Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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

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

Answer: D

NEW QUESTION 2
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 3
What are two QoS requirements for VoIP traffic?

A. Voice traffic must be marked "to DSCP EF.
B. Loss must be no more man 1 percent.
C. Voice traffic must be marked to DSCP AF41.
D. One-way latency must be no more than 200 ms.
E. Average one-way jitter is greater than 50 ms.

Answer: AB

NEW QUESTION 4
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 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
Refer to the exhibit.

```
controller t1 0/0/1
pri-group timeslots 1-24
clock source line
linecode b8zs
framing esf
```

An administrator must replace the T1 card with an E1 card. What is the correct configuration if the administrator was asked to configure 12 time slots?

A.
```
controller e1 0/0/1
pri-group timeslots 1-12
clock source network
linecode hdb3
framing crc4
```

B.
```
controller e1 0/0/1
pri-group timeslots 1-11, 12
clock source line
linecode hdb3
framing crc4
```

C.
```
controller e1 0/0/1
pri-group timeslots 1-12
clock source line
linecode hdb3
framing crc4
```

D.
```
controller e1 0/0/1
pri-group timeslots 1-12
clock source line
linecode crc4
framing hd3
```

Answer: C

NEW QUESTION 8
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 9
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 10
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 10
Refer to the exhibit. Which two codec permutations should be transcoded by this dspfarm? (Choose two.)

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

Answer: CE

NEW QUESTION 15
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 20
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 23
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 27
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 30
During the Cisco IP Phone registration process, the TFTP download (ails. What are two reasons (or this issue? (Choose two.)

A. The DNS server was not specified, which is needed to resolve the DHCP server IP address.
B. Option 100 string was not specified, or an incorrect Option 100 string was specified.
C. The Cisco IP Phone does not know the IP address of the TFTP server.
D. The Cisco IP Phone does not know the IP address of any of the Cisco UCM Subscriber nodes.

E. Option 150 string was not specified, or an incorrect Option 150 string was specified.

Answer: CE

NEW QUESTION 34
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 37
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 40
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 43
Where in Cisco UCM is restrictions on audio bandwidth configured?

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

Answer: C

NEW QUESTION 44
Which endpoint feature is supported using Mobile and Remote Access through Cisco Expressway?

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

Answer: A

NEW QUESTION 46
Which attribute contains an XMPP stanza?

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

Answer: A

NEW QUESTION 49
Which type of input is required when configuring a third-party SIP phone?

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

Answer: A

NEW QUESTION 50
What should be used to detect common issues on a Cisco IOS XE-based Local Gateway and generate an email?

A. Real-Time Monitoring Tool
B. diagnostic signatures
C. syslog
D. SNMP

Answer: B

Explanation:
Diagnostic signatures are a feature that proactively detects commonly observed issues in the IOS XE-based Local Gateway and generates email, syslog, or terminal message notification of the event12. You can also install the diagnostic signatures to automate diagnostics data collection and transfer collected data to the Cisco TAC case to accelerate resolution time12.

NEW QUESTION 54
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 55
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 57
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 62
Refer to the exhibit.

```
Via: SIP/2.0/TCP
10.10.10.2:5060;branch=a8bH5bK7954A198F
From:
<sip:012345678@10.10.10.2>;tag=8D79AF62-DB2
To: <sip:90123456@10.10.4.14>;
tag=811681~ffa80926-5fac-4cc5-b802-
2dbde74ae7w2-
v=0
o=CiscoSystemsCCM-SIP 811681 1 IN IP4
10.10.4.14
s=SIP Call
c=IN IP4 10.5.4.3
t=0 0
m=audio 27839 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Which Codec is negotiated?

A. G.729
B. ILBC
C. G.711ulaw
D. G.728

Answer: C

NEW QUESTION 65
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 70
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 73
Which location must be assigned to the SIP trunk to replicate enhanced location information via a SIP trunk?

A. phantom
B. replica
C. hub_none
D. shadow

Answer: D

NEW QUESTION 77
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 79
A collaboration engineer must configure Cisco Unified Border Element to support up to five concurrent outbound calls across an Ethernet Link with a bandwidth of 160 kb to the Internet Telephony service provider. Which set of commands allows the engineer to complete the task without compromising voice quality?

A)

