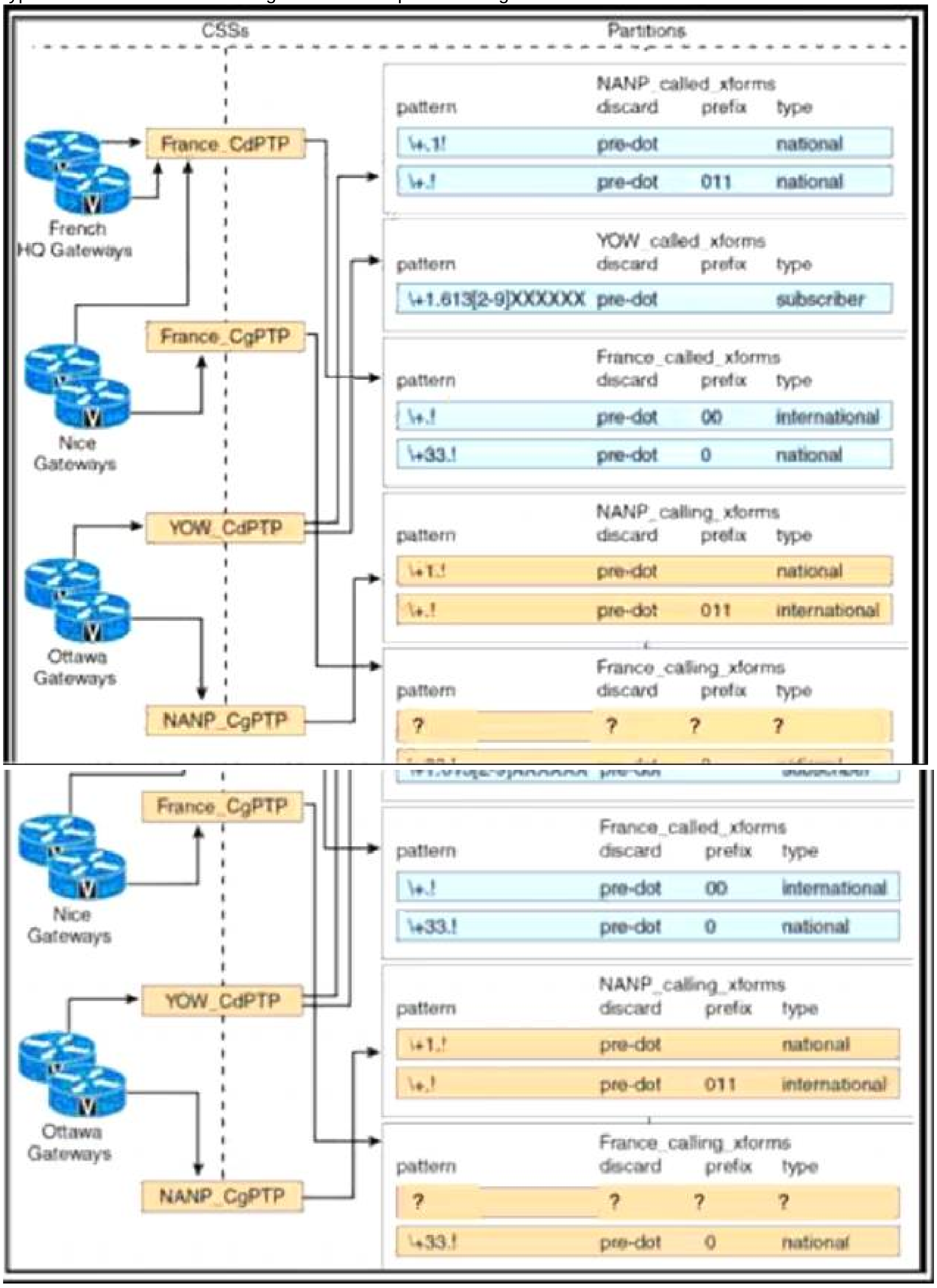
Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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



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

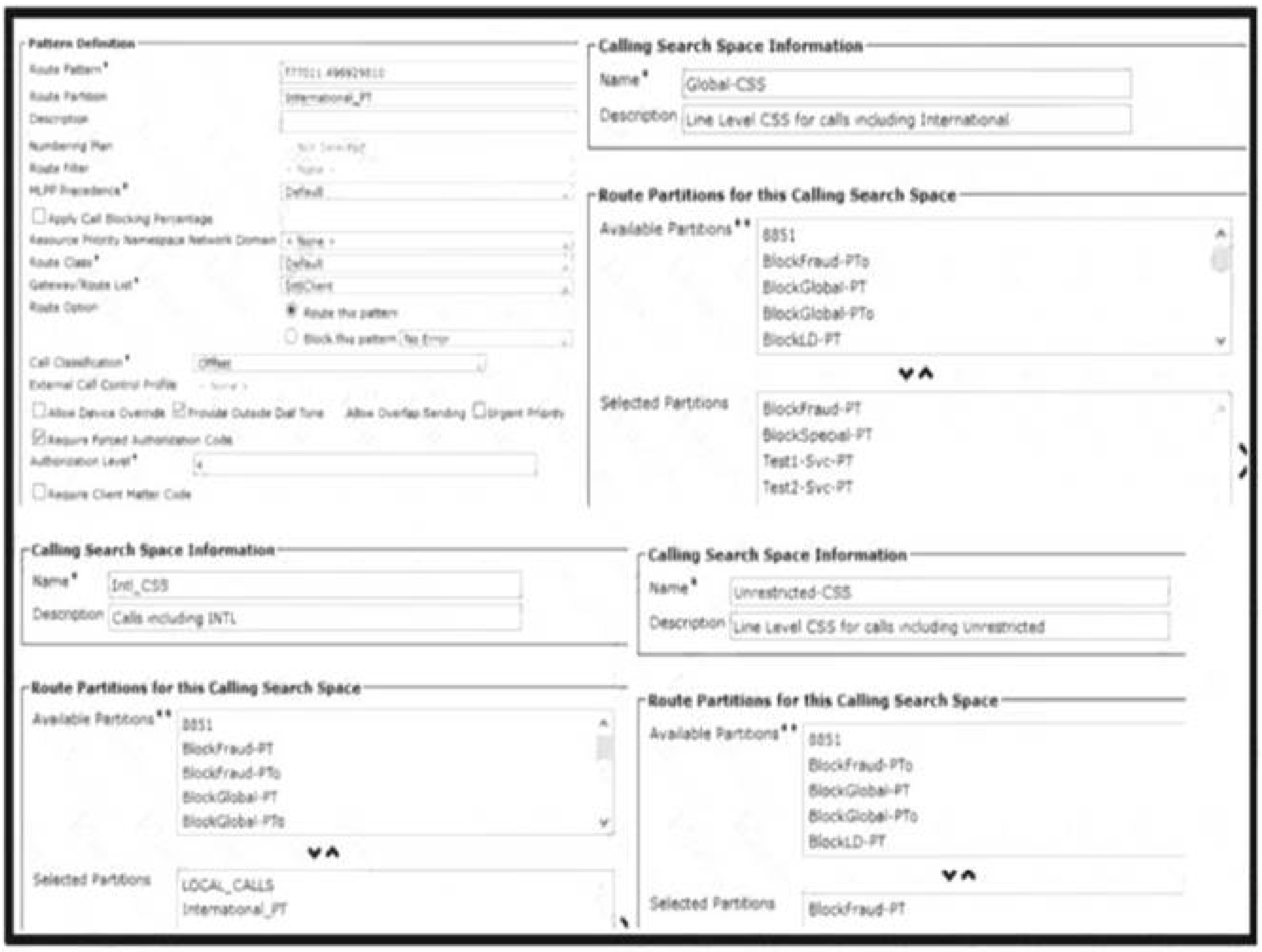
Answer: C

NEW QUESTION 2
What 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
Refer to the exhibit.



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

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

Pattern= \+.777011496929810, CSS=Intl_CSS

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

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

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

Answer: C

NEW QUESTION 4
A Cisco voice gateway is configured to use a sip-kpml DTMF relay in global settings. A new SIP dial peer is configured for a third-party application that only supports an in-band DTMF relay. Which commands must an engineer run on the dial peer?

