

Exam Questions 350-801

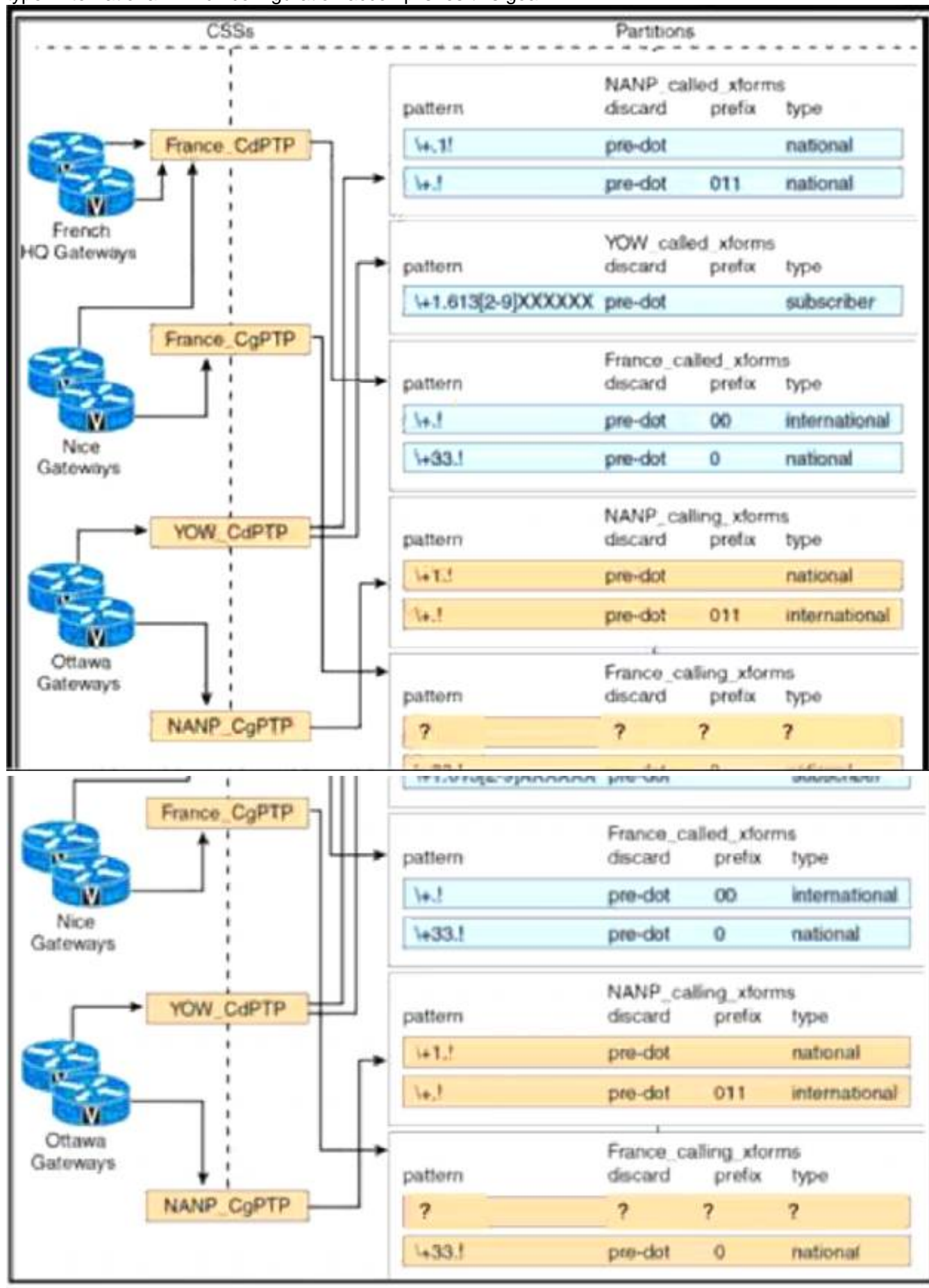
Implementing and Operating Cisco Collaboration Core Technologies

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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. 613XXXXXX none +011 internationa

Answer: C

NEW QUESTION 2

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 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

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

voice	AF11
interactive video	CS2
bulk data	EF
call-signaling	AF31/CS3
network management	AF41

- A. Mastered
- B. Not Mastered

Answer: A

Explanation:



NEW QUESTION 5

How are E.164 called-party numbers normalized on a globalized call-routing environment in Cisco UCM?

- A. Call ingress must be normalized before the call being routed.
- B. Normalization is not required.
- C. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.
- D. Normalization is achieved by setting up calling search spaces and partitions at the SIP trunks for PSTN connection.

Answer: C

NEW QUESTION 6

Which two configuration elements are part of the Cisco UCM toll-fraud prevention?(Choose two.)

- A. feature control policy
- B. partition
- C. SIP trunk security profile
- D. SUBSCRIBE Calling Search Space
- E. Calling Search Space

Answer: AE

Explanation:

The following are the configuration elements that are part of the Cisco UCM toll-fraud prevention:

- Feature control policy - This policy controls the features that are available to users. For example, you can use this policy to prevent users from making international calls.
- Calling Search Space - This space defines the numbers that users can call. For example, you can use this space to prevent users from calling premium-rate numbers.

NEW QUESTION 7

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 8

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 9

Which information is needed to restore the backup of a Cisco UCM publisher successfully?

- A. the TFTP server details
- B. the application credentials for Cisco UCM
- C. the security password for Cisco UCM
- D. the FTP server details

Answer: C

NEW QUESTION 10

An engineer must configure a SIP route pattern using domain routing with destination +13135551212. The domain ciscocm1.jupiter.com resolves to 192.168.1.3. How must the IPV4 Pattern be configured?

- A. +13135551212@192.168.1.3
- B. ciscocm1.jupiter.com
- C. \+13135551212@192.168.1.3
- D. 192.168.1.3

Answer: B

NEW QUESTION 10

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 12

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 17

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 18

Refer to the exhibit.

NAME	TTL	CLASS	TYPE	Priority	Weight	Port	Target Address
_sip._tcp.sample.com	86400	IN	SRV	10	60	5060	server1.sample.com
_sip._tcp.sample.com	86400	IN	SRV	10	30	5060	server2.sample.com
_sip._tcp.sample.com	86400	IN	SRV	5	20	5060	server3.sample.com

An administrator must fix the SRV records to ensure that server1. sample.com is always contacted first from the three servers. Which solution should the engineer apply to resolve this issue?

- A. Priority = 100, Weight = 90
- B. Priority = 10, Weight = 5
- C. Priority = 10, Weight = 10
- D. Priority = 5, Weight = 70

Answer: D

NEW QUESTION 19

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 24

Which command in the MGCP gateway configuration defines the secondary Cisco UCM server?

