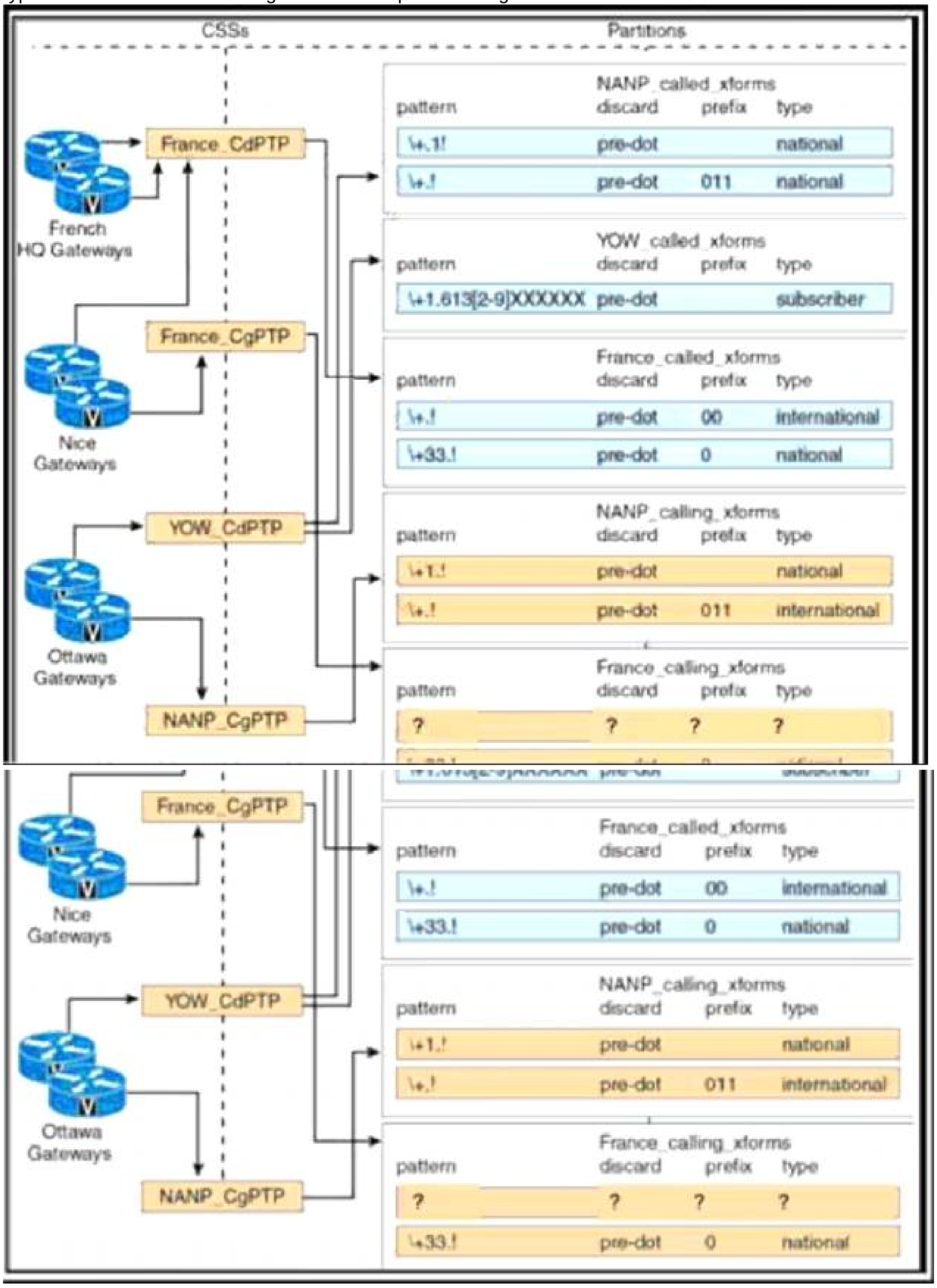
Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

NEW QUESTION 1
Refer to the exhibit A call from +1 613 555 1234 that is sent out through the Nice Gateways should result in a calling party of 001 613 555 1234 with the numbering type "international' Which configuration accomplishes this goal?



A. \+.001! pre-dot 1 international
B. \+1.1 none pre-dot 001 international
C. \+.! pre-dot 00 international
D. 613XXXXXXX none +011 internationa

Answer: C

NEW QUESTION 2
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 3
Refer to the exhibit.

```
Gateway1#show sccp
SCCP Admin State: UP
Gateway Local Interface: Loopback0
        IPv4 Address: 192.168.12.1
        Port Number: 2000

Gateway1#
Gateway1#show ccm-manager
% Call Manager Application is not enabled
Gateway1#

Gateway1#show mgcp
MGCP Admin State DOWN, Oper State DOWN - Cause Code NONE
MGCP call-agent: none Initial protocol service is MGCP 0.1
MGCP validate call-agent source-ipaddr DISABLED
MGCP validate domain name DISABLED
MGCP block-newcalls DISABLED
MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED
```

A collaboration engineer adds an analog gateway to a Cisco UCM cluster. The engineer chooses MGCP over SCCP as the gateway protocol. Which two actions ensure that the gateway registers? (Choose two.)

A. Enter "no seep" on the gateway in configuration mode.
B. Enter "ccm-manager mgcp" on the gateway in configuration mode.
C. Enter "mgcp" on the gateway in configuration mode.
D. Enter "ccm-manager config" on the gateway in configuration mode.
E. Delete and re-add the gateway configuration in Cisco UCM.

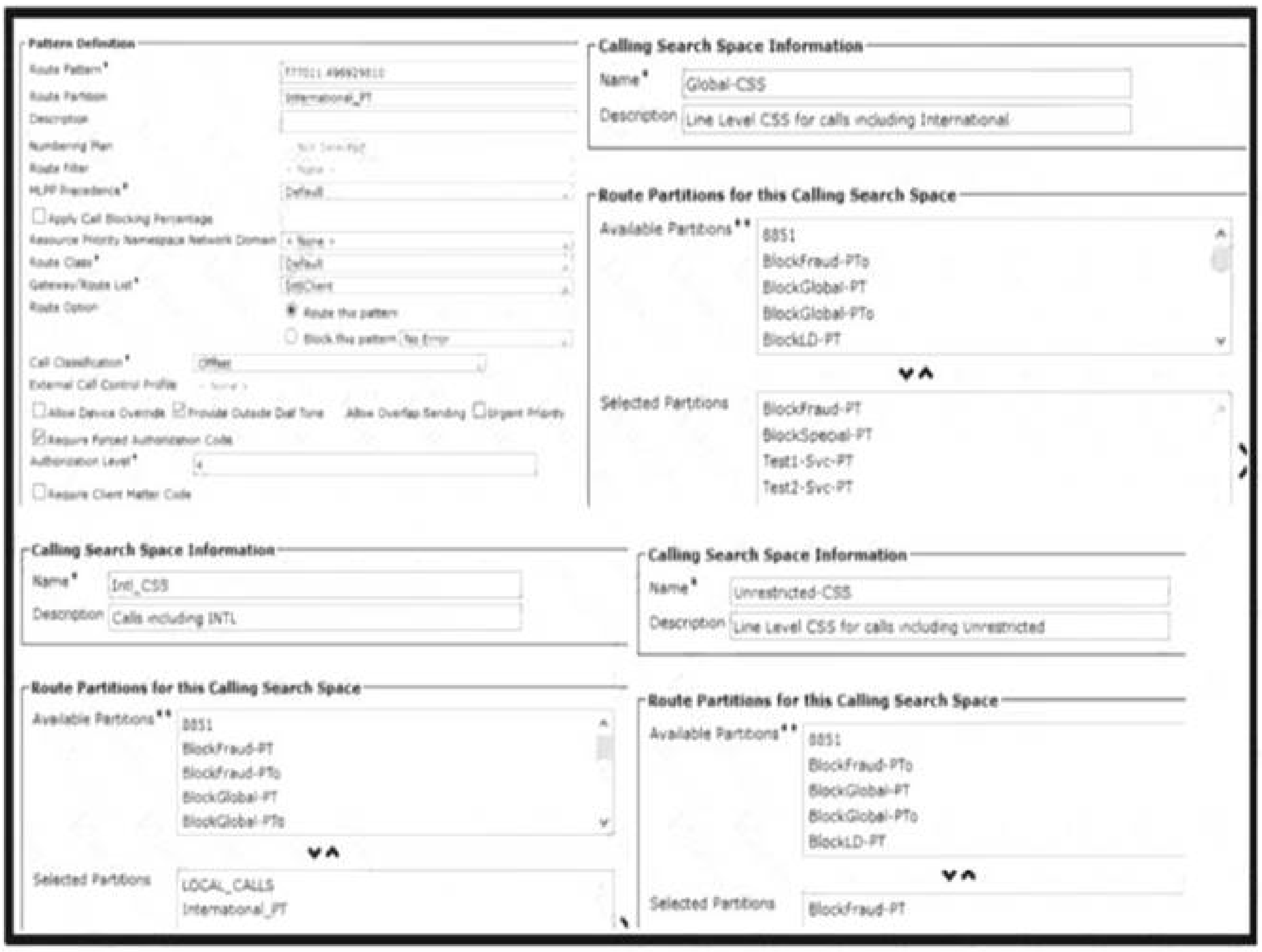
Answer: BC

NEW QUESTION 4
What is an indicator of network congestion in VoIP communications?

A. jitter increase due to variable delay
B. discards in the interface of routers and switches
C. video loss due to video frame corruption
D. gaps in the voice due to packet loss

Answer: A

NEW QUESTION 5
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

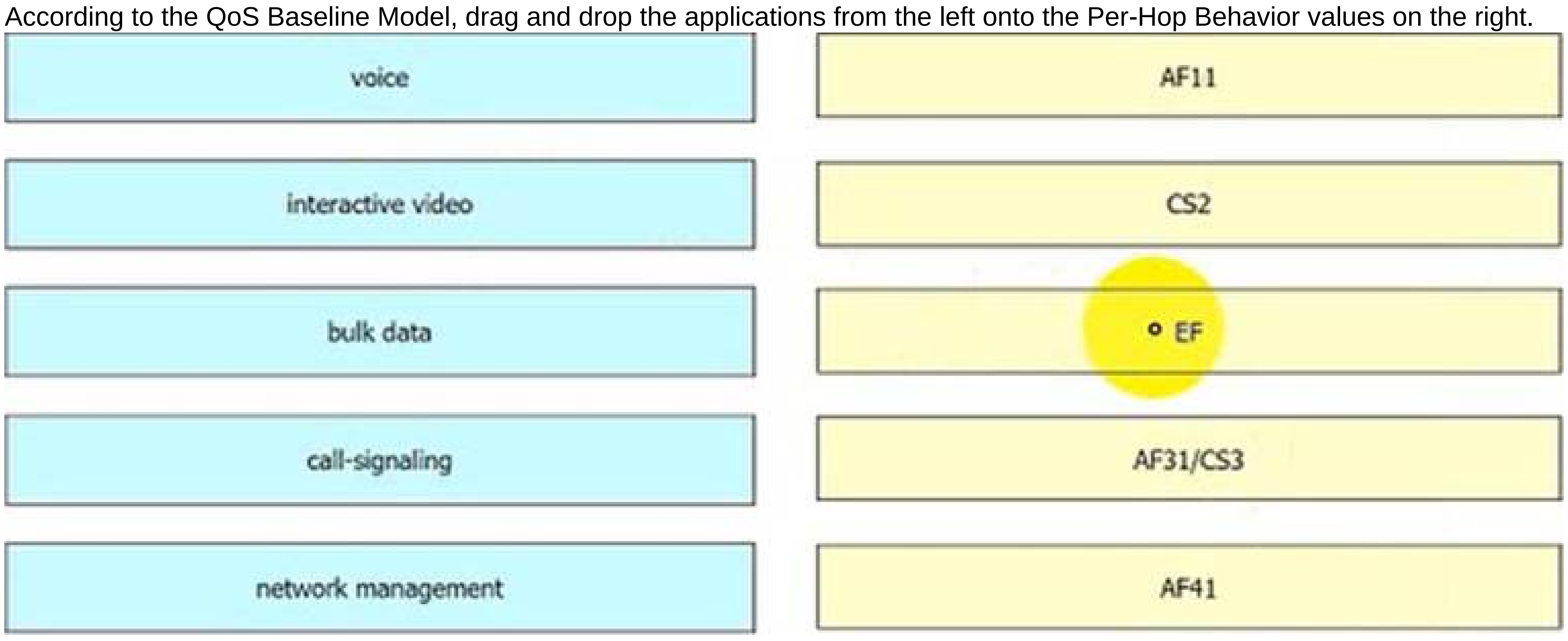
Answer: C

NEW QUESTION 6
A Cisco voice gateway is configured to use a sip-kpml DTMF relay in global settings. A new SIP dial peer is configured for a third-party application that only supports an in-band DTMF relay. Which commands must an engineer run on the dial peer?

A. dtmf-relay sip-info
B. dtmf-relay sip-notify
C. dtmf-relay rtp-net
D. no dtmf-relay sip-kpml

Answer: C

NEW QUESTION 7
According to the QoS Baseline Model, drag and drop the applications from the left onto the Per-Hop Behavior values on the right.



A. Mastered
B. Not Mastered

Answer: A

Explanation:

NEW QUESTION 8
What is an advantage of using Cisco Webex Control HuB?

