Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

https://www.2passeasy.com/dumps/350-801/

NEW QUESTION 1
A Cisco Unity Connection Administrator must set a voice mailbox so that it can be accessed from a secondary device. Which configuration on the voice mailbox makes this change?

A. Attempt Forward routing rule
B. Alternate Extensions
C. Alternate Names
D. Mobile User

Answer: A

NEW QUESTION 2
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 3
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 4
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 5
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 6
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 7
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 8
Which command is used in Cisco IOS XE TDM gateway to configure the voice T1/E1 controller to provide clocking?

A. clock source line
B. Cisco IOS XE TDM gateway T1/E1 controller cannot provide clocking.
C. clocking source internal
D. clocking source network

Answer: C

Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/routers/access/interfaces/NIM/software/configuration/guide/4gen-t1-e1-nim-guide.html

NEW QUESTION 9
Refer to the exhibit.

```
INVITE sip:4000@172.16.1.1:5061 SIP/2.0
Via: SIP/2.0/TLS 172.16.2.143:5061;branch=z9hG4bK8FD315E7
Remote-Party-ID: <sip:+14088335000@172.16.2.143>;party=calling;screen=no; privacy=off
From: <sip:+14088335000@172.27.2.143>;tag=7B42E5F6-9B8
To: <sip:4000@172.16.1.1>
Date: Tue, 06 Aug 2019 15:03:05 GMT
Call-ID: 4EA4363-B77111E9-8A4AFFCF-10B6D71B@172.16.2.143
Supported: 100rel,timer,resource-priority, replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 0082391505-3077640681-2319777743-0280418075
User-Agent: Cisco-SIPGateway/IOS-15.5.3.S4b
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Timestamp: 1565089565
Contact: <sip:+ 14088335000@172.16.2.143:5061;transport=tls>
Expires: 180
Allow-Events: telephone-event
Max-Forwards: 68
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 416
v=0
o=CiscoSystemsSIP-GW-UserAgent 8486 8298 IN IP4 172.16.2.143
s=SIP Call
c=IN IP4 172.16.2.143
t=0 0
m=audio 44612 RTP/SAVP 0 101
c=IN IP4 172.16.2.143
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
```

This INVITE is sent to an endpoint that only supports G729. What must be done for this call to succeed?

A. Nothing: both sides support G.729.
B. Add a transcoder that supports G711ulaw and G.729.
C. Add a media termination point that supports G.711ulaw and G.729.
D. Nothing: both sides support payload type 101.

Answer: D

NEW QUESTION 10
A user forwards a corporate number to an international number. What are two methods to prevent this forwarded call? (Choose two.)

A. Set the Call Classification to OnNet for the international route pattern.
B. Block international dial patterns in the SIP trunk CSS.
C. Configure a Forced Authorization Code on the international route pattern.
D. Set Call Forward All CSS to restrict international dial patterns.
E. Check Route Next Hop By Calling Party Number on the international route pattern.

Answer: BC

Explanation:
Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/cmetrans.html

NEW QUESTION 10
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/cpco_b_cisco-prime-collaboration-assurance-guide-advanced-12-1/cpco_b_cisco-prime-collaboration-assurance- guideadvanced-12-1_chapter_01111.html

NEW QUESTION 13
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 18
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 22
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=381c1aba7a78002c558eda31-12b8af63
To: <sip:1@10.10.10.219>
Call-ID: 381c1aba-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-maxcapturerate=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 23
An engineer wants to manually deploy a Cisco Webex DX80 video endpoint to an end user. Which type of provisioning can be configured on the endpoint?

A. CUBE
B. CMS
C. CUCM
D. Edge

Answer: C

Explanation:
Reference: https://www.cisco.com/c/en/us/products/collateral/collaboration-endpoints/desktop-collaboration-experience-dx600-series/datasheet-c78-731879.html

NEW QUESTION 25
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 27
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 31
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/ccmcfg/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

NEW QUESTION 32
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 37
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 41
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 43
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

NEW QUESTION 46
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 51
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 52
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 57
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 61
Refer to the exhibit.

```
000193: Dec 5 14:35:31.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:32.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:33.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
000193: Dec 5 14:35:34.054: ISDN Se0/1/0:15 Q921: User TX -> SABMEp sapi=0 tei=0
```

Given this "debug isdn q921" output, what is the problem with the PRI?

A. Layer 1 is down on the controller.
B. PRI does not have an IP address configured on the interface.
C. Nothing, the PRI is sending keepalives.
D. Layer 2 is down on the controller.

Answer: D

NEW QUESTION 66
A collaboration engineer must configure Cisco Unified Border Element to support up to five concurrent outbound calls across an Ethernet link with a bandwidth of 160 kb to the Internet Telephony Service Provider. Which set of commands allows the engineer to complete the task without compromising voice quality?

A. dial-peer voice 1 voip translation- profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]……$session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp-nte sip- notify sip-kpml
B. dial-peer voice 1 voip translation-profile outgoing Strip9 max- conn 5destination-pattern 91[2-9]..[2-9]……$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp- nte sip-notify sip-kpml codec ilbc mode 20
C. dial-peer voice 1 voip translation- profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]……$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp- nte sip-notify sip-kpml codec aacld
D. dial-peer voice 1 voip translation- profile outgoing Strip9 max-conn 5 destination-pattern 91[2-9]..[2-9]……$ session protocol sipv2 session target ipv4:142.45.10.1 dtmf-relay rtp- nte sip-notify sip-kpml codec mp4a- latm

Answer: B

NEW QUESTION 71
Which method is used to avoid toll fraud with Cisco Unified Communications Manager calls?

A. call policy service
B. TOLLFRAUD_APP
C. default zone access rules
D. class of service

Answer: D

NEW QUESTION 73
Which settings are needed to configure the SIP route pattern in Cisco Unified Communications Manager?

A. pattern usage, IPv6 pattern, and SIP trunk/Route list
B. pattern usage, IPv4 pattern, IPv6 pattern, and description
C. pattern usage, IPv4 pattern, and SIP trunk/Route list
D. SIP trunk/Route list, description, and IPv4 pattern

Answer: C

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.pdf

NEW QUESTION 74
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 79
A network administrator deleted a user from the LDAP directory of a company. The end user shows as Inactive LOAP Synchronized User in Cisco Unified Communications Manager. Which step is next to remove this user from Cisco Unified Communications Manager?

A. Delete the user directly from Cisco Unified Communications Manager
B. Restart the Dirsync service after the user is deleted from LDAP directory.
C. Execute a manual sync to refresh the local database and delete the end user.
D. Wait 24 hours for the garbage collector to remove the user.

Answer: B

NEW QUESTION 81
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 86
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 90
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 painting 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 95
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 98
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 100
An administrator is trying to change the default LINECODE for a voice ISDN T1 PRI. Which command makes this change?

A. linecode esf
B. linecode ami
C. linecode hdb3
D. linecode b8zs

Answer: D

Explanation:
Reference: https://www.cisco.com/en/US/docs/ios/dial/configuration/guide/dia_cfg_isdn_pri_external_docbase_0900e4b1806c752c_4container_external_docbase_0900e4b18216dd1b.html

NEW QUESTION 105
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 108
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 110
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 112
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

A. in the Calling Party Transformation Patterns section,
configure the Pattern as 9.011841234567
configure the Discard Digits as Predot 10-10-Dialing
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 116
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 121
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 124
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 126
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 129
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 aver 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 134
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

NEW QUESTION 138
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