connected.
C. Implement IP RTP header compression on the serial Interface to reduce the bandwidth required per voice call on point-to-point links.
D. Deploy RSVP to improve VoIP QoS only where it can have a positive impact on quality and functionality where there is limited bandwidth and frequent network congestion.
E. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.

Answer: AB

NEW QUESTION 40
What is the purpose of Mobile and Remote Access (MRA) in the Cisco UCM architecture?

A. MRA is used to access Webex cloud services only if authenticated with on-premises LDAP service.
B. MRA is used to make secure PSTN calls by Cisco UCM only while on-premises authentication.
C. MRA is used to make B2B calls through Expressway registration.
D. MRA is used to access the collaboration services offered by Cisco UCM from off-premises network connections

Answer: D

NEW QUESTION 44
Refer to the exhibit.

```
000193: Dec  5 14:35:31.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec  5 14:35:32.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec  5 14:35:33.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec  5 14:35:34.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
```

Given this "debug isdn q921" output, what is the problem with the PRI?

A. Nothing, the PRI is sending keepalives.
B. Layer 2 is down on the controller.
C. PRI does not have an IP address configured on the interface.
D. Layer 1 is down on the controller.

Answer: B

NEW QUESTION 47
Which external DNS SRV record must be present for Mobile and Remote Access?

A. _cisco-uds.Jcp.example.com
B. _collab-edge._tls.example.com
C. _collab-edge._tcp.example.com
D. _cisco-uds._tls.example.com

Answer: B

NEW QUESTION 52
Which configuration on Cisco UCM is required for SIP MWl to work?

A. Assign an MWl extension on the mailbox.
B. The line partition must be inside the inbound CSS assigned to the CUC SIP trunk.
C. The line partition must be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
D. Set the"Enable message waiting indicator" on the port group.

Answer: B

Explanation:
The line partition must be inside the inbound CSS assigned to the CUC SIP trunk. This ensures that the SIP MWI messages are sent to the correct destination.
The other options are incorrect because:
- Assigning an MWI extension on the mailbox is not required for SIP MWI to work.
- The line partition does not need to be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
- Setting the "Enable message waiting indicator" on the port group is not required for SIP MWI to work.

NEW QUESTION 53
An engineer roust deploy the Cisco Wet*x app to a Windows Virtual Desktop Infrastructure environment that has a roaming database named spark roaming_store stored In a user's AppData\Roaming directory, Which two command line arguments must be used when running the installer? (Choose two.)

A. ALLUSERS=0
B. ENABLEVDI=1
C. ALLUSERS=1
D. ENABLEVDI=2
E. ROAMINGENABLED=1

Answer: BE

Explanation:
The Cisco Webex app can be installed on a Windows Virtual Desktop Infrastructure (VDI) environment by using the following command-line arguments:
- ENABLEVDI=1 - This argument enables VDI mode for the Webex app.
- ROAMINGENABLED=1 - This argument enables roaming for the Webex app.

The ALLUSERS argument is not required when installing the Webex app on a VDI environment. The ENABLEVDI argument must be set to 1, and the ROAMINGENABLED argument must be set to 1.
The following is an example of the command that can be used to install the Webex app on a VDI environment:
Code snippet
msiexec /i WebexApp.msi ENABLEVDI=1 ROAMINGENABLED=1

NEW QUESTION 58
Which field is configured to change the caller ID information on a SIP route pattern?

A. Route Partition
B. Calling Party Transformation Mask
C. Called Party Transformation Mask
D. Connected Line ID Presentation

Answer: B

NEW QUESTION 61
A SIP phone has been configured in the system with MAC address 0030.96D2.D5CB. The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

A. SIP003096D2D5CB.cnf.xml
B. SEP003096D2D5CB.cnf.xml
C. SEP003096D2D5CB.cnf
D. SIP003096D2D5CB.cnf

Answer: B

NEW QUESTION 64
How are network devices monitored in a collaboration network?

A. The Cisco Discovery Protocol table is shared among devices.
B. Ping Sweep reports "unmanaged' state devices.
C. System logs are collected in a Cisco Prime Collaboration Server.
D. Simple Network Managed Protocol is enabled on each device to poll specific values periodically.

Answer: C

NEW QUESTION 69
A network administrator deleted a user from the LDAP directory of a company. The end user shows as Inactive LDAP Synchronized User in Cisco UCM. Which step is next to remove this user from Cisco UCM?

A. Delete the user directly from Cisco UCM.
B. Wait 24 hours for the garbage collector to remove the user.
C. Restart the Dirsync service after the user is deleted from LDAP directory.
D. Execute a manual sync to refresh the local database and delete the end user.

Answer: B

NEW QUESTION 72
Which DiffServe PHB preserves backward compatibility with any IP precedence scheme?

A. expedited forwarding
B. class selector
C. assured forwarding
D. default

Answer: B

NEW QUESTION 77
Which certificate does the Disaster Recovery System in Cisco UCM use to encrypt its communications?

