

Cisco

Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies



NEW QUESTION 1

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

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

Answer: D

NEW QUESTION 2

Refer to the exhibit.

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

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

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

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

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

Answer: BC

NEW QUESTION 3

Refer to the exhibit.

The screenshot displays three configuration panels for route partitions in Cisco IOS:

- Top Panel (Global-CSS):**
 - Pattern Definition:** Route Pattern: 777011.496929810, Route Partition: International_PT, Description: , Numbering Plan: , Route Filter: , HLR Precedence: Default, Apply Call Blocking Percentage: ☐, Resource Priority Namespace Network Domain: , Route Class: Default, Gateway/Route List: [No]Client, Route Option: ☒ Route this pattern, ☐ Block this pattern, No Empr, Call Classification: Other, External Call Control Profile: , Allow Device Override: ☐, Provide Outside Dial Tone: ☒, Allow Overlap Sending: ☐, Urgent Priority: ☐, Require Forward Authorization Code: ☒, Authorization Level: 4, Require Client Matter Code: ☐.
 - Calling Search Space Information:** Name: Global-CSS, Description: Line Level CSS for calls including International.
 - Route Partitions for this Calling Search Space:**
 - Available Partitions: 8851, BlockFraud-PT, BlockFraud-PT, BlockGlobal-PT, BlockGlobal-PT, BlockLD-PT.
 - Selected Partitions: BlockFraud-PT, BlockSpecial-PT, Test1-Svc-PT, Test2-Svc-PT.
- Bottom Left Panel (Intl_CSS):**
 - Calling Search Space Information:** Name: Intl_CSS, Description: Calls including INTL.
 - Route Partitions for this Calling Search Space:**
 - Available Partitions: 8851, BlockFraud-PT, BlockFraud-PT, BlockGlobal-PT, BlockGlobal-PT, BlockLD-PT.
 - Selected Partitions: LOCAL_CALLS, International_PT.
- Bottom Right Panel (Unrestricted-CSS):**
 - Calling Search Space Information:** Name: Unrestricted-CSS, Description: Line Level CSS for calls including unrestricted.
 - Route Partitions for this Calling Search Space:**
 - Available Partitions: 8851, BlockFraud-PT, BlockFraud-PT, BlockGlobal-PT, BlockGlobal-PT, BlockLD-PT.
 - 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

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 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

Which two protocols can be configured for the Cisco Unity Connection and Cisco UCM integration? (Choose two.)

- A. 323
- B. SIP
- C. SCCP
- D. MGCP
- E. RTP

Answer: BC

Explanation:

The two protocols that can be configured for the Cisco Unity Connection and Cisco UCM integration are SIP and SCCP. SIP, or Session Initiation Protocol, is a

signaling protocol used for initiating, maintaining, and terminating real-time sessions, including voice, video, and messaging applications. SCCP, or Skinny Client Control Protocol, is a Cisco proprietary signaling protocol used for controlling Cisco IP phones. H.323 is an older signaling protocol that is no longer widely used. MGCP, or Media Gateway Control Protocol, is a protocol used for controlling media gateways. RTP, or Real-time Transport Protocol, is a protocol used for transporting real-time data, such as voice and video

NEW QUESTION 15

Which option must be used when configuring the Local Gateway for a Cisco Webex Calling trunk?

- A. local authentication
- B. certificate-based
- C. mutual TLS
- D. Auth-based

Answer: B

Explanation:

A certificate-based trunk is a type of trunk that uses certificates to authenticate the connection between Webex Calling and the Local Gateway¹. A Local Gateway is a supported session border controller that terminates the trunk on the premises². A certificate-based trunk requires a certificate authority (CA) to issue and manage the certificates for both Webex Calling and the Local Gateway¹.

NEW QUESTION 19

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

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

Answer: C

NEW QUESTION 21

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 25

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 26

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 27

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

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

Answer:

C

NEW QUESTION 32

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 33

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 35

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 39

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 44

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 49

Which type of message must an administrator configure in the SIP Trunk Security Profile for a Message Waiting Indicator light to work with a SIP integration between Cisco UCM and Cisco Unity Connection?