- A. ccm-manager redundant-host
- B. ccm-manager fallback-mgcp
- C. mgcpapp
- D. mgcp call-agent

Answer: A

NEW QUESTION 27

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

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

Answer: BC

NEW QUESTION 28

When configure Cisco UCM, which configuration enables phones to automatically reregister to a Cisco UCM publisher 'when the connection to the subscriber is lost?

- A. SRST
- B. Route Group
- C. Cisco UCM
- D. Device Pool

Answer: A

Explanation:

SRST, or Survivable Remote Site Telephony, is a feature that allows Cisco IP phones to continue to function even when the connection to the Cisco UCM publisher is lost. When SRST is configured, the phones will automatically reregister to the publisher when the connection is restored. Route groups are used to route calls to different destinations based on the caller's phone number or other criteria. Cisco UCM is the call management system that controls the IP phones. Device pools are used to group phones together for administrative purposes.

NEW QUESTION 32

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 36

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 38

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 g711alaw preference 1 codec g711ulaw preference 2
- B. voice class codec 11j codec g711ulaw preferred codec g711alaw
- C. vice class codec 100 codec preference 1 g711ulaw codec preference 2 g711alaw
- D. voice class codec 100 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 43

When designing the capacity for a Cisco UCM 12.x cluster, an engineer must decide which VMware template will be used for each node. What is the lowest number of users supported in a template and the highest number of users in a template?

- A. 750 and 15,000 users
- B. 750 and 10,000 users
- C. 500 and 10,000 users
- D. 1000 and 10,000 users

Answer: D

NEW QUESTION 48

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

Answer: D

NEW QUESTION 53

Which attribute contains an XMPP stanza?

- A. iq
- B. message
- C. type
- D. presence

Answer: A

NEW QUESTION 55

An engineer is going to redesign a network, and while looking at the QoS configuration, the engineer sees that a portion of the network is marked with AF42. Which type of traffic is marked with this tag?

- A. signaling
- B. voice
- C. video conference
- D. streaming video

Answer: D

NEW QUESTION 59

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 63

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 66

An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

- A. linecode ami
- B. linecode b8zs
- C. linecode hdb3
- D. linecode esf

Answer: A

NEW QUESTION 71

Refer to the exhibit.

Time	Source	Destination	Info
18.683437	10.117.34.222	10.0.101.10	50310 → 5060 [SYN] Seq=0 Win=64240 Len=0 MSS=1460 WS=256 SACK
18.938881	10.117.34.222	10.0.101.10	50314 → 5060 [SYN] Seq=0 Win=64240 Len=0 MSS=1460 WS=256 SACK
21.686680	10.117.34.222	10.0.101.10	[TCP Retransmission] 50310 → 5060 [SYN] Seq=0 Win=64240 Len=0
21.941993	10.117.34.222	10.0.101.10	[TCP Retransmission] 50314 → 5060 [SYN] Seq=0 Win=64240 Len=0
27.687008	10.117.34.222	10.0.101.10	[TCP Retransmission] 50310 → 5060 [SYN] Seq=0 Win=64240 Len=0
27.942784	10.117.34.222	10.0.101.10	[TCP Retransmission] 50314 → 5060 [SYN] Seq=0 Win=64240 Len=0

An administrator is attempting to register a SIP phone to a Cisco UCM but the registration is failing. The IP address of the SIP Phone is 10.117.34.222 and the IP address of the Cisco UCM is 10.0.101.10. Pings from the SIP phone to the Cisco UCM are successful. What is the cause of this issue and how should it be resolved?

- A. An NTP mismatch is preventing the connection of the TCP session between the SIP phone and the Cisco UC
- B. The SIP phone and Cisco UCM must be set with identical NTP sources.
- C. The certificates on the SIP phone are not trusted by the Cisco UC
- D. The SIP phone must generate new certificates.
- E. DNS lookup for the Cisco UCM FQDN is failin
- F. The SIP phone must be reconfigured with the proper DNS server.
- G. An network device is blocking TCP port 5060 from the SIP phone to the Cisco UC
- H. This device must be reconfigured to allow traffic from the IP phone.

Answer: D

NEW QUESTION 75

An engineer configures a new phone in Cisco UCM The phone boots and gets IP when it connects to the network, however the phone fails to register With CISCO UCM, The engineer observes that the phone has a status Rejected In CISCO UCM What must be verified first 'Mien troubleshooting the issue?

- A. whether auto-registration is enabled in Cisco UCM
- B. whether the Initial Trust List and Certificate Trust List files on the phone are correct
- C. whether the phone ts in the correct VLAN
- D. whether the phone's MAC address is correct In Cisco UCM

Answer: A

Explanation:

This is the first thing that must be verified when troubleshooting the issue of phone status showing rejected in Cisco UCM1. Auto-registration allows new phones to register with Cisco UCM without manual configuration 1. If auto-registration is disabled, the phone will not be able to register and will show a rejected statu1s. The other options are not the first things that must be verified when troubleshooting the issue:

- B. whether the Initial Trust List and Certificate Trust List files on the phone are correct is not the first thing to verify, but it may be a possible cause of the issue if the phone has an ITL file from another cluster that prevents it from registering with Cisco UCM2. To resolve this issue, the ITL file needs to be deleted from the phone or exchanged between the clusters2.
- C. whether the phone is in the correct VLAN is not the first thing to verify, but it may be a possible cause of the issue if the phone is not in the same VLAN as the Cisco UCM server or cannot reach it due to network issues3. To resolve this issue, the network connectivity and VLAN configuration need to be checked and

fixed3.

➤ D. whether the phone's MAC address is correct in Cisco UCM is not the first thing to verify, but it may be a possible cause of the issue if the phone's MAC address does not match the one configured in Cisco UCM. To resolve this issue, the MAC address needs to be corrected and updated in Cisco UCM.

NEW QUESTION 78

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 79

A collaboration engineer adds a voice gateway to Cisco UCM. The engineer creates a new gateway device in Cisco UCM. selects VG320 as the device type and selects MGCP as the protocol What must be done next to add the gateway to the Cisco UCM database?

- A. Select the DTMF relay type for the gateway.
- B. Select a device pool for the new gateway.
- C. Add the FQDN or hostname of the device.
- D. Configure the module in slot 0 of the new gateway.

Answer: C

NEW QUESTION 80

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 outage
- B. Active/standby mode provides an always on standbysnode 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 outage
- E. Active/standby mode provides an always on standbysnode 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 85

Which two functions are provided by Cisco Expressway Series? (Choose two.)