A. Cisco Tomcat
B. CAPF
C. Cisco CallManager
D. IPsec

Answer: D

NEW QUESTION 79
A company wants to provide remote users with access to its on-premises Cisco collaboration features. Which components are required to enable Cisco Mobile and Remote Access for the users?

A. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server
B. Cisco Unified Border Element, Cisco IM and Presence Server and Cisco Video Communication Server
C. Cisco Expressway-E, Cisco Expressway-C, and Cisco UCM
D. Cisco Unified Border Element, Cisco UCM, and Cisco Video Communication Server

Answer: C

NEW QUESTION 84
An administrator is asked to implement toll fraud prevention in Cisco UCM, specifically to restrict off-net to off-net call transfers. How is this implemented?

A. Enforce ad-hoc conference restrictions.
B. Set the appropriate service parameter.
C. Implement time-of-day routing.
D. Use the correct route filters.

Answer: B

Explanation:
To restrict off-net to off-net call transfers, an administrator can set the "Block Offnet to Offnet Transfer" service parameter to "On". This will prevent users from transferring calls from one external number to another external number.
The other options are not correct because:
- A. Enforce ad-hoc conference restrictions: This will prevent users from creating ad-hoc conferences, but it will not prevent them from transferring calls.
- C. Implement time-of-day routing: This will allow calls to be routed to different destinations based on the time of day, but it will not prevent users from transferring calls.
- D. Use the correct route filters: This will allow calls to be filtered based on the destination, but it will not prevent users from transferring calls.

NEW QUESTION 89
Refer to the exhibit.

```
rule 1 /^\(0[25]..\)\-\(...\)\-\(....$\)/ /\1\2\3/
```

The translation rule is configured on the voice gateway to translate DNIS. What is the outcome if the gateway receives 0255-343-1234 as DNIS?

A. The translation rule is not matched because DNIS does not end with a "$".
B. The translation rule is matched and the translated number is 02553431234.
C. The translation rule is matched and the translated number is 025553431234.
D. The translation rule is not matched because DNIS contains"-".

Answer: B

NEW QUESTION 90
In which location does an administrator look to determine which subscriber the phone registers to if loses registration with the current Cisco UCM subscriber?

A. On Cisco UCM Administration Page Device > Phone > Phone Configuration page
B. On Cisco UCM Administrator Page server > Cisco UCM
C. On Cisco UCM Administrator page system > Device Pool > Cisco UCM group
D. On Cisco UCM Administrator page system > Enterprise Parameters

Answer: C

NEW QUESTION 95
What is the traffic classification for voice and video conferencing?

A. Voice is classified as CoS 4, and video conferencing is CoS 5.
B. Voice and video conferencing are both classified is CoS 3.
C. Voice is classified as CoS 5, and video conferencing is CoS 4.
D. Video conferencing is classified as CoS 1, and voice is CoS 2.

Answer: B

NEW QUESTION 98
On a cisco catalyst switch which command is required to send CDP packets on a switch port that configures a cisco IP phone to transmit voice traffic in 802.1q frames tagged with the voice VLAN ID 221?

A. Device(config-if)# switchport access vlan 221
B. Device(config-if)# switchport vlan voice 221
C. Device(config~if)# switchport trunk allowed vlan 221
D. Device(config-if)# switchport voice vlan 221

Answer: D

NEW QUESTION 103
Which Cisco Unified communications manager configuration is required for SIP MWI integration?

A. Select “Redirecting Diversion Header Delivery— Inbound’ on the SIP trunk
B. Enable “Accept presence subscription” on the SIP trunk security profile
C. Select “Redirecting Diversion Header Delivery – outbound” on the SIP trunk
D. Enable “Accept unsolicited notification” on the SIP Trunk security profile

Answer: D

NEW QUESTION 107
Which DSCP value and PHB equivalent are the default for audio calls?

A. 48 and EF
B. 34 and AF41
C. 32 and AF41
D. 32 and CS4

Answer: A

NEW QUESTION 110

Refer to the exhibit.

```
ROUTER-1(config)# policy-map LLQ_POLICY
ROUTER-1(config-pmap)# class VOICE
ROUTER-1(config-pmap-c)# bandwidth 170
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# class VIDEO
ROUTER-1(config-pmap-c)# bandwidth remaining percent 30
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# exit
```

An engineer must modify the existing QoS policy-map statement to implement LLQ for voice traffic. Which change must the engineer make in the configuration?