- A. Unsolicited NOTIFY
- B. 200 ok
- C. SIP Register
- D. TCP port 5060

Answer: A

NEW QUESTION 50

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

- A. 750 and 15.000 users
- B. 750 and 10.000 users
- C. 500 and 10.000 users
- D. 1000 and 10.000 users

Answer: D

NEW QUESTION 51

Where in Cisco UCM is restrictions on audio bandwidth configured?

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

Answer: C

NEW QUESTION 55

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

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

Answer: A

NEW QUESTION 56

On a Cisco Catalyst Switch, which command is required to send CDP packets on a switch port that configures a Cisco IP phone to transmit voice traffic in 802.10 frames, tagged with the voice VLAN ID 221?

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

Answer: A

NEW QUESTION 59

On which protocol and port combination does Cisco Prime Collaboration receive notifications (Traps and InformRequests) from several network devices in the Collaboration infrastructure for which it has requested notifications?

- A. UDP161
- B. TCP 161
- C. UDP 162
- D. TCP 80

Answer: C

NEW QUESTION 64

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

- A. Wait until 10 pm on March 4th when the user is automatically removed from Cisco UCM.
- B. Wait until 10 pm on March 5th when the user is automatically removed from Cisco UCM.
- C. Wait until 3:15 a.
- D. on March 6th for garbage collection to remove the user from Cisco UCM.
- E. Wait until 3:15am on March 5th for garbage collection to remove the user from Cisco UCM.

Answer: C

NEW QUESTION 68

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 73

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

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

Answer: AC

NEW QUESTION 74

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 79

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 80

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 84

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 85

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

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

Answer: B

NEW QUESTION 89

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 94

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 97

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 102

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 107

An engineer with ID012345678 must build an international dial plan in Cisco UCM. Which action is taken when building a variable-length route pattern?

- A. configure single route pattern for international calls
- B. set up all international route patterns to 0.!
- C. reduce the T302 timer to less than 4 seconds
- D. create a second route pattern followed by the # wildcard

Answer: D

Explanation:

When building a variable-length route pattern, you need to create a second route pattern followed by the # wildcard. This will allow the user to indicate the end of the number by dialing #. For example, if you want to create a route pattern for international calls, you would create a route pattern like this: 9.011!#

This route pattern will match any number that starts with 9.011, followed by any number of digits, and then ends with #.

The other options are incorrect because:

- Configuring a single route pattern for international calls will not allow the user to indicate the end of the number.
- Reducing the T302 timer to less than 4 seconds will not allow the user to indicate the end of the number.

NEW QUESTION 109

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 114

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 118

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 123

What is the purpose of Mobile and Remote Access (MRA) in the Cisco UCM architecture?

- A. MRA is used to access Webex cloud services only if authenticated with on-premises LDAP service.
- B. MRA is used to make secure PSTN calls by Cisco UCM only while on-premises authentication.
- C. MRA is used to make B2B calls through Expressway registration.
- D. MRA is used to access the collaboration services offered by Cisco UCM from off-premises network connections

Answer: D

NEW QUESTION 128

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 131

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 136

What are two reasons that AF41 is marked for the audio and video channels of a video call? (Choose two.)

- A. to prioritize video over other high -priority traffic classes
- B. to give video calls a higher priority than AP41 in the QoS policy
- C. to allow high-definition quality calls over low-speed links
- D. to preserve lip synchronization between the audio and video channels
- E. to provide separate classes for audio calls and video calls

Answer: DE

NEW QUESTION 137

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 141

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

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

Answer: B

Explanation:

The line partition must be inside the inbound CSS assigned to the CUC SIP trunk. This ensures that the SIP MWI messages are sent to the correct destination. The other options are incorrect because:

- Assigning an MWI extension on the mailbox is not required for SIP MWI to work.
- The line partition does not need to be inside the rerouting CSS assigned to the Cisco Unity Connection SIP trunk.
- Setting the "Enable message waiting indicator" on the port group is not required for SIP MWI to work.