- A. voice and video transcoding
- B. voice and video conferencing
- C. interworking of SIP and H.323
- D. intercluster extension mobility
- E. endpoint registration

Answer: AC

Explanation:

The Cisco Expressway Series provides the following functions:

- Voice and video transcoding
- Interworking of SIP and H.323
- Firewall traversal
- Session border controller (SBC) functionality
- Endpoint registration
- Call admission control (CAC)
- Quality of service (QoS)

> Security

The Cisco Expressway Series does not provide voice and video conferencing or intercluster extension mobility.

NEW QUESTION 90

What must be configured on a Cisco Unity Connection voice mailbox to access the mailbox from a secondary device?

- A. mobile user
- B. alternate names
- C. alternate extensions
- D. Attempt Forward routing rule

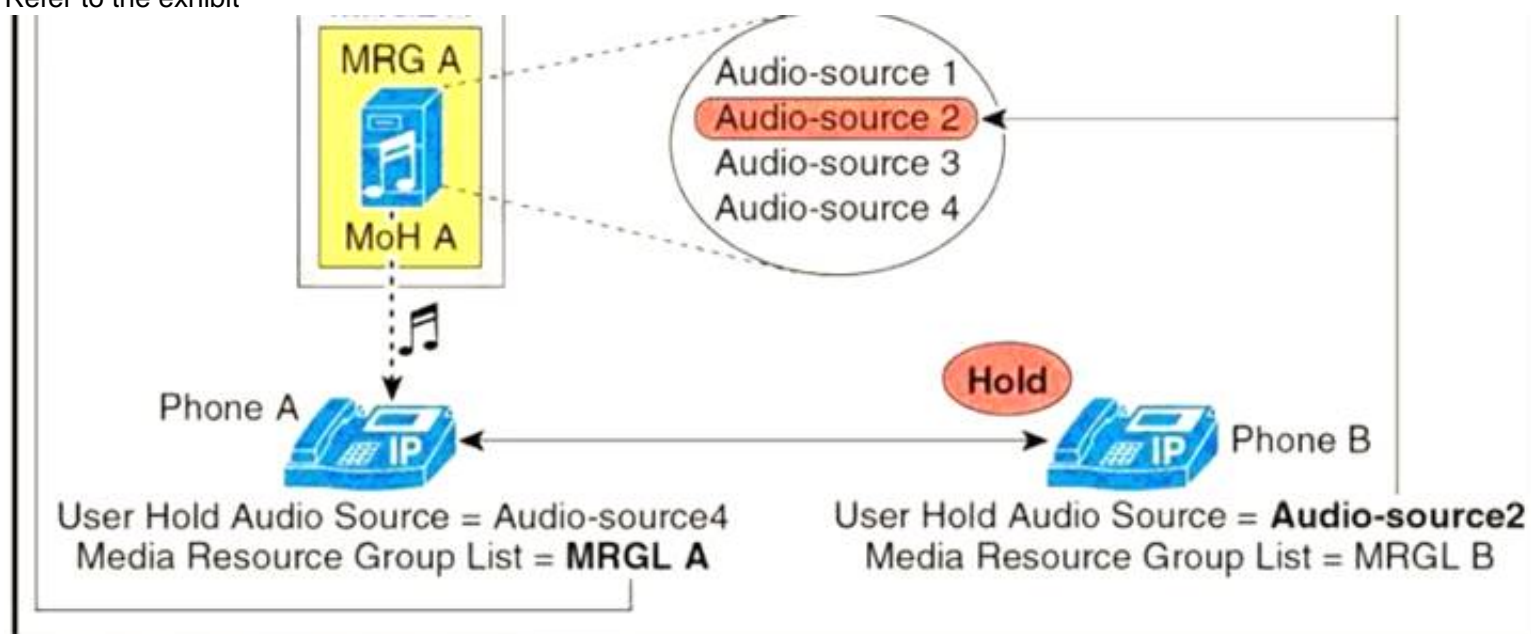
Answer: C

Explanation:

To access a Cisco Unity Connection voice mailbox from a secondary device, you must configure an alternate extension for the mailbox. This is a phone number that is different from the mailbox's primary extension. When you call the alternate extension, you will be prompted to enter the mailbox's PIN. Once you have entered the PIN, you will be able to access the mailbox just as you would if you were calling from the primary device.

NEW QUESTION 95

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 96

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 98

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 Element
- E. Cisco UCM, and Cisco Video Communication Server

Answer: B

NEW QUESTION 99

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 103

What are two Cisco UCM location bandwidths that are deducted when G 729 and G.711 codecs are used? (Choose two.)

- A. If a call uses G.729. Cisco UCM subtracts 16k.
- B. If a call uses G.711, Cisco UCM subtracts 64k
- C. If a call uses G.711, Cisco UCM subtracts 80k
- D. If a call uses G.729. Cisco UCM subtracts 24k.
- E. If a call uses G.729. Cisco UCM subtracts 40k

Answer: CD

NEW QUESTION 108

Which configuration on Cisco UCM is required for SIP MWI to work?

- A. Assign an MWI 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 113

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 114

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 115

The chief officer at a company must reduce collaboration infrastructure costs by onboarding all on-premises 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 boarding 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 118

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 119

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 121

Which version is used to provide encryption for SNMP management traffic in collaboration deployments?

- A. SNMPv1
- B. SNMPv3
- C. SNMPv2
- D. SNMPv2c

Answer: B

NEW QUESTION 125

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 127

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 130

What is required for Cisco UCM to accept SIP calls with a URI in the format of 'sip:2001@cucmpub.cisco.com'?

- A. Define Cluster Fully Qualified Domain Name under Servers in Cisco UCM.
- B. Change the Destination Address to a Fully Qualified Domain Name on the SIP trunk.
- C. Define Cluster Fully Qualified Domain Name in Enterprise Parameters.
- D. Set the SIPS URI Handling to True in CallManager Service Parameters.

Answer: C

NEW QUESTION 133

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 134

A customer enters no IP domain lookup on the Cisco IOS XE gateway to suppress the interpreting of invalid commands as hostnames Which two commands are needed to restore DNS SRV or A record resolutions? (Choose two.)

- A. ip dhcp excluded-address
- B. ip dhcp-sip
- C. ip dhcp pool
- D. transport preferred none
- E. ip domain lookup

Answer: DE

NEW QUESTION 136

What is a capability of a Cisco IOS XE media resource?

- A. It provides a hardware conferencing solution.
- B. It provides call forwarding capabilities.
- C. It provides redundancy for voice calls.
- D. It provides a voice packet optimization solution.

Answer: A

Explanation:

A Cisco IOS XE media resource provides a hardware conferencing solution. It can be used to mix multiple media streams, such as audio and video, into a single stream that can be sent to all participants in a conference call. This is done using a digital signal processor (DSP), which is a specialized processor that is designed to handle the processing of digital signals, such as audio and video.

