



**Cisco**

## **Exam Questions 350-801**

Implementing and Operating Cisco Collaboration Core Technologies

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

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

**Answer:** C

#### NEW QUESTION 3

Refer to the exhibit.

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

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

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

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

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

**Answer:** C

#### Explanation:

Reference: [https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/jabber/Windows/9\\_7/CJAB\\_BK\\_C606D8A9\\_00\\_cisco-jabber-dns-configuration-guide/CJAB\\_BK\\_C606D8A9\\_00\\_cisco-jabber-dns-configuration-guide\\_chapter\\_010.html](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/Windows/9_7/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide/CJAB_BK_C606D8A9_00_cisco-jabber-dns-configuration-guide_chapter_010.html)

#### NEW QUESTION 4

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 5

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 6

Where is the default for Maximum Session Bit Rate for a region configured?

- A. Service Parameter Configuration
- B. Enterprise Phone Configuration
- C. Enterprise Parameters Configuration
- D. Region Configuration

**Answer:** A

#### Explanation:

Reference: [https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/9\\_1\\_1/ccmcfg/CUCM\\_BK\\_A34970C5\\_00\\_admin-guide-91/CUCM\\_BK\\_A34970C5\\_00\\_admin-guide-91\\_chapter\\_0111.html](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmcfg/CUCM_BK_A34970C5_00_admin-guide-91/CUCM_BK_A34970C5_00_admin-guide-91_chapter_0111.html)

#### NEW QUESTION 7

How many DNS SRV entries can be defined in the SIP trunk destination address field in Cisco Unified Communications Manager?

- A. 1
- B. 8
- C. 16
- D. 4

**Answer:** C

#### Explanation:

Reference: [https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/11\\_5\\_1/sysConfig/CUCM\\_BK\\_SE5DAF88\\_00\\_cucm-system-configuration-guide-1151/CUCM\\_BK\\_SE5DAF88\\_00\\_cucm-system-configuration-guide1151\\_chapter\\_01110.html](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/11_5_1/sysConfig/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide-1151/CUCM_BK_SE5DAF88_00_cucm-system-configuration-guide1151_chapter_01110.html)

#### NEW QUESTION 8

What is a valid class included in the 8-Class QoS Strategy in a VoIP network?

- A. Assured Forwarding
- B. Broadcast Video
- C. Multimedia Conferencing
- D. Real-Time Interactive

**Answer:** C

#### Explanation:

Reference: <https://www.ciscopress.com/articles/article.asp?p=2756478&seqNum=8>

#### NEW QUESTION 9

Regarding SIP integrations with Cisco Unified Communications Manager, if the Cisco Unity Connection is configured to listen for incoming IPv4 and IPv6 traffic, how should the addressing mode be set up in the Cisco Unity Connection?

- A. Set up is not required.
- B. Set up for each group to use IPv4 and IPv6.
- C. Set up media ports for each port group to use IPv4.
- D. Set up IPv4 and IPv6 in Cisco Unified CM.

**Answer:** B

#### NEW QUESTION 10

Which DHCP option must be set up for new phones to obtain the TFTP server IP address?

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

**Answer:** C

#### Explanation:

Reference: <https://blog.router-switch.com/2013/03/dhcp-option-150-dhcp-option-66/>

#### NEW QUESTION 10

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*	<input type="text" value="\+."/>
Partition	<input type="text" value="PT_US_VG_CD_Out_xForm"/>
Description	<input type="text" value="US International calling"/>
Numbering Plan	<input type="text" value=" &lt; None &gt;"/>
Route Filter	<input type="text" value=" &lt; None &gt;"/>
<input checked="" type="checkbox"/> Urgent Priority	
<input type="checkbox"/> MLPP Preemption Disabled	

  

Called Party Transformations	
Discard Digits	<input type="text" value="PreDot"/>
Called Party Transformation Mask	<input type="text" value=""/>
Prefix Digits	<input type="text" value="9011"/>
Called Party Number Type*	<input type="text" value="International"/>
Called Party Numbering Plan*	<input type="text" value="ISDN"/>

B.



**Pattern Definition**

Pattern\*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

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**Called Party Transformations**

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type\*

Called Party Numbering Plan\*

**Pattern Definition**

Pattern\*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

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**Called Party Transformations**

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type\*

Called Party Numbering Plan\*

**Pattern Definition**

Pattern\*

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Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

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**Called Party Transformations**

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type\*

Called Party Numbering Plan\*

Answer: C

#### NEW QUESTION 12

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 14

How are E.164 called-party numbers normalized on a globalized call-routing environment in Cisco Unified Communications Manager?

