



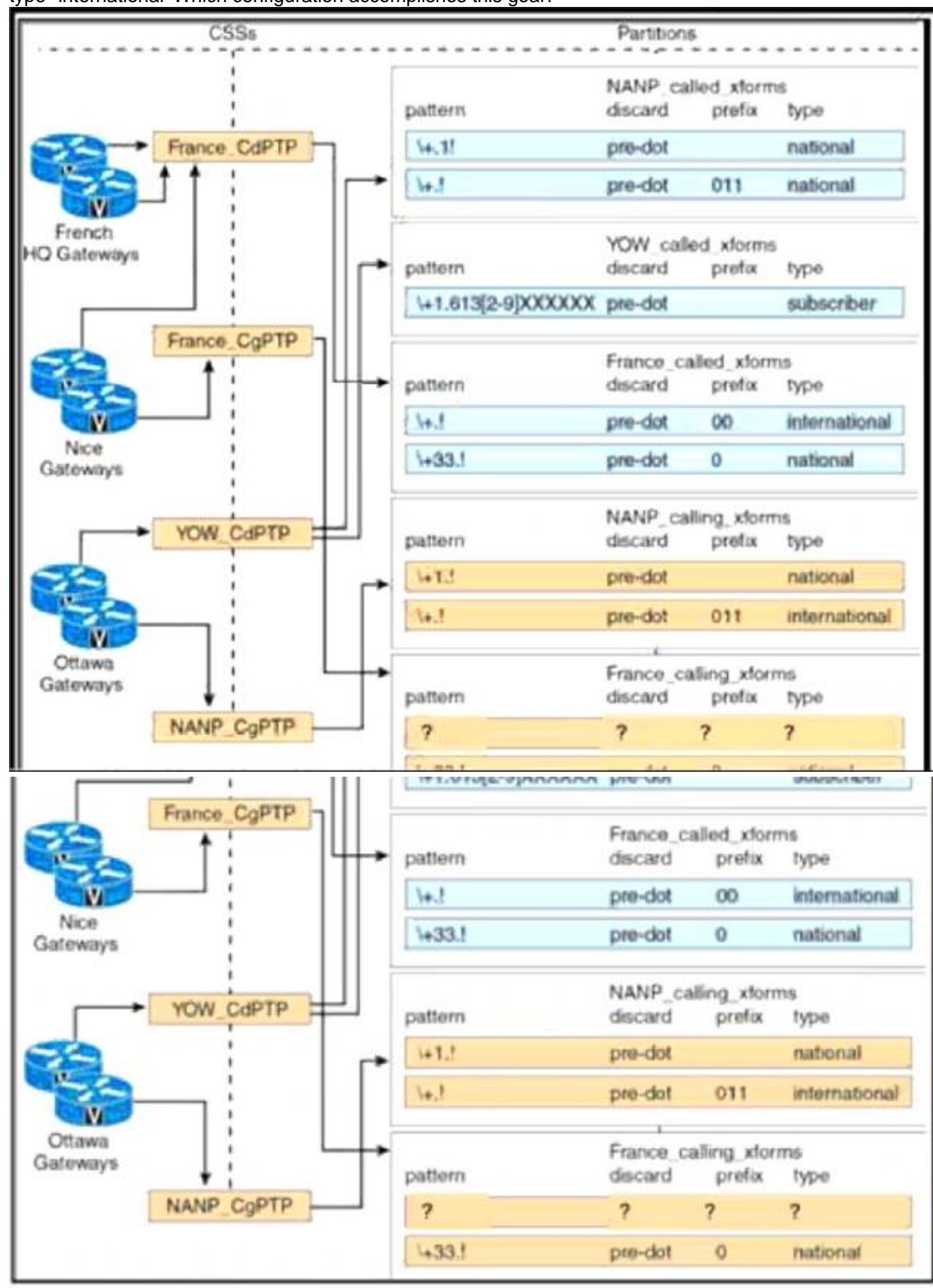
**Cisco**

## **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 international

Answer: C

### NEW QUESTION 2

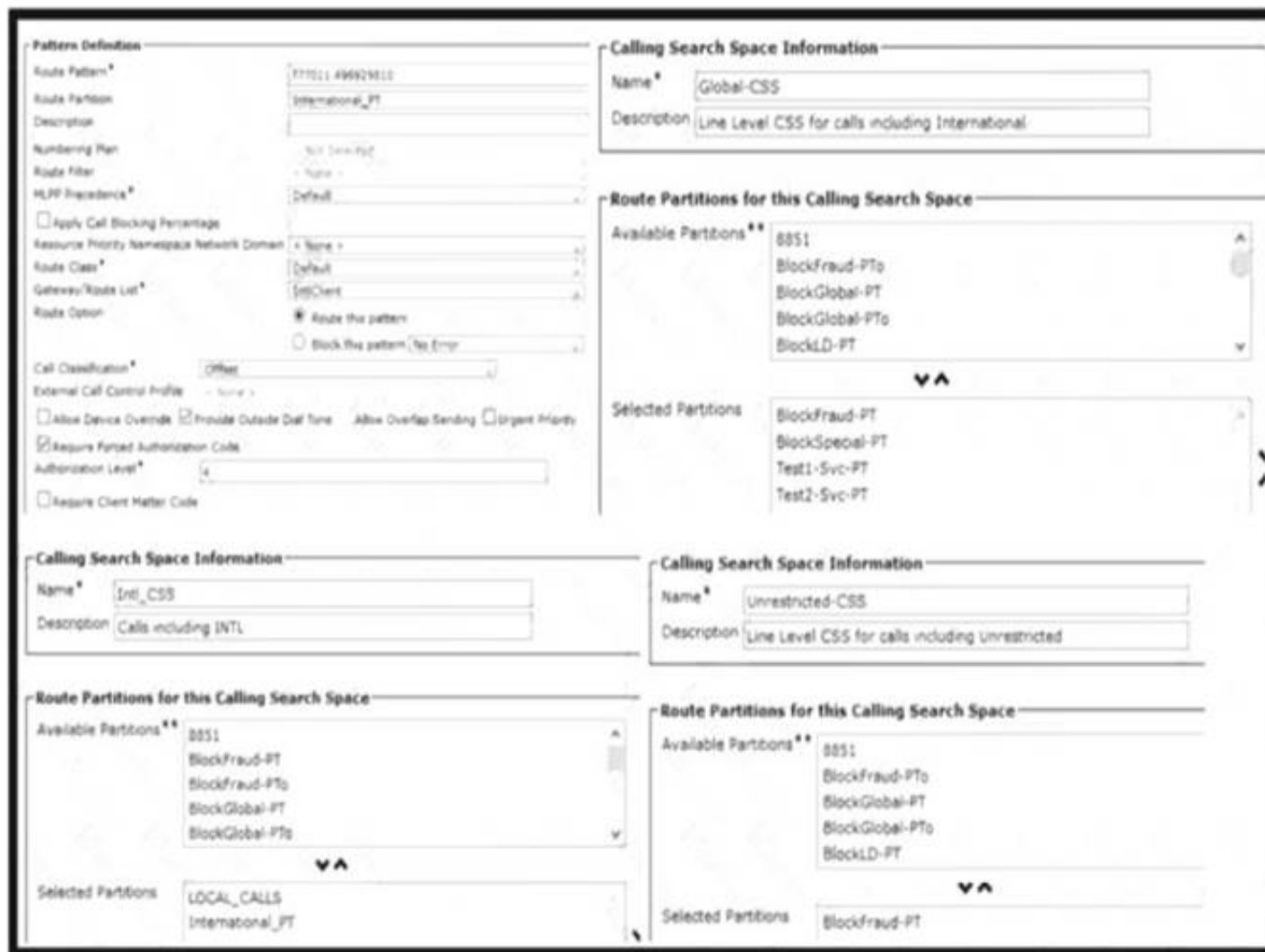
What happens when a Cisco IP phone loses connectivity to the cluster during an active call?

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

Answer: D

### NEW QUESTION 3

Refer to the exhibit.



The image shows four screenshots of Cisco IOS configuration pages:

- Route Pattern Configuration:** Shows the configuration for route pattern 777011.496929810. The route partition is set to 'International\_PT'. The route class is 'Default'. The route option is 'Route this pattern'.
- Calling Search Space Information:** Shows the configuration for 'Global-CSS' with the description 'Line Level CSS for calls including International'.
- Route Partitions for this Calling Search Space:** Shows the available partitions for 'Global-CSS' and the selected partitions: 'BlockFraud-PT', 'BlockSpecial-PT', 'Test1-Svc-PT', and 'Test2-Svc-PT'.
- Calling Search Space Information:** Shows the configuration for 'Intl\_CSS' with the description 'Calls including INTL'.
- Route Partitions for this Calling Search Space:** Shows the available partitions for 'Intl\_CSS' and the selected partitions: 'LOCAL\_CALLS' and 'International\_PT'.
- Calling Search Space Information:** Shows the configuration for 'Unrestricted-CSS' with the description 'Line Level CSS for calls including unrestricted'.
- Route Partitions for this Calling Search Space:** Shows the available partitions for 'Unrestricted-CSS' and the selected partitions: 'BlockFraud-PT'.

