

- **Vendor: Cisco**
- **Exam Code: 350-801**
- **Exam Name: Implementing and Operating Cisco Collaboration Core Technologies (CLCOR)**
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QUESTION 277

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

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

Answer: AC

Explanation:

https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/bat/9_0_1/CUCM_BK_C22BD805_00_cucm-bulk-administration-guide_chapter_01.pdf

QUESTION 278

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 CLI of the phone.

Which two actions in Cisco UCM resolve the issue? (Choose two)

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

Answer: CE

Explanation:

Enable ssh for phone, and then setup username and password so that engineer can login to phone via CLI.

<https://www.cisco.com/c/en/us/support/docs/collaboration-endpoints/ip-phone-7800-series/200850-Troubleshoot-Cisco-Phone-7800-8800-Serie.pdf>

QUESTION 279

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

- A. The packets arrive at varying time intervals
- B. The packets arrive out of sequence
- C. The packets arrive with frame errors

- D. The packets arrive at uniform time intervals.
- E. The packets never arrive due to tail drop

Answer: AB

QUESTION 280

Refer to the exhibit. An administrator configures a secure SIP trunk on Cisco UCM.

Which value is needed in the Secure Certificate Subject or Subject Alternate Name field to accomplish this task?

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

- 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 common name of the remote device certificates
- D. the fully qualified domain name of all Cisco UCM nodes that run the CallManager service

Answer: A

Explanation:

<https://www.cisco.com/c/en/us/support/docs/conferencing/meeting-server/213820-configure-cisco-meeting-server-and-cucm.html>

QUESTION 281

An administrator works with an ISDN PRI that is connected to a third-party PBX.

The ISDN link does not come up; and the administrator finds that the third-party PBX uses the QSIG signalling method. Which command enables the Cisco IOS Gateway to use QSIG signalling on the ISDN link?

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

Answer: C

Explanation:

The switch type configured must be QSIG:

`isdn switch-type primary-qsig`

<https://community.cisco.com/t5/collaboration-knowledge-base/qsig/ta-p/3126953>

QUESTION 282

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

- A. digest user

- B. serial number
- C. manufacturer
- D. authorization code

Answer: A

Explanation:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_0_1/sysConfig/CUCM_BK_C733E983_00_cucm-system-configuration-guide-1101/Configure___Third_Party_SIP_Phones.pdf

QUESTION 283

Refer to the exhibit. 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?

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

Answer: D

QUESTION 284

A company has an excessive number of call transfers to local and long-distance PSTN from Cisco Unity Connection voicemail.

Which action in the Cisco Unity Connection restriction table resolves this issue?

- A. Implement password complexity on voicemail boxes to prevent accounts from being compromised
- B. Block PSTN patterns on Default Transfer, Default Outdial, and Default System Transfer.
- C. Create a custom restriction table ***** and block it.
- D. Create a custom restriction table ?????????? and block it.

Answer: B

QUESTION 285

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

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

Answer: AC

Explanation:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/admin_guide/X14-0-2/exwy_b_cisco-expressway-administrator-guide-x1402/exwy_m_introduction.html

QUESTION 286

Which DiffServ marking is the most likely to drop packets?

- A. AF12
- B. AF32
- C. AF13
- D. AF11

Answer: C

QUESTION 287

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)#pri-group timeslots 1-7
```
- B.

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

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

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

Answer: D

Explanation:

Enable the ISDN Switch type as Primary-ni globally:

```
config t
#isdn switch-type primary-ni
```

Then the controller E1 and Pri group timeslot configuration:

```
#controller e1 0/0/0
#pri-group timeslots 1-10
```

QUESTION 288

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. Location Conveyance on intercluster SIP trunks
- B. intra-Cluster Communication Signaling
- C. Mobility Cross Cluster
- D. Intercluster Lookup Service

Answer: D

QUESTION 289

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. TCP port 5060
- B. SIP Register
- C. Unsolicited NOTIFY

D. 200 OK

Answer: C

Explanation:

<https://community.cisco.com/t5/collaboration-voice-and-video/understanding-troubleshooting-mwi-on-unity-connection/ta-p/3162948>

QUESTION 290

Drag and Drop Question

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

Answer:

	SET
	TRAP
	INFORM
	GET

QUESTION 291

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

Answer: BD

QUESTION 292

To provide high-quality voice and take advantage of the full voice feature set, which two access layer switches provide support? (Choose two.)

- A. Use multiple egress queues to provide priority queuing of RTP voice packet streams and the ability to classify or reclassify traffic and establish a network trust boundary.

- B. Deploy RSVP to improve VoIP QoS only where it can have a positive impact on quality and functionality where there is limited bandwidth
- C. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.
- D. Use 802.1Q trunking and 802.1p for proper treatment of Layer 2 CoS packet marking on ports with phones connected.
- E. Implement IP RTP header compression on the serial interface to reduce the bandwidth required per voice call on point-to-point links.

Answer: AD

Explanation:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/srnd/design/guide/cmestnd/nstrct.html

QUESTION 293

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

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

Answer: AE

QUESTION 294

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 Unified Border Element, Cisco IM and Presence Server, and Cisco Video Communication Server
- B. Cisco Unified Border Element, Cisco UCM, and Cisco Video Communication Server
