Exam Questions 350-801

Implementing and Operating Cisco Collaboration Core Technologies

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NEW QUESTION 1
What is the element of Cisco Collaboration infrastructure that allows Jabber clients outside of the network to register in Cisco Unified Communications Manager and use its resources?

A. Cisco IM and Presence node
B. Cisco Unified Border Element
C. Cisco Expressway
D. Cisco Prime Collaboration Provisioning server

Answer: C

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
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 5
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 6
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 7
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 8
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 9
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 10
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 10
What causes poor voice quality and video pixelization in a video call?

A. The QoS is configured incorrectly.
B. A firewall is blocking the RTP ports.
C. Cisco Unified Communications Manager is configured to use G.711 instead of G.729.
D. 1 Gbps network ports are used instead of 100 Mbps network ports.

Answer: A

NEW QUESTION 13
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 14
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 18
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-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=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 a ptime value.
C. The media stream is set to sendonly.
D. The RTP port is set to 0.

Answer: D

NEW QUESTION 21
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 22
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 26
Which DHCP option must be set up for new phones to obtain the TFTP server IP address?

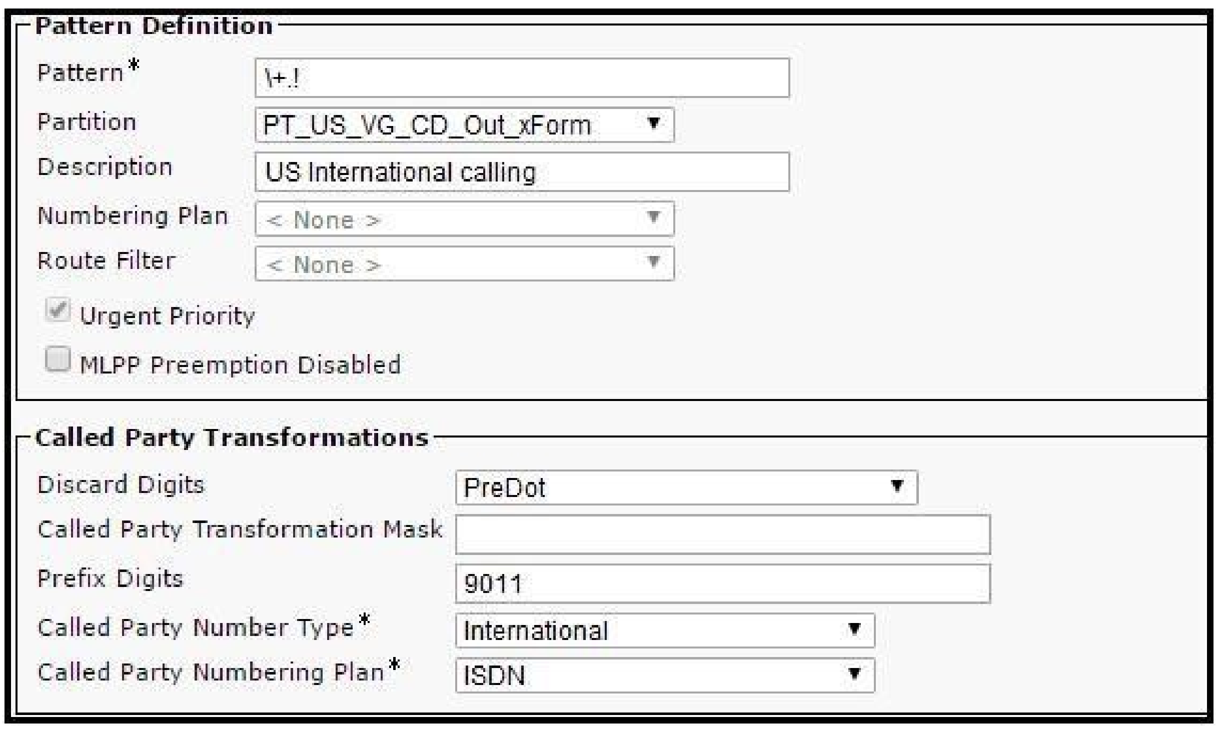
A. option 15
B. option 6
C. option 66
D. option 120