NEW QUESTION 138

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 143

An engineer must configure DTMF relay on a Cisco Unified Border Element by using RFC2833 as the preferred relay mechanism and KPML as the next preferred relay mechanism. The engineer logs in to the CUBE and enters the dial-peer configuration level. Which command should be run at dial-peer configuration level?

- A. dtmf-relay sip-kvmi rtp-nte
- B. dtmf-relay rtp-nte sip-kpml
- C. dtmf-relay sip-kgml rtp-inband
- D. dtmf-relay rtp-inband sip-kvmi

Answer: B

Explanation:

The dtmf-relay command is used to configure DTMF relay on a Cisco Unified Border Element. The rtp-nte option specifies that RFC2833 is the preferred relay mechanism, and the sip-kpml option specifies that KPML is the next preferred relay mechanism.

NEW QUESTION 146

What are two characteristics of jitter in voice and video over IP communications? (Choose two.)

- A. The packets arrive with frame errors.
- B. The packets arrive at varying time intervals.
- C. The packets arrive out of sequence.
- D. The packets never arrive due to tail drop.
- E. The packets arrive at uniform time intervals.

Answer: BC

NEW QUESTION 151

Which characteristic 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 152

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 154

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 158

A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP, and the phone is registered to Cisco UCM. However, the engineer cannot reach the CU of the phone. Which two actions in Cisco UCM resolve the issue? (Choose two)

- A. Enable SSH Access under Product Specific Configuration Layout in Cisco UCM
- B. Disable Web Access under Product Specific Configuration Layout in Cisco UCM
- C. Set a username and password under Secure Shell information in Cisco UCM
- D. Enable Settings Access under Product Specific Configuration Layout in Cisco UCM
- E. Enable FIPS Mode under Product Specific Configuration Layout in Cisco UCM

Answer: AB

NEW QUESTION 161

Which action prevents toll fraud in Cisco Unified Communication Manager?

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

Answer: A

NEW QUESTION 166

An administrator works with an ISDN PRI that is connected to a third-party PBX. The ISDN link does not come up, and the administrator finds that the third-party PBX uses the OSIG signaling method. Which command enables the Cisco IOS Gateway to use QSIG signaling on the ISDN link?

- A. isdn incoming-voice voice
- B. isdn switch-type basic-ni
- C. isdn switch-type basic-qsig
- D. isdn switch-type primary-qsig

Answer: D

NEW QUESTION 171

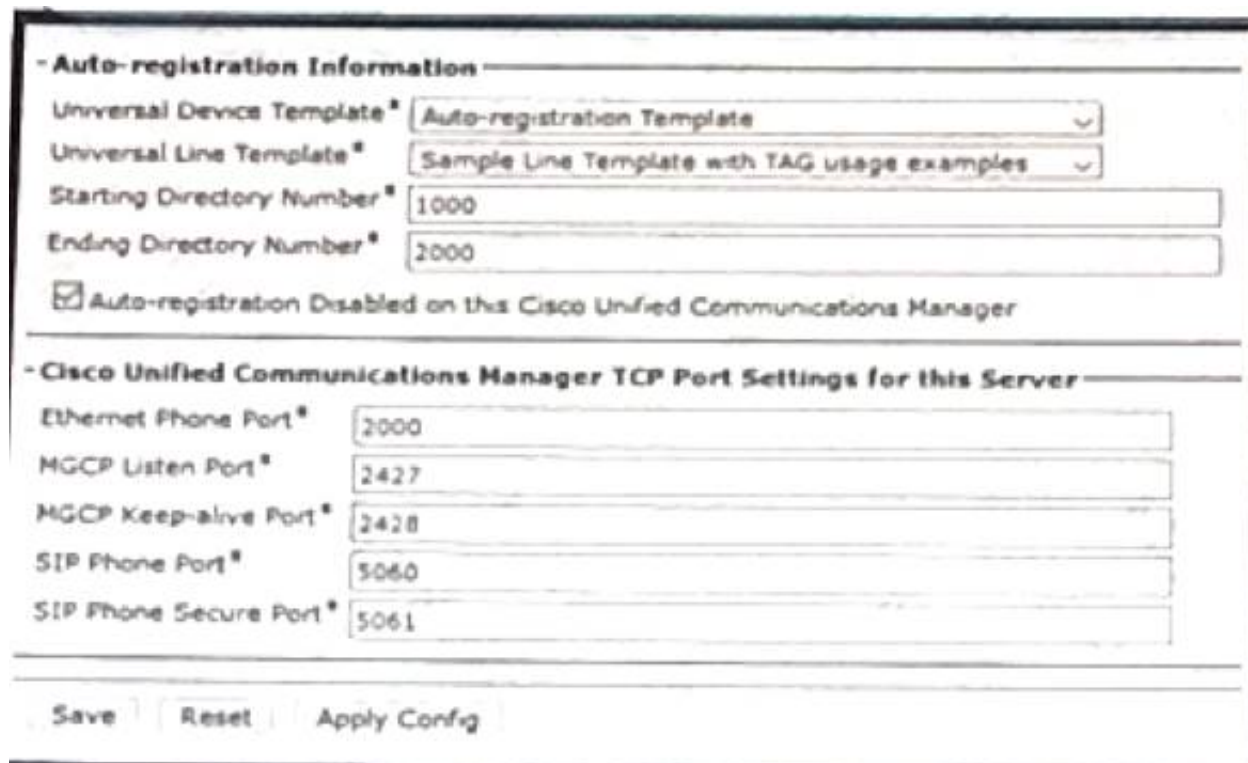
An end user at a remote site is trying to initiate an Ad Hoc conference call to an end user at the main site. The conference bridge is configured to support G.711. The remote user's phone only supports G.729. The remote end user receives an error message on the phone: "Cannot Complete Conference Call." 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 174

Refer to the exhibit.



The screenshot shows a configuration page for a Cisco Unified Communications Manager server. It is divided into two main sections:

- Auto-registration Information -**
 - Universal Device Template*: Auto-registration Template (dropdown)
 - Universal Line Template*: Sample Line Template with TAG usage examples (dropdown)
 - Starting Directory Number*: 1000 (text input)
 - Ending Directory Number*: 2000 (text input)
 - ☒ Auto-registration Disabled on this Cisco Unified Communications Manager (checkbox)
- Cisco Unified Communications Manager TCP Port Settings for this Server -**
 - Ethernet Phone Port*: 2000 (text input)
 - MGCP Listen Port*: 2427 (text input)
 - MGCP Keep-alive Port*: 2428 (text input)
 - SIP Phone Port*: 5060 (text input)
 - SIP Phone Secure Port*: 5061 (text input)

At the bottom, there are three buttons: Save, Reset, and Apply Config.

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 176

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 180

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 181

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 185

What is a description of the DiffServ model used for implementing QoS?

