

Exam Questions 350-801

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

A customer has Cisco Unity Connections that is integrated with LDAP. As a Unity Connection administrator, you have received a request to change the first name for VM user. Where must the change be performed?

- A. Cisco Unity Connection
- B. Cisco Unified Communications Manager end user
- C. Active Directory
- D. Cisco IM and Presence

Answer: C

NEW QUESTION 3

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

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

Answer: B

Explanation:

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

NEW QUESTION 4

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 5

An engineer must extend the corporate phone system to mobile users connecting through the internet with their own devices. One requirement is to keep that as simple as possible for end users. Which infrastructure element achieves these goals?

- A. Cisco Express Mobility
- B. Cisco Expressway-C and Expressway-E
- C. Cisco Unified Border Element
- D. Cisco Unified Instant Messaging and Presence

Answer: C

NEW QUESTION 6

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 7

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 8

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

Answer: A

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/net_mgmt/prime/collaboration/12-1/assurance/advanced/guide/cpcp_b_cisco-prime-collaboration-assurance-guide-advanced-12-1/cpcp_b_cisco-prime-collaboration-assurance-guideadvanced-12-1_chapter_01111.html

NEW QUESTION 9

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 10

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 10

Refer to the exhibit.


```
INVITE sip:1@10.10.10.219;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.10.10.84:50083;branch=z9hG4bK471df613
From: "1234 - My Phone" <sip:1234@10.10.10.219>;tag=381claba7a78002c558eda31-12b8af63
To: <sip:1@10.10.10.219>
Call-ID: 381claba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Allow-Events: kpml,dialog
Content-Type: application/sdp
Content-Length: 658

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
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=sendrecv
```

When a UC Administrator is troubleshooting DTMF negotiated by this SIP INVITE, which two messages should be examined next to further troubleshoot the issue? (Choose two.)

- A. REGISTER
- B. UPDATE
- C. PRACK
- D. NOTIFY
- E. SUBSCRIBE

Answer: DE

NEW QUESTION 12

An administrator recently upgraded a Cisco Webex DX80 through its web interface but discovered the next morning that the unit has received to its previous version. What must the administrator do to prevent this from happening again?

- A. Assign a phone security profile with secure SIP.
- B. Set the prepare cluster for rollback to pre-8-0 enterprise parameter to true.
- C. Confirm the phone load name in the phone configuration.
- D. Assign a universal device template to the phone.

Answer: C

NEW QUESTION 13

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 18

Refer to the exhibit.

```
rule 1 /^\(0[25]..\)\-\(\...\)\-\(\....$\)/ /\1\2\3/
```

The translation rule is configured on the voice gateway to translate DNIS. What is the outcome if the gateway receives 0255-343-1234 as DNIS?

- A. The translation is not matched because DNIS contains "-".

- B. The translation is not matched because DNIS does not end with a “\$”.
- C. The translation is matched and the translated number is 02553431234.
- D. The translation is matched and the translated number is 025553431234.

Answer: A

NEW QUESTION 19

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

- A. route pattern
- B. route group
- C. translation pattern
- D. partition
- E. CSS

Answer: CE

NEW QUESTION 24

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

NEW QUESTION 26

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 30

Which two conditions must a user meet to provision a new device using the Self-Provisioning feature? (Choose two.)

- A. The user must have a primary extension.
- B. At least two DNs must be assigned to the user device.
- C. The user must be part of “Standard CCM Super User”.
- D. The user must have the appropriate universal device template linked to the user profile.
- E. The user must have at least user device profile assigned.

Answer: AD

NEW QUESTION 34

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 37

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 42

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 Software Upgrades page in CUCM OS Administration
- B. The In-Room Control Editor on the webpage of the MX800
- C. The phone configuration page in CUCM Administration
- D. The SIP Trunk Security Profile page in CUCM Administration

Answer: A

NEW QUESTION 47

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 51

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 56

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="< None >"/>
Route Filter	<input type="text" value="< None >"/>
<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"/>
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

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

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

Called Party Transformations

Discard Digits

Called Party Transformation Mask

Prefix Digits

Called Party Number Type*

Called Party Numbering Plan*

Answer: C

NEW QUESTION 58

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 63

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 66

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 69

Which two types of device are supported by the Bulk Administration Tool? (Choose two.)

- A. H.322 clients
- B. Cisco Unified IP phones (all models)
- C. SIP trunks
- D. H.225 trunks
- E. music on hold servers

Answer: AB

NEW QUESTION 70

Which two functionalities does Cisco Expressway provide in the Cisco Collaboration architecture? (Choose two.)