A. bandwidth 170 to reserve 170
B. bandwidth 170 to LL1 170
C. bandwidth 170 to priority 170
D. bandwidth 170 to percent 170

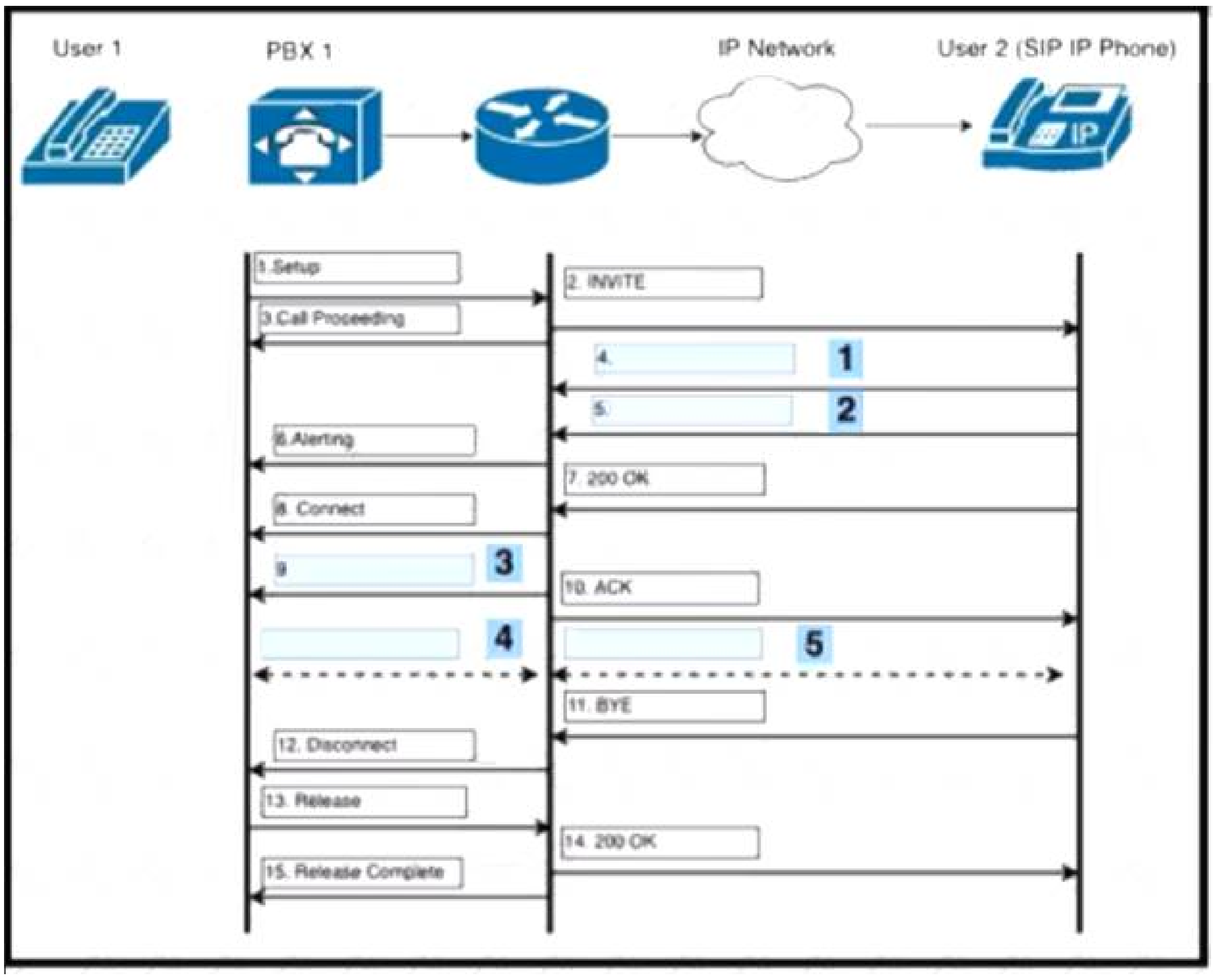
Answer: C

NEW QUESTION 112
An administrator executes the debug isdn q931 command while debugging a failed call. After a test call is placed, the logs return a disconnect cause code of 1. What is the cause of this problem?

A. The media resource is unavailable.
B. The destination number rejects the call.
C. The destination number is busy.
D. The dialed number is not assigned to an endpoint.

Answer: D

NEW QUESTION 114
Refer to the exhibit.



https://i.postimg.cc/wMYy0Fhm/image.png
Drag and drop the flow step labels from the left into the correct order on the right to establish this call flow:
• User 1 calls user 2.
• User 2 answers the call
• user 2 disconnects the call

two-way voice path

two-way RTP channel

100 Trying

Connect ACK

180 Ringing

A. Mastered
B. Not Mastered

Answer: A

Explanation:
* 1. 100 Trying
* 2. 180 Ringing
* 3. two-way voice path
* 4. Connect ACK
* 5. two-way RTP channel

NEW QUESTION 118
Which two recommendations are made to optimize Cisco UCM configuration to reduce the number of toll fraud incidents in an organization? (Choose two.)

A. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco UCM service parameters.
B. Classify all route patterns as on-net or off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.
C. Classify all route patterns as off-net and prohibit off-net to off-net call transfers in Cisco UCM service parameters.
D. Inbound CSS on any gateway typically should have access to internal destinations and PSTN destinations.
E. Inbound CSS on any gateway typically should have access to internal destinations only and not PSTN destinations.