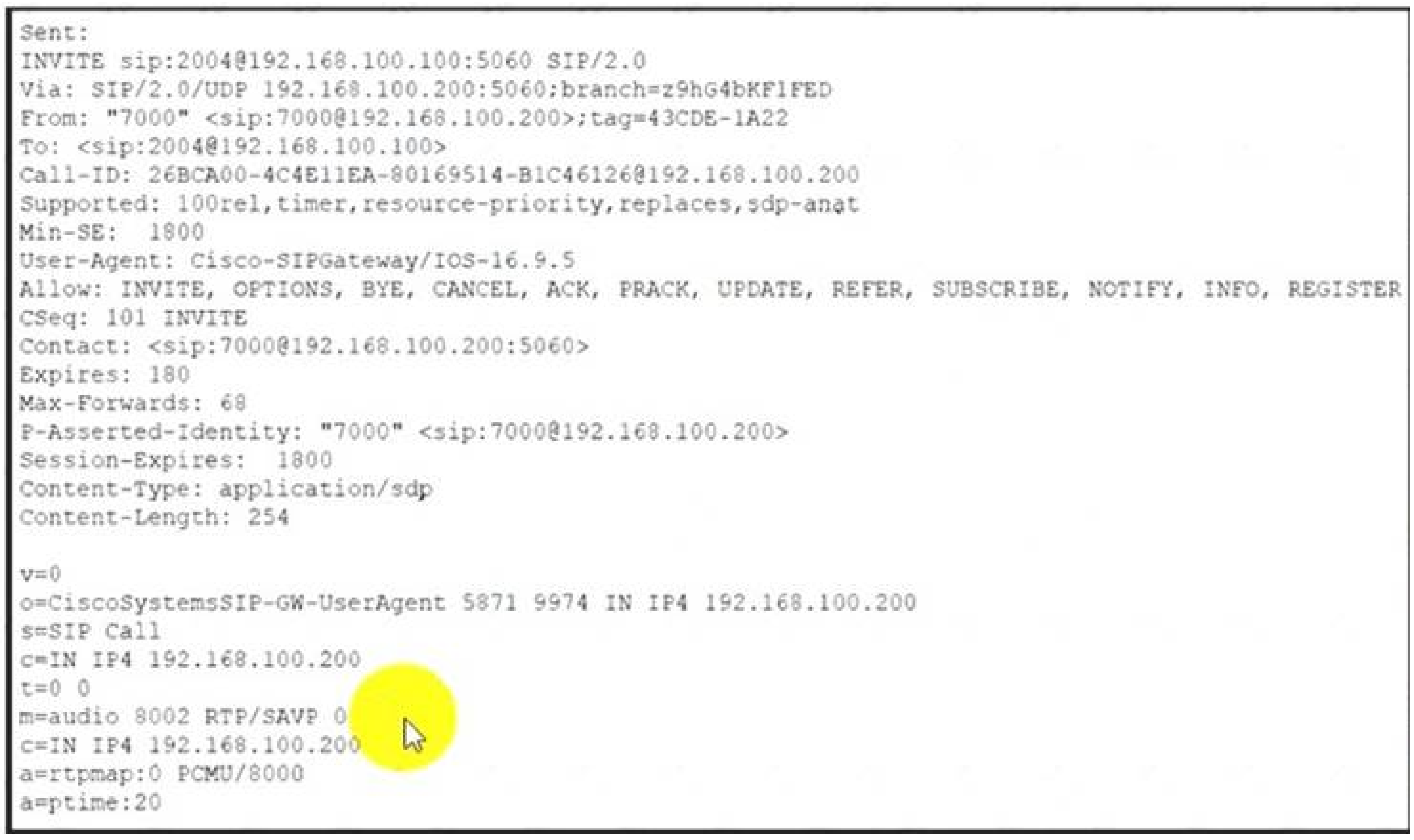
A. enables the provisioning, administration, and management of Webex services and Webex Hybrid Services
B. brings Video, audio, and web communication together to meet the collaboration needs of the modern workplace
C. provides streamlined communication and collaboration for a hybrid workforce
D. offers 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 9
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 10
What describes the outcome when the trust boundary is defined at the Cisco IP phone?

A. Packets or Ethernet frames are remarked at the distribution layer switch.
B. Packets or Ethernet frames are not remarked at the access layer switch.
C. Packets or Ethernet frames are not remarked by the IP phone.
D. Packets or Ethernet frames are remarked at the access layer switch.

Answer: B

NEW QUESTION 10
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 15
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 17
Which dial plan function restricts calls that are made by a lobby phone to internal extensions only?

A. manipulation of dialed destination
B. path selection
C. calling privileges
D. endpoint addressing

Answer: C

NEW QUESTION 22
An engineer is configuring Cisco Jabber for Windows and must implement desk phone control mode for some of the users. Which access control group must be configured for those users to enable this functionality?

A. Allow Control of Device from CTI
B. Standard CTI Secure Connection
C. Standard CTI Enabled
D. Standard CTI Allow Reception of SRTP Key Material

Answer: C

NEW QUESTION 26
An administrator is designing a new Cisco UCM for a company with many departments and firm structure on their communications policies. The administrator must make sure that these communication policies are reflected in the phone system setup. Certain departments cannot be accessed directly, even if they have dedicated DID numbers. Some phones, especially public phones, must not be able to dial international numbers Which type of function is configured to control which device is allowed to call another device in Cisco UCM?

A. partitions and calling search spaces
B. calling patterns and route patterns
C. regions and device pools
D. links and pipes

Answer: A

NEW QUESTION 28
A remote office has a less-than-optimal WAN connection and experiences packet loss, delay and jitter. Which VoIP codec is used in this situation?

A. G722.1
B. iLBC
C. G.711alaw
D. G.729A

Answer: B

NEW QUESTION 33
What are two common attributes of XMPP XML stanzas? (Choase two.)

A. from
B. to

C. destination
D. version
E. Source

Answer: AB

NEW QUESTION 34
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 35
Which two features of Cisco Prime Collaboration Assurance require advanced licensing? (Choose two.)

A. real lime alarm browse
B. multicluster support
C. call quality monitoring
D. email notifications
E. customizable events

Answer: BC

NEW QUESTION 40
What is set when using COS to mark an Ethernet frame?

A. Ipp bits
B. IP ECN bits
C. DCSP bits
D. 802.1 p User Priority bits

Answer: D

Explanation:
When using COS to mark an Ethernet frame, the 802.1 p User Priority bits are set. These bits are used to indicate the priority of the frame. The higher the priority, the more likely the frame is to be transmitted first.

NEW QUESTION 44
Which wildcard must an engineer configure to match a whole domain in SIP route patterns?

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

Answer: A

Explanation:
The asterisk (*) wildcard is used to match any sequence of characters, including an empty sequence. Therefore, it can be used to match any domain name in a SIP Route Pattern.
The other options are not correct because:
- C. !: The ! symbol is used to negate a character class.
- D. .: The . symbol is used to match any single character.

NEW QUESTION 48
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 51
Where in Cisco UCM is restrictions on audio bandwidth configured?

A. location
B. partition
C. region
D. serviceability

Answer: C

NEW QUESTION 52
How does Cisco UCM perform a digit analysis on-hook versus off-hook for an outbound call from a Cisco IP phone that is registered to Cisco UCM?

A. On-hoo
B. by pressing the digits and entering "#" to process the cal
C. UCM performs a digit-by-digit analysis; off-hoo
D. UCM analyzes all digits as a string.
E. On-hoo
F. no digit analysis is performed; off-hoo
G. UCM requires the '#" to start the digit analysis
H. On-hoo
I. UCM performs a digit-by-digit analysis; off-hoo
J. UCM considers all digits were dialed and does not wait for additional digits.
K. On-hoo
L. UCM considers all digits were dialed and does not wait for additional digits; off-hoo
M. UCM performs a digit-by-digit analysis.

Answer: D

NEW QUESTION 57
Which Webex Calling construct is used to organize calling features within a physical site?

A. client settings
B. locations
C. service settings
D. call routing

Answer: B

Explanation:
A location is a physical site that contains users, devices, and resources. Locations are used to organize calling features within a physical site. For example, you can create a location for each of your offices and then assign users, devices, and resources to that location. This will allow you to manage calling features for each office separately.

NEW QUESTION 62
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 802.10 frames, tagged with the voice VLAN ID 221?

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

Answer: A

NEW QUESTION 67
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 72
An employee of company ABC just quit. The IT administrator deleted the employee’s user id from the active directory at 10 a m. on March 4th The nightly sync occurs at 10 p.m. daily. The IT administrator wants to troubleshoot and find a way to delete the user id as soon as possible How is this issue resolved?

A. Wart until 10 pm on March 4th when the user is automatically removed from Cisco UCM.

B. Wait until 10 pm on March 5th when the user is automatically removed from Cisco UCM.
C. Wait until 3 15 a.
D. on March 6th for garbage collection to remove the user from Cisco UCM.
E. Wait until 315am on March 5th for garbage collection to remove the user from Cisco UCM.

Answer: C

NEW QUESTION 76
A customer wants to conduct B2B video calls with a partner using on-premises conferencing solution. Which two devices are needed to facilitate this request?

A. Expressway-C
B. Cisco Telepresence Management Suite
C. Expressway-E
D. MGCP gateway
E. Cisco Unified Border Element

Answer: AC

NEW QUESTION 81
Refer to the exhibit.

```
dial-peer voice 10 voip
          destination-pattern 1…
          session target ipv4:10.1.1.1
          no vad
```

An engineer configures a VoIP dial peer on a Cisco gateway. Which codec is used?

A. G711alaw
B. No codec is used (missing codec command)
C. G.711ulaw
D. G729r8

Answer: D

NEW QUESTION 85
According to QoS guidelines, what is the packet loss for streaming video?

A. Not more than 8%
B. Not more than 1%
C. Not more than 3%
D. Not more than 5%

Answer: B

NEW QUESTION 87
A company hosts a conference call with no local users. How does the administrator stop the conference from continuing?

A. modifies the Drop Ad Hoc Conference service parameter
B. modifies the Block OffNet to OffNet Transfer service parameter
C. removes the transcoder
D. changes the codecs that are supported on the conference resource

Answer: A

NEW QUESTION 91
What is the major difference between the two possible Cisco IM and Presence high-availability modes?

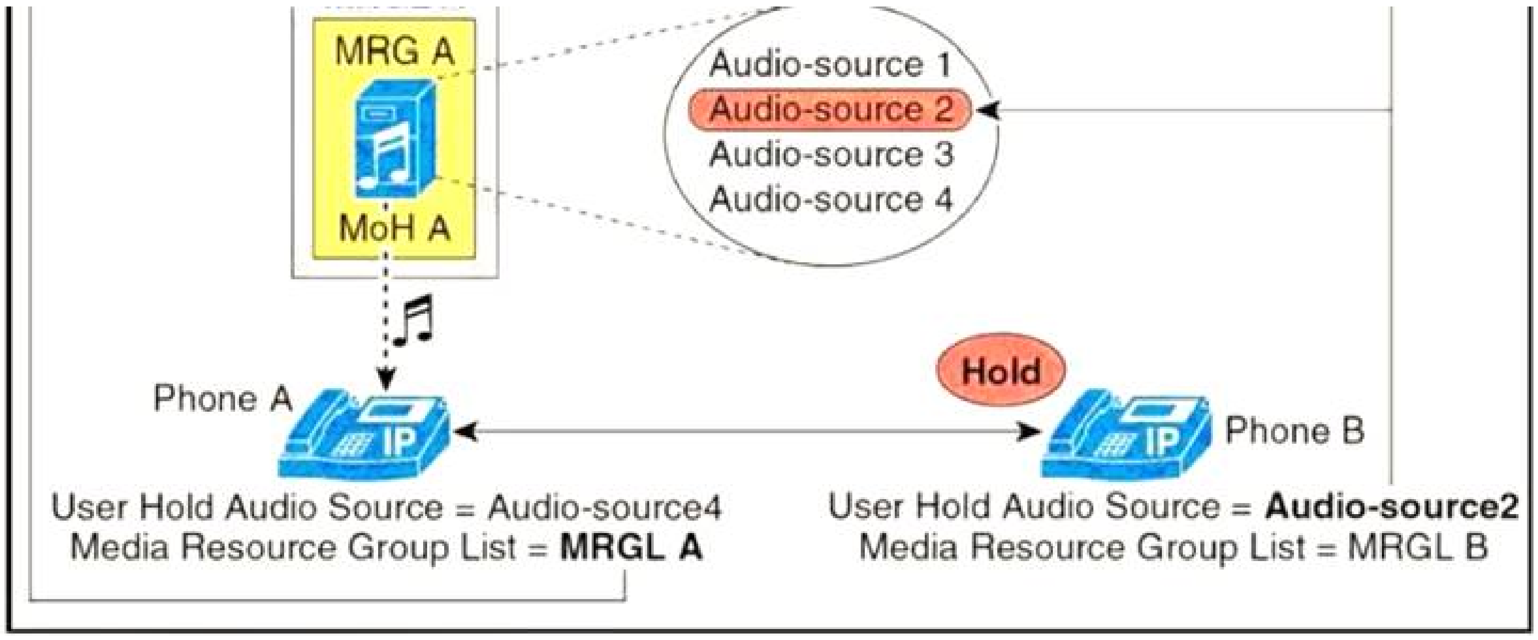
A. Balanced mode provides user load balancing and user failover in the event of an outag
B. Active/standby mode provides an always on standbynode in the event of an outage, and it also provides load balancing.
C. Balanced mode provides user load balancing and user failover only for manually generated failovers.Active/standby mode provides anunconfigured standby node in the event of an outage, but it does not provide load balancing.
D. Balanced mode provides user load balancing and user failover in the event of an outag
E. Active/standby mode provides an always on standbynode in the event of an outage, but it does not provide load balancing.
F. Balanced mode does not provide user load balancing, but it provides user failover in the event of an outag
G. Active/standby mode provides analways on standby node in the event of an outage, but it does not provide load balancing.