```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec aacld
```

B)

```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec ilbc mode 20
```

C)

```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
 codec mp4a-latm
```

D)

```
dial-peer voice 1 voip
 translation-profile outgoing Strip9
 max-conn 5
 destination-pattern 91[2-9]..[2-9]......$
 session protocol sipv2
 session target ipv4:142.45.10.1
 dtmf-relay rtp-nte sip-notify sip-kpml
```

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

Answer: B

NEW QUESTION 84
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 89
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 94
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 95
How does traffic policing respond to violations?

A. Excess traffic is dropped.
B. Excess traffic is retransmitted.
C. All traffic is treated equally.
D. Excess traffic is queued.

Answer: A

NEW QUESTION 96
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 99
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 104

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 109

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 112

Due to service provider restriction. Cisco UCM cannot send video in the SDR Which two options on Cisco UCM are configured to suppress video in the SDP in outgoing invites? (Choose two.)

A. Add the audio forced command to voice service VoIP on the Cisco Unified Border Element.
B. Check the Retry Video Call as Audio on the SIP trunk.
C. Set Video Bandwidth in the Region settings to 0.
D. Change the Video Capabilities dropdown on the endpoint to Disabled.
E. Check the Send send-receive SDP in mid-call INVITE check box on the SIP trunk SIP profile.

Answer: CD

NEW QUESTION 117

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 118
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 119
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 120

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 122
What are two functions of Cisco Expressway in the Collaboration Edge? (Choose two.)

A. Expressway-C provides encryption (or Mobile and Remote Access but not (or business-to-business communications.
B. The Expressway-C and Expressway-E pair can enable connectivity from the corporate network to the PSTN via a T1/E1 trunk.
C. The Expressway-C and Expressway-E pair can interconnect H.323-to-SIP calls for voice.
D. Expressway-E provides a VPN entry point for Cisco IP phones with a Cisco AnyConnect client using authentication based on certificates.
E. Expressway-E provides a perimeter network that separates the enterprise network from the Internet.

Answer: CE

NEW QUESTION 123
A customer is deploying a SIP IOS gateway for a customer who requires that in-band DTMF relay is first priority and out-of-band DTMF relay is second priority. Which 10$ entry sets the required priority?

A. dtmf-relay cisco-rtp
B. dtmf-relay sip-kpml cisco-rtp
C. sip-notify dtmf-relay rtp-nte
D. dtmf-relay rtp-nte sip-notify

Answer: D

NEW QUESTION 125
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 128
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 132
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 135
Which certificate does the Disaster Recovery System in Cisco UCM use to encrypt its communications?

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

Answer: D

NEW QUESTION 138
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 142
A company deploys centralized cisco ucm architecture for a hub location and two remote sites.