Answer: BE

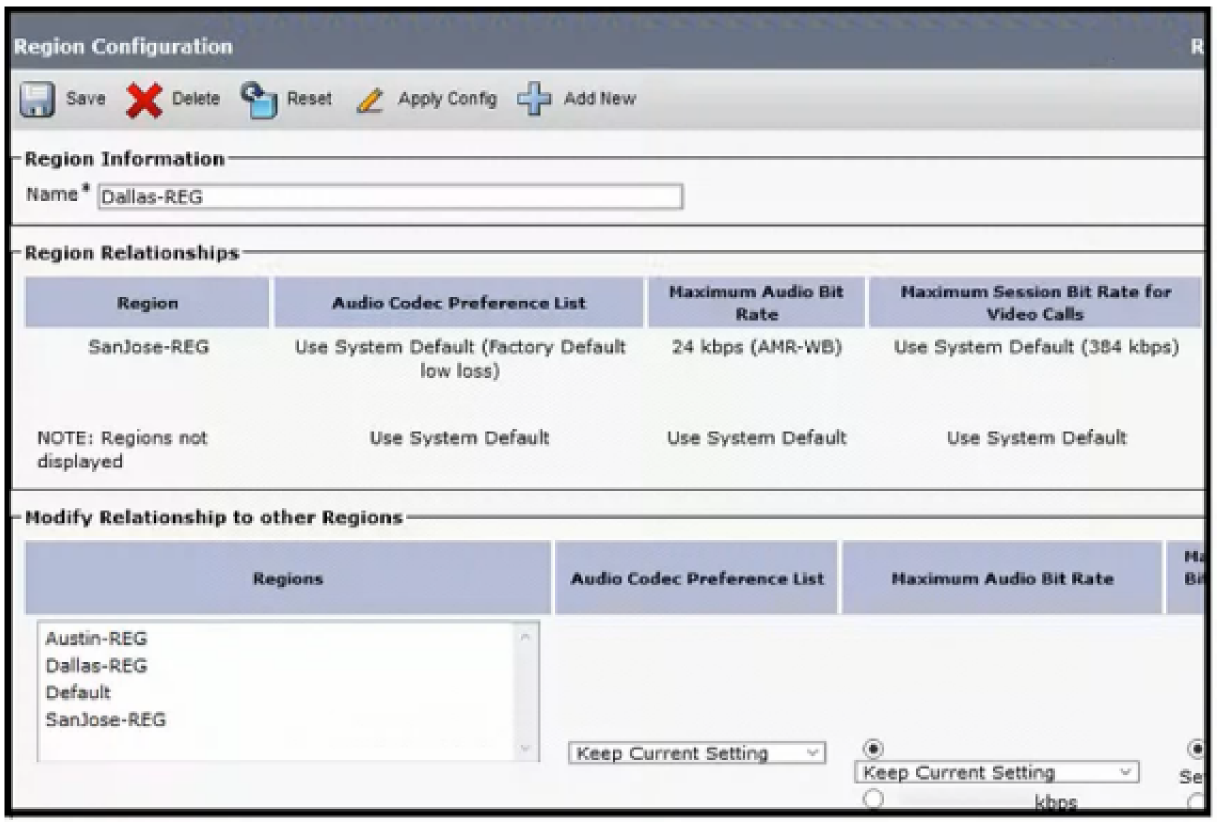
NEW QUESTION 119
Where is Directory Connector hosted in a Cisco Webex Hybrid Services deployment?

A. on a server in the Webex Data Center
B. on a dedicated on-premises server
C. on a Cisco Expressway-C connector host server
D. on an on-premises Microsoft Active Directory server

Answer: B

Explanation:
The Cisco Directory Connector is a software application that is installed on a dedicated on-premises server. It synchronizes user identities between the on-premises directory and the Cisco Webex cloud.

NEW QUESTION 124
Refer to the exhibit.



Which codec should an engineer select for a call made between "Dallas-REG' & "Austin-REG'?

A. MP4A-LATM
B. G.711
C. OPUS
D. G.729

Answer: D

Explanation:
The codec preference list for the "Dallas-REG" region is "Factory Default low loss". This list includes the following codecs in order of preference:
- G.729
- G.711
- OPUS
- MP4A-LATM

The codec preference list for the "Austin-REG" region is "Factory Default low loss". This list includes the following codecs in order of preference:
- G.729
- G.711
- OPUS
- MP4A-LATM

Since both regions have the same codec preference list, the codec that will be used for a call made between "Dallas-REG" and "Austin-REG" is G.729.
G.729 is a narrowband speech codec that was developed by the ITU-T in 1988. It is a low-bitrate codec that provides good quality speech at a bitrate of 8 kbps. G.729 is widely used in VoIP applications and is the default codec for many VoIP systems.
G.711 is a wideband speech codec that was developed by the ITU-T in 1972. It is a high-bitrate codec that provides excellent quality speech at a bitrate of 64 kbps. G.711 is not as widely used as G.729 due to its high bitrate requirements.
OPUS is a lossy audio codec that was developed by the IETF in 2012. It is a low-bitrate codec that provides good quality speech at a bitrate of 6 kbps. OPUS is widely used in VoIP applications and is the default codec for many VoIP systems.
MP4A-LATM is a lossy audio codec that was developed by the IETF in 1999. It is a high-bitrate codec that provides excellent quality speech at a bitrate of 24 kbps. MP4A-LATM is not as widely used as G.729 or OPUS due to its high bitrate requirements.

