

350-801 Dumps

Implementing and Operating Cisco Collaboration Core Technologies

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NEW QUESTION 1

Which Cisco Unified Communications Manager configuration is required for SIP MWI integrations?

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

Answer: A

NEW QUESTION 2

An engineer with troubleshoots poor voice quality on multiple calls. After looking at packet captures, the engineer notices high levels of jitter. Which two areas does the engineer check to prevent jitter? (Choose two.)

- A. The network meets bandwidth requirements.
- B. MTP is enabled on the SIP trunk to Cisco Unified Border Element.
- C. Cisco UBE manages voice traffic, not data traffic.
- D. All devices use wired connections instead of wireless connections.
- E. Voice packets are classified and marked.

Answer: AE

Explanation:

Reference: <https://www.cisco.com/c/en/us/support/docs/voice/voice-quality/20371-troubleshoot-qos-voice.html>

NEW QUESTION 3

Multiple route patterns match a number. How does Cisco Unified Communications Managers determine which pattern to use?

- A. the one that comes first in numerical order
- B. the one with the longest match
- C. the one with the closest match
- D. the one that discards everything PreDot

Answer: C

Explanation:

Reference: <https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/13920-call-routing.html#bcr>

NEW QUESTION 4

On which Cisco Unified Communications Manager nodes can the TFTP service be enabled?

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

Answer: C

Explanation:

You can configure the TFTP service on the first node or a subsequent node, but usually you should configure it on the first node. For small systems, the TFTP server can coexist with a Cisco Unified Communications Manager on the same server.

NEW QUESTION 5

Which issue causes slips on a PRI?

- A. incorrect clock source
- B. incorrect encapsulation
- C. incorrectly configured time zone
- D. change in the line code

Answer: A

NEW QUESTION 6

Calls are being delivered to the end user in the globalized format. Where does an engineer configure the calling number into a localized format?

- A. route pattern
- B. service parameters
- C. IP phone
- D. gateway

Answer: C

NEW QUESTION 7

Which command in the MGCP gateway configuration defines the secondary Cisco Unified Communications Manager server?

- A. mgcapp
- B. ccm-manager fallback-mgcp
- C. mgcp call-agent
- D. ccm-manager redundant-host

Answer: B

NEW QUESTION 8

Refer to the exhibit.

```
Endpoint A:  
m=audio 21796 RTP/AVP 108 9 104 105 101  
b=TIAS:64000  
a=extmap:14 http://protocols.cisco.com/timestamp#100us  
a=rtpmap:108 MP4A-LATM/90000  
a=fmtp:108 bitrate=64000;profile-level-id=24;object=23  
a=rtpmap:9 G722/8000  
a=rtpmap:104 G7221/16000  
a=fmtp:104 bitrate=32000  
a=rtpmap:105 G7221/16000  
a=fmtp:105 bitrate=24000  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=trafficclass:conversational.audio.immersive.aq:admitted
```

```
Endpoint B:  
m=audio 21796 RTP/AVP 105 0 8 18 101  
b=TIAS:64000  
a=extmap:14 http://protocols.cisco.com/timestamp#100us  
a=rtpmap:105 G7221/16000  
a=fmtp:105 bitrate=24000  
a=rtpmap:0 PCMU/8000  
a=rtpmap:8 PCMA/8000  
a=rtpmap:18 G729/8000  
a=fmtp:18 annexb=no  
a=rtpmap:101 telephone-event/8000  
a=fmtp:101 0-15  
a=trafficclass:conversational.audio.immersive.aq:admitted
```

Endpoint A calls endpoint B. What is the only audio codec that can be used for the call?

- A. Telephone-event/8000
- B. G7221/16000
- C. PCMA/8000
- D. G722/8000

Answer: B

NEW QUESTION 9

Which description of the Mobile and Remote Access feature is true?

- A. Collaboration Edge feature that enables remote individuals to perform international calls from Jabber with a VPN connection.
- B. Collaboration Edge feature that enables remote individuals to access all enterprise collaboration services using a PC within the corporate environment.
- C. Collaboration Edge feature that enables remote individuals to access enterprise collaboration services via Jabber without the use of a VPN connection.
- D. Collaboration Edge feature that enables remote individuals to access enterprise collaboration services via Jabber with the use of a VPN connection.