- A. Normalization is achieved by stripping or translating the called numbers to internally used directory numbers.
- B. Normalization is achieved by setting up calling search spaces and partitions at the SIP trunks for PSTN connection.
- C. Call ingress must be normalized before the call being routed.
- D. Normalization is not required.

Answer: A

#### NEW QUESTION 17

An engineer configures local route group to simplify a dial plan. Where does the engineer set the route groups according to the local route group names that are

configured?

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

**Answer:** D

#### NEW QUESTION 19

What is the major difference between the two possible Cisco IM and Presence high-availability modes?

- A. Balanced mode provides user load balancing and user failover in the event of an outag
- B. Active/standby mode provides an always on standby node in the event of an outage, and it also provides load balancing.
- C. Balanced mode provides user load balancing and user failover only for manually generated failover
- D. Active/standby mode provides an unconfigured standby node in the event of an outage, but it does not provide load balancing.
- E. Balanced mode provides user load balancing and user failover in the event of an outag
- F. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.
- G. Balanced mode does not provide user load balancing, but it provides in the event of an outag
- H. Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.

**Answer:** C

#### NEW QUESTION 21

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 IOS entry sets the required priority?

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

**Answer:** A

#### NEW QUESTION 24

A customer wants a video conference with five Cisco TelePresence IX5000 Series systems. Which media resource is necessary in the design to fully utilize the immersive functions?

- A. Cisco PVD4-128
- B. software conference bridge on Cisco Unified Communications Manager
- C. Cisco Webex Meetings Server
- D. Cisco Meeting Server

**Answer:** C

#### NEW QUESTION 28

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 31

An engineer is designing a high availability and failover solution for two Cisco Unified Border Element routers. The first router (cube1.ab?.com) takes 60% of the calls and the second router (cube2.abc.com) takes 40% of the calls. Assume all DNS A records have been created. Which two SRV records are needed for a load balanced solution? (Choose two.)

- A. \_sip.\_udp.abc.com 60 IN SRV 2 60 cube1.abc.com
- B. \_sip.\_udp.abc.com 60 IN SRV 60 1 cube1.abc.com
- C. \_sip.\_udp.abc.com 60 IN SRV 1 40 cube2.abc.com
- D. \_sip.\_udp.abc.com 60 IN SRV 3 60 cube2.abc.com
- E. \_sip.\_udp.abc.com 60 IN SRV 1 60 cube1.abc.com

**Answer:** CE

#### NEW QUESTION 35

How does an administrator make a Cisco IP phone display the last 10 digits of the calling number when the call is in the connected state, and also display the calling number in the E.164 format within call history on the phone?

- A. Configure a translation pattern that has a Calling Party Transform Mask of XXXXXXXXXX.
- B. On the inbound SIP trunk, change Significant Digits to 10.
- C. Change the service parameter Apply Transformations On Remote Number to True.

D. Configure a calling party transformation pattern that keeps only the last 10 digits.

**Answer:** D

#### NEW QUESTION 40

How can an administrator stop Cisco Unified Communications Manager from advertising the OPUS codec for recording enabled devices?

- A. Route recorded calls through Cisco Unified Border Element because it does not support OPUS.
- B. Go to the phone's configuration page and set "Advertise OPUS Codec" to be "false".
- C. Integrate the Cisco Unified CM with 3 recording solution that does not support OPUS.
- D. In CUCM Service Parameters set "Opus Codec Enabled" to "Enabled for all Devices Except Recording-Enabled Devices."

**Answer:** D

#### Explanation:

Reference: <https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/211297-Configure-Opus-Support-on-Cisco-Unified.pdf>

#### NEW QUESTION 43

Which configuration on Cisco Unified Communications Manager is required for SIP MWI to work?

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

**Answer:** C

#### NEW QUESTION 48

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

#### NEW QUESTION 52

Due to provider requirements, outgoing calls from the Enterprise to the PSTN must start with channel 1. Which ISDN command changes the channel selection an IOS to meet this requirement?

- A. isdn bchan-number-order decending
- B. isdn bchan-number-order ascending
- C. isdn protocol-emulate network
- D. isdn incoming-voice voice

**Answer:** B

#### NEW QUESTION 53

Which wildcard must an engineer configure to match a whole domain in SIP route patterns?

- A. .
- B. !
- C. @
- D. \*

**Answer:** D

#### Explanation:

Reference: [https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/admin/10\\_0\\_1/ccmcfg/CUCM\\_BK\\_C95ABA82\\_00\\_admin-guide-100/CUCM\\_BK\\_C95ABA82\\_00\\_admin-guide-100\\_chapter\\_0100111.html](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.html)

#### NEW QUESTION 55

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