Answer: C

Explanation:
Balanced mode provides user load balancing and user failover in the event of an outage. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.
Here is a more detailed explanation of the two modes:

- Balanced mode: In balanced mode, the IM and Presence Service nodes are configured to work together to provide high availability. The nodes are configured in a redundancy group, and the system automatically balances the load of users across the nodes in the group. If one of the nodes fails, the system automatically fails over the users to the other nodes in the group.
- Active/standby mode: In active/standby mode, one of the IM and Presence Service nodes is designated as the active node, and the other nodes are designated as standby nodes. The active node handles all of the user traffic, and the standby nodes are only used if the active node fails. If the active node fails, the system automatically fails over to one of the standby nodes.

NEW QUESTION 96
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 100
What is the maximum number of servers that are in an IM and Presence presence redundancy group?

A. 10
B. 6
C. 2
D. 4

Answer: C

NEW QUESTION 105
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 106
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 108
A company wants to provide remote user with access to its premises Cisco collaboration features. Which components are required to enable cisco mobile and remote access for the users?

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

E. Cisco UCM, and Cisco Video Communication Server

Answer: B

NEW QUESTION 112
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 115
Which behavior occurs when Cisco UCM has a CallManager group that consists of two subscribers?

A. Endpoints attempt to register with the bottom subscriber in the list.
B. Endpoints attempt to register with the top 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: B

NEW QUESTION 120
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 121
A user forwards a corporate number to an international number. What are two methods to prevent this forwarded call? (Choose two.)

A. Configure a Forced Authorization Code on the international route pattern.
B. Block international dial patterns in the SIP trunk CSS.
C. Set Call Forward All CSS to restrict international dial patterns.
D. Set the Call Classification to OnNet for the international route pattern.
E. Check Route Next Hop By Calling Party Number on the international route pattern.

Answer: AC

NEW QUESTION 122
Refer to the exhibit.

```
ISDN Serial1:23 interface
        dsl 1, interface ISDN Switchtype =
primary-5ess
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State =
TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Activated dsl 1 CCBs = 0
    The Free Channel Mask:  0x807FFFFF
    Total Allocated ISDN CCBs = 5
```

What causes the PRI issue?

A. The controller shut down
B. The cable is unplugged

C. The framing is configured incorrectly
D. The clock source is incorrect.

Answer: B

Explanation:
The show controller t1 command shows that the T1 interface is up but the line protocol is down. This indicates that the physical layer is working but the data link layer is not. The most likely cause of this is that the cable is unplugged.

NEW QUESTION 125
A user dials 9011841234567 to reach Vietnam. Which steps send the call to the PSTN provider as 011841234567?

A)

in the Called Party Transformation Pattern Configuration section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot

B)

in the Calling Party Transformation Patterns section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot 10-10-Dialing

C)

in the Calling Party Transformation Patterns section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot

D)

in the Called Party Transformation Pattern Configuration section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot 10-10-Dialing

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

Answer: A

NEW QUESTION 128
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 130
Why would we not include an end user’s PC device in a QoS trust boundary?

A. The end user could incorrectly tag their traffic to advertise their PC as a default gateway.
B. The end user could incorrectly tag their traffic to bypass firewalls.
C. There is no reason not to include an end user's PC device in a QoS trust boundary.
D. The end user may incorrectly tag their traffic to be prioritized over other network traffic.

Answer: D

NEW QUESTION 135
An engineer is notified that the Cisco TelePresence MX800 that is registered in Cisco Unified communications Manager shows an empty panel, and the Touch 10 shows a corresponding icon with no action when pressed. Where does the engineer go to remove the inactive custom panel?

A. The phone configuration page in CUCM Administration
B. The SIP Trunk security profile page in CUCM Administration
C. The software Upgrades page in CUCM OS Administration
D. The In-Room control Editor on the webpage of the MX800

Answer: D

NEW QUESTION 139
The chief officer at a company must reduce collaboration infrastructure costs by onbording all on-permises equipment to the cloud by using CISCO Webex Control Hub. Administrators need the ability to manage upgrades and set up hot desking for on-premises devices.
Which action must be taken before on boardng devices by using the Control Hub?

A. Configure tie Control Hub organization ID on the devices
B. Acquire a license for each device.
C. Allow HTTP traffic from each device to Control Hub.
D. Upgrade all the devices to software version CE9.15 or later

Answer: D

Explanation:
This is a prerequisite for using the Device Connector tool, which allows you to onboard and register several devices simultaneously to the Webex Control Hub1. The Device Connector tool creates a workspace, an activation code, and activates all of your devices in one go1. This way you don't need to be physically present in the same room to activate the devices.
The other options are not required before onboarding devices by using the Control Hub:
- Configuring the Control Hub organization ID on the devices is not necessary, as the Device Connector tool will send the device information to your Webex organization and generate activation codes for them 1.
- Acquiring a license for each device is not necessary, as you can assign licenses to users and devices after they are registered to the Webex Control Hub2.
- Allowing HTTP traffic from each device to Control Hub is not necessary, as HTTPS connectivity is required for the Device Connector tool to communicate with the devices1.

NEW QUESTION 141
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 144
Refer to the exhibit.
https://i.postimg.cc/C57TkczG/image.png

```
v=0
o=Cisco-SIPUA 13439 0 IN IP4 10.10.10.10
s=SIP Call
b=AS:4064
t=0 0
m=audio 0 RTP/AVP 114 9 124 113 115 0 8 116 18 101
c=IN IP4 10.10.10.10
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-
maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
```

A call is falling to establish between two SIP Devices The called device answers with these SOP Which SOP parameter causes issue?

A. The calling device did not offer a ptime value
B. The media stream is set to send only
C. The payload for G.711ulaw must be 18.

D. The RTP port is set to 0.

Answer: D

Explanation:
The RTP port is used to send and receive media packets during a call. If the RTP port is set to 0, the called device will not be able to send or receive media packets, and the call will fail.
The other options are not correct because:
A. The calling device did not offer a ptime value: The ptime value is used to specify the amount of time between each media packet. If the calling device does not offer a ptime value, the called device will use the default value of 20 milliseconds.
B. The media stream is set to sendonly: The media stream is set to sendonly when the called device is only able to send media packets, and not receive them. This is not a problem, and the call will still succeed.
C. The payload for G.711ulaw must be 18: The payload for G.711ulaw is the type of media packet that is used. The payload must be set to 18 for G.711ulaw, but this is not a problem, and the call will still succeed.

NEW QUESTION 145
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 146
An engineer implements a new Cisco UCM based telephony system per these requirements.
•The local Ethernet bandwidth is sized based on the total bandwidth per call
•A G 736 codec is used.
•The bit rate is 64 kbps
•The codec sample interval is 10 ms
•The voice payload size is 160 bytes per 20 ms
What should the size of the Ethernet bandwidth be per call?

A. 31.2 kbps
B. 38.4 kbps
C. 55.2 kbps
D. 87.2 kbps

Answer: D

NEW QUESTION 147
An administrator configures the voicemail feature in a Cisco collaboration deployment. The user mailboxes must be configured when the Cisco Unity Connection server is configured. Which action accomplishes this task?

A. Configure a SIP integration with Cisco UCM to sync users.
B. Configure an SCCP integration with Cisco UCM.
C. Configure an AXL server to access the Cisco UCM users.
D. Configure an active directory to sync the users who will have a voicemail box.

Answer: C

NEW QUESTION 149
Where in Cisco UCM are codec negotiations configured for endpoints?

A. under device profiles in device settings
B. in in-service parameters
C. under regions using preference lists
D. in enterprise parameters

Answer: C

NEW QUESTION 152
An administrator is in the process of moving Cisco Unity Connection mailboxes between mailbox stores. The administrator notices that some mailboxes have active Message Waiting Indicators. What happens to these mailboxes when they are moved?

A. The move will fail if MWI status is active.
B. The MWI status is retained after a mailbox is moved from one store to another.
C. If the source and target mailbox store are not disabled, MWI status is not retained.
D. Moving the mailboxes from one store to another fails if MWI is turned on.

Answer: B

NEW QUESTION 157
A Cisco UCM administrator sets up new route patterns to support phones in tour different locations, all with local gateways. The administrator wants to use the

same route pattern for all four locations. How must the system be configured to achieve this goal?

A. Use CSS alternate routing rules.
B. Use standard local route groups.
C. Add a CSS to each local gateway.
D. Use transforms in the route groups.

Answer: B

NEW QUESTION 161
Which two protocols are proxied over an Expressway-E/C pair when a Mobile and Remote Access login including phone services is performed? (Choose two.)

A. HTTPS
B. H.323
C. SIP
D. SCCP
E. SRTP

Answer: AC

NEW QUESTION 165
Refer to the exhibit.

```
05:50:14.102: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=21653 ai=127
05:50:14.134: ISDN BR0/1/1 Q921: User RX <- SABMEp sapi=0 tei=0
05:50:14.150: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=19004 ai=127
05:50:14.165: ISDN BR0/1/1 Q921 User RX <- SABMEp sapi=0 tei=0
```

A customer submits this debug output, captured on a Cisco IOS router. Assuming that an MGCP gateway is configured with an ISDN BRI interface, which BRI changes resolve the issue?

A.
```
interface BRI0/1/0
 no ip address
 isdn switch-type basic-net3
 isdn point-to-multipoint-setup
 isdn incoming-voice voice
 isdn send-alerting
 isdn static-tei 0
```

B.
```
interface BRI0/1/1
 no ip address
 isdn switch-type basic-net3
 isdn point-to-multipoint-setup
 isdn incoming-voice voice
 isdn send-alerting
 isdn static-tei 0
```

C.
```
interface BRI0/1/1
 no ip address
 isdn switch-type basic-net3
 isdn point-to-point-setup
 isdn incoming-voice voice
 isdn send-alerting
 isdn static-tei 0
```

D.
```
interface BRI0/1/1
 no ip address
 isdn switch-type basic-net3
 isdn incoming-voice voice
 isdn send-alerting
 isdn static-tei 0
```

Answer: C

NEW QUESTION 170
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 174
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 175
The security department will audit an IT department to ensure that the proper guidelines are being followed. The reports of the call detail records show unauthorized access to PSTN. Which two actions should an administrator check to prevent the unauthorized use of the telephony system? (Choose two.)

A. Ensure that ad hoc conference calls are dropped if an external user is add.
B. Call forward settings (ALL/Busy/No Answer) are restricted to internal extensions in the network
C. Add an additional firewall between the Cisco UCM server and the Expressway Core server.
D. For extension mobility, logged-out CSS is restricted to internal extensions and emergencies.
E. Forced authorization code is used to recognize a dialing extension and authorize an international call.

Answer: BE

NEW QUESTION 179
A customer asked to integrate Unity Connection with Cisco UCM using SIP protocol. Which two features must be enabled on SIP security profiles? (Choose two.)

A. accept presence subscription
B. allow changing header
C. accept unsolicited notification
D. enable application-level authorization
E. accept replaces header

Answer: CE

NEW QUESTION 182
Which actions required for a firewall configuration on a Mobile and Remote Access through Cisco Expressway deployment?

A. The traversal zone on Expressway-c points to Expressway-e through the peer address field on the traversal zone, which specifies the Expressway-e server addres
B. For dual NIC deployments, set the Expressway-e address using an FQDN that resolves the IP address of the internal interface
C. The external firewall must allow these inbound connections to Expressway: SIP: TCP 5061; HTTPS: TCP 8443; XMPP TCP 5222; media: UDP 36002 to 59999
D. Do not use a shared address for Expressway-e and Expressway-c, as the firewallcannot distinguish between the
E. If static NAT for IP addressing on Expressway-e is used, ensure that any NAT operation on expressway-c does not resolve the same traffic IP addres
F. Shared NAT IS not supported
G. The internal firewall must allow these inbound and outbound connections between expressway - c and Expressway-e :sip;HTTPS(tunneled over SSH between C and E.TCP 2222: TCP 7001: Traversal Media: UDP 2776 to 2777(or 36000 to 36011 for large VM/appliance);XMPP:TCP 7400

Answer: B

NEW QUESTION 183
Which Webex Calling dial plan settings restrict a user from placing a particular outbound call type?

A. Block
B. Transfer to Number
C. Reject
D. Restrict

Answer: D

Explanation:
The Restrict setting in the Webex Calling dial plan prevents users from placing certain types of outbound calls. For example, you can use the Restrict setting to prevent users from making international calls or calls to premium-rate numbers.
The Block setting in the Webex Calling dial plan prevents users from placing any outbound calls. The Transfer to Number setting in the Webex Calling dial plan transfers all outbound calls to a specified number. The Reject setting in the Webex Calling dial plan rejects all outbound calls.
Here is a table summarizing the different dial plan settings and their effects:
Dial Plan Setting Effect
Block
Prevents users from placing any outbound calls. Transfer to Number
Transfers all outbound calls to a specified number. Reject
Rejects all outbound calls. Restrict
Prevents users from placing certain types of outbound calls.

NEW QUESTION 188
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 192
Refer to the exhibit.

```
hostname GATEWAY
ccm-manager config
ccm-manager config server 192.168.1.100
ccm-manager mgcp

mgcp call-agent CCMSub1.domain.com 2427 service-type mgcp version 0.1
```

An engineer verifies the configured of an MGCP gateway. The commands are already configured. Which command is necessary to enable MGCP?