- C. Cisco Expressway-E, Cisco Expressway-C, and Cisco UCM
- D. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server

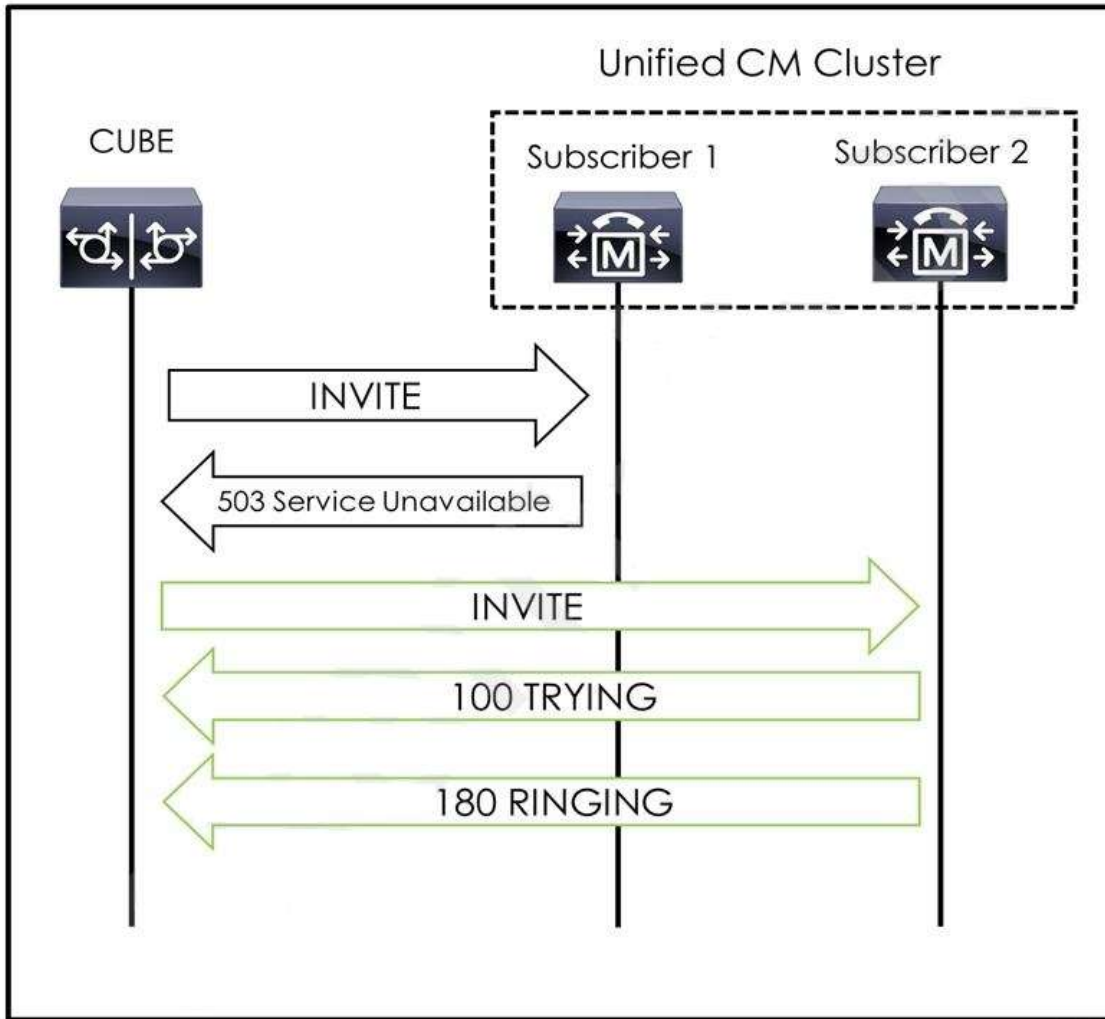
Answer: C

Explanation:

<https://www.cisco.com/c/en/us/td/docs/solutions/CVD/Collaboration/enterprise/12x/120/collbcvd/edge.html>

QUESTION 295

Refer to the exhibit. Cisco Unified Border Element is attempting to establish a call with Subscriber 1, but the call fails. Cisco Unified Border Element then retries the same call with Subscriber 2, and the call proceeds normally. Which action resolves the issue?



- A. Verify that the Run On All Active Unified CM Nodes checkbox is enabled.
- B. Verify that the correct calling search space is selected for the Inbound Calls section.
- C. Verify that the Significant Digits field for Inbound Calls is set to All.
- D. Verify that the PSTN Access checkbox is enabled.

Answer: A

Explanation:

Run On All Unified CM Nodes allows Subscribers NOT configured within the Call Manager Group that is applied to Route Lists and/or SIP Trunks to process calls.

There is a whole rabbit whole with how this feature works when only applied to one or the other but per best practice, unless there is a VERY specific reason, you want this check box enabled on ALL Route Lists and SIP Trunks.

QUESTION 296

Refer to the exhibit. A collaboration engineer is configuring the Cisco UCM IM and Presence Service. Which two steps complete the configuration of the SIP trunk security profile? (Choose two.)

SIP Trunk Security Profile Information

Name*	CUP Non Secure SIP Profile
Description	
Device Security Mode	Non Secure
Incoming Transport Type*	TCP +UDP
Outgoing Transport Type	TCP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application level authorization	
<input checked="" type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input type="checkbox"/> Accept unsolicited notification	
<input type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default filter

- A. Check the box to accept replaces header.
- B. Check the box to allow charging header.
- C. Check the box to enable application-level authorization.