NEW QUESTION 126
Which service must be enabled when LDAP on cisco UCM is used?

A. Cisco AXL Web Service
B. Cisco CallManager SNMP Service
C. Cisco DirSync
D. Cisco Bulk Provisioning Service

Answer: C

NEW QUESTION 129
Which two steps should be taken to provision a phone after the Self-Provisioning feature was configured for end users? (Choose two.)

A. Ask the Cisco UCM administrator to associate the phone to an end user.
B. Plug the phone into the network.
C. Dial the hunt pilot extension and associate the phone to an end user
D. Dial the self-provisioning IVR extension and associate the phone to an end user.
E. Enter settings menu on the phone and press * ,*,# (star, star, pound).

Answer: BD

NEW QUESTION 132
A Cisco Unity Connection Administrator must set a voice mailbox so that it is accessed from a secondary device. Which configuration on the voice mailbox makes

this change?

A. Attempt Forward routing rule
B. Mobile User
C. Alternate Extensions
D. Alternate Names

Answer: C

NEW QUESTION 135
An administrator must implement toll fraud prevention on Cisco UCM using these parameters:
• Enable Forced Authorization Code 112211.
• Set an authorization level of 3 for the route pattern 8005551212.
• Require no access code to dial 10-digit numbers. How must the route pattern be implemented?

A. Pattern = 1122113.8005551212
B. Pattern = 8005551212.1122113
C. Pattern = 8005xxxxxx
D. Pattern = 3.800xxxxxxx

Answer: A

Explanation:
To implement toll fraud prevention on Cisco UCM, an administrator can use the following parameters:
- Enable Forced Authorization Code 112211.
- Set an authorization level of 3 for the route pattern 8005551212.
- Require no access code to dial 10-digit numbers.

The route pattern must be implemented as follows: Pattern = 1122113.8005551212
This will require users to enter the authorization code 112211 followed by the number 8005551212 to dial this number. The authorization level of 3 will prevent users from transferring calls to this number.

NEW QUESTION 136
Which two technical reasons make QoS a necessity in a video deployment? (Choose Two)

A. Low response time between endpoints
B. Provisioned bandwidth of the link
C. Variable bit rate of the video stream
D. Bursly behavior of video traffic

Answer: CD

NEW QUESTION 137
What is the function of the Cisco Unity Connection Call Handler?

A. routes calls to a user based on caller input
B. queues calls
C. allows customized scripts for IVR capabilities
D. searches a list of extensions until the call is answered

Answer: A

Explanation:
A Cisco Unity Connection Call Handler is a software application that answers calls, plays greetings, and routes calls to users based on caller input. Call handlers can be used to create automated attendants, voice menus, and other interactive voice response (IVR) applications.
Call handlers are created and managed using the Cisco Unity Connection Administration interface. When creating a call handler, you can specify a variety of settings, including the greeting that is played, the caller input options that are available, and the destination that calls are routed to.
Call handlers are a powerful tool that can be used to create a variety of IVR applications. By using call handlers, you can improve the efficiency of your organization's communications and provide a better experience for your callers.
Here are some additional tips for using call handlers:
- Use call handlers to create automated attendants that can answer calls and route them to the appropriate person or department.
- Use call handlers to create voice menus that can provide callers with information or options.
- Use call handlers to create interactive voice response (IVR) applications that can collect information from callers and process their requests.

NEW QUESTION 142
What is the default TCP port for SIP OAuth mode in Cisco UCM?

A. 5011
B. 3174
C. 8443
D. 5090

Answer: D

Explanation:
The Cisco Unified Communications Manager (CUCM) uses SIP Phone OAuth Port (5090) to listen for SIP line registration from Jabber OnPremise devices over TLS. However, CUCM uses SIP Mobile Remote Access Port (default 5091) to listen for SIP line registrations from Jabber over Expressway through mTLS. Both of these ports are configurable.

NEW QUESTION 144
Why does Cisco UCM use DNS?

A. It provides certificate-based security for media
B. It resolves FQDN to IP address resolution for trunks
C. it connects endpoints to single sign-on services.
D. It provides SRV resolution to the endpoints registered

Answer: D

NEW QUESTION 145
Refer to the exhibit.

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

The SDP offer/answer has been completed successfully but there is no DTMF when users press keys. What is the cause of the issue?

A. Payload type 110 was negotiated rather than type 101.
B. DTMF was negotiated property in these messages.
C. DTMF was not negotiated on the call.
D. G.729 rather than G.711ulaw was negotiated.