- A. AF41 has higher drop precedence than AF42, which has higher drop precedence than AF43.
- B. Voice and video calls are marked with different DSCP values and placed in different queues.
- C. AF43 has higher drop precedence than AF42 but lower drop precedence than AF41.
- D. RTP traffic from voice and video calls is marked EF and placed in the same queue.

Answer: A

NEW QUESTION 186

Refer to the exhibit.

```
isdn switch-type primary-ni
controller t1 0/1/0
framing esf
linecode b8zs
pri-group timeslots 1-10
```

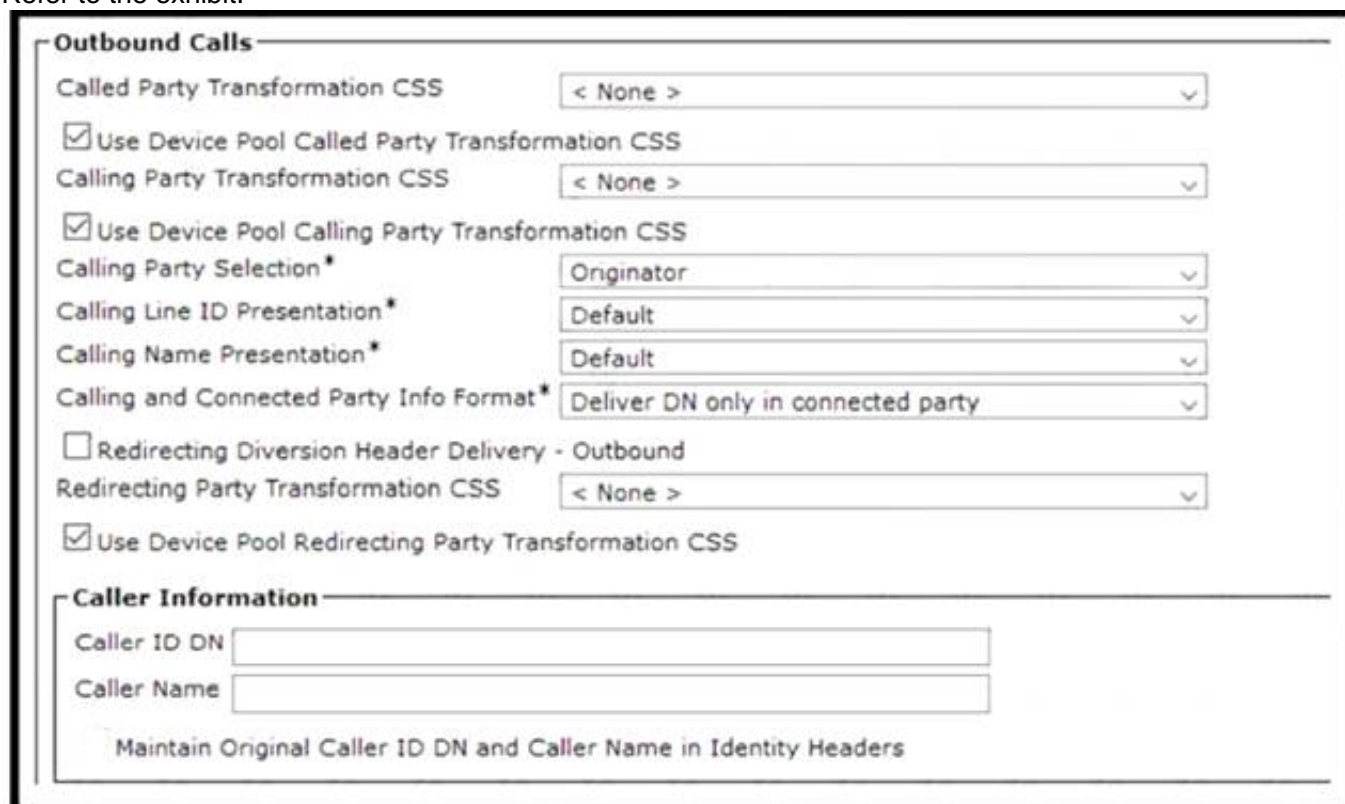
An engineer configures ISDN on a voice gateway. The provider confirms that the PRI is configured with 10 channels the engineer ordered and is working from the provider side, but the engineer cannot get a B-channel to carry voice. The rest of the configuration for the serial interface and voice network is functioning correctly. Which actions must be taken to carry voice?

- A. The engineer must activate the controller card on the voice gateway before configuring the device.
- B. The engineer used a T1 interface but must use an E1 interface.
- C. The pri-group timeslots command must be 0-9 for the 10 channels because all values on a router start with 0.
- D. The engineer must manually revert the order of using the channels.

Answer: A

NEW QUESTION 190

Refer to the exhibit.

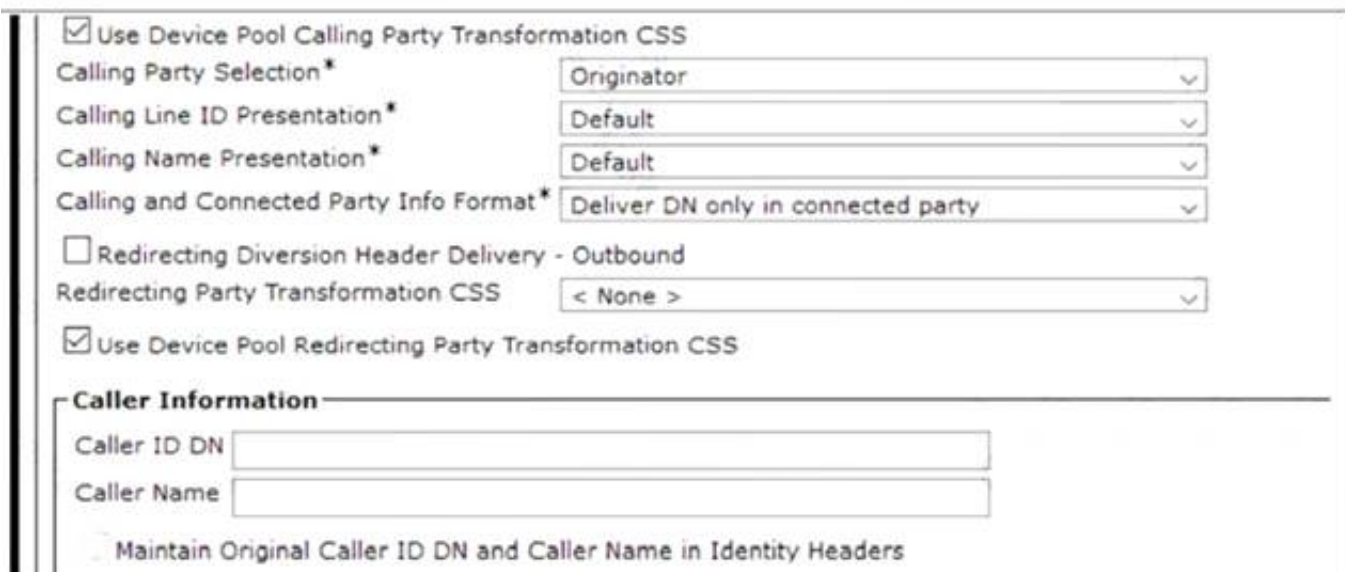


The exhibit shows the 'Outbound Calls' configuration page in Cisco IOS. The configuration includes:

- Called Party Transformation CSS: < None >
- ☒ Use Device Pool Called Party Transformation CSS
- Calling Party Transformation CSS: < None >
- ☒ Use Device Pool Calling Party Transformation CSS
- Calling Party Selection*: Originator
- Calling Line ID Presentation*: Default
- Calling Name Presentation*: Default
- Calling and Connected Party Info Format*: Deliver DN only in connected party
- ☐ Redirecting Diversion Header Delivery - Outbound
- Redirecting Party Transformation CSS: < None >
- ☒ Use Device Pool Redirecting Party Transformation CSS

Below the Outbound Calls section is the 'Caller Information' section:

- Caller ID DN: [Empty field]
- Caller Name: [Empty field]
- ☐ Maintain Original Caller ID DN and Caller Name in Identity Headers



This is a continuation of the configuration page from the previous exhibit, showing the same settings for Outbound Calls and Caller Information.

Unanswered calls do not reach the voicemail associated with the phones. Instead, callers receive the default greeting. Which action fixes the configuration?

- A. Reboot Cisco Unity Connection.
- B. Check the box "Redirecting Diversion Header Delivery - Outbound", then reset the trunk.
- C. Check the box 'Redirecting Diversion Header Delivery - Outbound'.
- D. Review the conversation manager logs on Cisco Unity Connection.

Answer: B

NEW QUESTION 192

An engineer troubleshoots outbound call failure on an ISDN-PRI circuit. The engineer is suspecting the 'Incomplete Destination'. Which debugs or commands are run in the voice gateway to troubleshoot the issue?

- A. debug isdn q921term mon
- B. debug voip ecapl inout show controller ti
- C. debug isdn q931 show isdn status
- D. debug isdn q921 debug voip ecapl inout

Answer: C

Explanation:

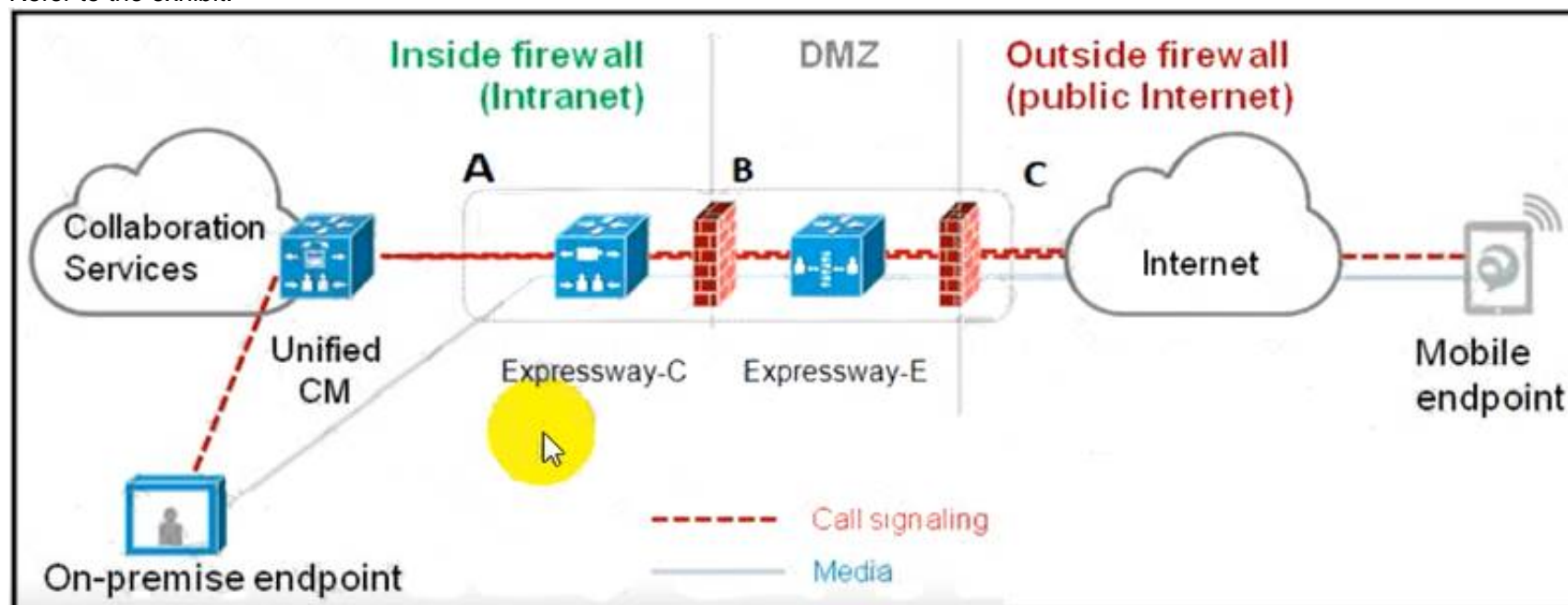
The engineer should run the following debugs or commands in the voice gateway to troubleshoot the issue: ➤ debug isdn q931 - This debug will show the ISDN Q.931 messages that are being exchanged between the voice gateway and the ISDN switch. This can be used to identify the cause of the "Incomplete Destination" error.

➤ show isdn status - This command will show the status of the ISDN PRI circuit. This can be used to verify that the circuit is up and running.

The other options are not correct. The debug isdn q921 command will show the ISDN Q.921 messages that are being exchanged between the voice gateway and the ISDN switch. This is not necessary for troubleshooting the issue. The term mon command will show the terminal monitor output. This is not necessary for troubleshooting the issue. The debug voip ecap inout command will show the VoIP ECAP messages that are being exchanged between the voice gateway and the VoIP server. This is not necessary for troubleshooting the issue. The show controller ti command will show the status of the T1 controller. This is not necessary for troubleshooting the issue.