A. Device(config)# mgcp enable
B. Device(config)# ccm-manager enable
C. Device (config) # com-manager active
D. Device (config)# mgcp

Answer: D

NEW QUESTION 193
How many minutes does it take for automatic fallback to occur in a Presence Redundancy Group if the primary node lost a critical service?

A. 5 min
B. 10 min
C. 30 min
D. 60 min

Answer: C

NEW QUESTION 195
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 200
What are two access management mechanisms in Cisco Webex Control Hub? (Choose two.)

A. multifactor authentication
B. Active Directory synchronization
C. attribute-based access control
D. single sign-on with Google
E. Client ID/Client Secret

Answer: AB

Explanation:
The correct answers are A and B.
The two access management mechanisms in Cisco Webex Control Hub are multifactor authentication and Active Directory synchronization.
Multifactor authentication is a security measure that requires users to provide two or more pieces of evidence to verify their identity. This can include something they know, such as a password, and something they have, such as a security token.
Active Directory synchronization is a process that allows Cisco Webex Control Hub to automatically synchronize user accounts from an Active Directory domain. This can simplify user management and provide users with single sign-on access to Cisco Webex Control Hub and other applications.

NEW QUESTION 201
Refer to the exhibit.

```
        Bearer Capability i = 0x8090A2
                Standard = CCITT
                Transfer Capability = Speech
                Transfer Mode = Circuit
                Transfer Rate = 64 kbit/s
        Channel ID i = 0xA98388
                Exclusive, Channel 8
        Calling Party Number i = 0x2181, '5125551212'
                Plan: ISDN, Type:National
        Called Party Number i = 0xA1, '2145551212'
                Plan:ISDN, Type:National
Mar  1 02:35:37: ISDN Se0/1/1:23 Q931: RX <- CALL_PROC pd = 8  callref = 0x809A
        Channel ID i = 0xA98388
                Exclusive, Channel 8

interface Serial0/1/1:23
 description PRI Circuit to R1
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn protocol-emulate network
 isdn incoming-voice voice
 no cdp enable
```

An engineer is troubleshooting why PSTN phones are not receiving the caller's name when called from a remote Cisco UCM site. An ISDN PRI connection is being used to reach the PSTN What must the administrator select to resolve the issue?

A. isdn supp-service name calling
B. isdn outgoing display-ie
C. isdn enable did
D. isdn send display le

Answer: B

NEW QUESTION 205
A collaboration engineer configures Global Dial Plan Replication for multiple Cisco UCM clusters. The local cluster acts as the hub cluster, and the remaining clusters act as spoke clusters Which service must the engineer configure on the local cluster'

A. Intercluster Lookup Service
B. Location Conveyance on intercluster SIP trunks
C. Intra-Cluster Communication Signaling
D. Mobility Cross Cluster

Answer: A

NEW QUESTION 206
Which characterstic of distributed class- based weighted fair queueing addresses jitter prevention?

A. It provides additional granularity by allowing a user to create classes
B. It minimizes jitter by implementing a priority queue for voice traffic
C. It uses a priority queue for voice traffic to avoid jitter.
D. It provides additional granularity by allowing a user to define custom class

Answer: B

NEW QUESTION 210
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 211
What are two key features of the Expressway series? (Choose two.)

A. VPN connection toward the internal UC resources
B. SIP header modification
C. B2B calls
D. device registration over the Internet
E. IP to PSTN call connectivity

Answer: CD

NEW QUESTION 215

Drag and drop the SNMPv3 message types from the left onto the corresponding definitions on the right.

TRAP	messages used to modify a value of an object variable
SET	unreliable messages that alert the SNMP manager to a condition on the network
GET	reliable messages that alert the SNMP manager to a condition on the network
INFORM	messages used to retrieve an object instance

A. Mastered
B. Not Mastered

Answer: A

Explanation:
Table Description automatically generated

NEW QUESTION 218
What is a characteristic of video traffic that governs QoS requirements for video?

A. Video is typically constant bit rate.
B. Voice and video are the same, so they have the same QoS requirements.
C. Voice and video traffic are different, but they have the same QoS requirements.
D. Video is typically variable bit rate.

Answer: D

NEW QUESTION 223
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 226
A customer reports that the Cisco UCM toll-fraud prevention does not work correctly, and the customer is receiving charges for unapproved international calls as a result. Which two configuration changes resolve the issues? (Choose two.)

A. Mark patterns as off-net or on-net.
B. Modify the Block OffNet to OffNet Transfer service parameter.
C. Disable call forwarding on the phone.
D. Use Cisco Unified Border Element to debug the calls.
E. Make the calls route through a firewall.

Answer: AB

NEW QUESTION 227
An engineer encounters third-party devices that do not support Cisco Discovery Protocol. What must be configured on the network to allow device discovery?

A. LLDP
B. TFTP
C. LACP
D. SNMP

Answer: A

Explanation:
LLDP (Link Layer Discovery Protocol) is a vendor-neutral network discovery protocol that is used to discover the topology of a network. LLDP is similar to CDP (Cisco Discovery Protocol), but it is not proprietary to Cisco. LLDP is supported by a wide range of network devices, including switches, routers, and firewalls.
To configure LLDP on a network, you must enable LLDP on the devices that you want to discover. You can then use a network management tool, such as Cisco Network Assistant, to view the topology of the network.

The other options are incorrect. TFTP (Trivial File Transfer Protocol) is a network protocol that is used to transfer files between devices. LACP (Link Aggregation Control Protocol) is a network protocol that is used to aggregate multiple network links into a single logical link. SNMP (Simple Network Management Protocol) is a network protocol that is used to manage network devices.

NEW QUESTION 232
What is a capability of Cisco Expressway?

A. It functions as an analog telephony adapter.
B. It has remote endpoint enrollment with Certificate Authority Proxy Function.
C. It gives directory access for remote users via Cisco Directory Integration.
D. It provides access to on-premises Cisco Unified Communications infrastructure for remote endpoints.

Answer: D

NEW QUESTION 233
An end user at a remote site is trying to initiate an Ad Hoc conference call to an end user at the main site. The conference bridge is configured to support G.711. The remote user's phone only supports G.729. The remote end user receives an error message on the phone: "Cannot Complete Conference Call." What is the cause of the issue?

A. The remote phone does not have the conference feature assigned.
B. A software conference bridge is not assigned.
C. A Media Termination Point is missing.
D. The transcoder resource is missing.

Answer: D

NEW QUESTION 237
When a call Is delivered to a gateway, the calling and called party number must be adapted to the PSTN service requirements of the trunk group. If a call is destined locally, the + sign and the explicit country code must be replaced with a national prefix. For the same city or region, the local area code must be replaced by a local prefix as applicable. Assuming that a Cisco UCM has a SIP trunk to a New York gateway (area code 917). which two combinations of solutions localize the calling and called party for a New York phone user? (Choose two.)

A. Configure two calling party transformation patterns:
\+1917.XXXXXXX, strip pre-dot, numbering type: subscriber
\+1.!, strip pre-dot, numbering type: national

B. Configure the gateway to translate called numbers and apply it to the dial peer. Combine it with a translation profile for calling nu

```
!
voice translation-rule 1
rule 1 /^1917/ //
rule 2 /^[+]1917/ //
!
voice translation-profile strip+1
translate called 1
!
```

C. Configure the gateway to translate the calling number and apply it to the dial peer. Combine it with a translation profile for called nu

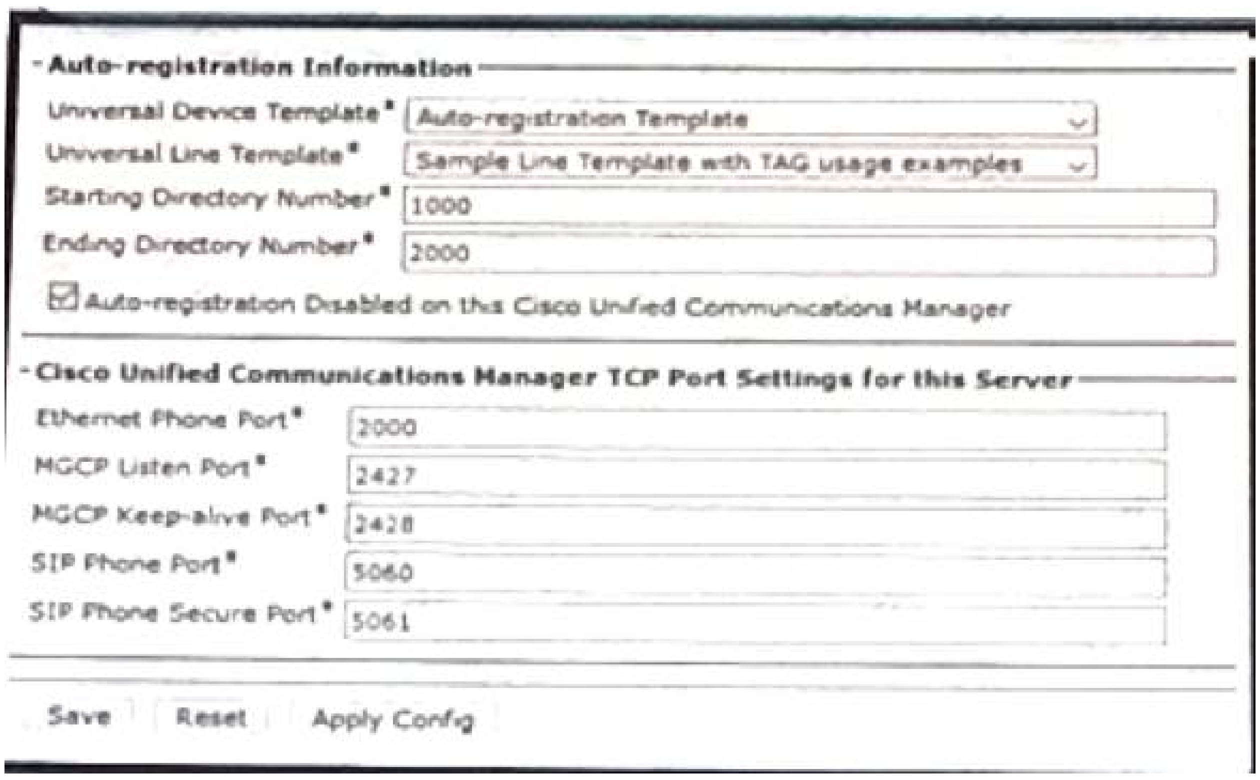
```
!
voice translation-rule 1
rule 1 /^1917/ //
rule 2 /^[+]1917/ //
!
voice translation-profile strip+1
translate calling 1
!
```

D. Configure two called party transformation patterns:
\+1917.XXXXXXX, strip pre-dot, numbering type: subscriber
\+1.!, strip pre-dot, numbering type: national

E. Configure two calling party transformation patterns:
\+1917.CCCCCC, strip pre-dot, numbering type: subscriber
\+!, strip pre-dot, numbering type: national

Answer: BC

NEW QUESTION 238
Refer to the exhibit.

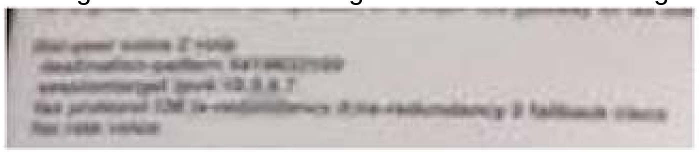


Which action must an engineer take to implement self-provisioning on a primary communications manager server?

A. Select a different Universal Line Template.
B. Change the SIP Phone Secure Port.
C. Uncheck the auto-registration Disabled checkbox.
D. Select a different Universal Device Template.

Answer: C

NEW QUESTION 243
An engineer builds the configuration on a Cisco IOS gateway for the dial-peers:



Which command is required to complete the configuration?

A. Codec g726r32
B. Codec g729cr81
C. Codec g723ar63
D. Codec g711ulaw

Answer: D

NEW QUESTION 248
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 253
An administrator needs to help a remote employee make a free call to an international destination. The administrator calls the employee, then conferences in the international party. The administrator drops the call, and the employee and the international party continue their conversation. Which action prevents this type of toll fraud in the Cisco UCM?

A. Set service parameter 'Advanced Ad Hoc Conference" to FALSE.
B. Set service parameter "Drop Ad Hoc Conference" to "When Conference Controller leaves."
C. Set service parameter "Advanced Ad Hoc Conference" to 2.
D. Set service parameter "Drop Ad Hoc Conference" to "Do not allow outside parties."

Answer: B

NEW QUESTION 257
An administrator is developing an 8-class QoS baseline model. The CS3 standards-based marking recommendation is used for which type of class?

A. Scavenger
B. best effort
C. voice

D. call signaling

Answer: A

NEW QUESTION 262
Which IP Precedence value is used to classify a call signalling packet?