*The company has only one ITSP connection at the hub connection, and ITSP supports only G.711 calls
*Remote site A has a 1-Gbps fiber connection to the hub connection and calls to and from remote side A use G.711 codec
*Remote site B has a 1 T1 connection to the hub location and calls to and from remote site B use G.729 codec Based on the provided guidance, a Cisco voice engineer must design media resource management for the
customer What is the method that needs to be followed?

A. configure the hardware transcoder on the site B router
B. configure the hardware transcoder on the site A router
C. configure the hardware transcoder on the hub location router
D. configure the software transcoder on Cisco UCM to support voice calls to and from both remote sites

Answer: C

NEW QUESTION 146
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 150
Users dial a 9 before a 10-digit phone number to make an off-net call All 11 digits are sent to the Cisco Unified Border Element before going out to the PSTN The PSTN provider accepts only 10 digits. Which configuration is needed on the Cisco Unified Border Element for calls to be successful?

A. voice translation-rule 1 rule 1 /^9/ //
B. voice translation-rule 1 rule 1 /^9(..........)/ //
C. voice translation-rule 1 rule 1 /^9.+/ //
D. voice translation-rule 1 rule 1 /^9........./ //

Answer: A

NEW QUESTION 152
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 153
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.
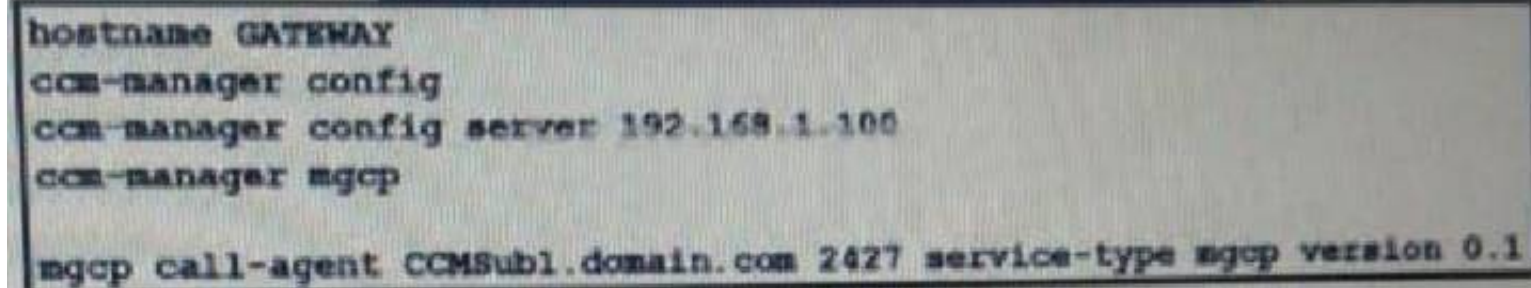

NEW QUESTION 158
In which location does an administrator look to determine which subscriber the phone registers to if loses registration with the current Cisco UCM subscriber?

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 161
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 166
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 167
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 170
Refer to the exhibit.
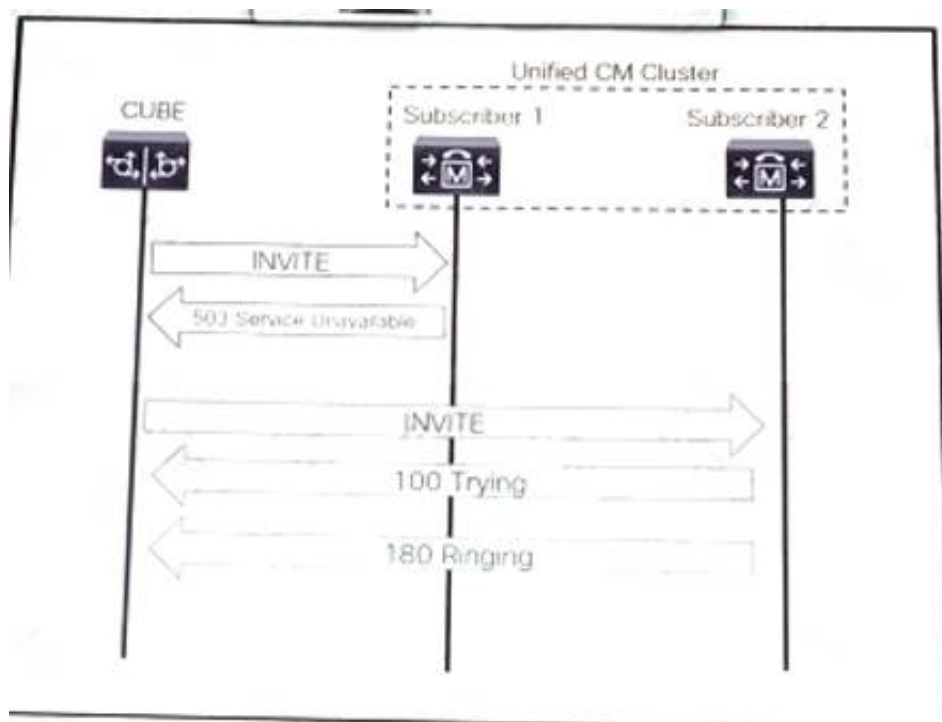
Cisco Unified element is attempting to establish a call with Subcribers1, but the call fails. Cisco Unified Border Element then retries the same call with Subcribers2, and the call proceeds normally.
Which action resolves the issue?

A. Verify that the correct calling search space is selected for the inbound Calls section
B. Verify that the run on all active United CM Nodes checkbox is enabled
C. Verify that the Significant Digits field for inbound Calls is set to All.
D. Verify that the PSTN Access checkbox is enabled.

Answer: B

NEW QUESTION 174
Refer to the exhibit.

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 177
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 181
Which Cisco Unified communications manager configuration is required for SIP MWI integration?

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

Answer: D

NEW QUESTION 184
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 188
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 191
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 195
Refer to the exhibit.

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

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

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

Answer: C

NEW QUESTION 196
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 201
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 205
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 206
Refer to the exhibit.