NEW QUESTION 143

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 145

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 149

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 154

A SIP phone has been configured in the system with MAC address 0030.96D2.D5CB. The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

- A. SIP003096D2D5CB.cnf.xml
- B. SEP003096D2D5CB.cnf.xml
- C. SEP003096D2D5CB.cnf
- D. SIP003096D2D5CB.cnf

Answer: B

NEW QUESTION 159

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 164

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 169

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

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

Answer: D

NEW QUESTION 173

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 176

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 177

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 181

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 182

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

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

Answer: A

NEW QUESTION 186

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 added.
- 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 189

An engineer is configuring IP telephony. The network relies on DHCP to provide TFTP server addresses to the endpoints. Policy requires the endpoints to receive two server addresses. Which DHCP option must be configured?

- A. 66
- B. 143
- C. 150
- D. 166

Answer: C

NEW QUESTION 193

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 195

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

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

Answer: D

Explanation:

The Restrict setting in the Webex Calling dial plan prevents users from placing certain types of outbound calls. For example, you can use the Restrict setting to prevent users from making international calls or calls to premium-rate numbers.

The Block setting in the Webex Calling dial plan prevents users from placing any outbound calls. The Transfer to Number setting in the Webex Calling dial plan transfers all outbound calls to a specified number. The Reject setting in the Webex Calling dial plan rejects all outbound calls.

Here is a table summarizing the different dial plan settings and their effects:

Dial Plan Setting	Effect
Block	Prevents users from placing any outbound calls.
Transfer to Number	Transfers all outbound calls to a specified number.
Reject	Rejects all outbound calls.
Restrict	Prevents users from placing certain types of outbound calls.

NEW QUESTION 199

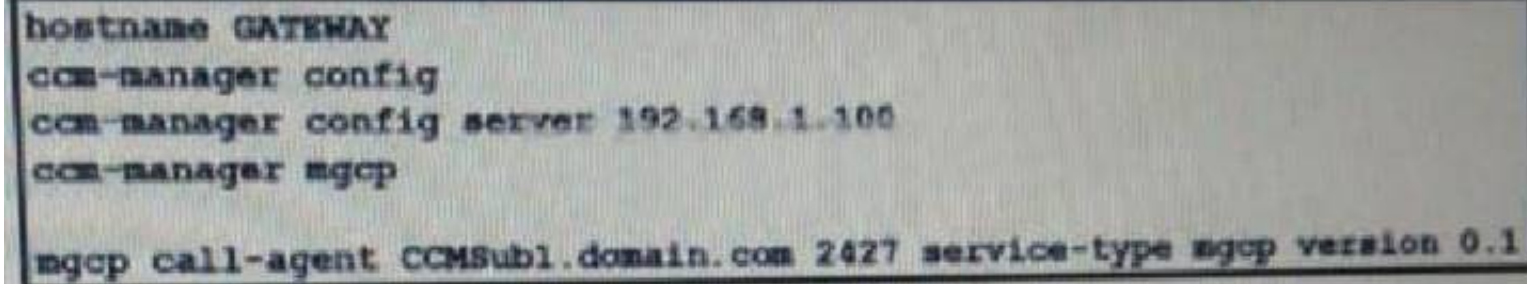
In which location does an administrator look to determine which subscriber the phone registers to if it loses registration with the current Cisco UCM subscriber?

- A. On Cisco UCM Administration Page Device > Phone > Phone Configuration page
- B. On Cisco UCM Administrator Page server > Cisco UCM
- C. On Cisco UCM Administrator page system > Device Pool > Cisco UCM group
- D. On Cisco UCM Administrator page system > Enterprise Parameters