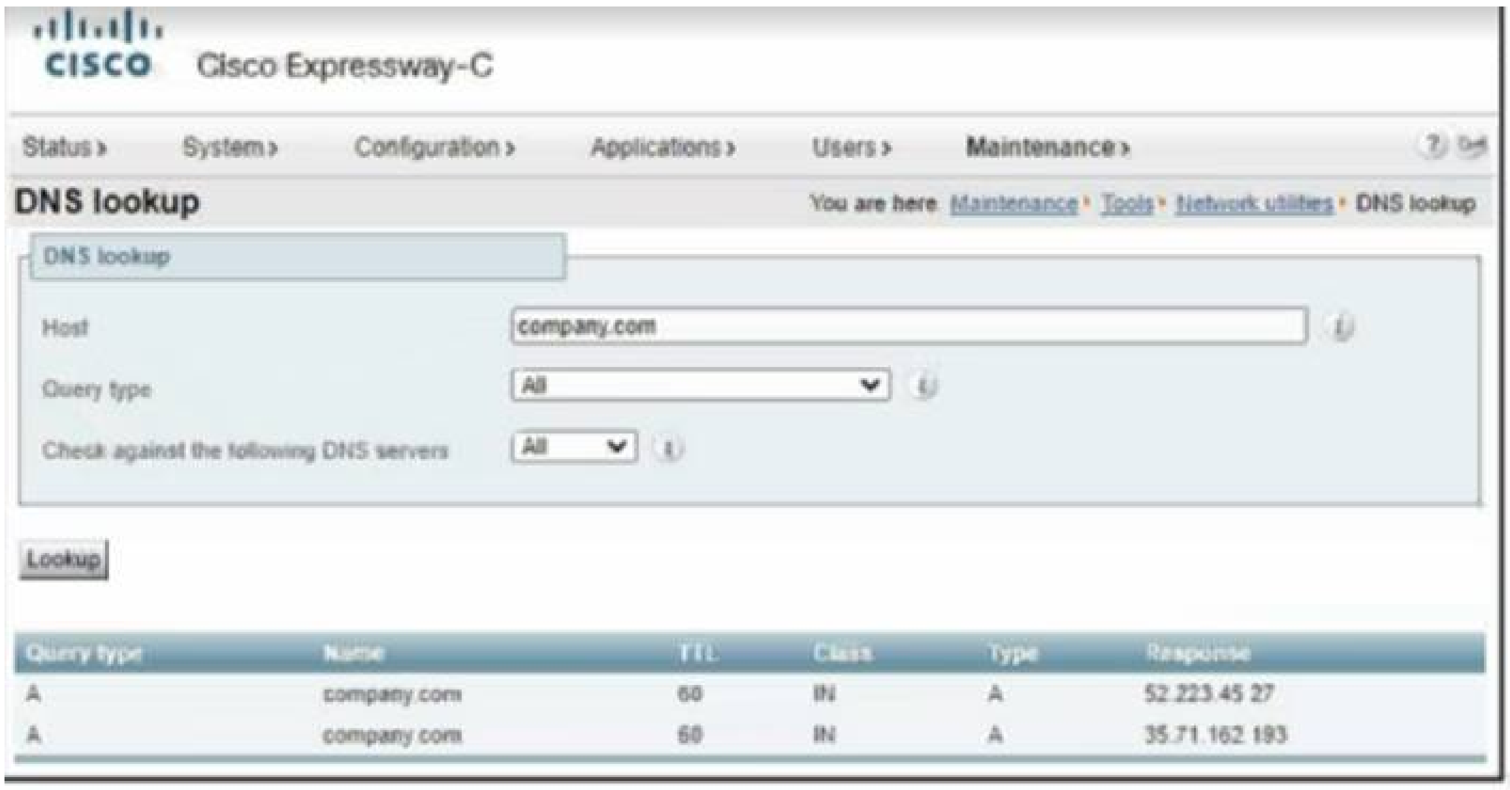
Answer: C

NEW QUESTION 148
An engineer is asked to implement on-net/off-net call classification in Cisco UCM. Which two components are required to implement this configuration? (Choose two.)

A. CTI route point
B. SIP route patterns
C. route group
D. route pattern
E. SIP trunk

Answer: DE

NEW QUESTION 153
Refer to exhibit.



A company recently deployed CISCO Jabber Users log in to Jabber by using their email address in a domain named company com. The users report that they cannot register their telephony services when working from unless they use a VPN. An engineer runs DNS lookup tool in Cisco Expressway-C to troubleshoot tie What IS me cause of the issue?

A. The company com domain must be resolved only in Expressway-E
B. There is a missing SRV record for the company.com domain.
C. The TTL value for the company.com is too short.
D. There must be only response for the company.com domain

Answer: B

NEW QUESTION 155
Which behavior occurs when Cisco UCM has a Call Manager group that consists of two subscribers?

A. Endpoints attempt to register with the top subscriber in the list.
B. Endpoints attempt to register with (he bottom subscriber in the list.
C. Endpoints attempt to register with both subscribers in a load-balanced method.
D. If a subscriber is rebooted, endpoints deregister until the rebooted system is back in service.

Answer: A

NEW QUESTION 157
Which call flow matches traffic from a Mobile and Remote Access registered endpoint to central call control?