```
INVITE sip:1@10.10.10.219;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.10.10.84:50083;branch=z9hG4bK471df613
From: "1234 - My Phone" <sip:1234@10.10.10.219>;tag=381c1aba7a78002c558eda31-12b8af63
To: <sip:1@10.10.10.219>
Call-ID: 381c1aba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml,dialog
Content-Type: application/sdp
Content-Length: 658

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
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
```

When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages are examined next to further troubleshoot the issue? (Choose two.)

A. REGISTER
B. SUBSCRIBE
C. PRACK

D. NOTIFY
E. UPDATE

Answer: BD

NEW QUESTION 208
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 211
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 214
Which Cisco UCM configuration is required for SIP MWI integrations?

A. Enable "Accept presence subscription" on the SIP Trunk Security Profile.
B. Select "Redirecting Diversion Header Delivery - Outbound" on the SIP trunk.
C. Enable "Accept unsolicited notification" on the SIP Trunk Security Profile.
D. Select "Redirecting Diversion Header Delivery - Inbound" on the SIP trunk.

Answer: C

NEW QUESTION 217
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 220
A Cisco UCM administrator wants to enable the Self-Provisioning feature for end users. Which two prerequisites must be met first? (Choose two.)

A. End users must have a secondary extension.
B. Cisco Extended Functions service must be running
C. End users must belong to Standard CCM Admin Users group, the Standard CCM End Users group, and the Standard CCM Self-Provisioning group.
D. End users must have a primary extension.
E. End users must be associated to a user profile or feature group template that includes a universal line template and universal device template.

Answer: DE

NEW QUESTION 222
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 223
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 227