A. 6
B. 5
C. 4
D. 3

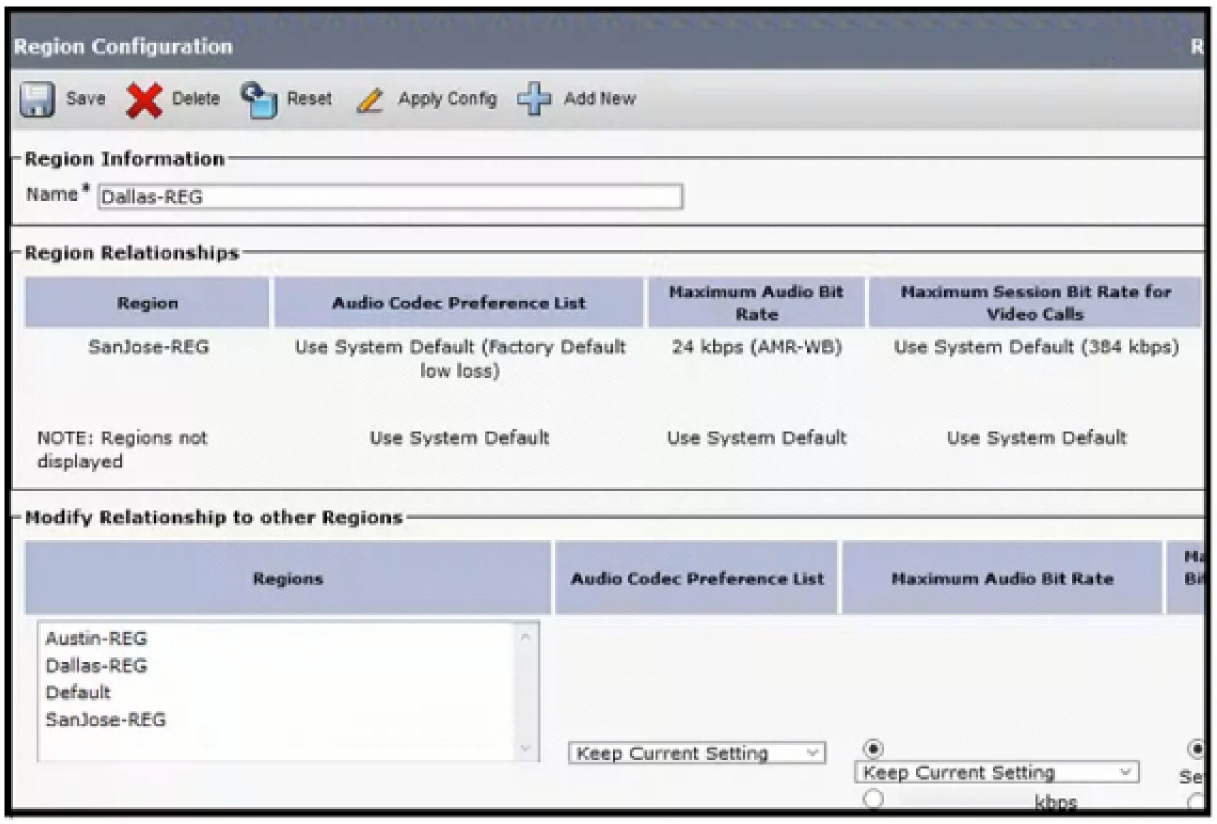
Answer: D

NEW QUESTION 266
A company has an excessive number of call transfers to local and long-distance PSTN from Cisco Unity Connection voicemail. Which action in the Cisco Unity Connection restriction table resolves this issue?

A. Block PSTN patterns on Default Transfe
B. Default Outdia
C. and Default System Transfer.
D. Implement password complexity on voicemail boxes to prevent accounts from being compromised.
E. Create a custom restriction table ??????????? and block it.
F. Create a custom restriction table **********and block it.

Answer: A

NEW QUESTION 270
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 272
An engineer must enable onboarding of on-premises devices by using activation to a Cisco UCM server The engineer activated the CISCO Device Activation Service and set the default registration method to use the codes. Which action completes the configuration?

A. Set Enable Activation Code enterprise parameter to True
B. Manually provision new phones that have an activation code requirement
C. Create a Bulk Administration Tool provisioning template.
D. Generate 16-digit codes by using the Bilk Administration Tool

Answer: A

Explanation:
The engineer must set the Enable Activation Code enterprise parameter to True. This will enable the use of activation codes for onboarding on-premises devices to a Cisco UCM server. The other options are not necessary to complete the configuration.
Here are the steps to complete the configuration:
- Log in to the Cisco Unified Communications Manager (CUCM) Administration interface.
- Go to System > Enterprise Parameters.
- Set the Enable Activation Code enterprise parameter to True.
- Click Save.

The activation code onboarding feature is now enabled. You can use it to onboard new phones to the CUCM server.

NEW QUESTION 277
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).

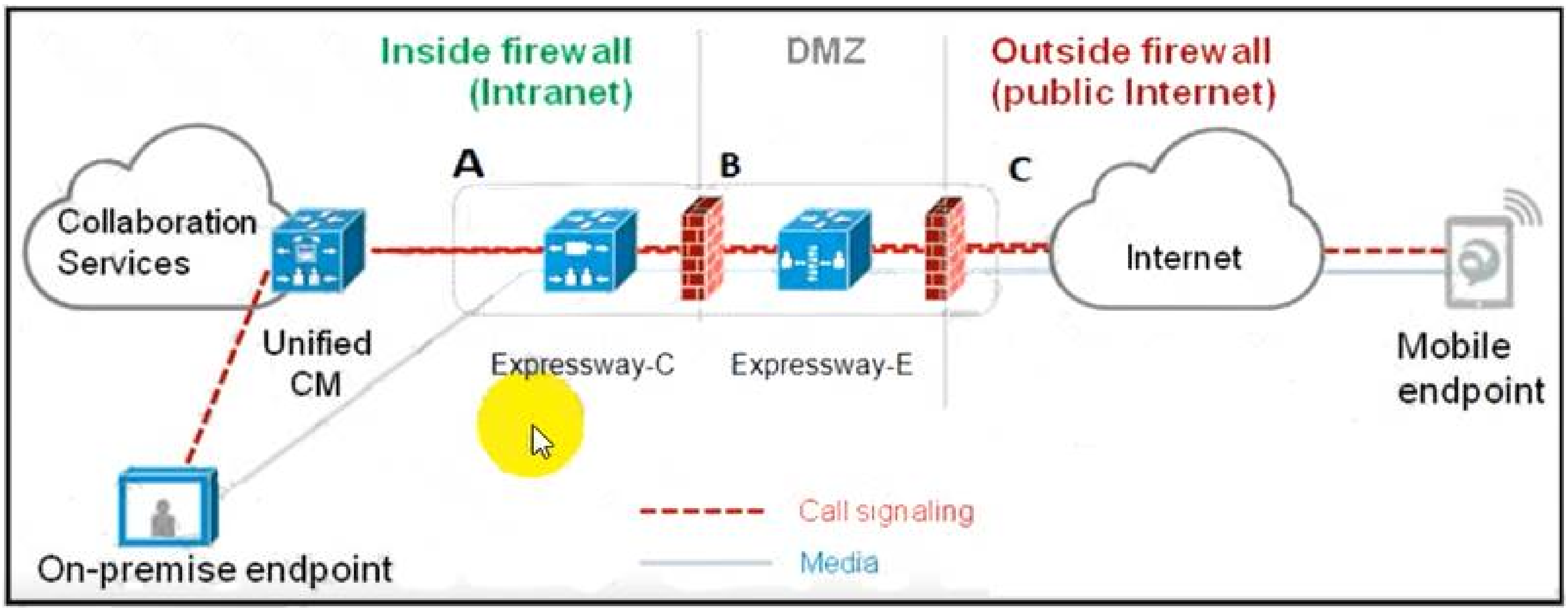
Answer: BD

NEW QUESTION 278
Cisco UCM delays routing of a call during digit analysis with an overlapping dial plan. How long is the default wait time?

A. 5 seconds
B. 10 seconds
C. 15 seconds
D. 20 seconds

Answer: C

NEW QUESTION 283
Refer to the exhibit.



When making a call to a Mobile and Remote Access client, what are the combinations of protocol on each of the different sections A-B-C?

A. IP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TLS (C)
B. SIP TCP/TLS (A) + SIP TCP/TLS (B) + SIP TCP/TLS (C)
C. SIP TLS (A) + SIP TLS (B) + SIP TLS (C)
D. SIP TCP/TLS (A) + SIP TLS (B) + SIP TLS (C)

Answer: D

NEW QUESTION 284
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 285
An administrator troubleshoots call flows and suspects that there are issues with the dial plan. Which tool enables a quick analysis of the dial plan and provides call flows of dialled digits?

A. Cisco Dial Plan Analyzer
B. Dial Plan Analyzer
C. Digit Analysis Analyzer
D. Dialed Number Analyzer

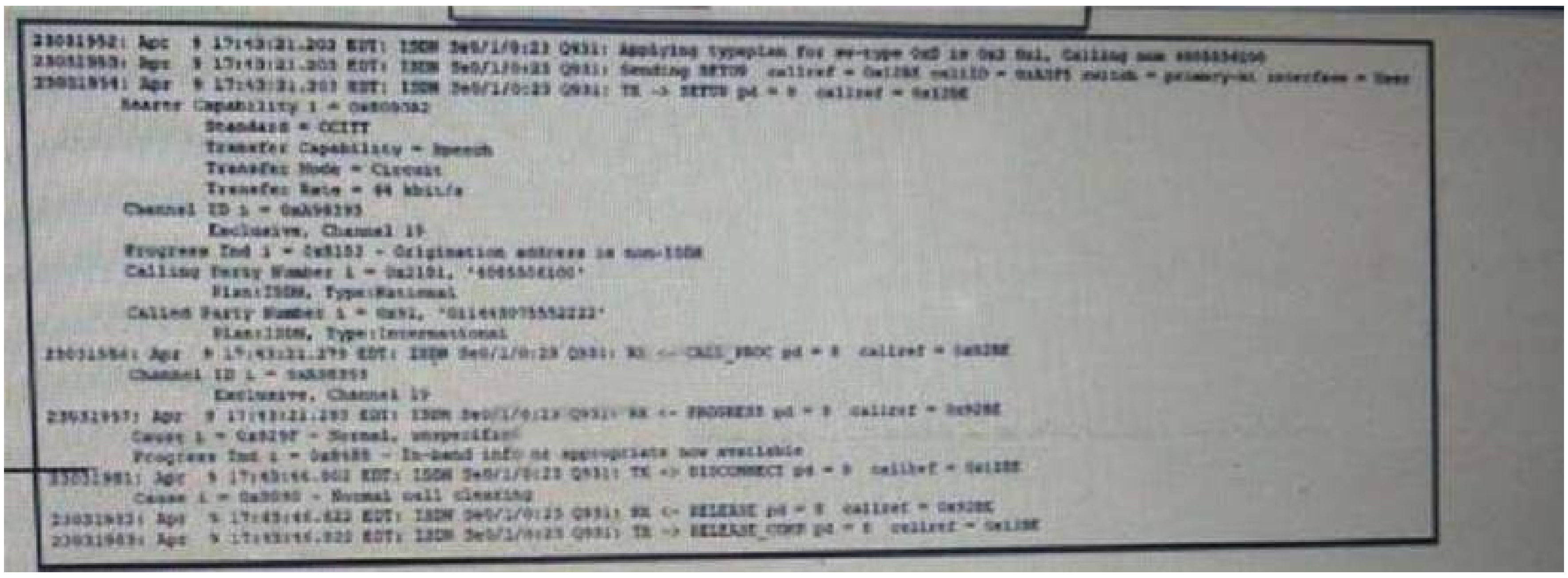
Answer: D

NEW QUESTION 288
An engineer is configuring a Cisco Unified Border Element to allow the video endpoints to negotiate without the Cisco Unified Border Element interfering in the process. What should the engineer configure on the Cisco Unified Border Element to support this process?

A. Configure path-thru content sdp on the voice service.
B. Configure a hardcoded codec on the dial peers.
C. Configure a transcoder for video protocols.
D. Configure codec transparent on the dial peers.

Answer: D

NEW QUESTION 290
Refer to the exhibit.



A call to an international number has failed. Which action corrects this problem?

A. Assign a transcoder to the MRGL of the gateway.
B. Strip the leading 011 from the called party number
C. Add the bearer-cap speech command to the voice port.
D. Add the isdn switch-type primart-dms100 command to the serial interface.

Answer: B

NEW QUESTION 292
When a remote office location is set up with limited bandwidth resources, which codec would allow the most voice calls with the limited bandwidth?

A. G.722
B. G.711
C. G.729
D. G.723

Answer: C

NEW QUESTION 294
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 296
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 299
Where is urgent priority enabled to bypass the T302 timer?

A. route partition
B. transformation pattern
C. directory number
D. CTI port

Answer: C

Explanation:
Urgent priority is enabled on the directory number configuration page. This allows the call to be routed at once to the fully qualified DN without any necessity to wait for inter-digit-timeout. If the Urgent Priority checkbox is disabled and you have overlap patterns configured, then CUCM waits for the user to dial further digits.
The other options are incorrect because:
- Route partitions are used to group route patterns and route lists.
- Transformation patterns are used to convert dialed digits into a different format.
- CTI ports are used to connect Cisco Unified Communications Manager to third-party applications. https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callman

NEW QUESTION 303
The IP phones al a customer site do not pick an IP address from the DHCP An engineer must temporarily disable LLDP on all ports of the switch to leave only CDP. Which two commands accomplish this task? (Choose two.)

A. Switch# copy running-config startup-config
B. Switch(config)# no lldp run
C. Switch# configure terminal
D. Switch(conlicj)# interface GigabitEthernet1/0/1
E. Switch(config># no lldp transmit

Answer: BC

NEW QUESTION 306

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 308
Which DSCP marking is represented as 101110 in an IP header?