A. Endpoint>Expressway-C>Expressway-E>Cisco UCM
B. Endpoint>Expressway-E>Expressway-C> Cisco UCM
C. Endpoint>Expressway-E> Cisco UCM
D. Endpoint>Expressway-C> Cisco UCM

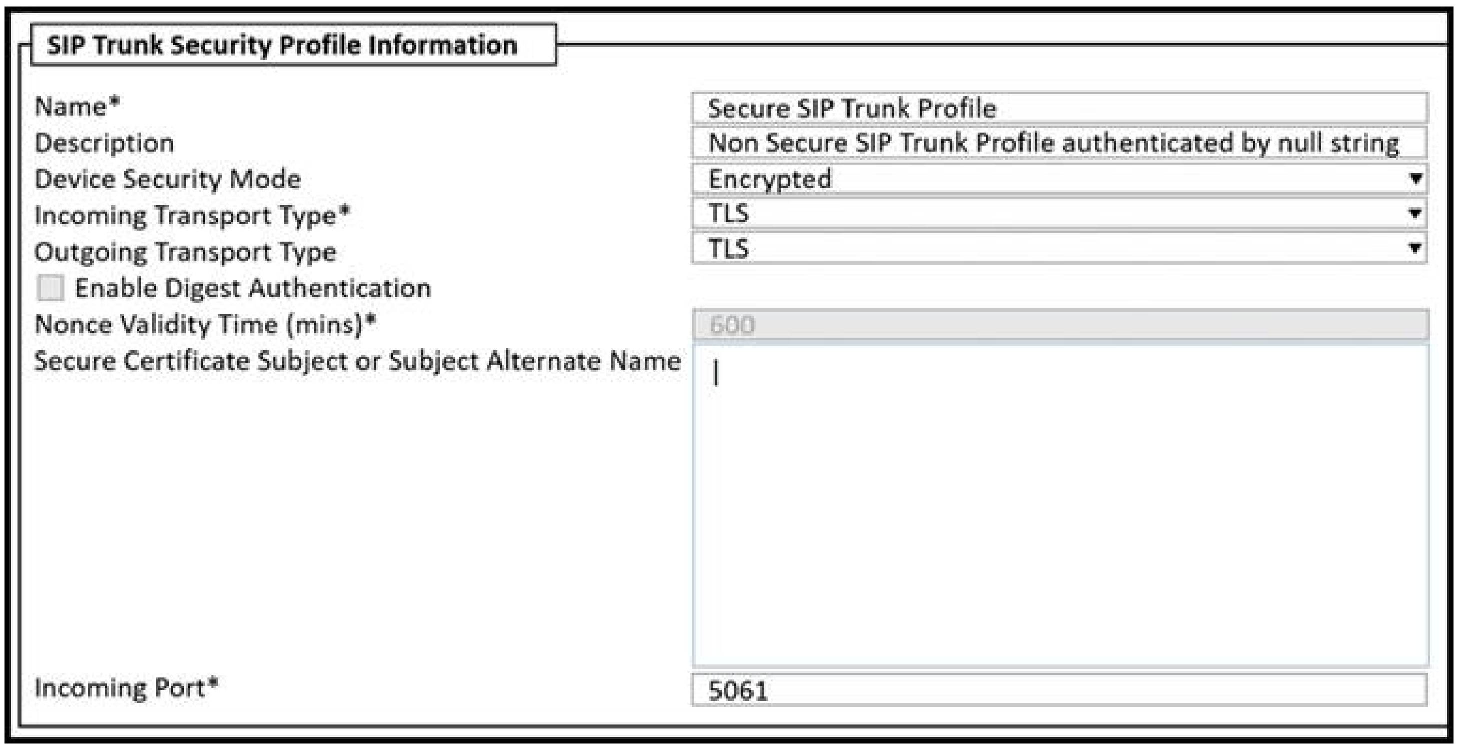
Answer: A

Explanation:
The call flow for a Mobile and Remote Access registered endpoint to central call control is as follows:
- The endpoint registers with the Expressway-C.
- The Expressway-C forwards the registration request to the Expressway-E.
- The Expressway-E forwards the registration request to the Cisco UCM.
- The Cisco UCM registers the endpoint.

When the endpoint places a call, the call flow is as follows:
- The endpoint sends the call request to the Expressway-C.
- The Expressway-C forwards the call request to the Expressway-E.
- The Expressway-E forwards the call request to the Cisco UCM.
- The Cisco UCM places the call.

The Expressway-C and Expressway-E are used to provide secure access to the Cisco UCM for endpoints that are not located on the corporate network. The Expressway-C is located on the corporate network, and the Expressway-E is located in the DMZ.

NEW QUESTION 161
Refer to the exhibit.