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 230
An engineer troubleshoots outbound can failure on an ISDN-PRI circuit. The engineer ts 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 ecapi inout show controller ti
C. debug isdn q931 show isdn status
D. debug isdn q921 debug voip ecapi 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
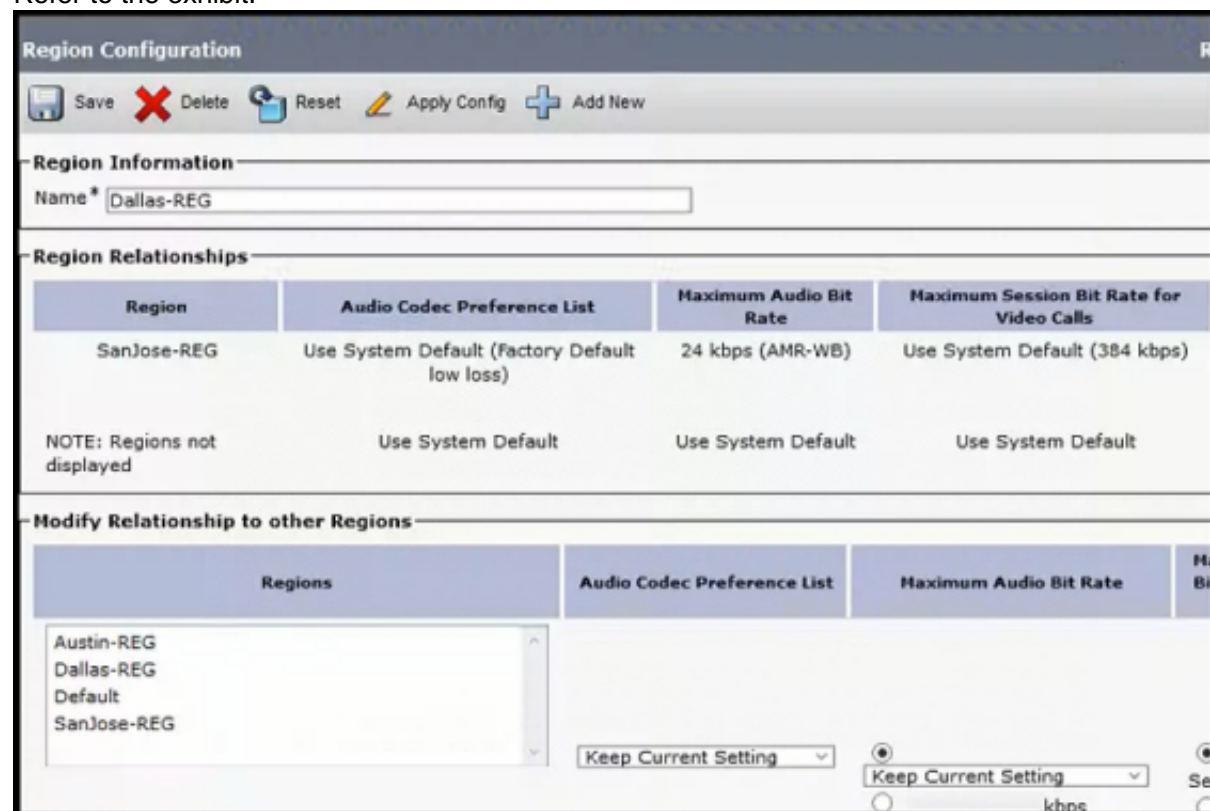the voice gateway and the ISDN switch. This can be used to identify the cause of the "Incomplete Destination" error.
➢ show isdn status - This command will show the status of the ISDN PRI circuit. This can be used to verify that the circuit is up and running.
The other options are not correct. The debug 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 ecapi 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 231
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 235
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 238
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 241
What is the purpose of a hybrid Local Gateway?

A. to handle calls between Webex Calling and Cisco Calling Plans
B. to handle calls between Webex Calling and Cloud Connected PSTN
C. to handle calls between Cisco IJCM and Webex Calling
D. to handle calls between the Public Switched Telephone Network and Webex Calling

Answer: D

Explanation:
A hybrid local gateway handles calls between the Public Switched Telephone Network (PSTN) and Webex Calling. It is commonly deployed on the customer's premises but can also be hosted by a partner. The local gateway registers with Webex Calling and handles all calls between the PSTN and Webex Calling. It gives customers the flexibility to bring their own service provider or continue using their existing provider for a smooth and effective transition to the cloud.

NEW QUESTION 246
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 249
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 251
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 256