Answer: C

NEW QUESTION 200

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 203

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 208

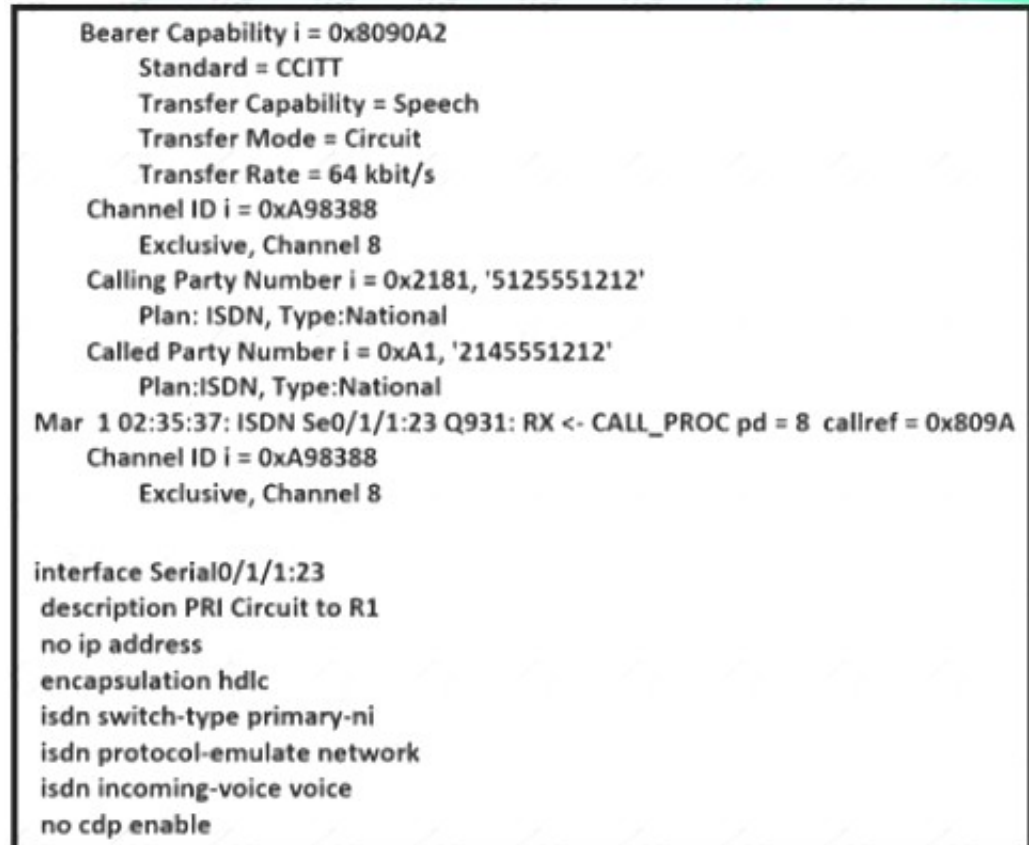
How many minutes does it take for automatic fallback to occur in a Presence Redundancy Group if the primary node lost a critical service?

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

Answer: C

NEW QUESTION 213

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 215

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 220

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 224

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 227

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

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

- A. Mastered
- B. Not Mastered

Answer: A

Explanation:

Table Description automatically generated

NEW QUESTION 231

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 236

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

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

Answer: D

NEW QUESTION 238

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 241

What is a characteristic of a SIP endpoint configured in Cisco UCM with 'Use Trusted Relay Point' set to 'On'?

- A. It creates a trust relationship with the called party.
- B. It enables the Use Trusted Relay Point setting from the associated common device configuration.
- C. It enables Cisco UCM to insert an MTP or transcoder designated as a TRP.
- D. If TRP is allocated and MTP is also required for the endpoint
- E. calls fail.