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

Answer: C

NEW QUESTION 5
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:

voice	interactive video
interactive video	network management
bulk data	voice
call-signaling	call-signaling
network management	bulk data

NEW QUESTION 6
Refer to the exhibit.

```
C:\Users\CISCO>nslookup
Default Server:   dns.example.com
Address:   192.168.100.1

> set type=SRV
> _collab-edge._tcp.example.com
Server:   dns.example.com
Address:   192.168.100.1

Non-authoritative answer:
_collab-edge._tcp.example.com        SRV service location:
          priority       = 10
          weight         = 10
          port           = 8443
          svr hostname   = expe.example.com
```

You deploy Mobile and Remote Access for Jabber and discover that Jabber for Windows does not register to cisco Unified Communications Manager while outside of the office. What is a cause of this issue?

A. The DNS record should be created for _ cisco-uds._tcp example.com.
B. The DNS record should be changed from _collab-edge._tls example.com.
C. The DNS record type should be changed from SRV to A.
D. Server 4.2.2.2 is not a valid DNS server.

Answer: B

NEW QUESTION 7
What is an advantage of using Cisco Webex Control HuB?

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

Answer: A

Explanation:

Cisco Webex Control Hub is a cloud-based management platform that enables you to provision, administer, and manage Webex services and Webex Hybrid Services. It provides a single pane of glass for managing all of your Webex services, including Webex Meetings, Webex Teams, and Webex Calling.
Webex Control Hub offers a number of features and benefits, including:

- A single pane of glass for managing all of your Webex services
- Centralized user management
- Simplified provisioning and administration
- Real-time analytics and reporting
- Enhanced security and compliance

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

NEW QUESTION 8
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 9
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 10
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 10
What describes the outcome when the trust boundary is defined at the Cisco IP phone?

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

Answer: B

NEW QUESTION 14
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 16
An engineer configures a SIP trunk for MWI between a Cisco UCM cluster and Cisco Unity Connection. The Cisco UCM cluster fails to receive the SIP notify messages. Which two SIP trunk settings resolve this issue? (Choose two.)

A. accept out-of-dialog refer
B. accept out-of-band notification
C. transmit security status
D. allow changing header
E. accept unsolicited notification

Answer: AE

NEW QUESTION 21
Callers from a branch report getting busy tones intermittetly when trying to reach colleagues in other office branches during peak hours. An engineer collects Cisco CallManager service traes to examine the situation. The traces show:

```
50805567.000 |07:35:39.676 |Sdl Sig |StationOoutputDisplayNotify |restart0
|StaatinD(1,100,63,6382) |StionCdpc(1,100,64,4725) |1,100,40,6.709919^*^*
|[R:N-H:0,L:0,V:0,Z:0,D:0] TimeOutValue=10 Status=x807 Unicode Status=Locale=1
50805567.001 |07:35:39.676 |AppInfo |StationD: (0006382) DisplayNotify
timeOutValue=10 notify='x807' content='Not Enough Bandwidth' ver=85720014.
```

What should be fixed to resolve the issue?

A. class of service configuration
B. region configuration
C. geolocation configuration
D. codec configuration

Answer: B

NEW QUESTION 25
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 29
Which Cisco IM and Presence service handles failover and state changes in the cluster?

A. XCP Sync Agent
B. Cisco Server Recovery Manager
C. Cisco XCP Connection Manager
D. XCP router

Answer: B

NEW QUESTION 33
Refer to the exhibit.

```
voice class codec 20
codec preference 1 g722-64
codec preference 2 ilbc mod 30
!
dial-peer voice 200 voip
destination-pattern ^408555...$
session target ipv4:10.2.3.4
incoming called-number 9T
dtmf-relay h245-alphanumeric rtp-nte
no vad
!
```

An administrator configured a codec preference list with 0,122 and ILBC codecs. Which change must the administrator make in the dial-peer section of the configuration to use this list?

A. add voice-codecs 20
B. add session codec 20
C. add codec preference 20
D. add voice-class codec 20

Answer: D

NEW QUESTION 35
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 37
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 40
A remote office has a less-than-optimal WAN connection and experiences packet loss, delay and jitter. Which VoIP codec is used in this situation?

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

Answer: B

NEW QUESTION 45

What are two common attributes of XMPP XML stanzas? (Chose two.)

A. from
B. to
C. destination
D. version
E. Source

Answer: AB

NEW QUESTION 46
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 48
Which command must be defined before an administrator changes the linecode value on an ISDN T1 PRI in slot 0/2 on an IOS-XE gateway?

A. isdn incoming-voice voice
B. pri-group timeslots 1-24
C. card type t1 0 2
D. voice-port 0/2/0:23

Answer: C

NEW QUESTION 51
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 53
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 55
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 57

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

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

Answer: D

NEW QUESTION 58
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 59
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 63
Refer to the exhibit.