Answer: C

NEW QUESTION 10

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

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

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcf/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

NEW QUESTION 10

What is a software-based media resource that is provided by the Cisco IP Voice Media Streaming Application?

- A. video conference bridge
- B. auto-attendant
- C. transcoder
- D. annunciator

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09/media.html

NEW QUESTION 11

When a user dials a number with a phone that is registered to the Cisco Unified Communications Manager, what is the default timeout before the number is sent?

- A. 15 seconds
- B. 5 seconds
- C. 10 seconds
- D. 3 seconds

Answer: C

Explanation:

Reference: <https://www.cisco.com/c/en/us/support/docs/voice-unified-communications/unified-communications-manager-callmanager/13920-call-routing.html>

NEW QUESTION 15

Which action is required if an engineer wants to have Cisco Unified Communications Manager control the configuration for an MGCP gateway?

- A. Apply the ccm-manager configuration commands to the gateway.
- B. Upload the custom configuration in the TFTP server in Cisco Unified CM.
- C. From Cisco Unified CM > Device > Gateway > Add gateway, check the auto-configuration check box.
- D. Configure the Cisco Unified CM's IP in voice service VoIP.

Answer: C

NEW QUESTION 18

Which protocols does Cisco IM and Presence use to authenticate Jabber?

- A. XMPP
- B. SOAP
- C. TCP
- D. LDAP
- E. QBE

Answer: AB

NEW QUESTION 20

Which call flow matches traffic from a Mobile and Remote Access registered endpoint to central call control?

- A. Endpoint > Expressway-E > Expressway-C > Cisco Unified CM
- B. Endpoint > Expressway-E > Cisco Unified CM
- C. Endpoint > Expressway-C > Cisco Unified CM
- D. Endpoint > Expressway-C > Expressway-E > Cisco Unified CM

Answer: A

NEW QUESTION 22

Which recommendation is the best practice for marking video and voice media in a Cisco Unified Communications network?

- A. Voice Cos 5 (IP Precedence 6, PHB AF41, or DSCP 16) Video Cos 4 (IP Precedence 5, PHB EF, or DSCP 32)
- B. Voice Cos 6 (IP Precedence 4, PHB AF41, or DSCP 24) Video Cos 5 (IP Precedence 4, PHB EF, or DSCP 34)
- C. Voice Cos 5 (IP Precedence 2, PHB EF, or DSCP 48) Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 46)
- D. Voice Cos 5 (IP Precedence 5, PHB EF, or DSCP 46) Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 34)

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/netstruc.html

NEW QUESTION 26

Refer to the exhibit.

```
v=0
o=Cisco-SIPUA 13439 0 IN IP4 10.10.10.10
s=SIP Call
b=AS:4064
t=0 0
m=audio 0 RTP/AVP 114 9 124 113 115 0 8 116 18 101
c=IN IP4 10.10.10.10
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

A call is failing to establish between two SIP Devices The called device answers with this SOP. Which SDP parameter causes this issue?

- A. The payload for G.711ulaw must be 18.
- B. The calling device did not offer aptime value.
- C. The media stream is set to sendonly.
- D. The RTP port is set to 0.

Answer: D

NEW QUESTION 31

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

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

Answer: C

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_classn/configuration/xr-16/qos-classn-xr-16-book/qos-classn-mrkg-ntwk-trfc-xr.html

NEW QUESTION 33

Given the H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
voice translation-rule 40
 rule 1 /3...$/ /408555&/
!
voice translation-profile INT
 translate calling 40
!
dial-peer voice 9011 pots
 translation-profile outgoing INT
 destination-pattern 9011T
 port 0/1/0:23
```

A.

Pattern Definition

Pattern* | \+!

Partition | PT_US_VG_CD_Out_xForm

Description | US International calling

Numbering Plan | < None >

Route Filter | < None >

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits | PreDot

Called Party Transformation Mask |

Prefix Digits | 9011

Called Party Number Type* | International

Called Party Numbering Plan* | ISDN

B. Pattern Definition

Pattern* | \+!