Answer: C

NEW QUESTION 243

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 247

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 251

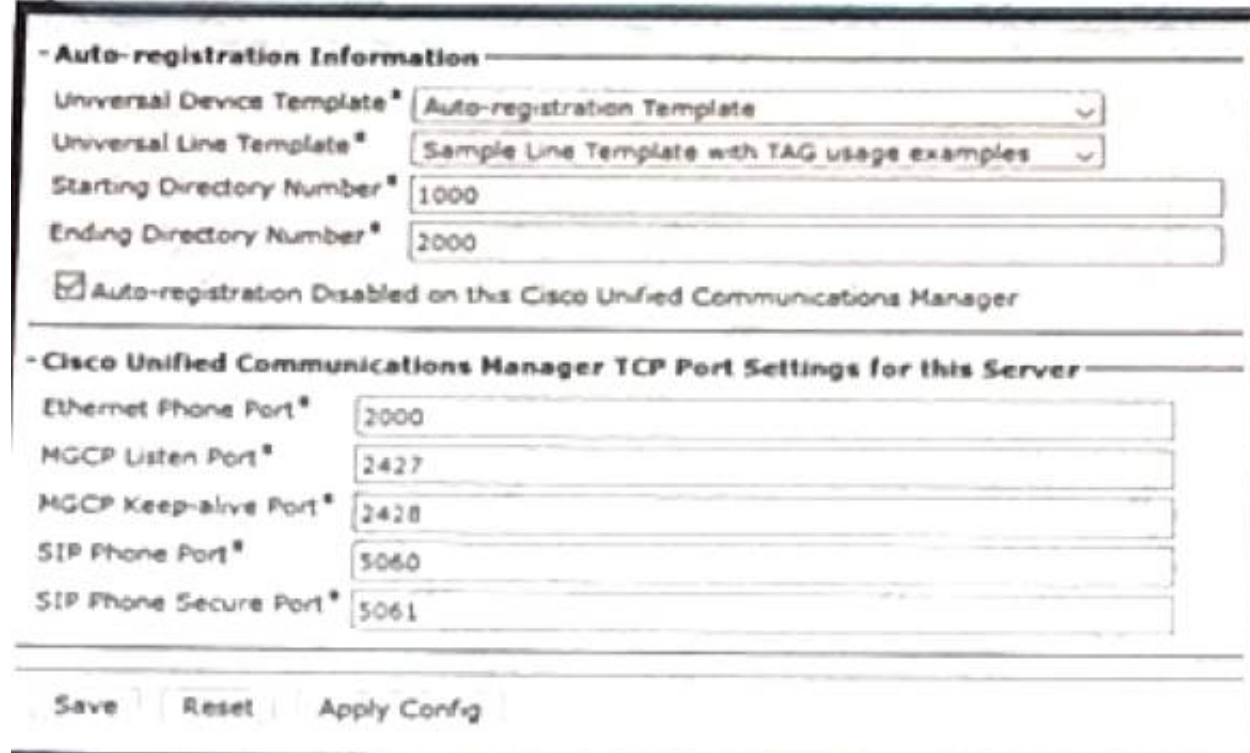
A customer reports that the Cisco UCM toll-fraud prevention does not work correctly, and the customer is receiving charges for unapproved international calls as a result. Which two configuration changes resolve the issues? (Choose two.)

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

Answer: AB

NEW QUESTION 254

Refer to the exhibit.



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

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

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

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

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

Answer: C

NEW QUESTION 258

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 260

What is a reason for using a Diffserv value of AF41 for video traffic?

- A. Video traffic cannot tolerate any packet loss and has a latency of 150 milliseconds
- B. Video traffic can tolerate up to 10% packet loss and latency of 10 seconds
- C. Video traffic can tolerate up to 5% packet loss and latency of 5 seconds
- D. Video traffic can tolerate a packet loss of up to 1% and latency of 150 milliseconds

Answer: D

NEW QUESTION 265

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

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

Answer: D

NEW QUESTION 270

Which QoS marking is used when an administrator configures voice call signaling?

- A. AF41
- B. CS3
- C. CS4
- D. EF

Answer: B

NEW QUESTION 275

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 Bulk 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 276