- A. Survivable Remote Site Telephony functionality
- B. customer interaction management services
- C. secure firewall and NAT traversal for mobile or remote Cisco Jabber and TelePresence Video endpoints
- D. MGCP gateway registration
- E. Secure business-to-business communications

Answer: CE

NEW QUESTION 75

How does Cisco Unified Communications Manager perform a digit analysis on-hook versus off-hook for an outbound call from a Cisco IP phone that is registered to Cisco Unified CM?

- A. On-hook, Unified CM performs a digit-by-digit analysis, off-hook, Unified GM considers all digits were dialed and does not wait for additional digits.
- B. On-hook, Unified CM considers all digits were dialed and does not wait for additional digits, off-hook, Unified CM performs a digit-by-digit analysis.
- C. On-hook, by pressing the digits and entering "#" to process the call, Unified CM performs a digit-by-digit analysis; off-hoo
- D. Unified CM analyzes all digits as a string.
- E. On-hook, no digit analysis is performed, off-hoo
- F. Unified CM requires the "*" to start the digit analysis.

Answer: C

NEW QUESTION 79

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 81

You are adding regions in Cisco Unified Communications Manager. Which codec(s) are selected when a call is placed if you set up the max audio bit rate to use 8 kbps?

- A. G.729
- B. G 729 and G.711ulaw
- C. G.711ulaw and G 711alaw
- D. G.722

Answer: A

NEW QUESTION 82

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 84

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 85

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 87

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 is a possible cause of the PRI issue?

- A. The cable is unplugged.
- B. The clock source is incorrect.
- C. The controller shut down.
- D. The framing is configured incorrectly.

Answer: D

NEW QUESTION 90

Which call routing pattern is used for phone numbers that are in the E.164 format?

- A. \+.! Route Pattern
- B. \+.! Translation Pattern
- C. /+! Route Pattern
- D. \+1.[2-9]XX[2-9]XXXXXX Called Party Transformation Pattern

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html#pgfId-1591747

NEW QUESTION 91

Which two DNS records must be created to configure Service Discovery for or premises Jabber? (Choose two.)

- A. _cisco-uds._tls.cisco.com pointing to the IP address of Cisco Unified Communications Manager
- B. _cuplogin._tcp.cisco.com pointing to a record of IM&P
- C. _cuplogin._tls.cisco.com pointing to the IP address of IM&P
- D. _cisco-uds._tcp.cisco.com pointing to a record of Cisco Unified CM
- E. _xmpp._tls.cisco.com pointing to a record of IM&P

Answer: AB

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

NEW QUESTION 93

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 95

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 96

Which statement describes the outcome when the trust boundary is defined at the Cisco IP phone?

- A. Packets or Ethernet frames are remarked at the access layer switch.
- B. Packets or Ethernet frames are not remarked by the IP phone.
- C. Packets or Ethernet frames are not remarked at the layer switch.
- D. Packets or Ethernet frames are remarked at the distribution layer switch.

Answer: C

Explanation:

Reference: <https://networklessons.com/quality-of-service/how-to-configure-qos-trust-boundary-on-cisco-switches>

NEW QUESTION 99

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

Answer: B

NEW QUESTION 104

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 109

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 113

When a phone is registered over Mobile and Remote Access, where does it register?

- A. Cisco Unified Presence Server
- B. Expressway-E
- C. Cisco Unified Communications Manager
- D. Expressway-C

Answer: B

Explanation:

Reference: 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_01011010.html QUESTION

NEW QUESTION 116

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 120

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 121

Which description of the function of call handlers in Cisco Unity Connection is true?

- A. They answer calls, take messages, and provide menus of options.
- B. They provide access to a corporate directory by playing an audio list that users and outside callers use to reach users and leave messages.
- C. They collect information from callers by playing a series of questions and recording the answers.
- D. They control outgoing calls by allowing you to specify the numbers that Cisco Unity Connection can dial to transfer calls, notify users of messages, and deliver faxes.

Answer: A

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/connection/10x/administration/guide/10xcucsagx/10xcucsag080.html

NEW QUESTION 126

An engineer must configure an MGCP gateway and register it to Cisco Unified Communications Manager. Which prerequisite must be met before applying the gateway commands to enable MGCP?

- A. The MGCP gateway and the Cisco Unified CM must be able to communicate over ports 5060 and 5061.
- B. Cisco Unified CM and the MGCP gateway must utilize the SIP OPTIONS ping feature to monitor status.
- C. The MGCP gateway must have voice service VoIP configured.
- D. The MGCP gateway and the Cisco Unified CM must be able to communicate over ports 2427, 2428, and 2727.

Answer: D

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/ios/voice/cminterop/configuration/guide/12_4t/vc_12_4t_book/vc_ucm_mgcp_gw.html

NEW QUESTION 130

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