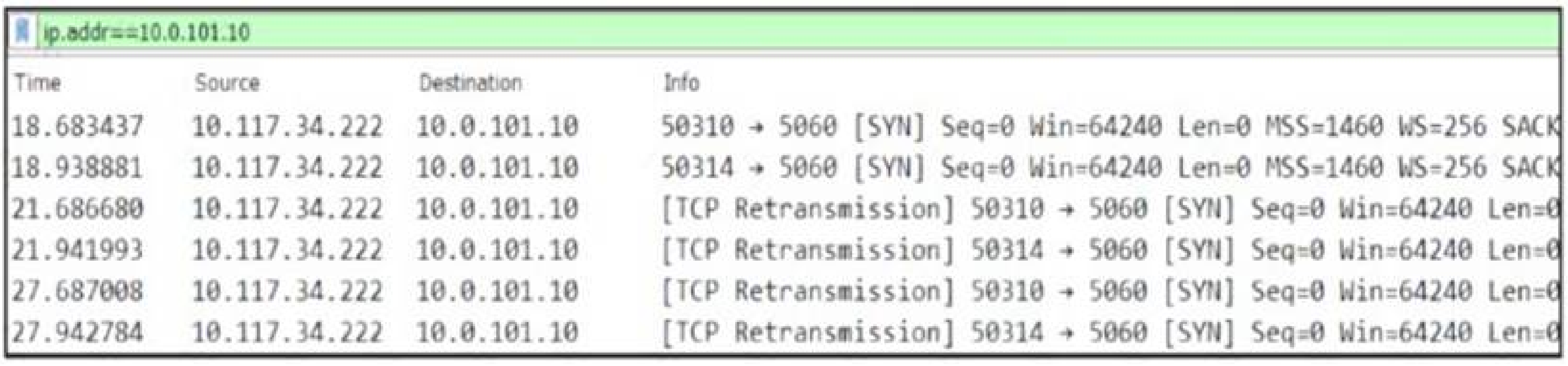
```
Via: SIP/2.0/TCP
10.10.10.2:5060;branch=a8bH5bK7954A198F
From:
<sip:012345678@10.10.10.2>;tag=8D79AF62-DB2
To: <sip:90123456@10.10.4.14>;
tag=811681~ffa80926-5fac-4cc5-b802-
2dbde74ae7w2-
v=0
o=CiscoSystemsCCM-SIP 811681 1 IN IP4
10.10.4.14
s=SIP Call
c=IN IP4 10.5.4.3
t=0 0
m=audio 27839 RTP/AVP 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 64
Refer to the exhibit.


ip.addr==10.0.101.10

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 66
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

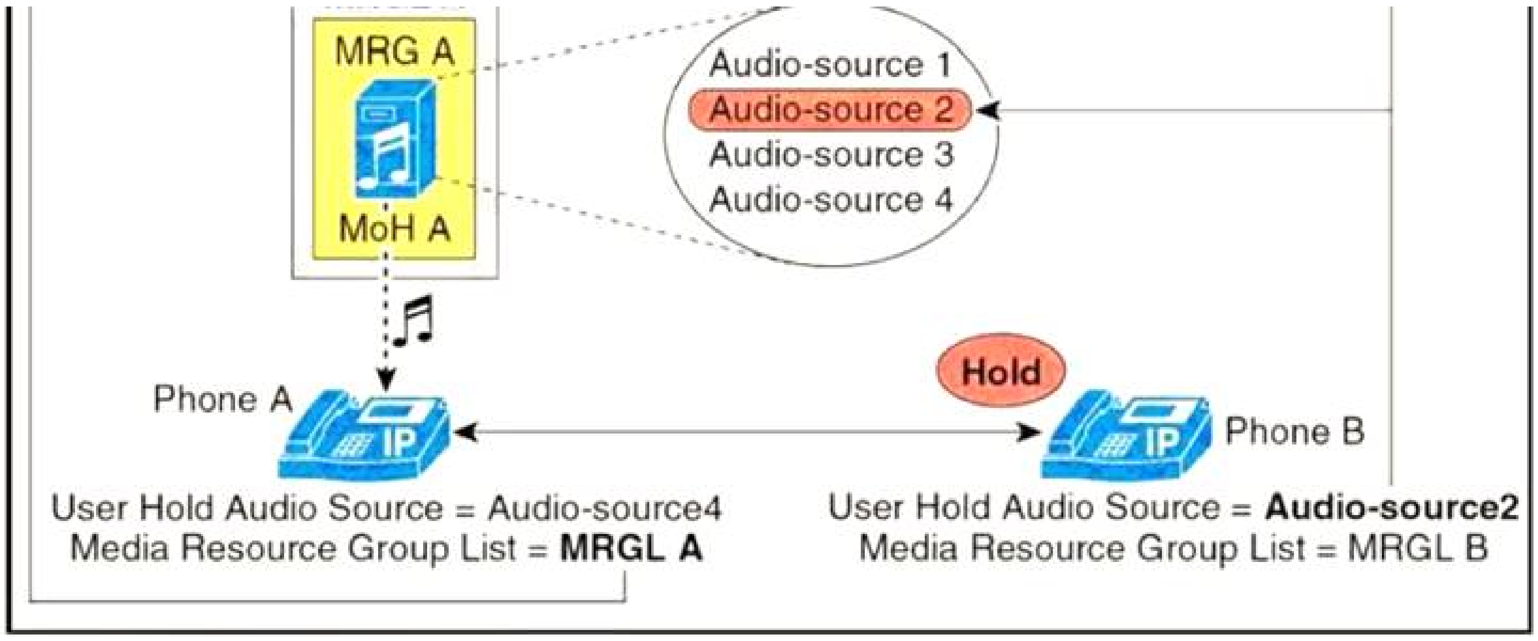
Answer: D

NEW QUESTION 67
An administrator is configuring a new Cisco UCM with PSTN capabilities. Due to bandwidth constraints, audio compression is used on the codec. DTMF must work as expected because the customer is calling many call centers where the users must select options in the call. Where is DTMF out-of-band in a CCM 12.5 with SIP-based gateway configured?

A. in the DTMF setting under SIP profile on the Cisco Unified Border Element
B. in the dial peer on the Cisco IOS router
C. in regions on the Cisco UCM where the appropriate codec to use is set
D. in DTMF settings in the audio codec preference list under regions in the Cisco UCM

Answer: B

NEW QUESTION 71
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 76

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

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

Answer: C

NEW QUESTION 78
An administrator has been asked to implement toll fraud prevention in Cisco UCM Which tool is used to begin this process?

A. Cisco UCM class of service
B. Cisco Unified Mobility
C. Cisco UCM Access Control Group restrictions
D. Cisco Unified Real-Time Monitoring Tool

Answer: A

NEW QUESTION 82
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 84
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 85
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 90
Refer to the exhibit.

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

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

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

Answer: B

NEW QUESTION 93
Refer to the exhibit.

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

What causes the PRI issue?

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

Answer: B

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

NEW QUESTION 97
Which external DNS SRV record must be present for Mobile and Remote Access?

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

Answer: B

NEW QUESTION 98
An administrator uses the Cisco Unified Real-Time Monitoring Tool to investigate recent calls on a Cisco UCM cluster. The SIP trace for an on-net. direct-media call shows two 180 Ringing and two 11 BYE messages. Why are there multiples of each message type in the trace?

A. The source phone sends a 180 Ringing signal to the Cisco UC
B. which sends a 180 Ringing signal to the destination phon
C. The same process applies to 11 BYE messages.
D. The source phone must signal to the destination phone that it is ringing, and the destination phone signals back with a 180 Ringing messag
E. The same process applies to 11 BYE messages.
F. The calls have an MTP in the call path due to different codec suppor
G. The calls are subsequently split into two call legs.
H. The destination phone signals back to the Cisco UCM that it is ringing, and the Cisco UCM signals back to the source phone.

Answer: A

NEW QUESTION 99
Which configuration on Cisco UCM is required for SIP MWl to work?

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

Answer: B

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

NEW QUESTION 100
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 105
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 108
The chief officer at a company must reduce collaboration infrastructure costs by onbording all on-permises equipment to the cloud by using CISCO Webex Control Hub. Administrators need the ability to manage upgrades and set up hot desking for on-premises devices.
Which action must be taken before on boardng devices by using the Control Hub?

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

Answer: D

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

NEW QUESTION 109
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 110