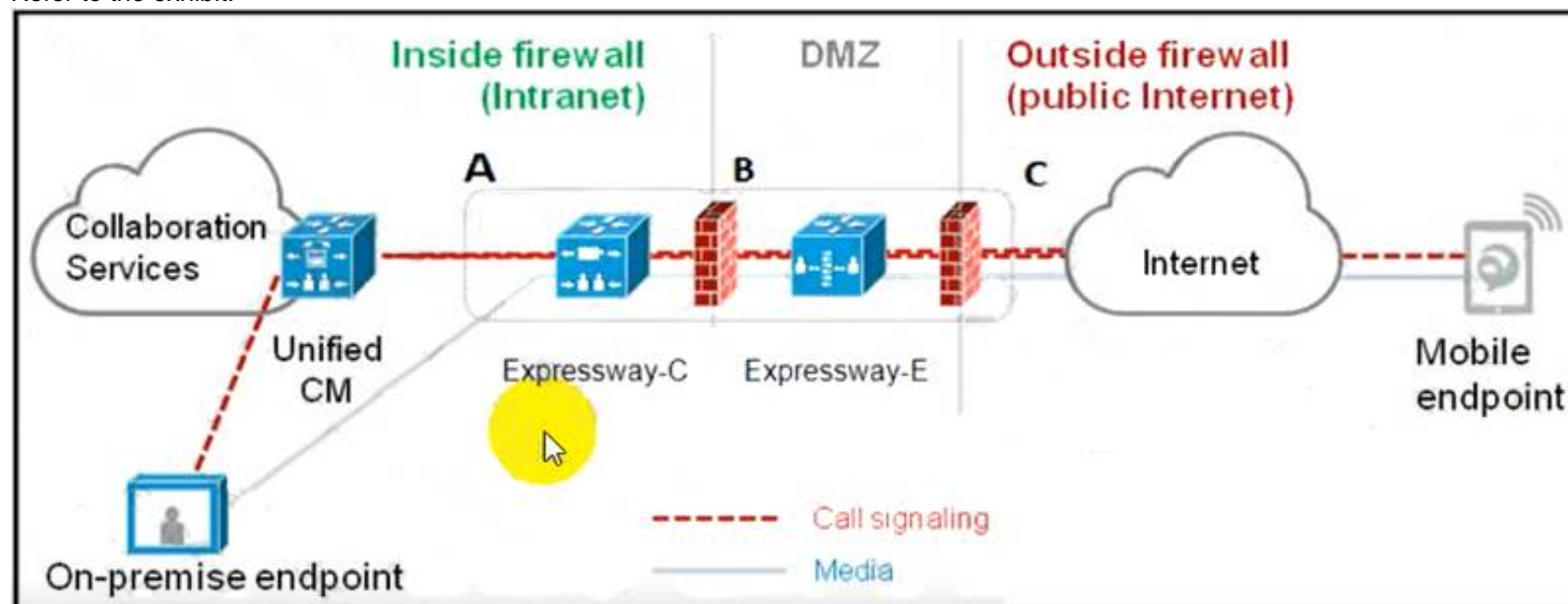
Which two steps should be taken to provision a phone after the Self-Provisioning feature was configured for end users? (Choose two.)

- 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 277

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 281

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 286

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

- A. Configure path-thru content sdp on the voice service.

- B. Configure a hardcoded codec on the dial peers.
- C. Configure a transcoder for video protocols.
- D. Configure codec transparent on the dial peers.

Answer: D

NEW QUESTION 288

Refer to the exhibit. An engineer is confining 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 291