NEW QUESTION 196

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 200

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 205

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 206

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 211

Refer to the exhibit.

```
!
voice service voip
 ip address trusted list
  ipv4 192.168.100.101
  ipv4 192.168.101.0 255.255.255.128
!
dial-peer voice 1 voip
 destination-pattern +T
 session protocol sipv2
 session target ipv4:192.168.102.102
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

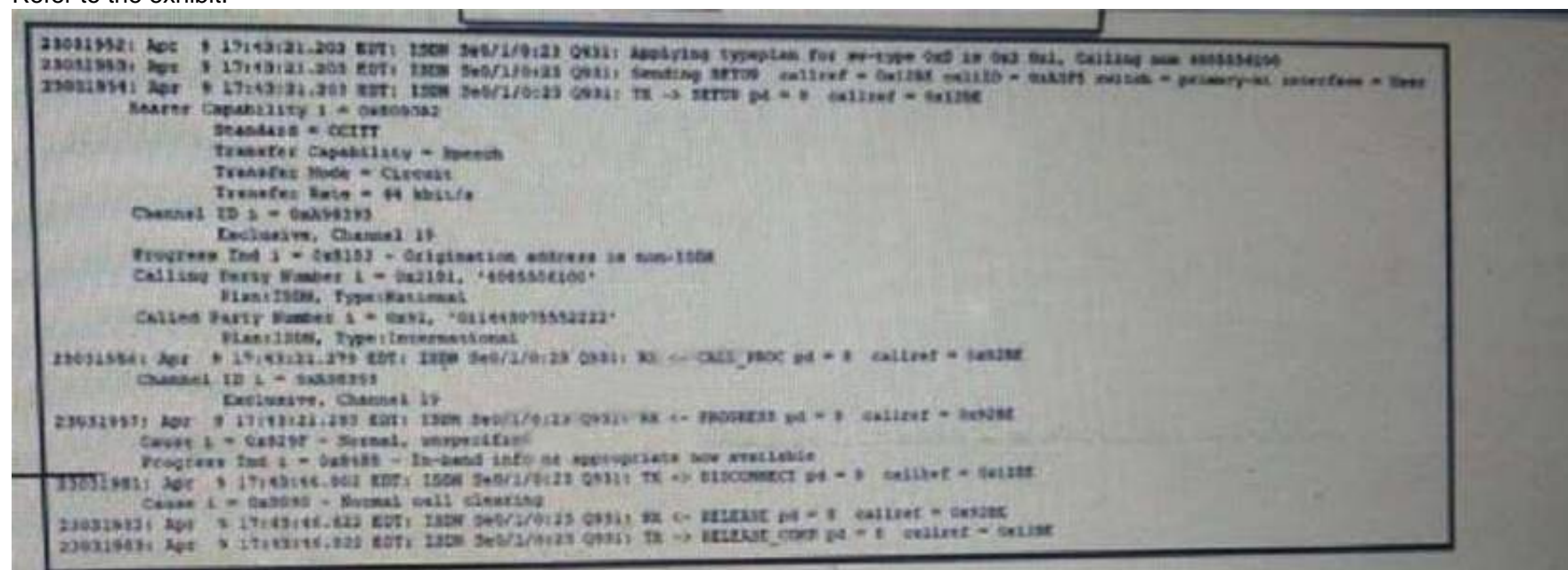
When a call is received on Cisco Unified Border Element. from which IP does it allow a connection?

- A. 192.168.100.103
- B. 192.168.102.102
- C. 192.168.100.102
- D. 192.168.101.201

Answer: B

NEW QUESTION 212

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 213

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 215

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.

Pattern Definition	
Pattern *	\+.
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> 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 *	Private

B.

Pattern Definition	
Pattern *	\+.
Partition	PT_US_VG_CD_Out_xForm
Description	US International calling
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> 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

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

D. **Pattern Definition**

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

Answer: C

NEW QUESTION 220

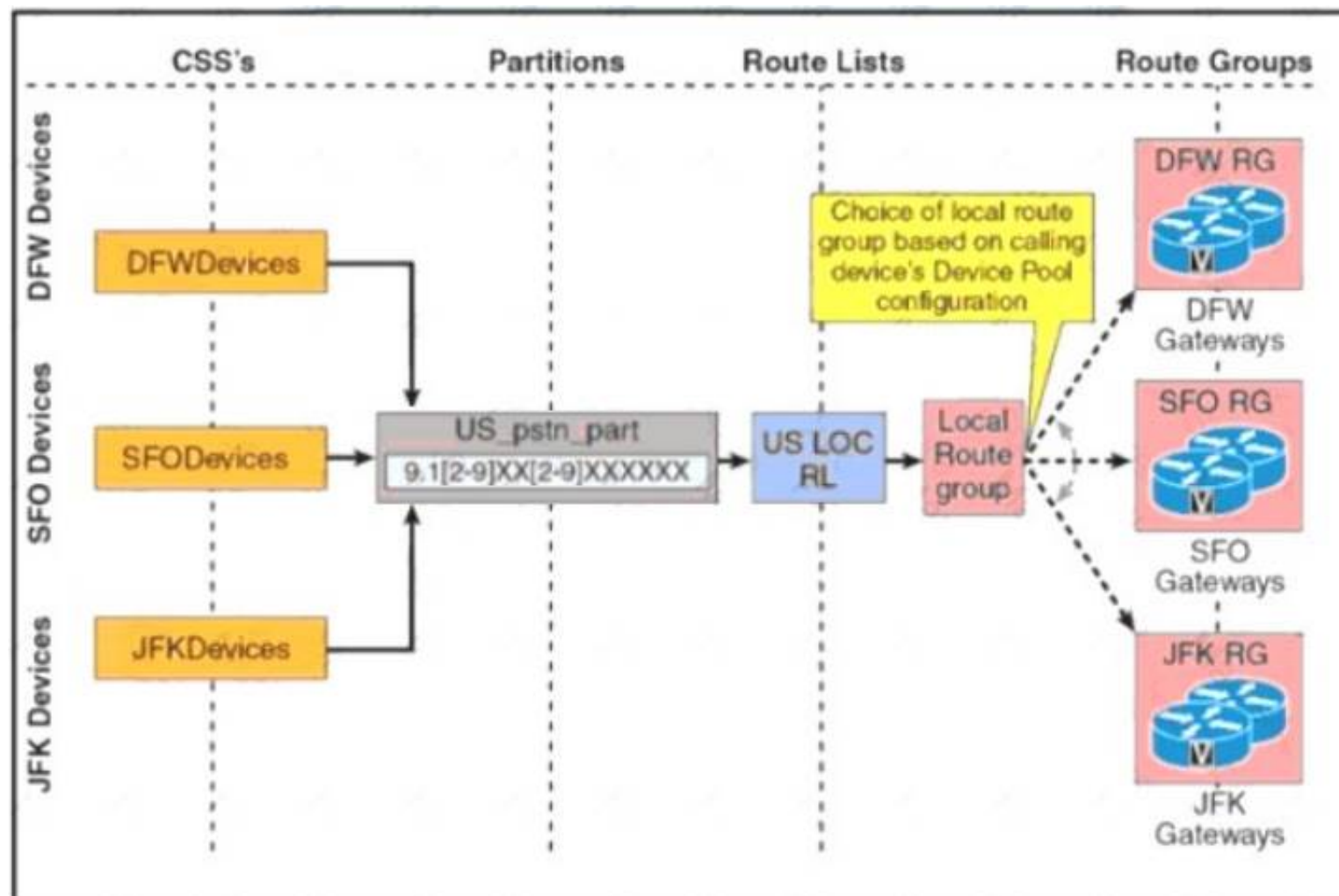
An administrator is configuring LDAP for Cisco UCM with Active Directory integration. A customer has requested to use "ipphone" instead of "telephoneNumber" as the phone number attribute. Where does the administrator specify this attribute mapping in Cisco UCM?

- A. LDAP Custom Filter
- B. LDAP Directory user fields
- C. LDAP Directory custom user fields
- D. LDAP Authentication

Answer: B

NEW QUESTION 222

Refer to the exhibit.



A user takes a phone from San Francisco to New York for a short reassignment. The phone was set up to use the San Francisco device pool, and device mobility is enabled on the Cisco UCM. The user makes a call that matches a route pattern in a route list that contains the Standard Local Route Group. To where does the call retreat?

- A. The call fails because device mobility is turned on, and the phone is not configured in New York
- B. The engineer must configure which sites the device should be roaming to.
- C. The call egresses in San Francisco because the user uses device mobility and is allowed to roam while still keeping the number and resources assigned in San Francisco.
- D. The call fails because the Standard Local Route Group is being used only if no configuration is set for the device pools.
- E. The call egresses in New York because the device automatically is assigned a New York device pool and uses the local gateway.

Answer: B

NEW QUESTION 226

A Cisco Telepresence SX80 suddenly has issues displaying main video to a display over HDMI. Which command can you use from the SX80 admin CLI to check the video output status to the monitor?

- A. xStatus HDMI Output
- B. xStatus video Output
- C. xconfiguration video Output
- D. xcommand video status

Answer: B

NEW QUESTION 229

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 230

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 232

Why isn't an end user's PC device in a QoS trust boundary included?

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

Answer: B

NEW QUESTION 235

An engineer is integrating Unity Connection with Cisco UCM. Which two actions must be configured so that recording and playback from the IP phones works at all times, including peak traffic hours? (Choose two.)

- A. Increase the number of voice ports.
- B. If it's a Unity Connection Cluster, ensure that replication is fine and not in split-brain mode.
- C. The phone system to which the phones are registered in Unity Connection has the Default Trap Switch check box enabled.
- D. Add dedicated dial-out ports with the allow trap connections setting selected.
- E. Ensure that you have set up SIP Digest Authentication on the SIP trunk security profile.

Answer: AC

NEW QUESTION 238

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 the 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 240

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 245

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 prompted 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_LOCAL_MACHINE .SoftwareCiscoCollabHost€Eula 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 249