An engineer configures Cisco UCM to prevent toll fraud. At which two points does the engineer block the pattern in Cisco UCM to complete this task? (Choose two.)

A. partition
B. route partem
C. translation pattern
D. CSS
E. route group

Answer: AD

NEW QUESTION 258
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 260
An administrator configures Cisco UCM to use UDP for SIP signaling and finds that an endpoint cannot make calls. Which action resolves this issue?

A. Change the common phone profile.
B. Change the SIP dial rules.
C. Change the SIP profile.
D. Change the phone security profile.

Answer: D

NEW QUESTION 265
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 269
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 >

☑ 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*	Private

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 271
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 dialpeer 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 274
An administrator is configuring LDAP tor 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 278
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 283
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 288
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 292
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 295
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.
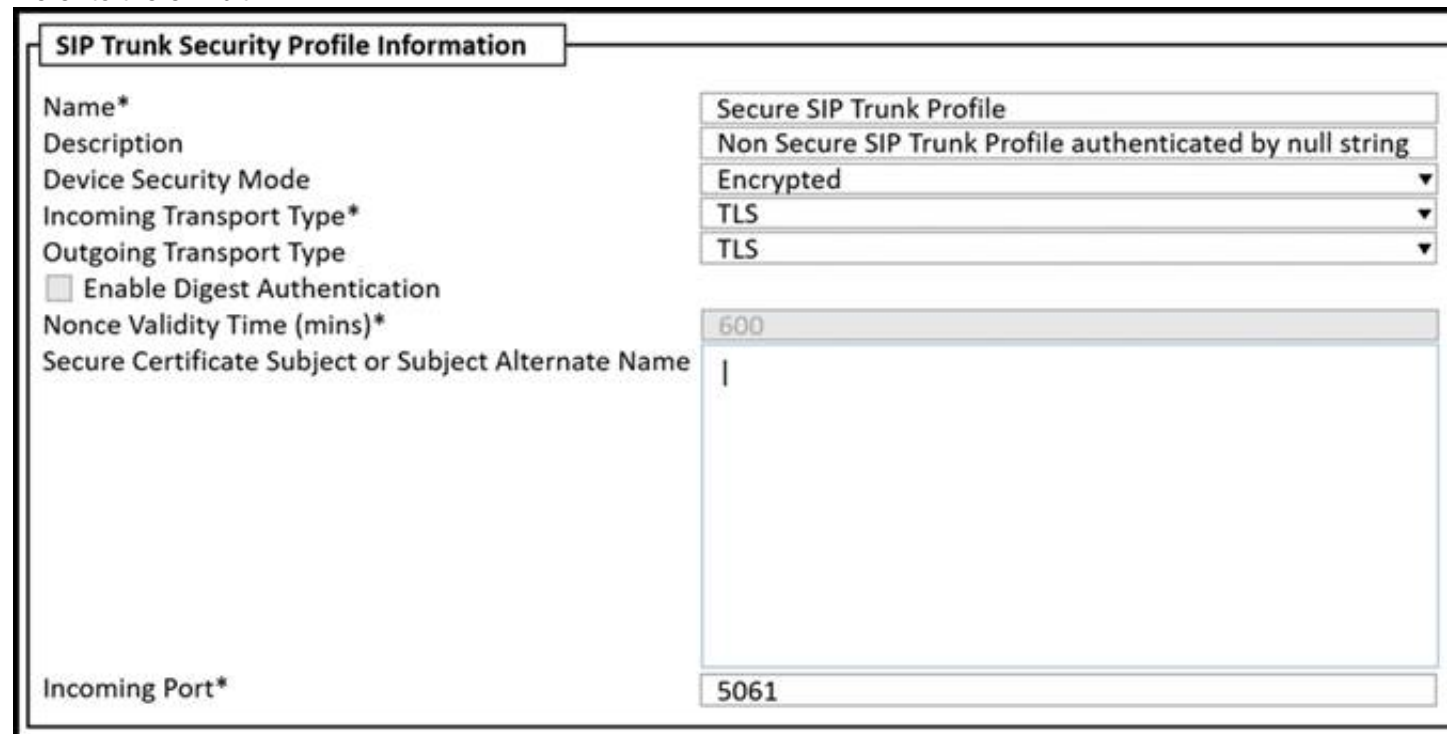
To avoid hardware oversubscription, you should carefully plan your VM deployments. You should also monitor the performance of the VMs to make sure that they are not overusing the resources.

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 299
Refer to the exhibit.

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

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

Answer: B

NEW QUESTION 301
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 303
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 308
Refer to the exhibit.