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 296

Refer to the exhibit.

```
23031952: Apr 9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Applying typeplan for se-type Q93 in Q93 Set, Calling num 0055554100
23031953: Apr 9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Sending SETUP called= 0x128 called= 0x128 switch = primary-wi interface = User
23031954: Apr 9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: TX -> SETUP p4 = 0 called = 0x128
  Caller Capability 1 = 0x00000002
  Standard = G.711
  Transfer Capability = Speech
  Transfer Mode = Circuit
  Transfer Rate = 64 kbit/s
  Channel ID 1 = 0x00000003
  Exclusive, Channel 1:
  Progress Ind 1 = 0x101 - Origination address is non-ISDN
  Calling Party Number 1 = 0x101, '0055554100'
  Plan:ISDN, Type:National
  Called Party Number 1 = 0x91, '01144005552222'
  Plan:ISDN, Type:International
23031954: Apr 9 17:43:21.279 EDT: ISDN Se0/1/0:23 Q931: RX -> CALL_PROC p4 = 0 called = 0x128
  Channel ID 1 = 0x00000003
  Exclusive, Channel 1:
23031957: Apr 9 17:43:21.383 EDT: ISDN Se0/1/0:23 Q931: RX -> PROGRESS p4 = 0 called = 0x128
  Cause 1 = 0x2107 - Normal, unspecified
  Progress Ind 1 = 0x0000 - In-band info not appropriate now available
23031961: Apr 9 17:43:44.942 EDT: ISDN Se0/1/0:23 Q931: TX -> DISCONNECT p4 = 0 called = 0x128
  Cause 1 = 0x0000 - Normal call clearing
23031963: Apr 9 17:43:44.942 EDT: ISDN Se0/1/0:23 Q931: RX -> RELEASE p4 = 0 called = 0x128
23031963: Apr 9 17:43:44.942 EDT: ISDN Se0/1/0:23 Q931: TX -> RELEASE_COMPLETE p4 = 0 called = 0x128
```

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 298

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

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

Answer: C

NEW QUESTION 301

What is the function of the Cisco Unity Connection Call Handler?

- A. routes calls to a user based on caller input
- B. queues calls
- C. allows customized scripts for IVR capabilities
- D. searches a list of extensions until the call is answered

Answer: A

Explanation:

A Cisco Unity Connection Call Handler is a software application that answers calls, plays greetings, and routes calls to users based on caller input. Call handlers can be used to create automated attendants, voice menus, and other interactive voice response (IVR) applications.

Call handlers are created and managed using the Cisco Unity Connection Administration interface. When creating a call handler, you can specify a variety of settings, including the greeting that is played, the caller input options that are available, and the destination that calls are routed to.

Call handlers are a powerful tool that can be used to create a variety of IVR applications. By using call handlers, you can improve the efficiency of your organization's communications and provide a better experience for your callers.

Here are some additional tips for using call handlers:

- Use call handlers to create automated attendants that can answer calls and route them to the appropriate person or department.
- Use call handlers to create voice menus that can provide callers with information or options.
- Use call handlers to create interactive voice response (IVR) applications that can collect information from callers and process their requests.

NEW QUESTION 302

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

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

Answer: D

NEW QUESTION 305

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 310

The IP phones at 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(config)# interface GigabitEthernet1/0/1
- E. Switch(config)# no lldp transmit

Answer: BC

NEW QUESTION 311

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

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

Answer: A

NEW QUESTION 313

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 315

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 317

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 318

An engineer is asked to implement on-net/off-net call classification in Cisco UCM. Which two components are required to implement this configuration? (Choose two.)

- A. CTI route point
- B. SIP route patterns
- C. route group
- D. route pattern
- E. SIP trunk

Answer: DE

NEW QUESTION 323

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

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

Answer: A

NEW QUESTION 326

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 327

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 328

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 330

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, 1
<---

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:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-220, a:G.729:G.729a:G.729b, b:8, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-110, a:G.726-16:G.728, b:16, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-70, a:G.726-24, b:24, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:10-50, a:G.726-32, b:32, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-270, a:G.723.1-H:G.723:G.723.1a-H, b:6, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
L: p:30-330, a:G.723.1-L:G.723.1a-L, b:5, e:on, qc:1, s:on, t:10, r:g, nt:IN, v:T:G:D:L:H:R:ATM:SST:PRE
M: sendonly, recvonly, sendrecv, inactive, loopback, 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 334

Which action prevents toll fraud in Cisco UCM?

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

Answer: D

NEW QUESTION 336

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 341

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

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

Answer: CD

NEW QUESTION 342

Which DHCP option must be set up for 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 343

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 346

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

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

Answer: BE

NEW QUESTION 350

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 353

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 358

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 363

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

- Answer: AB**

Refer to the exhibit.


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Status

 Status: Ready

Audio Codec Preference Information

Name*

CCNP COLLAB

Description*

CCNP COLLAB

Codecs in List

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

What is the cause of this issue?

- Answer: C**

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