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 250

Refer to the exhibit.

SIP Trunk Security Profile Information	
Name*	Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null string
Device Security Mode	Encrypted
Incoming Transport Type*	TLS
Outgoing Transport Type	TLS
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	
Incoming Port*	5061

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 251

When a new SIP phone is registered to Cisco Unified communications Manager, it keeps failing and showing an “unprovisioned” error message in the phone display. Which problem is a possible cause of this issue?

- A. Auto-registration is disabled on the Cisco Unified Communications Manager nodes and the phone device does not have a DN configured.
- B. The DHCP settings are incorrectly and the phone does not have an alternate TFTP defined.
- C. The phone cannot download and install the latest firmware.
- D. The DN assigned to the phone is already in use by another SIP phone.
- E. The DN configuration for this phone is shared with SCCP phone, which is not supported.

Answer: B

NEW QUESTION 254

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 259

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 264

Users want their mobile phones to be able to access their cisco unity connection mailboxes with only having to enter their voicemail pin at the login prompt calling pilot number where should an engineer configure this feature?

- A. transfer rules
- B. message settings
- C. alternate extensions
- D. greetings

Answer: C

NEW QUESTION 265

Which two steps are required for bulk configuration transactions on the Cisco UCM database utilizing BAT? (Choose two.)

- A. A data file in Abstract Syntax Notation One format must be uploaded to Cisco UCM
- B. A server template must be created in Cisco UCM
- C. A data file in comma-separated values format must be uploaded to Cisco UCM
- D. A data file in Extensible Markup Language format must be uploaded to Cisco UCM
- E. A device template must be created in Cisco UCM

Answer: CE

NEW QUESTION 270

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 273

After an engineer implements the FAC and CMC features together, users report that calls take almost one minute to complete and that they occasionally hear the reorder tone. Which two actions address this issue? (Choose two)

- A. Adjust the T302 timer from the default of 15 seconds to 5 seconds to shorten the interdigit timer
- B. Change the code if the problem persists
- C. Advise the user to hang up and try again
- D. Do not wait for the tones immediately dial the FAC and CMC
- E. Advise the user to press the "#" button after dialing the FAC and CMC codes

Answer: AB

NEW QUESTION 278

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 279

An engineer must configure switch port 5/1 to send CDP packets to configure an attached Cisco IP phone to trust tagged traffic on its 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 280

Refer to the exhibit.

Cisco Unified CM Administration
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Navigation: Cisco Unified CM Administration
ccmadministrator | Search Documentation | About

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Region configuration Related Links: [Back to Find/List](#)

Save Delete Copy Reset Apply Config Add New

Region Information

Name* REGION1

Region Relationships

Region	Audio Codec Preference List Configuration	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
REGION1	CCNP COLLAB	60 kbps	384 kbps	2147483647 kbps
REGION2	CCNP COLLAB	64 kbps (G.722, 6.711)	Use System Default (384 kbps)	Use System Default (2900000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Cisco Unified CM Administration
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Audio Codec Preference List Configuration Related Links: [Back to Find/List](#)

Save Delete Copy Add New

Status

Status: Ready

Audio Codec Preference Information

Name* CCNP COLLAB

Description* CCNP COLLAB

Codecs in List

- G.722 48k
- G.711 U-Law 64k
- G.729 8k
- G.711 A-Law 56k

An engineer is troubleshooting this video conference issue:

*A video call between a Cisco 9971 in Region1 and another Cisco 9971 in Region1 works.

*As soon as the Cisco 9971 in Region1 conferences in a Cisco 8945 in Region2. the Region1 endpoint cannot see the Region2 endpoint video.
What is the cause of this issue?

- A. Maximum Audio Bit Rate must be increased.
- B. Maximum Session Bit Rate for Immersive Video Calls is too low.
- C. Maximum Session Bit Rate for Video Calls is too low.
- D. Cisco 8945 does not have a camera connected.

Answer: C

NEW QUESTION 281

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