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

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 5

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 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

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 8

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 9

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 10

Callers from a branch report getting busy tones intermittently 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 |StationOutputDisplayNotify |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 10

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 12

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 14

An engineer is configuring a phone system CISCO UCM and wants to activate TFTP service. The engineer selects the serviceability page for configuration. Which nodes configurable for TFTP?

- A. any two nodes
- B. any node
- C. only nodes that have Cisco UCM service enabled
- D. any subscriber nodes

**Answer:** C

#### Explanation:

TFTP is a network protocol that is used to transfer files between devices. It is often used to transfer firmware and configuration files to network devices. In order to use TFTP, the device must have a TFTP server configured. In Cisco UCM, the TFTP server is configured on the serviceability page. The TFTP server can be configured on any node that has Cisco UCM service enabled. The TFTP server cannot be configured on nodes that do not have Cisco UCM service enabled.

#### NEW QUESTION 16

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

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

**Answer:** A

#### NEW QUESTION 18

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 19

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

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

**Answer:** D

#### Explanation:

According to the Cisco Community website<sup>1</sup>, the phone uses DNS to resolve the hostname of the CUCM server (cucm1.cisco.com) to its IP address. DNS is a network service that translates domain names into IP addresses.

#### NEW QUESTION 24

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

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

**Answer:** BC

#### NEW QUESTION 27

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

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

**Answer:** A

#### Explanation:

SRST, or Survivable Remote Site Telephony, is a feature that allows Cisco IP phones to continue to function even when the connection to the Cisco UCM publisher is lost. When SRST is configured, the phones will automatically reregister to the publisher when the connection is restored.

Route groups are used to route calls to different destinations based on the caller's phone number or other criteria. Cisco UCM is the call management system that controls the IP phones. Device pools are used to group phones together for administrative purposes.

#### NEW QUESTION 28

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 31

Which attribute contains an XMPP stanza?

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

**Answer:** A

#### NEW QUESTION 36

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 event<sup>12</sup>. You can also install the diagnostic signatures to automate diagnostics data collection and transfer collected data to the Cisco TAC case to accelerate resolution time<sup>12</sup>.

#### NEW QUESTION 41

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 46

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 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 47

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 50

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 51

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 56

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 when 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 is 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 UCM. 1. Auto-registration allows new phones to register with Cisco UCM without manual configuration. If auto-registration is disabled, the phone will not be able to register and will show a rejected status. 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 UCM. To resolve this issue, the ITL file needs to be deleted from the phone or exchanged between the clusters.
- 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 issues. To resolve this issue, the network connectivity and VLAN configuration need to be checked and fixed.
- 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 60

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 65

Refer to the exhibit.