An engineer wants to manually deploy a CISCO Webex DX80 Video endpoint to a remote user. Which type of provisioning is configured on the endpoint?

A. Cisco Unified Border Element
B. Cisco Unity Connection
C. Cisco Meeting Server
D. Edge

Answer: D

Explanation:
The Cisco Webex DX80 Video endpoint can be provisioned in two ways:

- Automatically, using the Cisco Unified Communications Manager (CUCM) or Cisco Video Communication Server (VCS)
- Manually, using the Edge provisioning mode

The Edge provisioning mode is used when the endpoint is not connected to the CUCM or VCS. In this mode, the endpoint is configured with the necessary settings, such as the IP address, SIP/H.323 parameters, and time and date.
The Cisco Unified Border Element (Cisco UBE) is a network element that provides security and call control for IP telephony networks. The Cisco Unity Connection is a unified messaging system that provides voicemail, email, and fax services. The Cisco Meeting Server is a video conferencing system that provides high-quality video and audio conferencing.

NEW QUESTION 114
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 115
An administrator configures the voicemail feature in a Cisco collaboration deployment. The user mailboxes must be configured when the Cisco Unity Connection server is configured. Which action accomplishes this task?

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

Answer: C

NEW QUESTION 119
An administrator is in the process of moving Cisco Unity Connection mailboxes between mailbox stores. The administrator notices that some mailboxes have active Message Waiting Indicators. What happens to these mailboxes when they are moved?

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

Answer: B

NEW QUESTION 121
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 124
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 127
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 128
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 129
The security department will audit an IT department to ensure that the proper guidelines are being followed. The reports of the call detail records show unauthorized access to PSTN. Which two actions should an administrator check to prevent the unauthorized use of the telephony system? (Choose two.)

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

Answer: BE

NEW QUESTION 130
When multiple potential patterns are present, which two things are considered when Cisco UCM selects a destination pattern? (Choose two.)

A. The pattern matches the shortest explicit prefix.
B. The pattern does not match the dialed string.
C. The pattern represents the smallest number of endpoints.
D. The pattern matches the dialed string.
E. The pattern represents the largest number of endpoints.

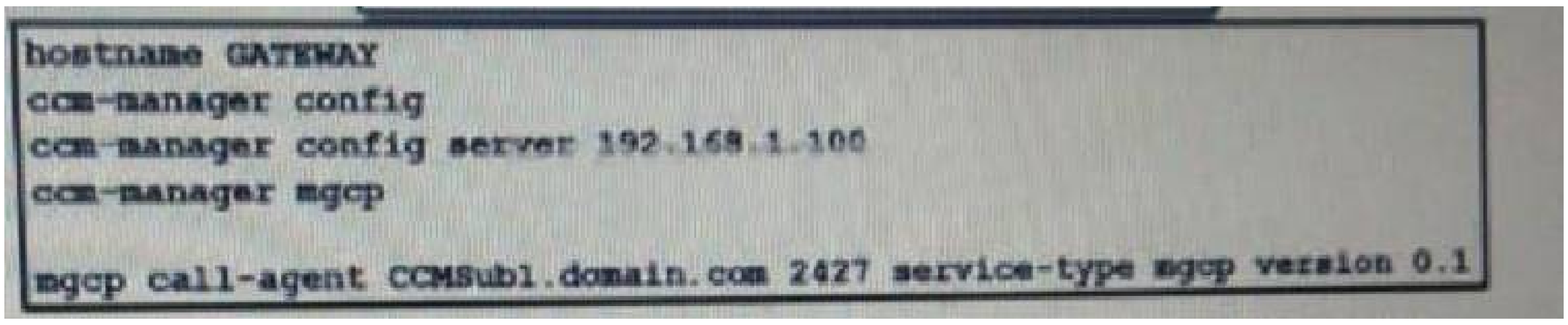
Answer: AD

NEW QUESTION 135
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 138
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 142
An engineer roust 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
Which task is required when configuring self-provisioning for an end user in Cisco UCM?

A. Enable Auto-Registration.
B. Associate the end user to the Standard CCM Super Users group
C. Associate the end user to a SIP Profile.
D. Disable Auto-Registration.

Answer: A

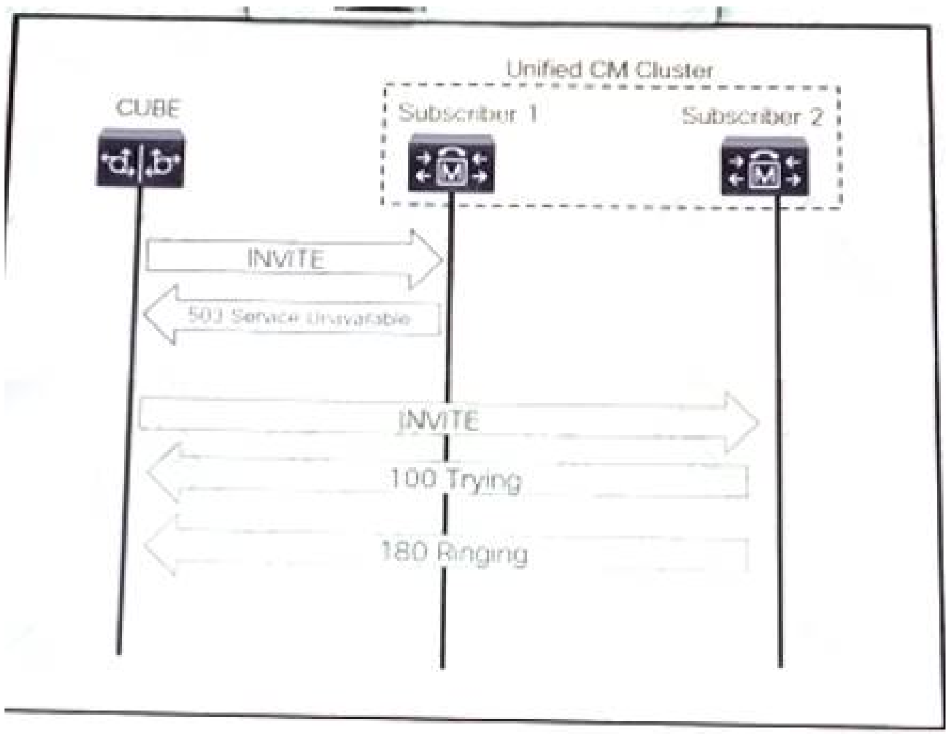
NEW QUESTION 150
What are two access management mechanisms in Cisco Webex Control Hub? (Choose two.)

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

Answer: AB

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

NEW QUESTION 151
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 152
An engineer configures local roule group names to simplify a dial plan. Where does the engineer set the route groups according to the local route group names that are configured?

A. route list
B. device pool
C. CSS
D. route pattern

Answer: B

NEW QUESTION 157
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 158
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 162
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 164
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 168
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.

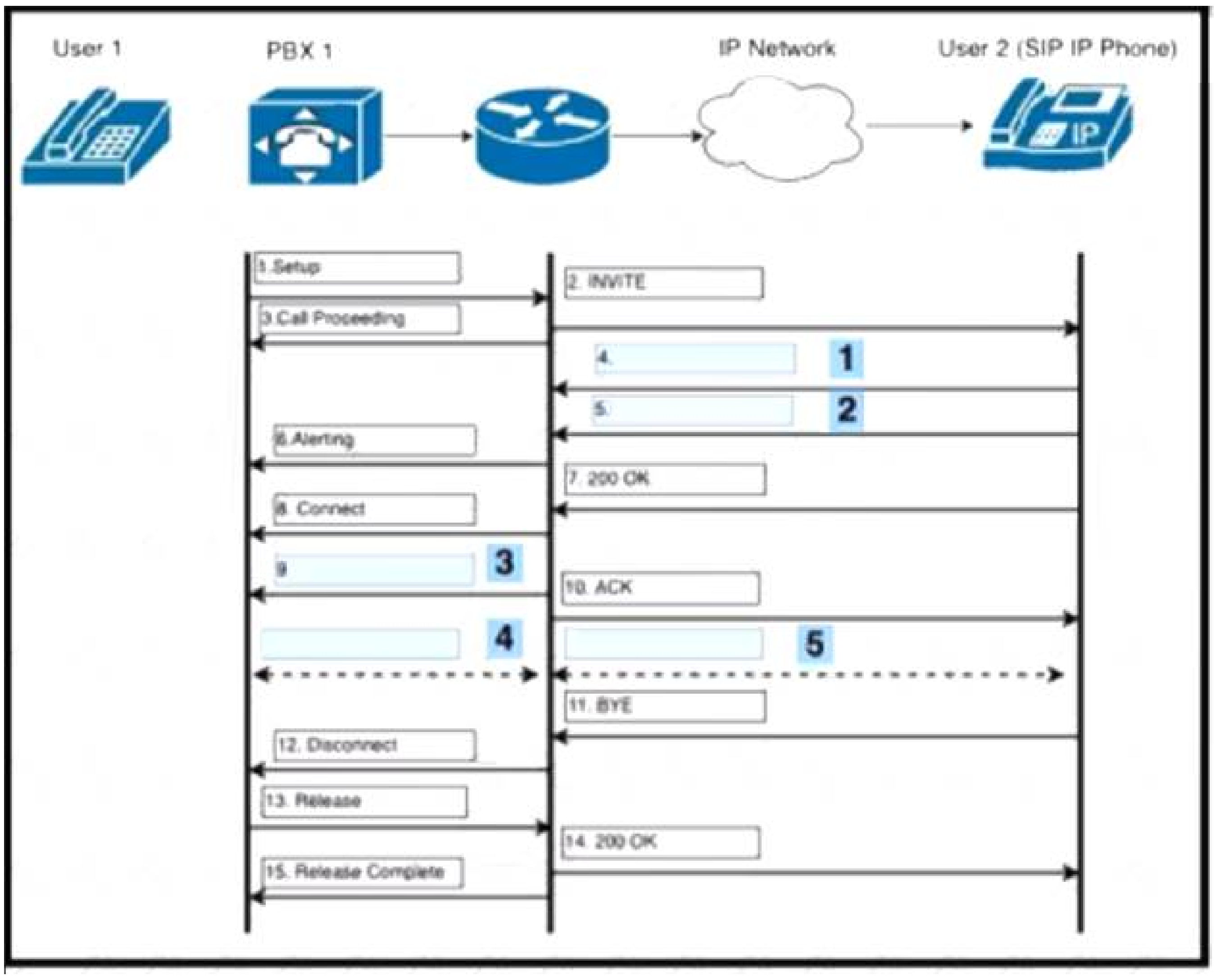
Answer: D

NEW QUESTION 169