- D. Check the box to transmit security status.
- E. Check the box to accept unsolicited notification.

Answer: AE

Explanation:

Check the following check boxes:

- Accept Presence Subscription
- Accept Out-of-Dialog REFER
- Accept Unsolicited Notification
- Accept Replaces Header

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/11_5_1/cup0_b_config-and-admin-guide-1151su5/cup0_b_imp-system-configuration-1151su5_chapter_0111.html#task_441822D5A33E7A9AEB98901FCE5DA3D8

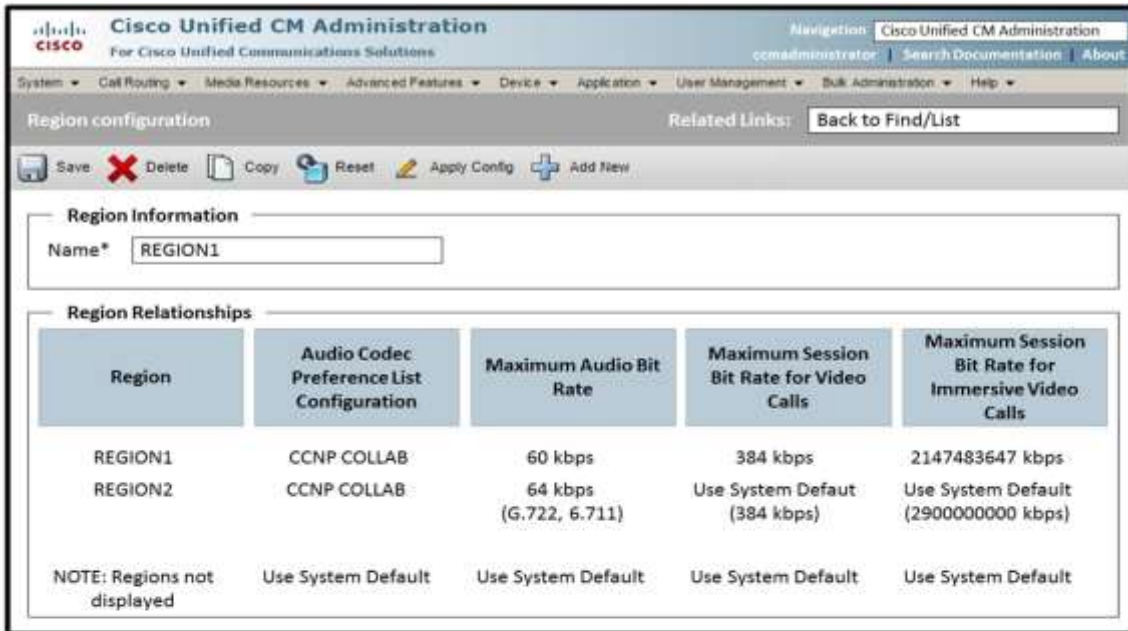
QUESTION 297

Refer to the exhibit. An engineer is troubleshooting this video conference issue:

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

- As soon as the Cisco 9971 in Region1 conferences in a Cisco 8945 in Region2, the Region1 endpoint cannot see the Region2 endpoint video.

What is the cause of this issue?



The screenshot shows the 'Region configuration' page in Cisco Unified CM Administration. The 'Region Information' section shows the name 'REGION1'. The 'Region Relationships' table is as follows:

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



The screenshot shows the 'Audio Codec Preference List Configuration' page. The 'Status' is 'Ready'. The 'Audio Codec Preference Information' section shows:

- 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

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

Answer: C

Explanation:

The section in the question of "when the 9971 CONFERENCE IN the 8945..." indicating that the 8945 is being added to an already in-progress video call, totaling three or more video streams. This would mean 384kbps is definitely way too low.

QUESTION 298

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

- A. SMTP

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

Answer: C

QUESTION 299

An engineer configures a Cisco Unified Border Element and must ensure that the codecs negotiated meet the ITSP requirements. The ITSP supports G.711ulaw and G.729 for audio and H.264 for video. The preferred voice codec is G.711. Which configuration meets this requirement?

A.