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

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

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

**Answer: D**

#### NEW QUESTION 70

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

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

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

**Answer: AD**

#### NEW QUESTION 75

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 78

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 79

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

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

**Answer:** A

#### NEW QUESTION 84

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

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

**Answer:** AC

#### Explanation:

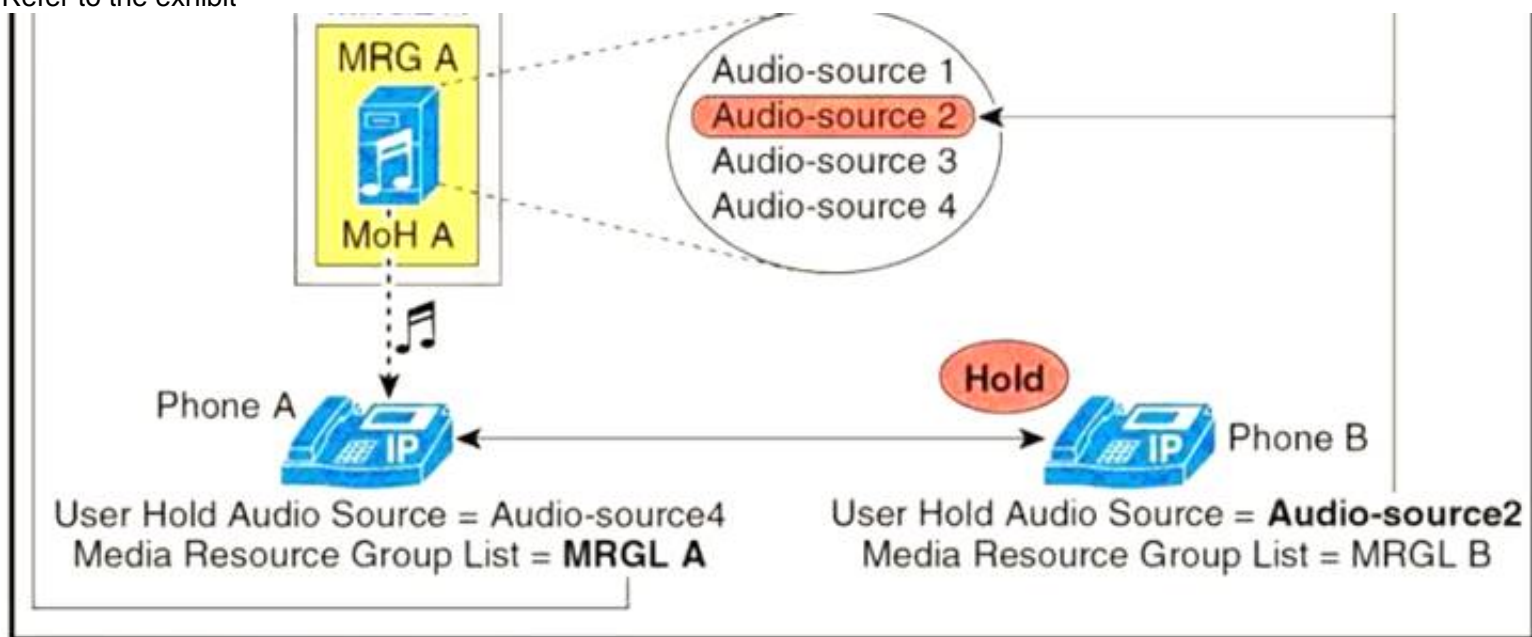
The Cisco Expressway Series provides the following functions:

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

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

#### NEW QUESTION 85

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 86

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 87

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

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

**Answer: B**

#### NEW QUESTION 91

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 94

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 96

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 101

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 102

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 107

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 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

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 117

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 119

The chief officer at a company must reduce collaboration infrastructure costs by onboarding 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 boarding devices by using the Control Hub?

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

**Answer:** D

#### Explanation:

This is a prerequisite for using the Device Connector tool, which allows you to onboard and register several devices simultaneously to the Webex Control Hub1. The Device Connector tool creates a workspace, an activation code, and activates all of your devices in one go1. This way you don't need to be physically present in the same room to activate the devices.

The other options are not required before onboarding devices by using the Control Hub:

- Configuring the Control Hub organization ID on the devices is not necessary, as the Device Connector tool will send the device information to your Webex organization and generate activation codes for them 1.
- Acquiring a license for each device is not necessary, as you can assign licenses to users and devices after they are registered to the Webex Control Hub2.
- Allowing HTTP traffic from each device to Control Hub is not necessary, as HTTPS connectivity is required for the Device Connector tool to communicate with the devices1.

#### NEW QUESTION 121

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

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

**Answer:** B

#### NEW QUESTION 123

An engineer implements QoS in the enterprise network. Which command is used to verify the classification and marking on a Cisco IOS switch?

- A. show class-map interface GigabitEthernet 1/0/1
- B. show policy-map interface GigabitEthernet 1/0/1
- C. show access-lists
- D. show policy-map

**Answer:** B

#### NEW QUESTION 127

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-
maxcapture=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 failing to establish between two SIP Devices. The called device answers with these SIP. Which SIP parameter causes the issue?

- A. The calling device did not offer aptime 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 aptime value: Theptime value is used to specify the amount of time between each media packet. If the calling device does not offer aptime 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 131

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 134**

How are network devices monitored in a collaboration network?