Partition | PT_US_VG_CD_Out_xForm

Description | US International calling

Numbering Plan | < None >

Route Filter | < None >

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits | PreDot

Called Party Transformation Mask |

Prefix Digits | 9011

Called Party Number Type* | Unknown

Called Party Numbering Plan* | Unknown

C. Pattern Definition

Pattern* | \+!

Partition | PT_US_VG_CD_Out_xForm

Description | US International calling

Numbering Plan | < None >

Route Filter | < None >

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits | PreDot

Called Party Transformation Mask |

Prefix Digits | 9011

Called Party Number Type* | Cisco CallManager

Called Party Numbering Plan* | Cisco CallManager

C. Pattern Definition

Pattern* | \+!

Partition | PT_US_VG_CD_Out_xForm

Description | US International calling

Numbering Plan | < None >

Route Filter | < None >

Urgent Priority

MLPP Preemption Disabled

Called Party Transformations

Discard Digits | PreDot

Called Party Transformation Mask |

Prefix Digits | 9011

Called Party Number Type* | International

Called Party Numbering Plan* | Private

Answer: C

NEW QUESTION 37

A user reports transfer failure from an IP phone for calls received from a PSTN to another PSTN number. What is a reason for these failures?

A. The IP phone is configured with the wrong region.

- B. The incoming calling search space of the SIP trunk does not include the partition of the line in the IP phone.
- C. The service parameter related to Offnet to Offnet Call Transfer is set to TRUE.
- D. The gateway is configured with the wrong device pool.

Answer: D

NEW QUESTION 39

An engineer with ID012345678 must build an international dial plan in Cisco Unified Communications Manager. Which action should be taken when building a variable-length route pattern?

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

Answer: A

NEW QUESTION 42

A remote office has a less-than-optimal WAN connection and experiences packet loss, delay, and jitter. Which VoIP codec should be used in this situation?

- A. G.711ulaw
- B. iLBC
- C. G.722.1
- D. G.729A

Answer: D

Explanation:

Reference: <https://community.cisco.com/t5/collaboration-voice-and-video/summary-of-cucm-supported-codecs/ta-p/3162905>

NEW QUESTION 46

Which configuration tells a switch part to send Cisco Discovery Protocol packets that configure an attached Cisco IP phone to trust tagged traffic that is received from a device that is connected to the access port on the Cisco IP phone?

- A. Router# configure terminalRouter(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos trust extend
- B. Router# configure terminalRouter(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos trust extend cos 3
- C. Router# configure terminalRouter(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos trust extend cos 5
- D. Router# configure terminalRouter(config)# interface gigabitethernet 5/1 Router(config-if)# platform qos extend trust

Answer: A

NEW QUESTION 47

Which access control group is required on an end user to allow Jabber to do deskphone mode?

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

Answer: B

NEW QUESTION 50

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 Video Output
- B. xCommand Video Status
- C. xConfiguration Video Output
- D. xStatus HDMI Output

Answer: C

NEW QUESTION 55

Which issue can occur if QoS is not deployed on a Cisco Collaboration architecture across the WAN?

- A. 403 Forbidden errors on SIP calls
- B. excessive jitter
- C. unexpected shut-down on Cisco Unified Communications Manager
- D. packet fragmentation

Answer: B

NEW QUESTION 56

Which transport protocol does the application layer protocol SNMP use?

- A. XML
- B. UDP
- C. SIP
- D. HTTP

Answer: B

Explanation:

Reference: <https://www.geeksforgeeks.org/simple-network-management-protocol-snmp/>

NEW QUESTION 61

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

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

Answer: AD

Explanation:

Reference: https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-11/Cisco-Meeting-Server-2-4-with-Cisco-Expressway-Deployment-Guide_X8-11-4.pdf

NEW QUESTION 64

Which Cisco Unified Communications Manager service parameter should be enabled disconnect a multiparty call when the call initiator hangs up?

- A. Drop Ad Hoc Conference
- B. H.225 Black Setup Destination
- C. Block OffNet To OffNet Transfer
- D. Enterprise Feature Access Code for Conference

Answer: A

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmsys/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100/CUCM_BK_SE5FCFB6_00_cucm-system-guide-100_chapter_011000.html#CUCM_TK_DFC66444_00

NEW QUESTION 65

Which protocol does Cisco Prime Collaboration Assurance use to poll the health status of different systems in the Collaboration environment?