End users report bad video quality and voice choppiness on Cisco Collaboration endpoints. The engineer changed the device pool the users were in but did not correct the problem. Which action should be taken to troubleshoot this issue?

A. Use direct IP address calls between two endpoints to troubleshoot call quality issues.
B. Restart the Cisco Location Bandwidth Manager service on the Cisco UCM publisher.
C. Check for duplex/speed mismatches between the network port settings of the system and network switch.
D. Set the service parameter Use Video Bandwidth Pool for Immersive Video Calls to "false".

Answer: D

NEW QUESTION 171
Refer to the exhibit.



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

two-way voice path

two-way RTP channel

100 Trying

Connect ACK

180 Ringing

A. Mastered
B. Not Mastered

Answer: A

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

NEW QUESTION 173
An engineer encounters third-party devices that do not support Cisco Discovery Protocol. What must be configured on the network to allow device discovery?

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

Answer: A

Explanation:
LLDP (Link Layer Discovery Protocol) is a vendor-neutral network discovery protocol that is used to discover the topology of a network. LLDP is similar to CDP (Cisco Discovery Protocol), but it is not proprietary to Cisco. LLDP is supported by a wide range of network devices, including switches, routers, and firewalls.

To configure LLDP on a network, you must enable LLDP on the devices that you want to discover. You can then use a network management tool, such as Cisco Network Assistant, to view the topology of the network.
The other options are incorrect. TFTP (Trivial File Transfer Protocol) is a network protocol that is used to transfer files between devices. LACP (Link Aggregation Control Protocol) is a network protocol that is used to aggregate multiple network links into a single logical link. SNMP (Simple Network Management Protocol) is a network protocol that is used to manage network devices.

NEW QUESTION 176
Which action prevent 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 180
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 182
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 186
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

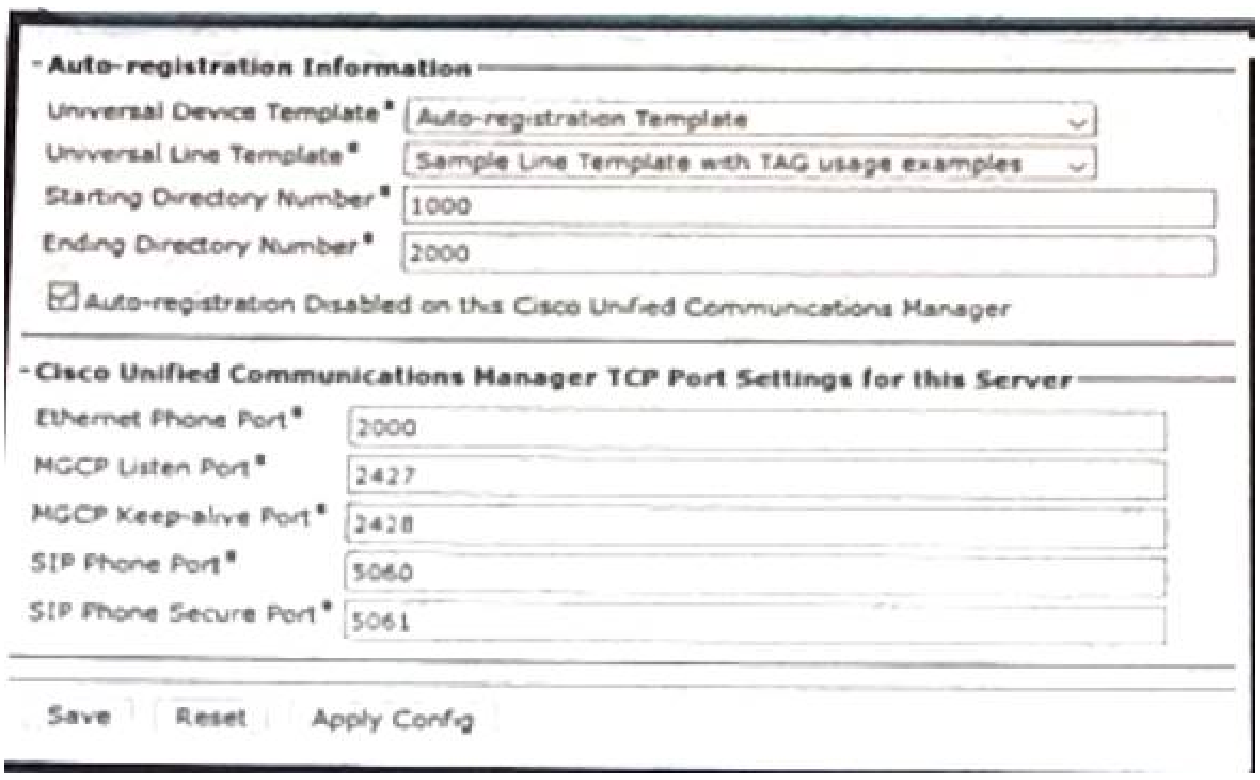
```
!
voice translation-rule 1
rule 1 /^1917/ //
rule 2 /^[+]1917/ //
!
voice translation-profile strip+1
translate calling 1
!
```

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 190
Refer to the exhibit.

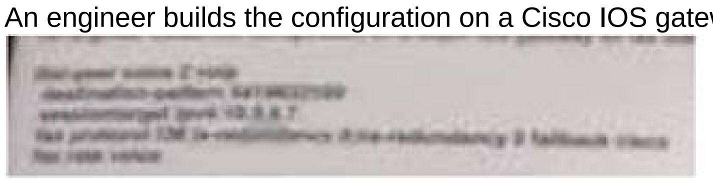


Which action must an engineer take to implement self-provisioning on a primary communications manager server?

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

Answer: C

NEW QUESTION 194
An engineer builds the configuration on a Cisco IOS gateway for the dial-peers:



Which command is required to complete the configuration?

A. Codec g726r32

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

Answer: D

NEW QUESTION 195
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 199
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

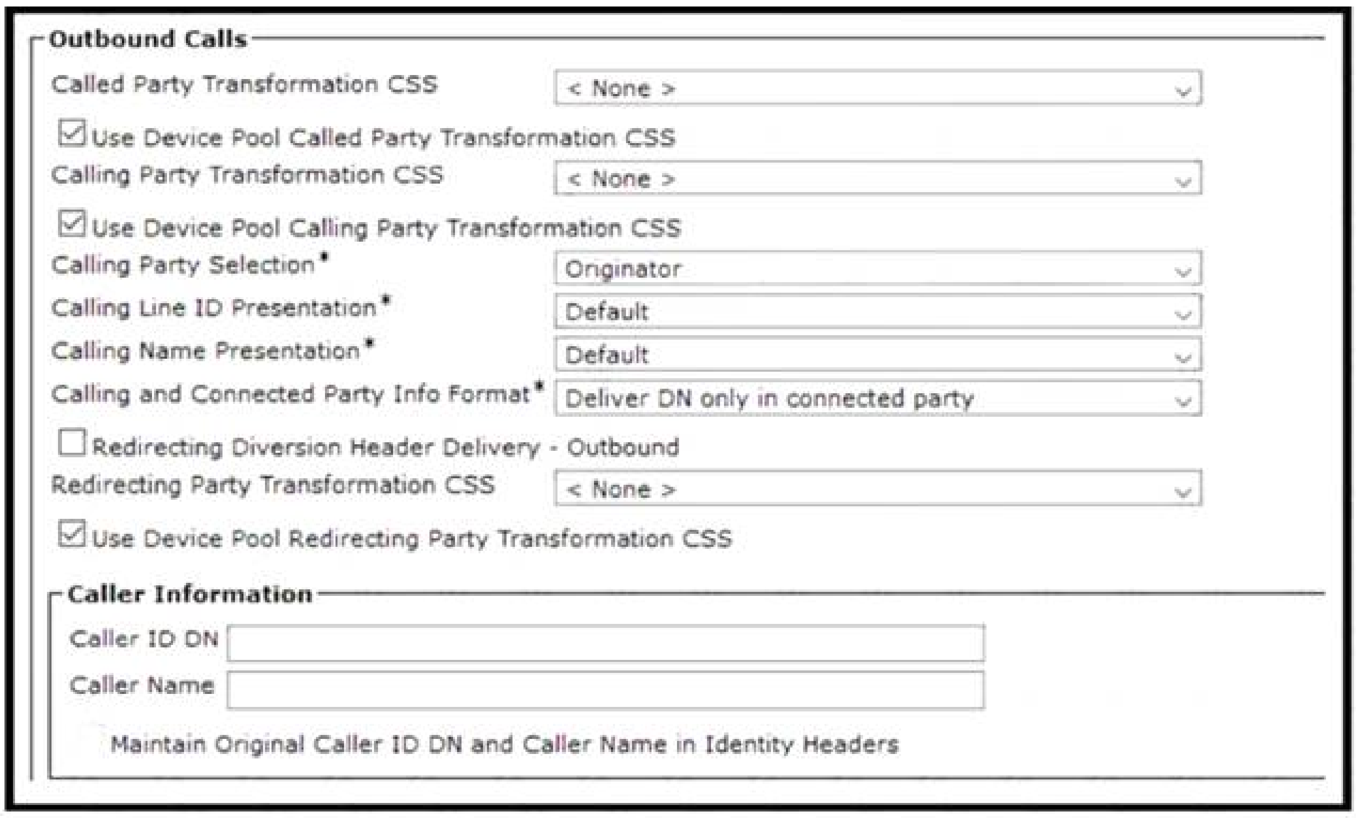
Answer: A

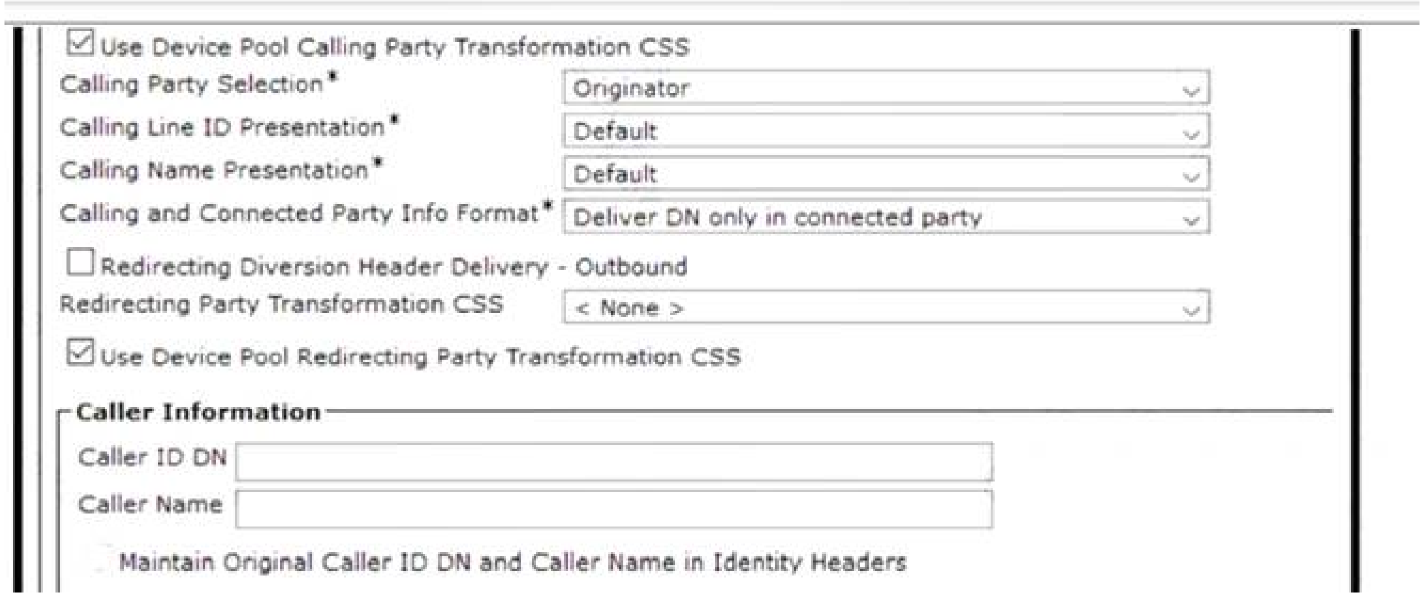
NEW QUESTION 202
Which IP Precedence value is used to classify a call signalling packet?

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

Answer: D

NEW QUESTION 207
Refer to the exhibit.





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 208
An engineer troubleshoots a Cisco Jabber login problem on a Windows PC. The login fails with the error message "Cannot find your services automatically. Click advanced settings to set up manually " Which action should the engineer take first?

A. Verify whether the cup-xmpp certificates are valid.
B. Verify the username and password and try again.
C. Perform a manual DNS lookup of SRV record _cisco-uds._tcp.domain.com.
D. Perform a manual DNS lookup of SRV record _collab-edge._tls.domain.com.

Answer: C

NEW QUESTION 211
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 213
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 214
A Company s employees have been complaining that they have been unable to select options on the internal IVR of the help desk. IT support has been given Cisco UCM traces and below is the snippet of the SDP of the INVITE packet.