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

**Answer: C**

**NEW QUESTION 137**

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 141**

A Cisco UCM administrator sets up new route patterns to support phones in four 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 145**

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

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

**Answer: AC**

**NEW QUESTION 147**

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 152**

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 157

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

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

**Answer:** B

#### Explanation:

To restrict off-net to off-net call transfers, an administrator can set the "Block Offnet to Offnet Transfer" service parameter to "On". This will prevent users from transferring calls from one external number to another external number.

The other options are not correct because:

- A. Enforce ad-hoc conference restrictions: This will prevent users from creating ad-hoc conferences, but it will not prevent them from transferring calls.
- C. Implement time-of-day routing: This will allow calls to be routed to different destinations based on the time of day, but it will not prevent users from transferring calls.
- D. Use the correct route filters: This will allow calls to be filtered based on the destination, but it will not prevent users from transferring calls.

#### NEW QUESTION 161

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 162

SIP proxies have operations defined in RFC 3261 and supporting extensions. Though no IETF RFC completely defines how SBCs must function. SBCs evolved over the years.

Which two operations demonstrate the high-level differences between SBCs and SIP proxies? (Choose two.)

- A. Stateful proxies are context-aware and can terminate communication sessions by themselves
- B. SIP proxies add a Via header and optionally a Record-Route header, and the rest of the headers are left untouched
- C. SBCs can modify headers such as To, From, Contact, and Call-ID. It can introduce new headers into the SIP message
- D. SBCs are capable of interworking completely different protocols to set up, modify, and tear down communication session
- E. It includes SIP, H.323, and MGCP protocols
- F. SIP proxies are SDP-aware and can change the SDP bodies

**Answer:** BD

#### NEW QUESTION 163

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

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

**Answer:** CE

#### NEW QUESTION 168

An engineer must manually provision a Cisco IP Phone 8845 using SIP. Which two fields must be configured for a successful provision? (Choose two.)

- A. media resources group list
- B. CSS
- C. location
- D. device security profile
- E. SIP profile

**Answer:** DE

#### NEW QUESTION 171

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

- A. dtmf-relay sip-kvml 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-nre 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 172**

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 174**

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

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

**Answer:** C

**NEW QUESTION 175**

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 176**

What is the traffic classification for voice and video conferencing?

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

**Answer:** B

**NEW QUESTION 180**

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 184**

Refer to the exhibit.