- A. SIP
- B. SNMP
- C. SCCP
- D. SMTP

Answer: B

Explanation:

Reference: https://www.cisco.com/c/en/us/products/collateral/cloud-systems-management/prime-collaboration/guide-c07-736946.html#_Toc446633083

NEW QUESTION 67

Which two configuration elements are part of the Cisco Unified Communications Manager toll-fraud prevention? (Choose two.)

- A. SIP trunk security profile
- B. Calling Search Space
- C. SUBSCRIBE Calling Search Space
- D. feature control policy
- E. partition

Answer: BE

NEW QUESTION 69

Which configuration step is necessary for a Cisco SIP phone to synchronize its time with a specific source?

- A. Add a Phone NTP Reference to the Date/Time Group.
- B. Assign the device to the correct region.
- C. Change the Time Format from 24-hour to 12-hour.
- D. Change the Time Zone from "America/Los_Angeles" to "Etc/GMT+8".

Answer: A

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcf/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0110.html

NEW QUESTION 73

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

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

Answer: C

NEW QUESTION 78

A DTMF mismatch is occurring between an MGCP gateway registered FXS port and a Cisco Unified Communications Manager SIP trunk. Which media resource can be leveraged to interwork this mismatch?

- A. Conference Bridge
- B. Trusted Relay Point
- C. Media Termination Point
- D. Annunciator

Answer: C

NEW QUESTION 83

As a voice engineer, which two recommendations do you to make to your company to optimize Cisco Unified Communications Manager configuration to reduce the number of toll fraud incidents? (Choose two.)

- A. Classify all route patterns as on-net and prohibit on-net to on-net call transfers in Cisco Unified CM service parameters.
- B. Classify all route patterns as off-net and prohibit off-net to off-net call transfers in Cisco Unified CM service parameters.
- C. Classify all route patterns as on-net or off-net and prohibit off-net to off-net call transfers in Cisco Unified CM service parameters.
- D. Inbound CSS on any gateway typically should have access to internal destinations and PSTN destinations.
- E. Inbound CSS on any gateway typically should have access to internal destinations only and not PSTN destinations.

Answer: BE

NEW QUESTION 84

DRAG DROP

According to the QoS Baseline Model, drag and drop the applications from the left onto the correct 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:



NEW QUESTION 87

After an engineer runs the `utils ntp status` command on the Cisco Unified Communications Manager publisher, the `stratum` value is 16. Which issue can the Cisco Unified CM cluster experience?

- A. Unified CM sends an NTPV4 packet.
- B. Database replication is not synchronized on the Unified CM nodes.
- C. The cluster loses access to port 124 at the firewall.
- D. The date/time group on all phones defaults to the time zone of the engineer.

Answer: B

NEW QUESTION 91

Which packet delay is the maximum supported between Cisco Unified Communications Manager nodes for clustering over WAN deployments?

- A. 150 ms round trip
- B. 510 ms round trip
- C. 40 ms round trip
- D. 80 ms round trip

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/callpros.html

NEW QUESTION 93

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 Called Party Transformation Pattern Configuration section, configure the Pattern as 9.011841234587
configure the Discard Digits as Predot 10-10-Dialing
- C. in the Calling Party Transformation Patterns section, configure the Pattern as a 011841234557
configure the Discard Digits as Predot

Answer: A

NEW QUESTION 95

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. G.711ulaw
- B. No codec is used (missing codec command).
- C. G.711alaw
- D. G.729r8

Answer: A

Explanation:

Reference: <https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/dialpeer/configuration/15-mt/vd-15-mt-book/vd-dp-cfg-examp.pdf>

NEW QUESTION 98

When a new SIP phone is registered to Cisco Unified Communications Manager, it keeps failing and showing an "unprovisioned" error message in the phone display. Which problem is a possible cause of this issue?

- A. Auto-registration is disabled on the Cisco Unified Communications Manager nodes and the phone device does not have a DN configured.
- B. The DN assigned to the phone is already in use by another SIP phone.
- C. The phone cannot download and install the latest firmware.
- D. The DHCP settings are set incorrectly and the phone does not have an alternate TFTP defined.
- E. The DN configuration for this phone is shared with an SCCP phone.
- F. which is not supported.

Answer: C

NEW QUESTION 102

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