```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  video codec h264

dial-peer voice 101 voip
  session protocol sipv2
  destination e164-pattern-map 1
  voice-class codec 100
```

B.

```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8

dial-peer voice 101 voip
  session protocol sipv2
  destination el64-pattern-map 1
  voice-class codec 10
```

C.

```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  video codec h264

dial-peer voice 101 voip
  session protocol sipv2
  destination el64-pattern-map 1
  voice-class codec 10
```

D.

```
voice class codec 10
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  video codec h264

dial-peer voice 101 voip
  session protocol sipv2
  destination e164-pattern-map 1
  voice-class codec 10
```

Answer: C

Explanation:

<https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/cube/configuration/cube-book/cube-codec-basic.html>

QUESTION 300

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

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

Answer: A

Explanation:

A Trusted Relay Point (TRP) is an MTP or transcoder that Cisco Unified Communications Manager can insert into the media stream to act as a control point for call media. The TRP can provide further processing on the stream and can ensure that the stream follows a specific path.

QUESTION 301

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

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

Answer: C

QUESTION 302

Refer to the exhibit. 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.)

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

Answer: BE

Explanation:

To enable the gateway to communicate with Cisco CallManager through the Media Gateway Control Protocol (MGCP) and to supply redundant control agent services, use the "ccm-manager mgcp" command in global configuration mode. http://www.cisco.com/en/US/docs/ios/12_3t/voice/command/reference/vrht_c4_ps5207_TSD_Products_Command_Reference_Chapter.html#wp1072910

Do a mgcp / no mgcp once its added. Make sure that the domain name on cucm is the same as it appears in the 'show ccm-manager' output on the gateway.

QUESTION 303

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

QUESTION 304

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

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

Answer: B

Explanation:

Cisco Expressway supports Mobile and Remote Access with multiple external domains. With this deployment, you will have more than one external domain where your MRA clients may reside. Expressway-E must be able to connect to all of them. To configure this deployment, do the following:

For Expressway-E:

On Expressway-E, configure `_collab-edge._tls.<domain>` and `_sips_tcp.<domain>` DNS SRV records for each Edge domain.

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X14-0-1/mra/exwy_b_mra-deployment-guide-x1401/exwy_m_requirements-for-mra.html

QUESTION 305

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 the gateway to translate the calling number and apply it to the dial peer. Combine it with a translation profile for called numbers.

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

- B. Configure two calling party transformation patterns:

```
\+1917.CCCCCC, strip pre-dot, numbering type: subscriber  
\+!, strip pre-dot, numbering type: national
```

- C. Configure two called party transformation patterns:

```
\+1917.XXXXXXX, strip pre-dot, numbering type: subscriber  
\+1.!, strip pre-dot, numbering type: national
```

- D. Configure the gateway to translate called numbers and apply it to the dial peer. Combine it with a translation profile for calling numbers.

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

- E. Configure two calling party transformation patterns:

```
\+1917.XXXXXXX, strip pre-dot, numbering type: subscriber  
\+1.!, strip pre-dot, numbering type: national
```

Answer: AE

Explanation:

<https://www.cisco.com/c/en/us/support/docs/voice/call-routing-dial-plans/61083-voice-transla-rules.html>

QUESTION 306

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. Use Cisco Unified Border Element to debug the calls.
B. Disable call forwarding on the phone.
C. Make the calls route through a firewall.

D. Use CSS alternate routing rules.

Answer: B

QUESTION 309

What are the last two bits of a DS field in DiffServe Byte used for?

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

Answer: B

Explanation:

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_dfsrv/configuration/15-mt/qos-dfsrv-15-mt-book/qos-dfsrv.html

QUESTION 310

What are two QoS requirements for VoIP traffic? (Choose two.)

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

Answer: AD

QUESTION 311

In an eleven-class queueing model following realtime, best effort, and scavenger queueing rules, what recommended percentage of bandwidth must be allocated for interactive video?

- A. 10%
- B. 12%
- C. 15%
- D. 20%

Answer: C

Explanation:

<https://www.ciscopress.com/articles/article.asp?p=357102&seqNum=7#:~:text=If%20Scavenger%20and%20Bulk%20trffic,illustrated%20in%20Figure%2012%2D10>

QUESTION 312

What is an indicator of network congestion in VoIP communications?

- A. jitter increase due to variable delay
- B. video loss due to video frame corruption
- C. gaps in the voice due to packet loss
- D. discards in the interface of routers and switches

Answer: A

QUESTION 313

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

- A. Configure an SCCP integration with Cisco UCM.
- B. Configure a SIP integration with Cisco UCM to sync users.

- C. Configure an AXL server to access the Cisco UCM users.
- D. Configure an active directory to sync the users who will have a voicemail box.

Answer: C

Explanation:

SIP integration doesn't sync users automatically.

AXL server can be configured in CUC and then sync CUCM users to it.

Another way is LDAP / AD configuration.

QUESTION 314

What are the predefined call handlers in Cisco Unity Connection?

- A. opening greeting, operator, and goodbye
- B. opening greeting, welcome, and default system
- C. greetings, operator, and closed
- D. caller input, greetings, and transfer

Answer: A

Explanation:

https://www.cisco.com/en/US/docs/voice_ip_comm/connection/1x/administration/guide/acm030.html

QUESTION 315

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

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

Answer: BE

QUESTION 316

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

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

Answer: BE

Explanation:

https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-10/Mobile-Remote-Access-via-Expressway-Deployment-Guide-X8-10.pdf

QUESTION 317

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

Explanation:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/12_0_1/systemConfig/cucm_b_system-configuration-guide-1201/cucm_b_system-configuration-guide-1201_chapter_0110010.html

QUESTION 318

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

Which command is required to complete the configuration?