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

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

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

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

**Answer: B**

#### NEW QUESTION 189

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 194

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 195

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 200

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 203

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 204

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  

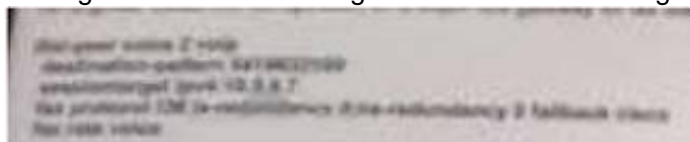
```

!
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 209

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 211

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 214

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 217

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

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

**Answer:** A

#### NEW QUESTION 218

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 221

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

- A. debug isdn q921term mon
- B. debug voip 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 223

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 227

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 230

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 234

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 237

Refer to the exhibit. An engineer is configuring class of control for a user in Cisco UCM. Which change will ensure that the user is unable to call 2143?

- A. Change line partition to Partition\_A
- B. Change line CSS to only contain Partition\_B
- C. Set the user's line CSS to <None>
- D. Set the user's device CSS to <None>

**Answer:** D

#### NEW QUESTION 242

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 243

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 244

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

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

**Answer:** A

#### NEW QUESTION 247

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

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

**Answer:** AE

#### NEW QUESTION 251

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
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 253

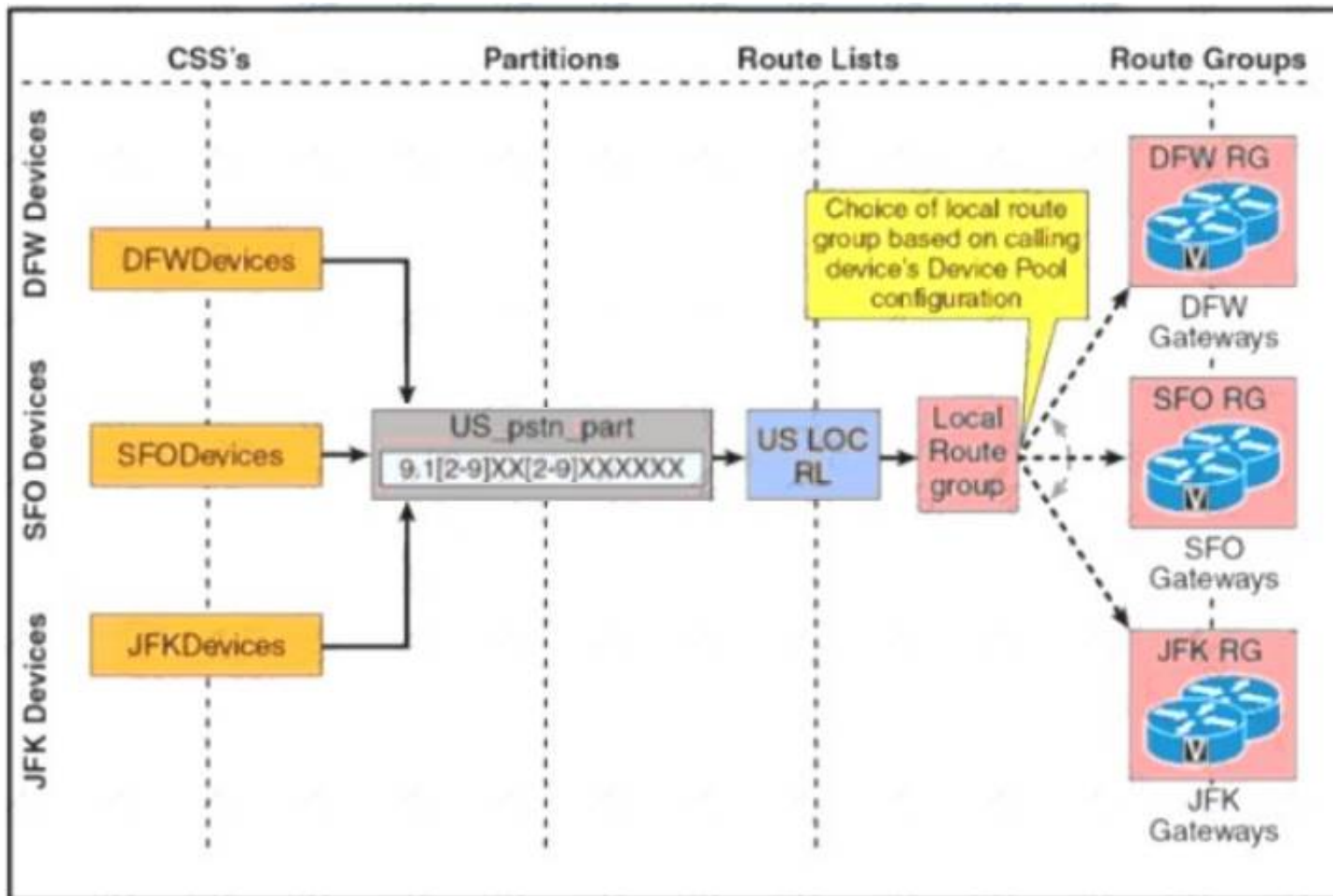
An administrator is configuring LDAP for Cisco UCM with Active Directory integration. A customer has requested to use "ipphone" instead of "telephoneNumber" as the phone number attribute. Where does the administrator specify this attribute mapping in Cisco UCM?

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

**Answer:** B

#### NEW QUESTION 257

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 262

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 properly 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.

```
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 264

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 268

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





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

**Answer: B**

#### NEW QUESTION 271


Refer to the exhibit.

### DHCP Server Configuration

 Save
  Delete
  Copy
  Add New

---





#### Status

 Add successful

---

#### DHCP Server Information

Host Server *	192.168.10.240
Primary DNS IPv4 Address	192.168.99.1
Secondary DNS IPv4 Address	
Primary TFTP Server IPv4 Address (Option 150)	192.168.10.244
Secondary TFTP Server IPv4 Address (Option 150)	
Bootstrap Server IPv4 Address	
Domain Name	
TFTP Server Name (Option 66)	
ARP Cache Timeout (sec) *	0
IP Address Lease Time (sec) *	0
Renewal (T1) Time (sec) *	0
Rebinding (T2) Time (sec) *	0

 Save
  Delete
  Copy
  Add New

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 273

Refer to the exhibit.

```
admin:utils ntp status
ntpd (pid 17428) is running...
```

remote	refid	st	t	when	poll	reach	delay	offset	jitter
*192.168.1.1	17.253.14.125	2	u	36	64	377	0.435	0.039	0.047
192.168.1.2	.INIT.	16	u	-	64	0	0.000	0.000	0.000

A collaboration engineer adds a redundant NTP server to an existing Cisco Collaboration solution On the Cisco UCM OS Administration page, the new NTP server shows as "Not Accessible" Which action resolves this issue?