```
m=audio 25268 RTP/AVP 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

How is this issue resolved?

A. Configure CODEC for G.729.
B. Configure DTMF for KPML.
C. Configure CODEC for G.722.
D. Configure DTMF for RFC 2833.

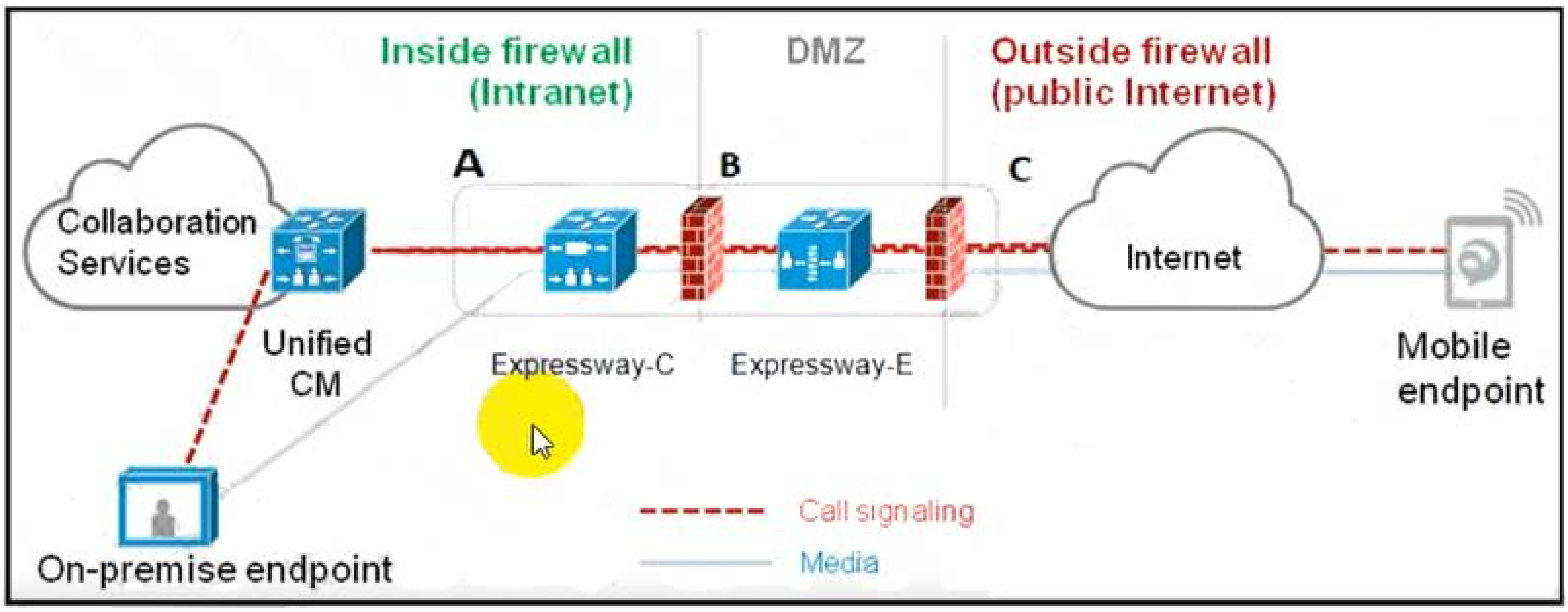
Answer: B

NEW QUESTION 218
Cisco UCM delays routing of a call during digit analysis with an overlapping dial plan. How long is the default wait time?

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

Answer: C

NEW QUESTION 223
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 227
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 228
An administrator must implement toll fraud prevention on Cisco UCM using these parameters:
• Enable Forced Authorization Code 112211.
• Set an authorization level of 3 for the route pattern 8005551212.
• Require no access code to dial 10-digit numbers. How must the route pattern be implemented?

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

Answer: A

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

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

NEW QUESTION 232
Which two DNS records must be created to configure Service Discovery for on-premises Jabber? (Choose two.)

A. _cisco-uds._tls.cisco.com pointing to the IP address of Cisco UCM
B. _cuplogin_tcp.cisco.com pointing to a record of IM and Presence
C. _cuplogin._tls.cisco.com pointing to the IP address of IM and Presence
D. _cisco-uds.tcp.cisco.com pointing to a record of Cisco UCM
E. _xmpp.tls.cisco.com pointing to a record of IM and Presence

Answer: BD

NEW QUESTION 236
Refer to the exhibit.