```
Router# configure terminal
Router(config)# interface gigabitethernet 5/1
Router(config-if)# description Cube E42.421-5965
```

- A. platform qos trust extend cos 3
- B. platform qos trust extend
- C. platform qos extend trust
- D. platform qos trust extend cos 5

Answer: B

QUESTION 319

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

QUESTION 320

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

Answer: D

Explanation:

The ringing message actually comes from the destination phone and not from the source one.

<https://community.cisco.com/t5/collaboration-voice-and-video/taking-sip-call-trace-on-cisco-unified-cm-using-rtmt/tap/3161200>

QUESTION 321

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

- A. partition
- B. route partem
- C. translation pattern
- D. CSS

E. route group

Answer: BC

QUESTION 322

Which dial plan function restricts calls that are made by a lobby phone to internal extensions only?

- A. manipulation of dialed destination
- B. path selection
- C. calling privileges
- D. endpoint addressing

Answer: C

QUESTION 323

Which Cisco UCM configuration is required for SIP MWI integrations?

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

Answer: C

Explanation:

In case of SIP integration, CUC sends SIP notify messages to the phone system to turn the MWI on or OFF so "Accept unsolicited notification" should be checked under SIP security profile assigned to the trunk.

<https://community.cisco.com/t5/collaboration-voice-and-video/understanding-troubleshooting-mwi-on-unity-connection/ta-p/3162948>

QUESTION 324

An administrator must make a pattern to route calls to two different destinations john.doe@company.com and jane.doe@company.com.

Which type of patterns are needed in the Cisco UCM, and what must the pattern look like?

- A. A SIP route pattern that looks like this: *@company.com
- B. A SIP route pattern that looks like this: company.com
- C. A regular route pattern with the URI feature enabled in the configuration page.
The pattern must look like this: (*@company.com)
- D. A regular route pattern with the URI feature enabled in the configuration page.
The pattern must look like this: MATCH(*.doe@company.com)

Answer: C

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: C

Explanation:

XCP Router in both types of chat

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/10_5_1/CUP0_BK_CE43108E_00_config-admin-guide-imp-105/CUP0_BK_CE43108E_00_config-admin-guide-imp-105_chapter_00.html

XCP Text Conference Manager only in external DB which is for persistent chat which is only applicable for group chat
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/11_5_1/cup0_b_config-and-admin-guide-1151su5/cup0_b_imp-system-configuration-1151su5_chapter_010000.html#task_2BA3E8B8D67B3AA5E200B9703818BB39

QUESTION 326

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

QUESTION 327

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

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

Answer: B

QUESTION 328

Which value should be changed when each Cisco UCM node does not allow for more than 5000 phones to be registered?

- A. Maximum Number of Registered and Unregistered Devices service parameter on each node
- B. Minimum Number of Phones service parameter on each node
- C. Maximum Number of Registered Devices service parameter on each node
- D. Maximum Number of Phones service parameter on the Publisher

Answer: C

QUESTION 329

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