A. EF
B. CS3
C. AF41
D. AF31

Answer: A

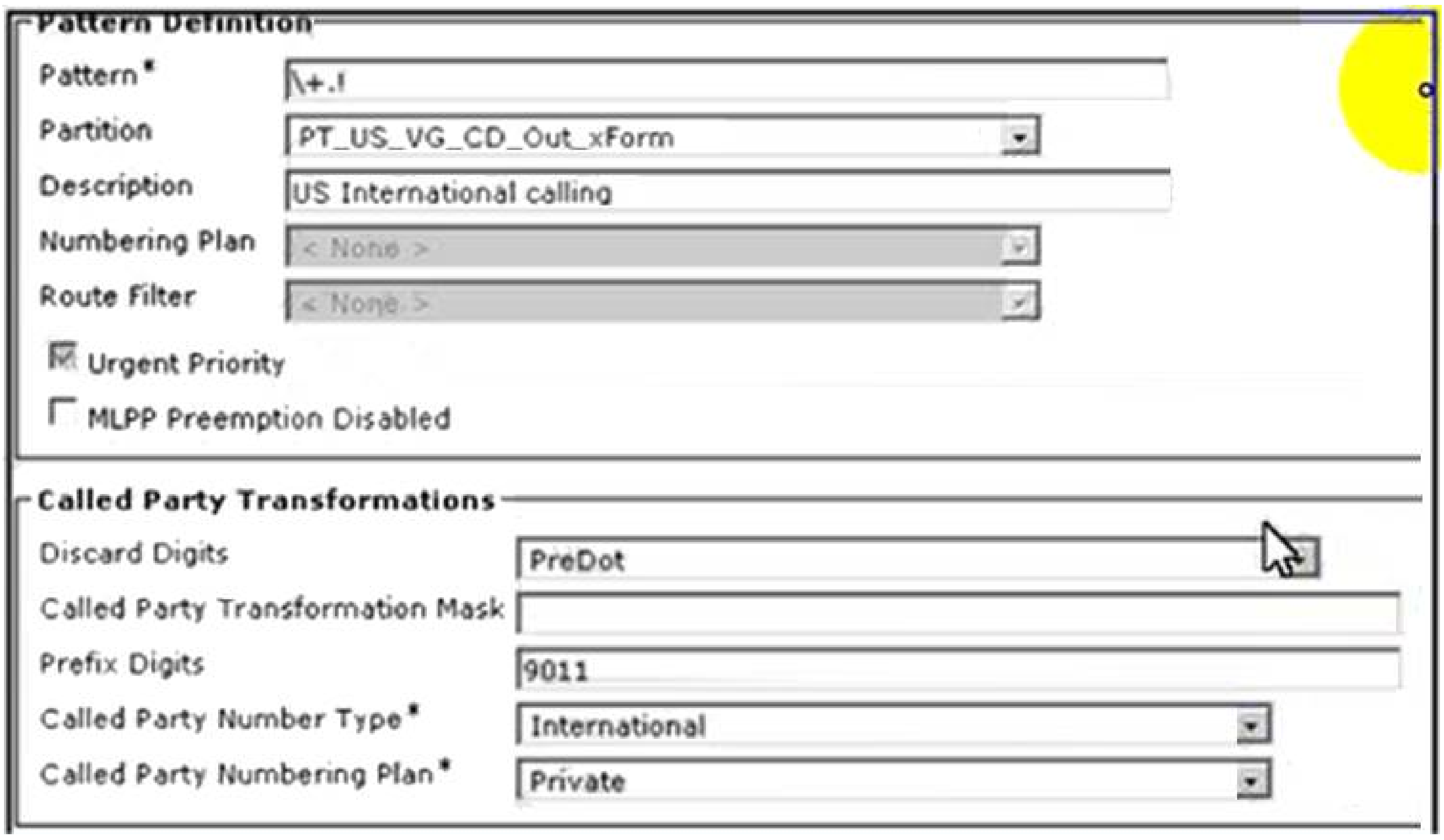
NEW QUESTION 310
Which value should be changed when each Cisco UCM node does not allow for more than 5000 phones to be registered?

A. Maximum Number of Registered and Unregistered Devices service parameter on each node
B. Minimum Number of Phones service parameter on each node
C. Maximum Number of Registered Devices service parameter on each node
D. Maximum Number of Phones service parameter on the Publisher

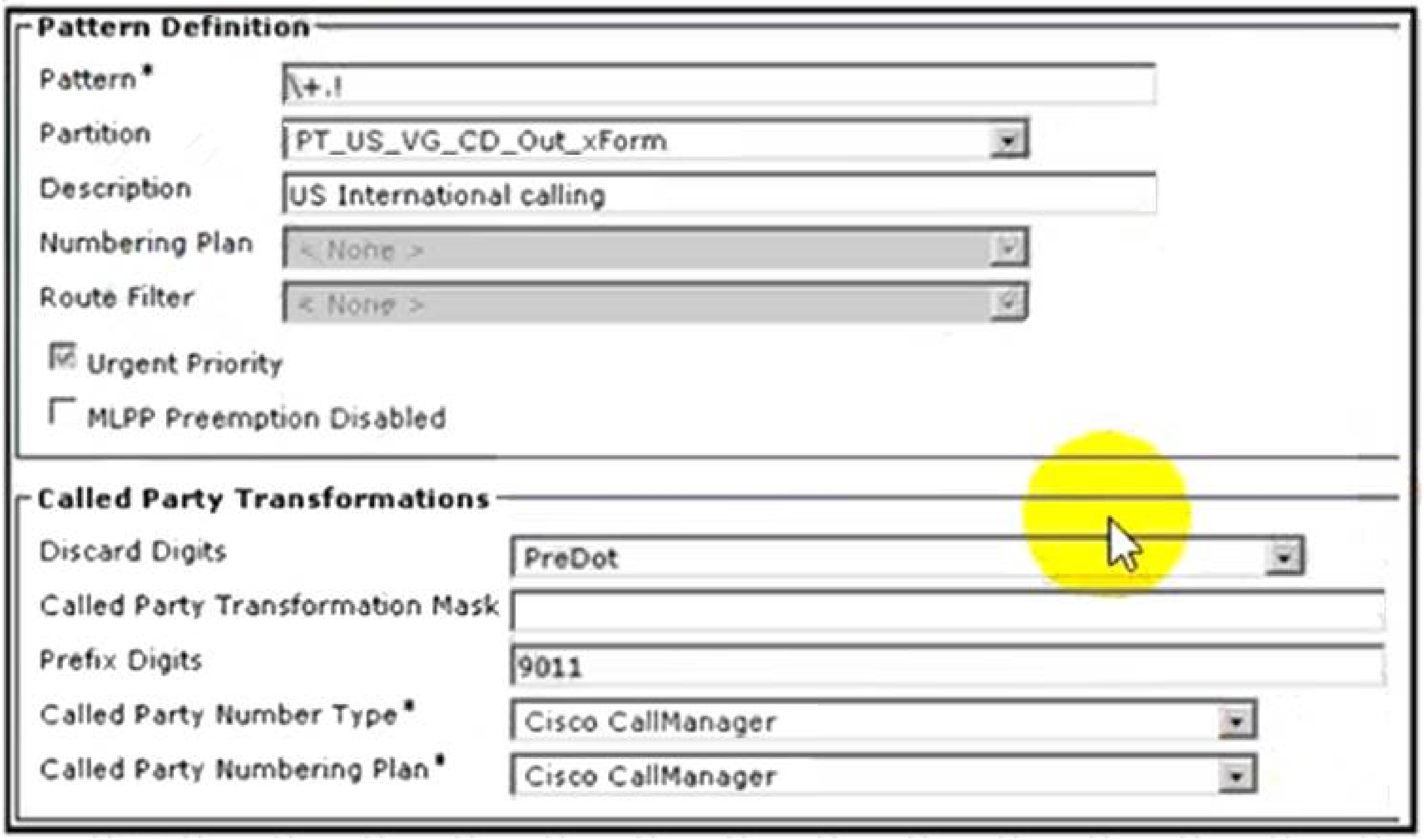
Answer: C

NEW QUESTION 314
Given this H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
voice translation-rule 40
 rule 1 /3...$/ /408555&/
!
voice translation-profile INT
 translate calling 40
!
dial-peer voice 9011 pots
 translation-profile outgoing INT
 destination-pattern 9011T
 port 0/1/0:23
```

A. 

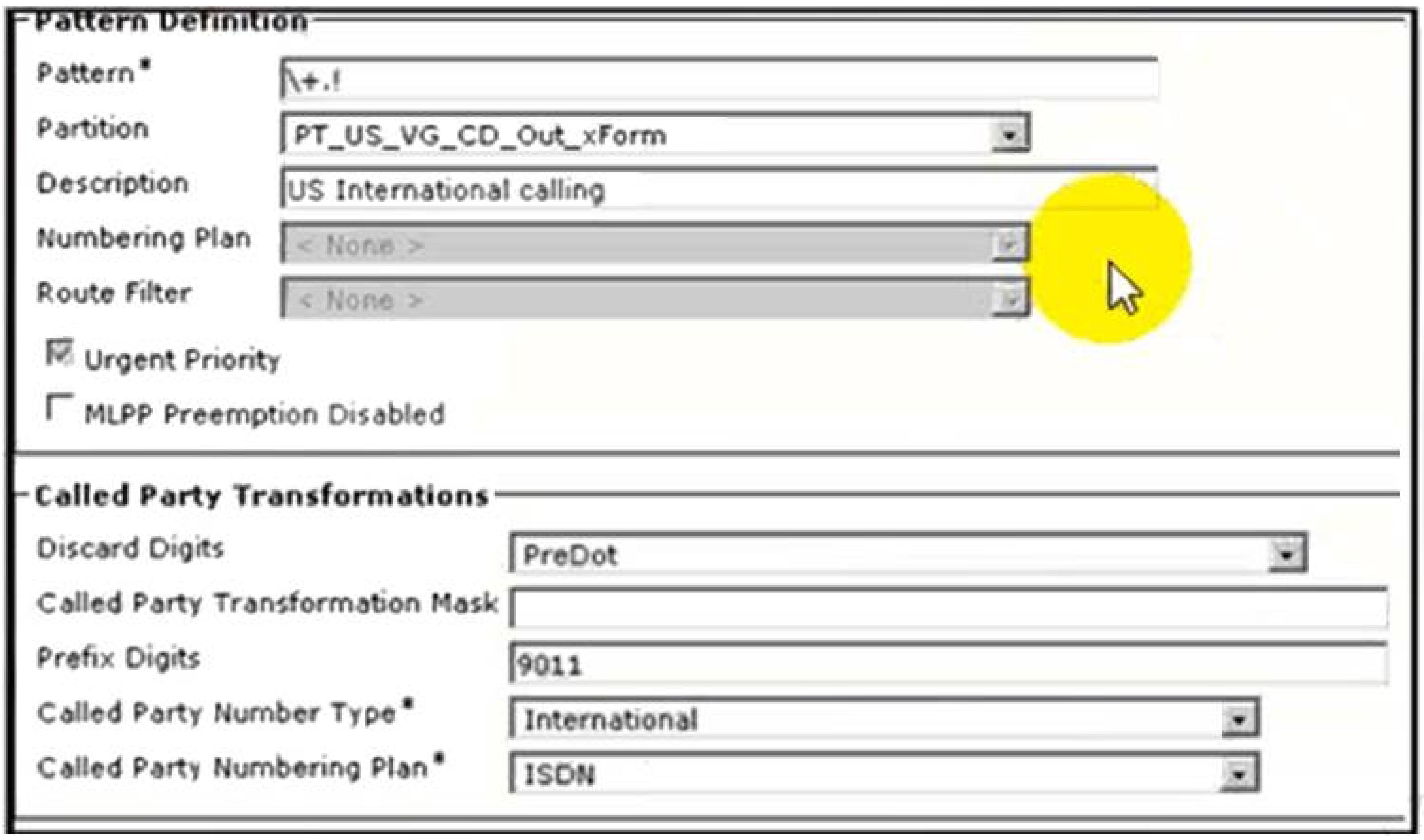
B.



Pattern Definition
Pattern* \+.!
Partition PT_US_VG_CD_Out_xForm
Description US International calling
Numbering Plan < None >
Route Filter < None >
☑ Urgent Priority
☐ MLPP Preemption Disabled

Called Party Transformations
Discard Digits PreDot
Called Party Transformation Mask
Prefix Digits 9011
Called Party Number Type* Cisco CallManager
Called Party Numbering Plan* Cisco CallManager

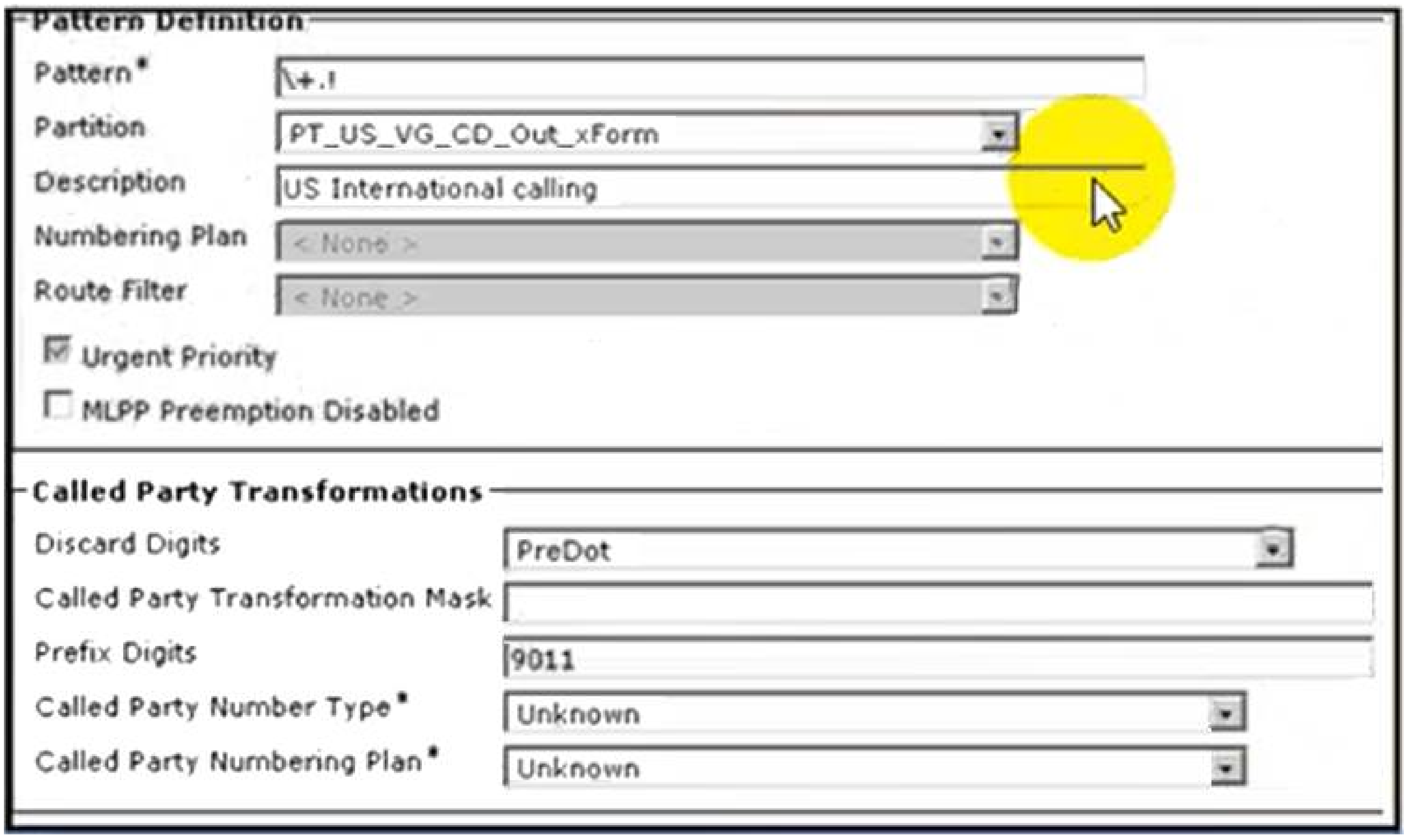
C.



Pattern Definition
Pattern* \+.!
Partition PT_US_VG_CD_Out_xForm
Description US International calling
Numbering Plan < None >
Route Filter < None >
☑ Urgent Priority
☐ MLPP Preemption Disabled

Called Party Transformations
Discard Digits PreDot
Called Party Transformation Mask
Prefix Digits 9011
Called Party Number Type* International
Called Party Numbering Plan* ISDN

D.



Pattern Definition
Pattern* \+.!
Partition PT_US_VG_CD_Out_xForm
Description US International calling
Numbering Plan < None >
Route Filter < None >
☑ Urgent Priority
☐ MLPP Preemption Disabled

Called Party Transformations
Discard Digits PreDot
Called Party Transformation Mask
Prefix Digits 9011
Called Party Number Type* Unknown
Called Party Numbering Plan* Unknown

Answer: C

NEW QUESTION 319
An administrator installed a Cisco Unified IP 8831 Conference Phone that is failing to register. Which two actions should be taken to troubleshoot the problem? (Choose two.)

A. Verify that the switch port of the phone is enabled.
B. Verify that the RJ-11 cable is plugged into the PC port.
C. Disable HSRP on the access layer switch.
D. Check the RJ-65 cable.
E. Verify that the phone's network can access the option 150 server.

Answer: AE

NEW QUESTION 324
Refer to the exhibit.