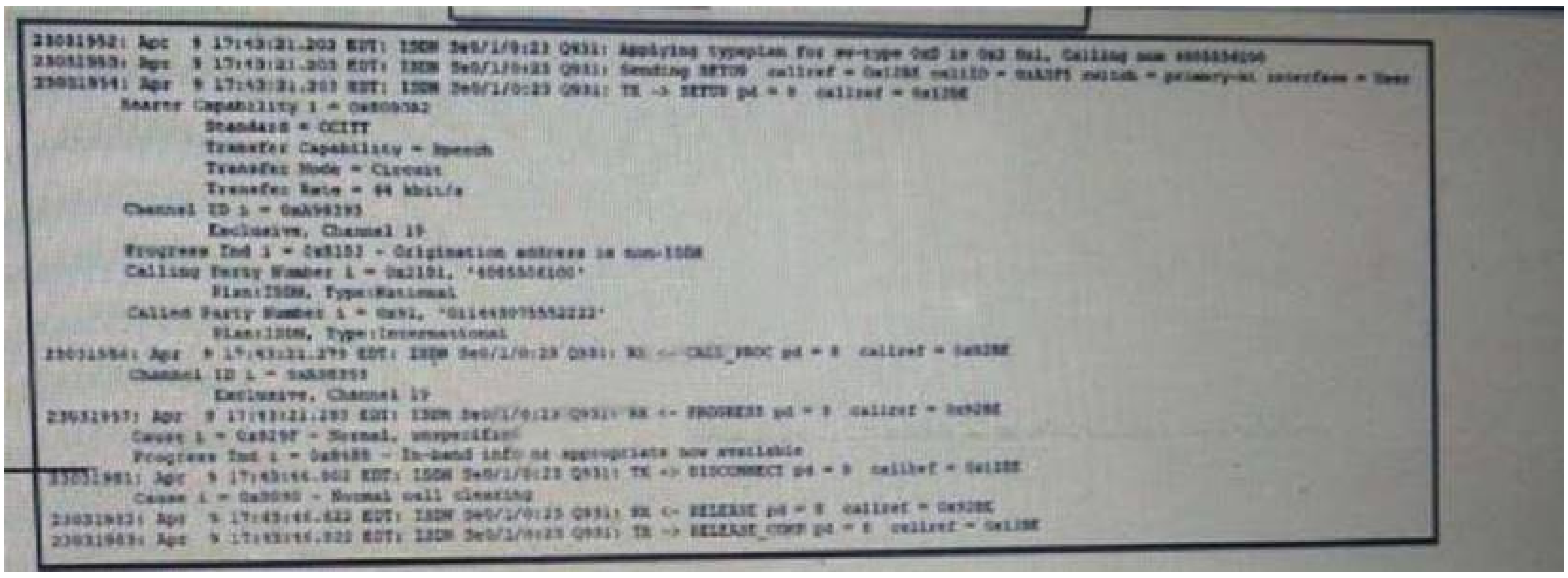
```
!
voice service voip
 ip address trusted list
  ipv4 192.168.100.101
  ipv4 192.168.101.0 255.255.255.128
!
dial-peer voice 1 voip
 destination-pattern +T
 session protocol sipv2
 session target ipv4:192.168.102.102
 dtmf-relay rtp-nte
 codec g711ulaw
 no vad
!
```

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 239
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 240
Which two technical reasons make QoS a necessity in a video deployment? (Choose Two)

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

Answer: CD

NEW QUESTION 243
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 244
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 245
The IP phones al a customer site do not pick an IP address from the DHCP An engineer must temporarily disable LLDP on all ports of the switch to leave only CDP. Which two commands accomplish this task? (Choose two.)

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

Answer: BC

NEW QUESTION 249

When setting a new primary DNS server in the Cisco UCM CLI what is required for the change to take affect?

A. restart of CallManager service
B. restart of DirSync service
C. restart of the network service
D. restart of TFTP service

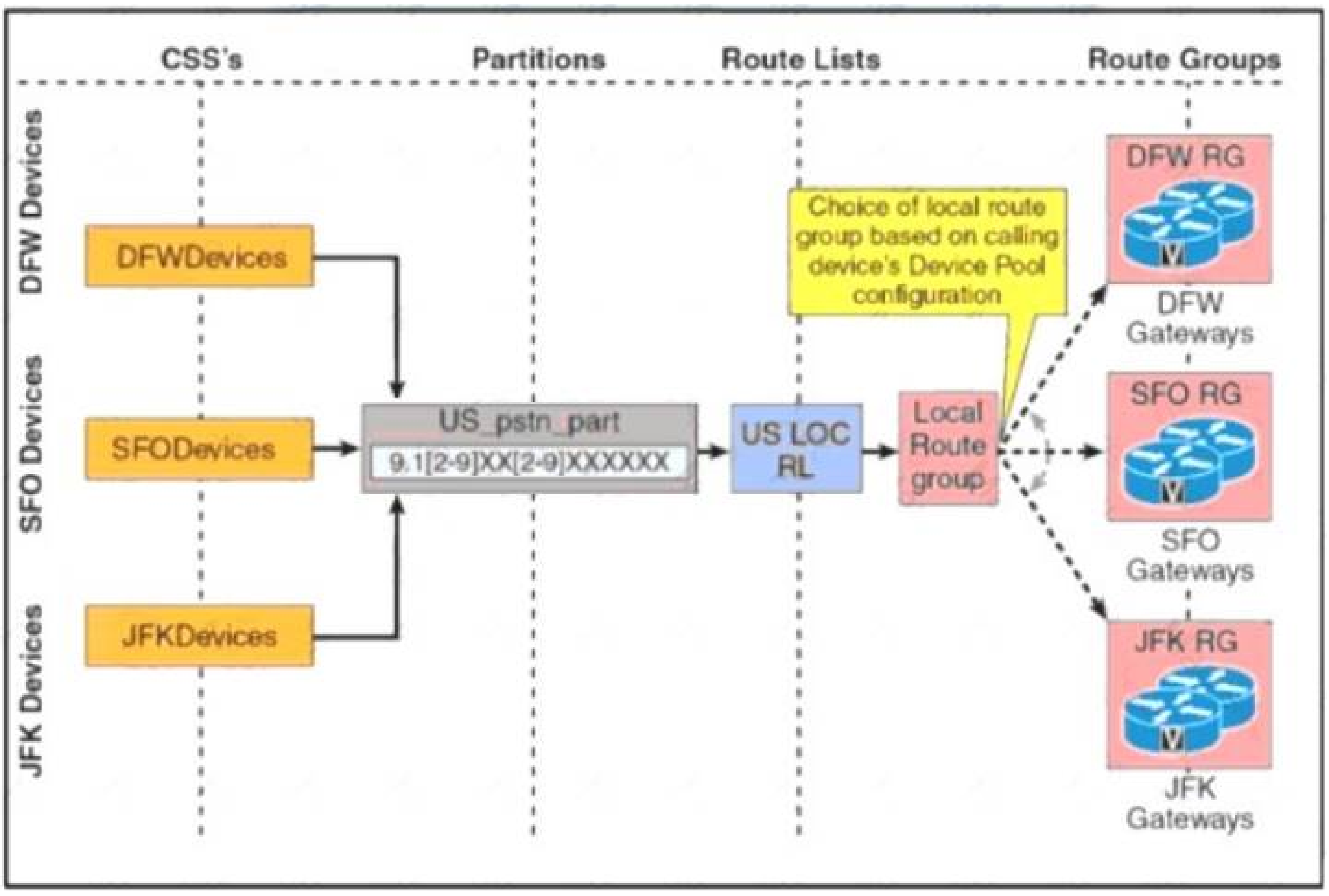
Answer: C

NEW QUESTION 253
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 256
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 Yor
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 whilestill keeping the number and resources assigned in San Francisco.
D. The call fails because the Standard Local Route Group is being used only it 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 259
Refer to the exhibit.

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

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

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

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

Answer: C

NEW QUESTION 263
Refer to the exhibit.

```
Server: Cisco-SIPGateway/IOS-15.4.3.S4
CSeq: 101 OPTIONS
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
Allow-Events: telephone-event
Accept: application/sdp
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Content-Type: application/sdp
Content-Length: 369

v=0
o=CiscoSystemsSIP-GW-UserAgent 6414 4717 IN IP4 10.8.140.23
s=SIP Call
c=IN IP4 10.8.140.23
t=0 0
m=audio 0 RTP/AVP 18 0 8 4 15
c=IN IP4 10.8.140.23
m=image 0 udptl t38
c=IN IP4 10.8.140.23
a=T38FaxVersion:0
a=T38MaxBitRate:9600
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:320
a=T38FaxUdpEC:t38UDPRedundancy
```

A customer wants the SIP 200 OK shown to advertise codecs in the following order:

- G.729
- G.711u
- G.711a
- G.723
- G.728

After correcting the codec preferences. What should the audio payload show in the SIP Traces?