```
hqcucmpub.pkinane.com - PuTTY
login as: admin
admin@hqcucmpub.pkinane.com's password:
Command Line Interface is starting up, please wait ...

    Welcome to the Platform Command Line Interface

VMware Installation:
        2 vCPU: Intel(R) Xeon(R) CPU E5-2699 v3 @ 2.30GHz
        Disk 1: 110GB, Partitions aligned
        8192 Mbytes RAM
        WARNING: DNS unreachable

admin:
```

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 310
Refer to the exhibit.

```
voice translation-rule 1
  rule 1 /^[2-9]......$/ /\0/ type any subscriber
  rule 2 /^1[2-9]..[2-9]......$/ /\0/ type any subscriber
```

What is the result of applying these two rules to a voice translation profile for use with an ISDN T1 PRI on a Cisco Voice Gateway?

A. The leading Plus is stripped from the numeric phone number.
B. The ISDN Plan is modified to the administrator's defined value.
C. Any zero is stripped from the numeric phone number.
D. The ISDN Type is modified to the administrator's defined value.

Answer: D

NEW QUESTION 313
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 318
An engineer is deploying the Webex app on a Microsoft Windows computer that has multiple user accounts. Which CLI argument allows the engineer to Install the application as a "per machine" Installation?

A. ACCEPT_EULA
B. INSTALL ROOT
C. ALLUSERS
D. FORCELOCKDOVN

Answer: C

Explanation:
The ALLUSERS property is a command-line argument that can be used to install the Webex app as a "per machine" installation. This means that the app will be installed for all users on the computer, regardless of which user account is currently logged in.
The ACCEPT_EULA property is a command-line argument that can be used to accept the end-user license agreement (EULA) for the Webex app. The INSTALL_ROOT property is a command-line argument that can be used to specify the installation directory for the Webex app. The FORCELOCKDOWN property is a command-line argument that can be used to prevent users from uninstalling the Webex app.

NEW QUESTION 323
Which two devices are supported by the Flexible DSCP Marking and Video Promotion feature? (Choose two.)

A. MGCP devices
B. SCCP devices
C. pass-through MTPs
D. H.323 trunks
E. DX80

Answer: BC

NEW QUESTION 328
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 333
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 335
A collaboration engineer is configuring the QoS trust boundary for Cisco UCM voice and video conferencing. Which two trust boundary configurations are valid? (choose two)

A. QoS trust boundaries include all the devices directly attached to the access switch ports
B. QoS trust boundaries can be extended to Jabber running on a PC
C. QoS trust boundaries exclude Jabber softphone running on a PC
D. QoS trust boundaries can be extended to voice and video devices if the connected PCs are included
E. QoS trust boundaries can be extended to voice and video devices exclusively

Answer: CD

NEW QUESTION 340
An engineer deploys a Cisco Expressway-E server for a customer who wants to utilize all features on the server. Which feature does the engineer configure on the Expressway-E?

A. H.323 endpoint registrations
B. Mobile and Remote Access
C. SIP gateway for PSTN providers
D. VTC bridge

Answer: A

NEW QUESTION 345
An administrator needs to create a partial PRI consisting of the first seven timeslots available. Which configuration snippet configures the ISDN E1 PRI for this task?

A.
```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#interface Serial0/0/0:15
2900(config-controller)#pri-group timeslots 1-7
```

B.
```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-timeslots 1-7
```

C.

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-group timeslots 1-7
```