```
Voice class codec 1
codec preference 1 g711alaw
codec preference 2 g711ulaw
codec preference 3 g729r8

dial-peer voice 13 voip
description incoming dialpeer from ITSP
incoming called-number .
voice-class codec 1

dial-peer voice 19 voip
description outgoing dialppeer to CUCM
destination-pattern .T
session protocol sipv2
session-target ipv4:3.3.3.3
voice-class codec 1

Incoming SDP from ITSP

v=0
o=sip:test@2.2.2.2 1 16 IN IP4 2.2.2.2
s=sip:test@2.2.2.2
c=IN IP4 2.2.2.2
t=0 0
m=audio 5000 RTP/AVP 18 0
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
```

Which outgoing m-line SDP is sent to Cisco UCM after matching the appropriate dial peers via Cisco Unified Border Element?

A.
```
m=audio 16550 RTP/AVP 8 0 18
a=rtpmap:0 PCMU/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:18 G729/8000/1
```

B.
```
m=audio 16550 RTP/AVP 18 0
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
```

C.
```
m=audio 16550 RTP/AVP 18 0
a=rtpmap:8 PCMA/8000/1
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
```

D.
```
m=audio 16550 RTP/AVP 0 8 18
a=rtpmap:0 PCMU/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:18 G729/8000/1
```

Answer: B

NEW QUESTION 326
What is a possible cause of the PRI issue?

```
ISDN Serial1:23 interface
        dsl 1, interface ISDN Switchtype =
primary-5ess
    Layer 1 Status:
        ACTIVE
    Layer 2 Status:
        TEI = 0, Ces = 1, SAPI = 0, State =
TEI_ASSIGNED
    Layer 3 Status:
        0 Active Layer 3 Call(s)
    Activated dsl 1 CCBs = 0
    The Free Channel Mask:  0x807FFFFF
    Total Allocated ISDN CCBs = 5
```

A. The cable is unplugged.
B. The controller shut down.
C. The clock source is incorrect.
D. The framing is configured incorrectly.

Answer: D

NEW QUESTION 328
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 329
Refer to the Exhibit.

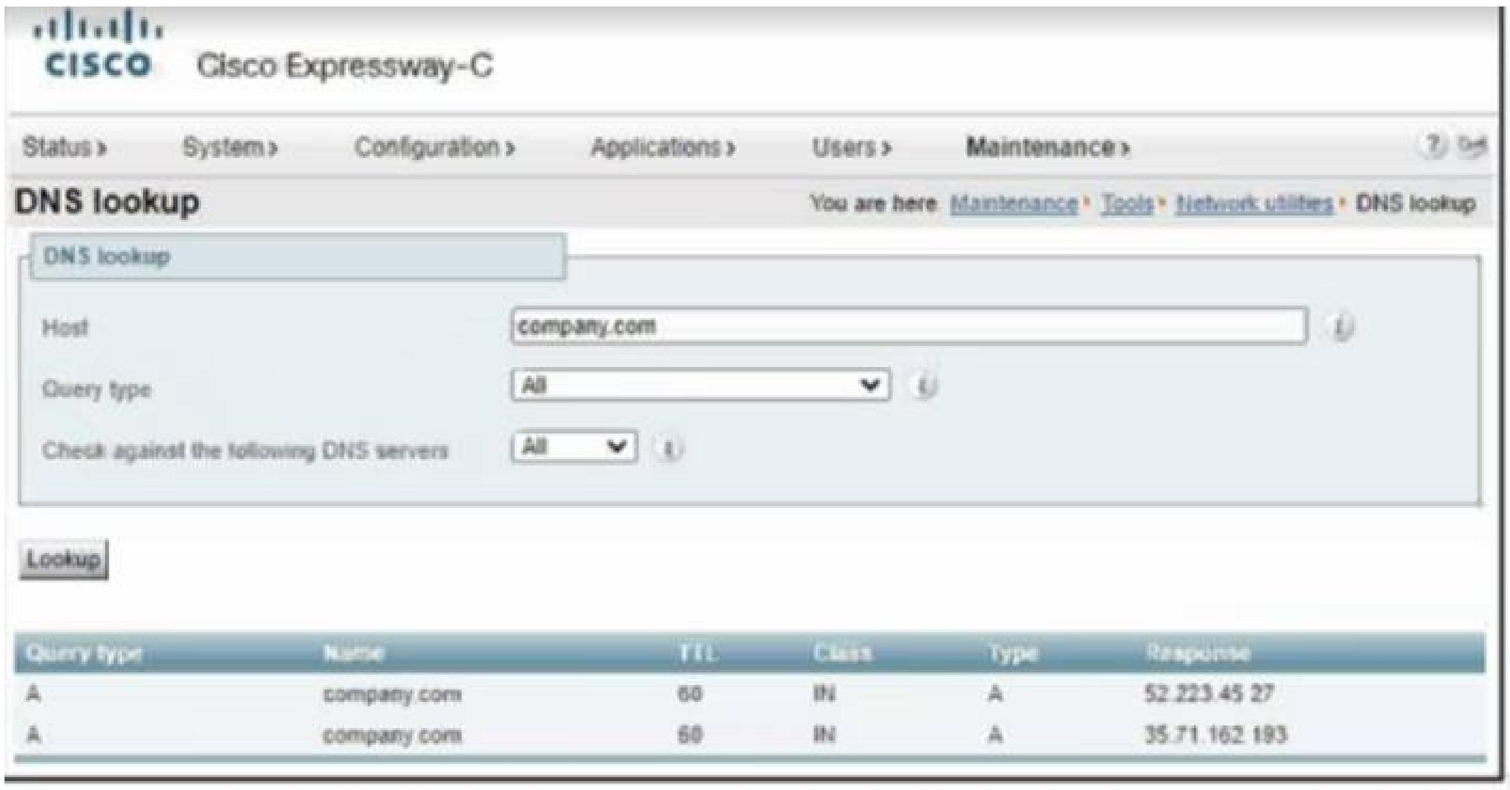
```
dspfarm profile 1 mtp
 codec g711ulaw
 maximum sessions software 50
 associate application SCCP
```

Which command is required to allow this media resource to handle Video Media streams?

A. maximum sessions hardware 50
B. video codec h264
C. codec pass-through
D. associate application Cisco unified border element

Answer: C

NEW QUESTION 331
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 334
Refer to the exhibit.

```
Endpoint A:
m=audio 21796 RTP/AVP 108 9 104 105 101
 b=TIAS:64000
 a=extmap:14 http://protocols.cisco.com/timestamp#100us
 a=rtpmap:108 MP4A-LATM/90000
 a=fmtp:108 bitrate=64000;profile-level-id=24;object=23
 a=rtpmap:9 G722/8000
 a=rtpmap:104 G7221/16000
 a=fmtp:104 bitrate=32000
 a=rtpmap:105 G7221/16000
 a=fmtp:105 bitrate=24000
 a=rtpmap:101 telephone-event/8000
 a=fmtp:101 0-15
 a=trafficclass:conversational.audio.immersive.aq:admitted

Endpoint B:
m=audio 21796 RTP/AVP 105 0 8 18 101
 b=TIAS:64000
 a=extmap:14 http://protocols.cisco.com/timestamp#100us
 a=rtpmap:105 G7221/16000
 a=fmtp:105 bitrate=24000
 a=rtpmap:0 PCMU/8000
 a=rtpmap:8 PCMA/8000
 a=rtpmap:18 G729/8000
 a=fmtp:18 annexb=no
 a=rtpmap:101 telephone-event/8000
```