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

Answer: D

NEW QUESTION 267
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

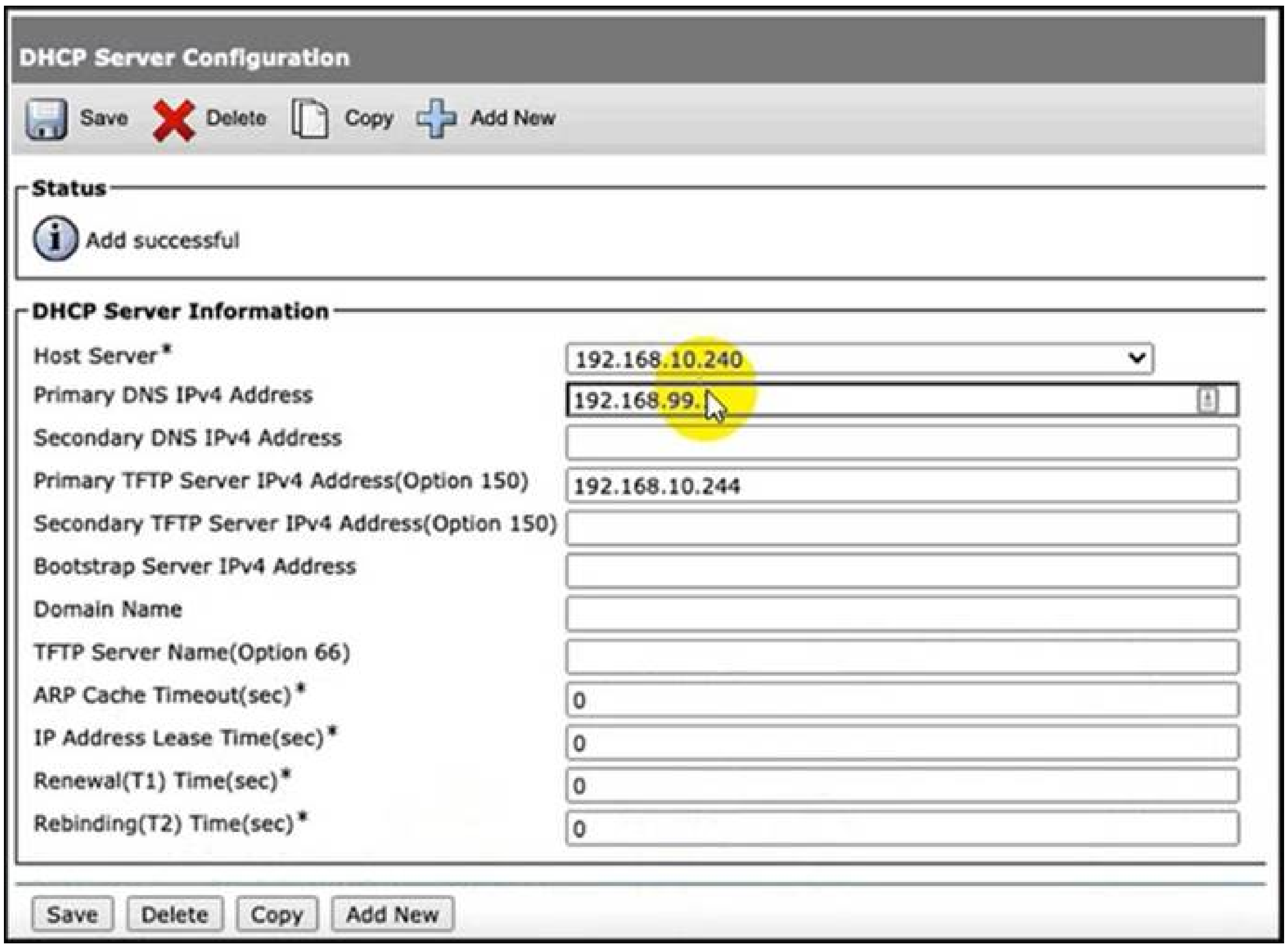
NEW QUESTION 270
What is a capability of the call forwarding feature in a Cisco Webex dial plan?

A. device pool selection
B. Call Admission Control
C. business continuity
D. ringtone selection

Answer: C

Explanation:
Call forwarding is a feature that allows users to forward incoming calls to another number. This can be useful in a number of situations, such as when a user is not available to take a call, or when a user wants to forward calls to a different number during certain times of the day.
Call forwarding can be used to improve business continuity by ensuring that calls are always answered, even if the user is not available. For example, if a user is out of the office, they can forward their calls to their voicemail or to another employee. This ensures that customers and clients can always reach someone, even if the user is not available.

NEW QUESTION 274
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 278
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 283
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 286
What are the last two bits of a DS field in DiffServe Byte used for?

A. INC
B. AFxy
C. ECN
D. RMI

Answer: C

NEW QUESTION 290
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 295
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 299
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

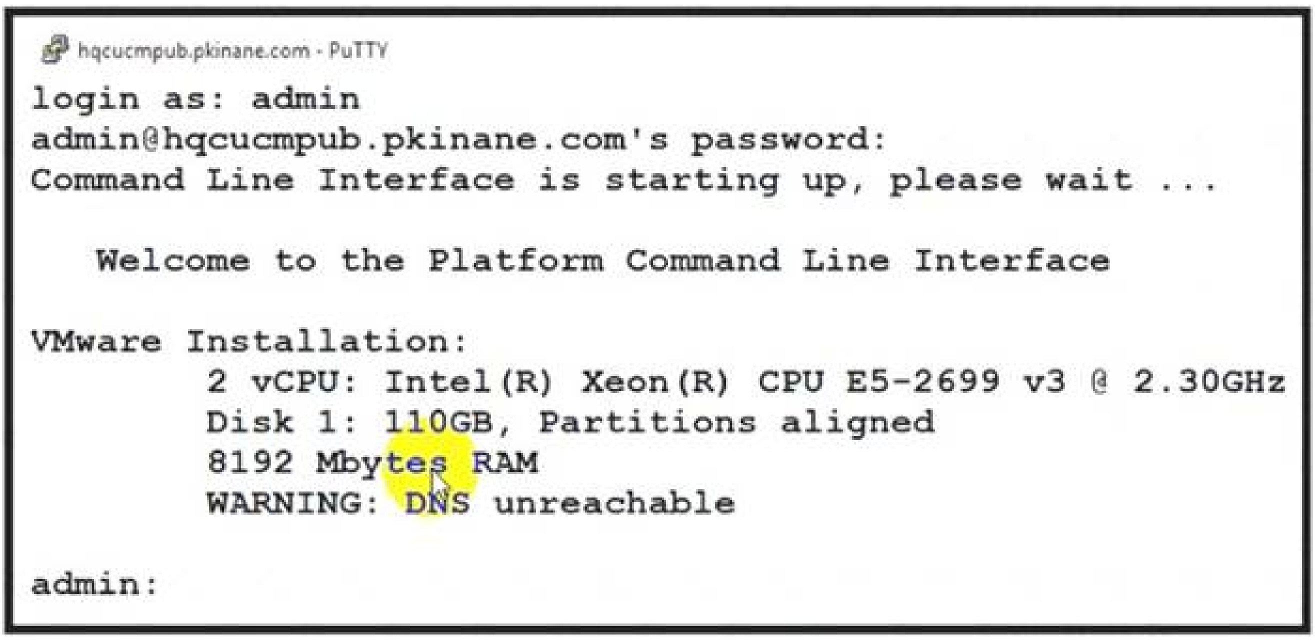
Answer: B

NEW QUESTION 301
What happens to voice packets from a Cisco 8845 IP phone in the QoS trust boundary?

A. The voice packets are not trusted, and the access layer switch reclassifies the packets.
B. The voice packets are classified by the phone, and the classification is accepted
C. The voice and access layer switch negotiate the classification of packets
D. Cisco UCM determines how the voice packers are classified.

Answer: B

NEW QUESTION 306
Refer to the exhibit.



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

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

Answer: DE

NEW QUESTION 307
Which DHCP option must be set up tor new phones to obtain the TFTP server IP address?

A. option 66
B. option 15
C. option 6
D. option 120

Answer: A

NEW QUESTION 312
Refer to the exhibit.

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no;privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B8
To: <sip:4000@172.16.1.1>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4EA4363-B77111E9-8A4AFFCF-10B6D71B@172.16.2.143
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE:  1800
Cisco-Guid: 0082391505-3077640681-2319777743-0280418075
User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8486 8298 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
c=IN IP4 172.16.2.143
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

This INVITE is sent to an endpoint that only supports G.729. What must be done for this call to succeed?

A. Add a transcoder that supports G.711ulaw and G.729.
B. Nothing; both sides support G.729.
C. Add a media termination point that supports G.711ulaw and G.729.
D. Nothing both sides support payload type 101.

Answer: A

NEW QUESTION 313
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 314
Which two types of trunks can be used when configuring a hybrid Local Gateway for Cisco Webex Calling? (Choose Two.)

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

Answer: AC

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

NEW QUESTION 318
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 323
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 326
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 331
......

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