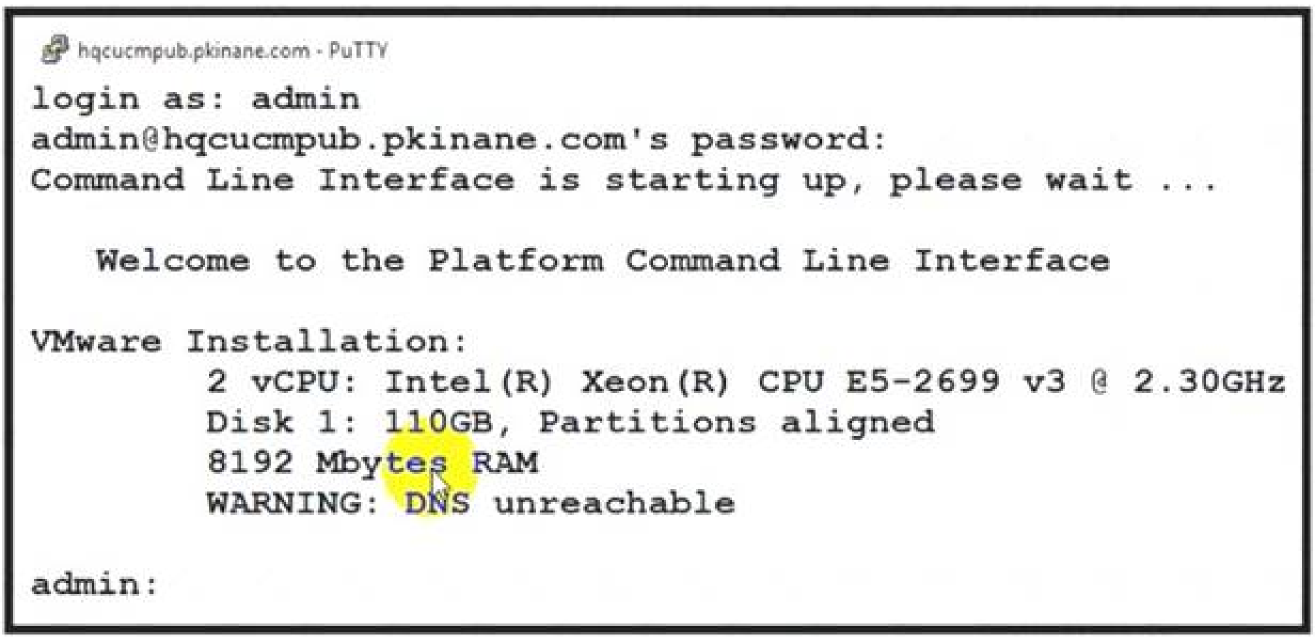


An administrator configures a secure SIP trunk on Cisco UCM.
Which value is needed in the secure certificate subject or subject alternate name field to accomplish this task?

A. The fully qualified domain name of the remote device that is configured on the SIP trunk.
B. The common name of the Cisco UCM CallManager certificate.
C. The full qualified domain name of all Cisco UCM nodes that run the CallManager service.
D. The common name of the remote device certificates.

Answer: B

NEW QUESTION 166
Refer to the exhibit.



An administrator accesses the terminal of a Cisco UCM and starts a packet capture. Which two commands must the administrator use on Cisco UCM to generate DNS traffic? (Choose two.)

A. utils ntp status
B. show cdp neighbor
C. show version active
D. utils diagnose test
E. autils diagnose module validate Network

Answer: DE

NEW QUESTION 168
An administrator configures international calling on a Cisco UCM cluster and wants to minimize the number of route patterns that are needed. Which route pattern enables the administrator to match variable-length numbers?

A. 9.011#
B. 9.011@
C. 9.011!
D. 9.011*

Answer: C

NEW QUESTION 173
Which two devices are supported by the Flexible DSCP Marking and Video Promotion feature? (Choose two.)

A. MGCP devices
B. SCCP devices
C. pass-through MTPs

D. H.323 trunks
E. DX80

Answer: BC

NEW QUESTION 177
Which two types of trunks can be used when configuring a hybrid Local Gateway for Cisco Webex Calling? (Choose Two.)

A. TLS-based
B. certificate-based
C. registration-based
D. authentication-based
E. OAuth-based

Answer: AC

Explanation:
These are the two types of trunks that can be used when configuring a hybrid local gateway for Cisco Webex Calling1. A TLS-based trunk uses Transport Layer Security (TLS) to secure the SIP signaling between the hybrid local gateway and Webex Calling1. A registration-based trunk uses SIP registration to authenticate the hybrid local gateway with Webex Calling and receive calls from the cloud1.

NEW QUESTION 182
An administrator installs a new Cisco TelePresence video endpoint and receives this error:"AOR is not permitted by Allow/Deny list. Which action should be taken to resolve this problem?

A. Reboot the VCS server and attempt reregistration.
B. Change the SIP trunk configuration.
C. Correct the restriction policy settings.
D. Upload a new policy in VCS.

Answer: C

Explanation:
The error message "AOR is not permitted by Allow/Deny list" indicates that the endpoint is not allowed to register with the VCS server because it is not on the Allow List or it is on the Deny List. To resolve this problem, you must correct the restriction policy settings.

NEW QUESTION 185
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