```
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=trafficclass:conversational.audio.immersive.aq:admitted
```

Endpoint A calls endpoint B. What is the only audio codec that is used for the call?

A. G722/8000
B. Telephone-event/8000
C. G7221/16000
D. PCMA/8000

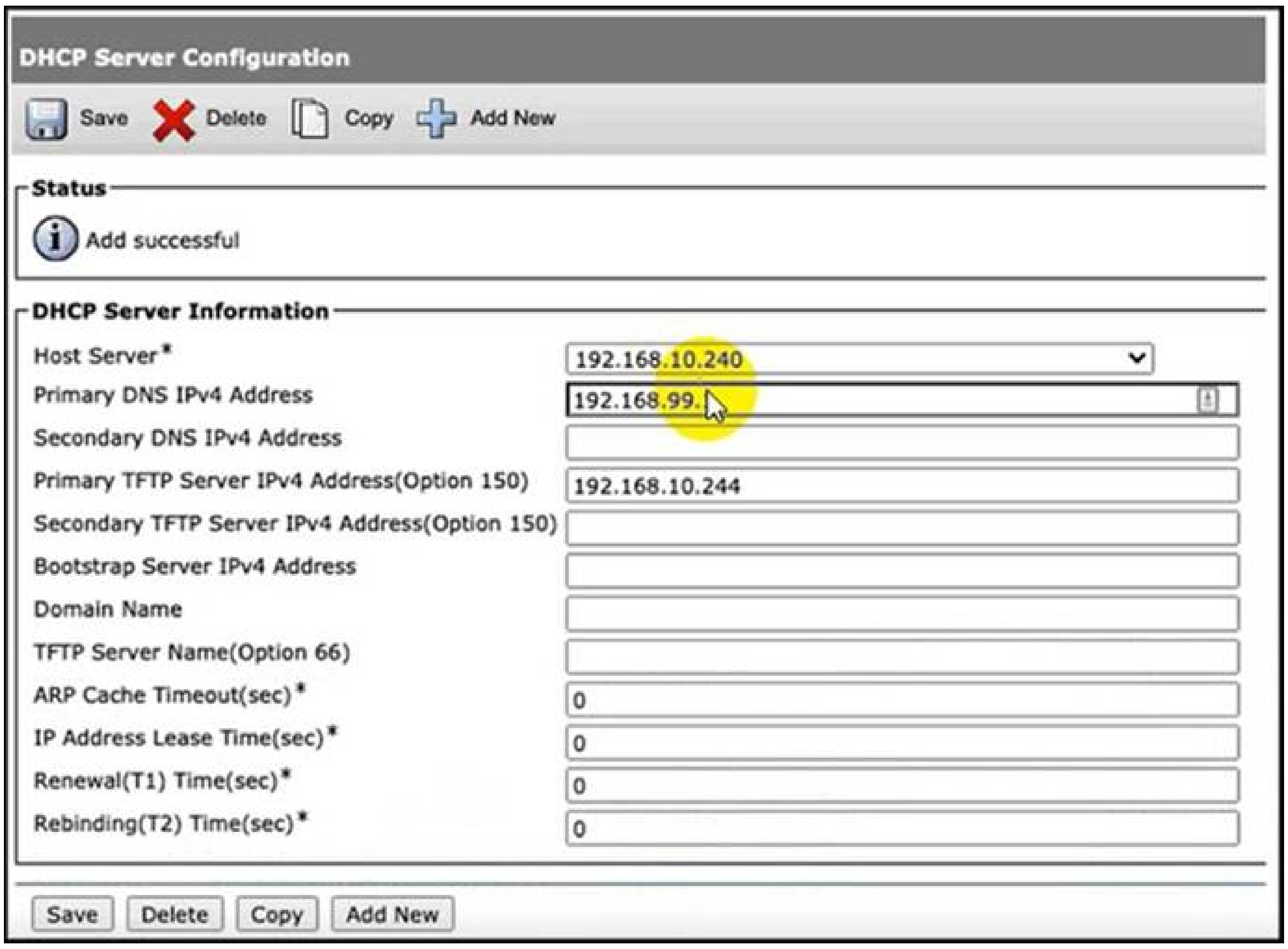
Answer: C

NEW QUESTION 337
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 341
Refer to the exhibit.



A collaboration engineer configures Cisco UCM to act as a DHCP server. What must be done next to configure the DHCP server?

A. Restart the Cisco DHCP Monitor Service under Cisco Unified Serviceability
B. Add the new DHCP server to the primary DNS server
C. Restart the TFTP service under Cisco Unified Serviceability.
D. Add a DHCP subnet to the DHCP server under Cisco UCM Administration.

Answer: D

NEW QUESTION 346
An engineer is deploying Webex app on Microsoft Windows computers. The engineer wants to ensure that the end users do not receive pop-IQ dialogues %'hen they start the application 'Much two actions ensure the end users are not promted to accept the end-user license (Choose two)

A. Set the DELETEUSERDATA=r installation argument
B. Set the "HKEY_LOCAL_MACHINE,.Software'«WOW6432Node .CiscoCollabHost -Eula_disable
C. Set the "HKEY_LCX^M._MACHINE .SoftwareCiscoCollabHo$t€ula Setting registry Eula_disable
D. Set the DEFAULT^'THEMEs-Dark"" installation argument
E. Set the "/quiet installation argument

Answer: BC

Explanation:
The correct answers are B and C.
To ensure that end users are not prompted to accept the end-user license agreement (EULA) when they start the Webex app, the engineer must set the following

two registry keys:
- HKEY_LOCAL_MACHINE\Software\WOW6432Node\CiscoCollabHost\Eula_disable
- HKEY_LOCAL_MACHINE\Software\CiscoCollabHost\Eula Setting\Eula_disable

Setting these registry keys will disable the EULA prompt for all users who start the Webex app.
The other options are not valid actions to ensure that end users are not prompted to accept the EULA.

NEW QUESTION 348
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 353
What are the predefined call handlers in Cisco Unity Connection?

A. opening greeting, welcome, and default system
B. caller input, greetings, and transfer
C. greetings, operator, and closed
D. opening greeting, operator, and goodbye

Answer: D

NEW QUESTION 354
What is required when deploying co-resident VMs by using Cisco UCM?

A. Provide a guaranteed bandwidth of 10 Mbps.
B. Deploy the VMs to a server running Cisco UCM.
C. Avoid hardware oversubscription.
D. Ensure that applications will perform QoS.

Answer: C

Explanation:
When deploying co-resident VMs by using Cisco UCM, it is important to avoid hardware oversubscription. This means that you should not assign more resources to the VMs than the physical hardware can provide. For example, if you have a server with 16 CPU cores, you should not assign more than 16 CPU cores to the VMs.
If you oversubscribe the hardware, the VMs will not be able to get the resources they need to run properly. This can lead to performance problems and even outages.
To avoid hardware oversubscription, you should carefully plan your VM deployments. You should also monitor the performance of the VMs to make sure that they are not overusing the resources.
Here are some additional tips for deploying co-resident VMs by using Cisco UCM:
- Use a virtualization platform that supports Cisco UCM.
- Make sure that the VMs have the correct operating system and software installed.
- Configure the VMs to use the correct network settings.
- Monitor the performance of the VMs to make sure that they are running properly.

NEW QUESTION 355
Refer to the exhibit.

```
000142: *Apr 23 19:41:49.050: MGCP Packet received from 192.168.100.100:2427--->
AUEP 4 AALN/S0/SU0/0@VG320.cisco.local MGCP 0.1
F: X, A, I
<---

000143: *Apr 23 19:41:49.050: MGCP Packet sent to 192.168.100.101:2427--->
200 4
I:
X: 2
L: p:10-20, a:PCMU:PCMA:G.nX64, b:64, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-220, a:G.729:G.729a:G.729b, b:8, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-110, a:G.726-16:G.728, b:16, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-70, a:G.726-24, b:24, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-50, a:G.726-32, b:32, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-270, a:G.723.1-H:G.723:G.723.1a-H, b:6, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-330, a:G.723.1-L:G.723.1a-L, b:5, e:on, gc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, conttest, data, netwloop, netwtest
<---
```

What is the registration state of the analog port in this debug output?

A. The analog port failed to register to Cisco UCM with an error code 200.
B. The MGCP Gateway is not communicating with the Cisco UCM.
C. The analog port is currently shut down.
D. The analog port is registered to Cisco UCM.

Answer: D

NEW QUESTION 360
Which action prevents toll fraud in Cisco UCM?

A. Implement route patterns in Cisco UCM.
B. Implement toll fraud restriction in the Cisco IOS router.
C. Allow off-net to off-net transfers.
D. Configure ad hoc conference restriction.

Answer: D

NEW QUESTION 364
Which call routing pattern is used for phone numbers that are in the E.164 format?

A. /+.! Route Pattern
B. \+.! Route pattern
C. \+.! Translation Pattern
D. \+1.[2-9]XX[2-9]XXXXXXX called Party Transformation Pattern

Answer: B

NEW QUESTION 366
An engineer is configuring a BOT device for a Jabber user in Cisco Unified Communication Manager. Which phone type must be selected?

A. Cisco Dual Mode for Android
B. Cisco Unified Client Services Framework
C. Cisco Dual Mode for iPhone
D. third-party SIP device

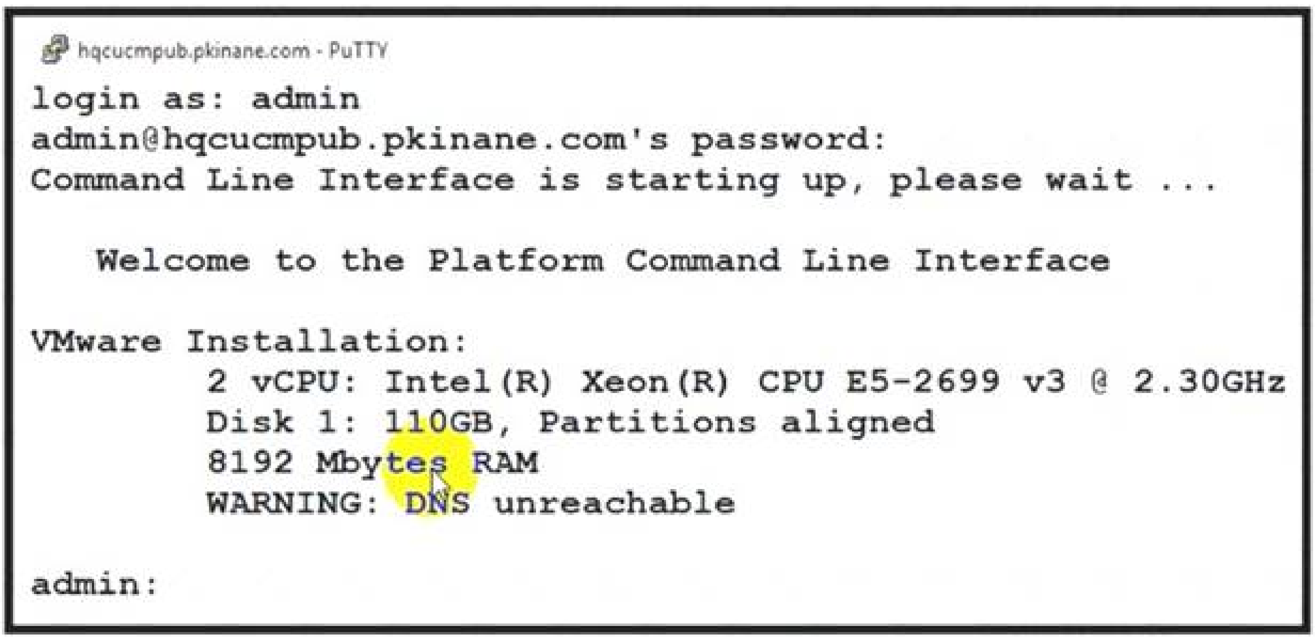
Answer: A

NEW QUESTION 370
User A Calls user. The call gets connected, but the quality is bed. What are two reasons for this issue? (Choose two)

A. Incorrect partition
B. No region relationship
C. Network congestion
D. Incorrect QoS
E. Incompatible codec

Answer: CD

NEW QUESTION 374
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 376
Which Cisco unity Connection handler plays a greeting at announces the option to dial a user extension by default?

A. the operator call handler
B. the Interview handler
C. the Goodbye call handler
D. the Directory handler

Answer: A

NEW QUESTION 381
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 384
A customer wants a video conference with five Cisco Telepresence 1X5000 Series systems. Which media resource is necessary in the design to fully utilize the immersive functions?

A. Cisco Webex Meetings Server
B. software conference bridge on Cisco UCM
C. Cisco Meeting Server
D. Cisco PVDM4-128

Answer: C

NEW QUESTION 385
In the cisco expressway solution, which two features does mobile and Remote access provide? (Choose two)

A. VPN-based enterprise access for a subset of Cisco Unified IP Phone models
B. secure reverse proxy firewall traversal connectivity
C. the ability 10 register third-party SIP or H 323 devices on Cisco UCM without requiring VPN
D. the ability of Cisco IP Phones to access the enterprise through VPN connection
E. the ability for remote users and their devices to access and consume enterprise collaboration applications and services

Answer: BE

NEW QUESTION 390
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 394
Which service on the Presence Server is responsible for maintaining the point-to-point chat connections between Jabber clients?

A. Cisco SIP Proxy
B. Cisco XCP Text Conference Manager
C. Cisco XCP Router
D. Cisco XCP XMPP Federation Manager

Answer: B

NEW QUESTION 397
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 401
An administrator would like to set several Cisco Jabber configuration parameters to only apply to mobile clients (iOS and Android). How does the administrator accomplish this with Cisco Jabber 12.9 and Cisco UCM 12.5?

A. Assign the desired configuration file to "Mobile" Jabber Client Configuration in the Service Profile.
B. Upload the jabber-config.enc file to TFTP
C. Create a user profile in Jabber Policies.
D. Deploy jabber-config-user.xml on iOS and Android devices.

Answer: A

NEW QUESTION 403
An engineer must configure switch port 5/1 to send CDP packets to configure an attached Cisco IP phone to trust tagged traffic on it's access port. Which command is required to complete the configuration?

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router config-if)# description Cube E41.228-0097
```

A. platform qos trust extend cos 3
B. platform qos trust extend
C. platform qos extend trust
D. platform qos trust extend cos 5

Answer: B

NEW QUESTION 404
How is bandwidth allocated to traffic flows in a flow-based WFQ solution?

A. All the bandwidth is divided based on the QoS marking of the packets.
B. Each type of traffic flow has equal bandwidth.
C. Bandwidth is divided among traffic flow
D. Voice has priority.
E. Voice has priority and the other types of traffic share the remaining bandwidth.

Answer: D

Explanation:
In a flow-based WFQ solution, bandwidth is allocated to traffic flows based on the following criteria:
- The priority of the traffic flow
-

- The amount of bandwidth that is available
- The number of traffic flows that are competing for bandwidth

Voice traffic is typically given a higher priority than other types of traffic, such as data traffic. This is because voice traffic is more sensitive to latency and jitter than data traffic.
When there is not enough bandwidth to accommodate all of the traffic flows, the WFQ algorithm will prioritize the traffic flows based on their priority. The traffic flows with the highest priority will be given the most bandwidth, and the traffic flows with the lowest priority will be given the least bandwidth.
If there is still not enough bandwidth to accommodate all of the traffic flows, the WFQ algorithm will start to drop packets. The packets that are dropped will be the packets from the traffic flows with the lowest priority.

NEW QUESTION 409

......

Thank You for Trying Our Product

We offer two products:

1st - We have Practice Tests Software with Actual Exam Questions

2nd - Questons and Answers in PDF Format

350-801 Practice Exam Features:

* 350-801 Questions and Answers Updated Frequently

* 350-801 Practice Questions Verified by Expert Senior Certified Staff

* 350-801 Most Realistic Questions that Guarantee you a Pass on Your FirstTry

* 350-801 Practice Test Questions in Multiple Choice Formats and Updatesfor 1 Year

100% Actual & Verified — Instant Download, Please Click
Order The 350-801 Practice Test Here