- A. Restart NTPD on the Cisco UCM server.
- B. Delete and re-add the new NTP server via the Cisco UCM command-line interface
- C. Start the NTP service on the new NTP server
- D. Configure the "reach" value as "377" for the new NTP server.

**Answer: C**

#### NEW QUESTION 275

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 276

Which two parameters influence the total number of supported conference participants on a Cisco IOS XE router that has DSP modules? (Choose two.)

- A. voice codec
- B. session capacity of the PVDM module
- C. number of protocol data units
- D. software version Of the router
- E. license types

**Answer:** AB

**Explanation:**

These are two parameters that influence the total number of supported conference participants on a Cisco IOS XE router that has DSP modules<sup>2</sup>. A voice codec is a software algorithm that compresses and decompresses voice signals for transmission over a network<sup>3</sup>. Different voice codecs have different bandwidth requirements and quality levels<sup>3</sup>. A PVDM module is a hardware component that provides digital signal processing (DSP) resources for voice applications such as conferencing and transcoding<sup>1</sup>. A PVDM module has a fixed session capacity, which is the maximum number of voice channels that it can support simultaneously<sup>1</sup>.

**NEW QUESTION 277**

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 279**

Refer to the exhibit.

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

000143: *Apr 23 19:41:49.050: MGCP Packet sent to 192.168.100.101:2427--->
200 4
I:
X: 2
L: p:10-20, a:PCMU:PCMA:G.nX64, b:64, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-220, a:G.729:G.729a:G.729b, b:8, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-110, a:G.726-16:G.728, b:16, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-70, a:G.726-24, b:24, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-50, a:G.726-32, b:32, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-270, a:G.723.1-H:G.723:G.723.1a-H, b:6, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-330, a:G.723.1-L:G.723.1a-L, b:5, e:on, qc:l, s:on, t:l0, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, contest, data, netwloop, netwtest
<---
```

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

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

**Answer:** D

**NEW QUESTION 284**

Refer to the exhibit.

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

An administrator configures a secure SIP trunk on Cisco UCM.

Which value is needed in the secure certificate subject or subject alternate name field to accomplish this task?

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

**Answer:** B

#### NEW QUESTION 288

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 292

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 295

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. utils diagnose module validate Network



Answer: DE

#### NEW QUESTION 298

Refer to the exhibit.

```
voice translation-rule 1
 rule 1 /^[2-9].....$/ /\0/ type any subscriber
 rule 2 /^[2-9]..[2-9].....$/ /\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 300

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 301

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-B77111E9-8A4AFCF-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:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtptime:0 PCMU/8000
a=rtptime: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 304

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 308

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 311

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 316

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 321

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 322

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 325

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 327

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 330

A Cisco IP Phone 7841 that is registered to a Cisco Unified Communications Manager with default configuration receives a call setup message. Which codec is negotiated when the SDP offer includes this line of text?

M=audio 498181 RTP/AVP 0 8 97

- A. G.711ulaw
- B. iLBC
- C. G.711alaw
- D. G.722

**Answer:** A

#### Explanation:

The SDP offer includes the following line of text: M=audio 498181 RTP/AVP 0 8 97

This line of text indicates that the following codecs are available:

- 0: G.711ulaw
- 8: G.711alaw
- 97: iLBC

The Cisco IP Phone 7841 is registered to a Cisco Unified Communications Manager with default configuration. This means that the phone will negotiate the G.711ulaw codec.

The G.711ulaw codec is a standard codec that is used for voice communication. It is a low-bandwidth codec that provides good quality.

The iLBC codec is a newer codec that is designed for use in low-bandwidth environments. It provides good quality, but it is not as widely supported as the G.711ulaw codec.

The G.722 codec is a high-quality codec that is used for voice communication. It provides excellent quality, but it requires more bandwidth than the G.711ulaw codec.

#### NEW QUESTION 331

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

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

**Answer:** AB

#### NEW QUESTION 332

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 334**

.....



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