D.
```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#pri-group timeslots 1-7
```

Answer: C

NEW QUESTION 347
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 351
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 354
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 357
Refer to the exhibit.
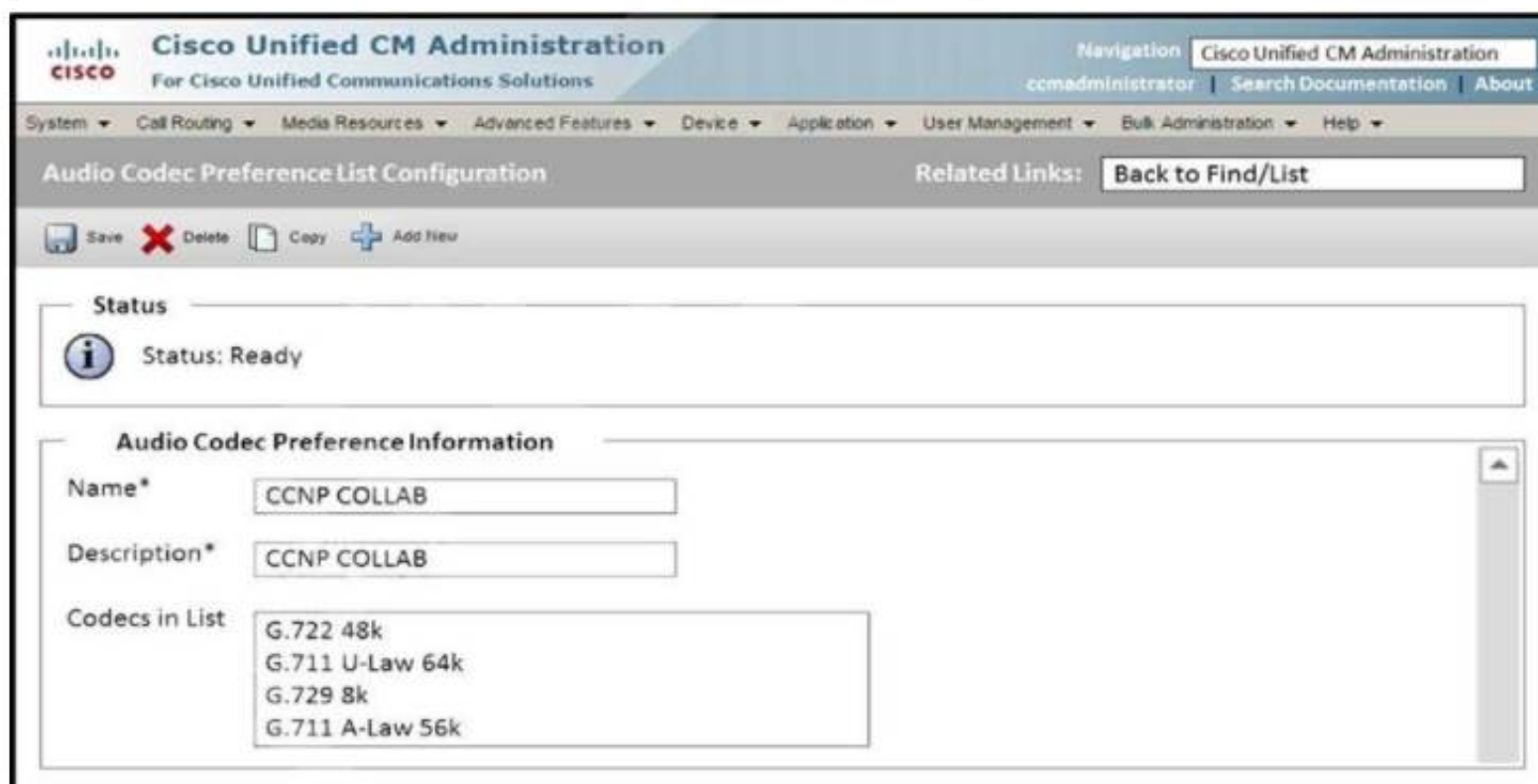
An engineer is troubleshooting this video conference issue:

*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 Regio1l 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 360
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 361
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