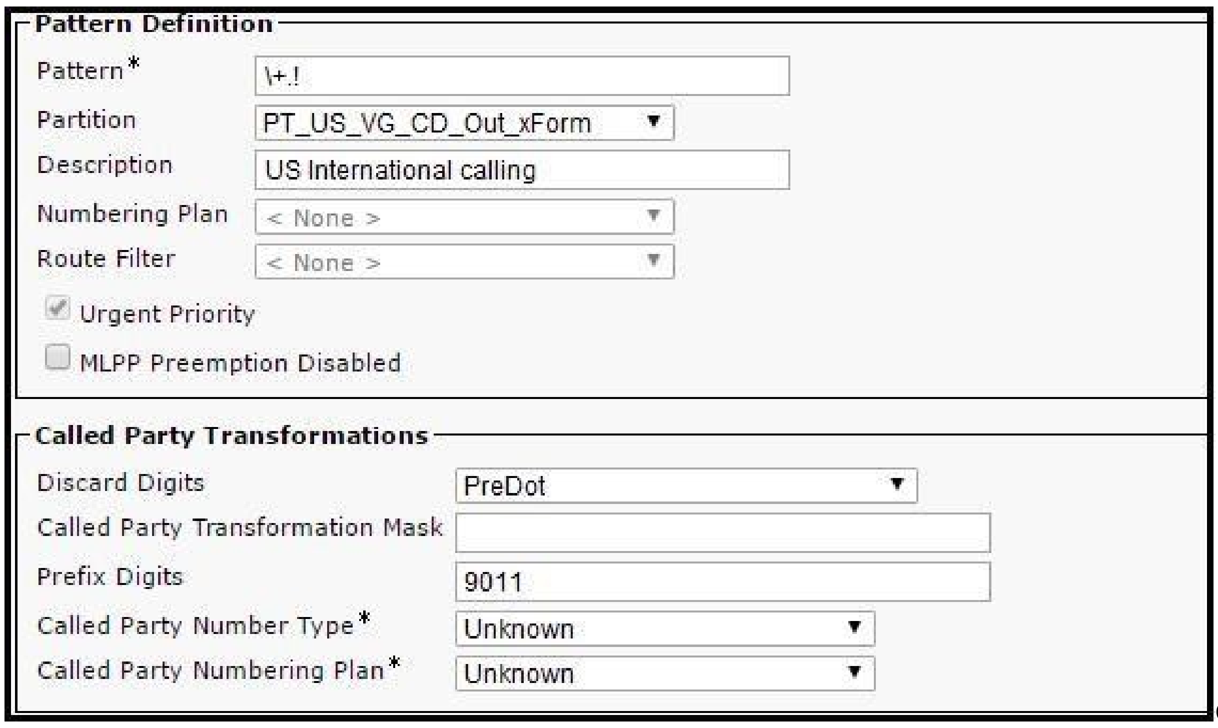
Answer: C

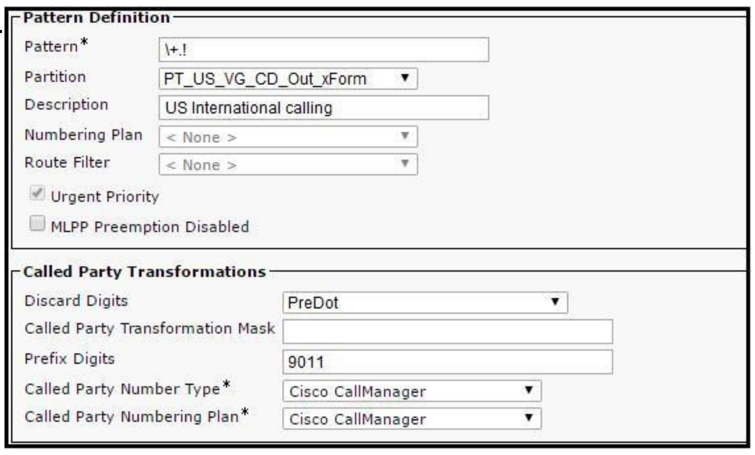
Explanation:
Reference: https://blog.router-switch.com/2013/03/dhcp-option-150-dhcp-option-66/

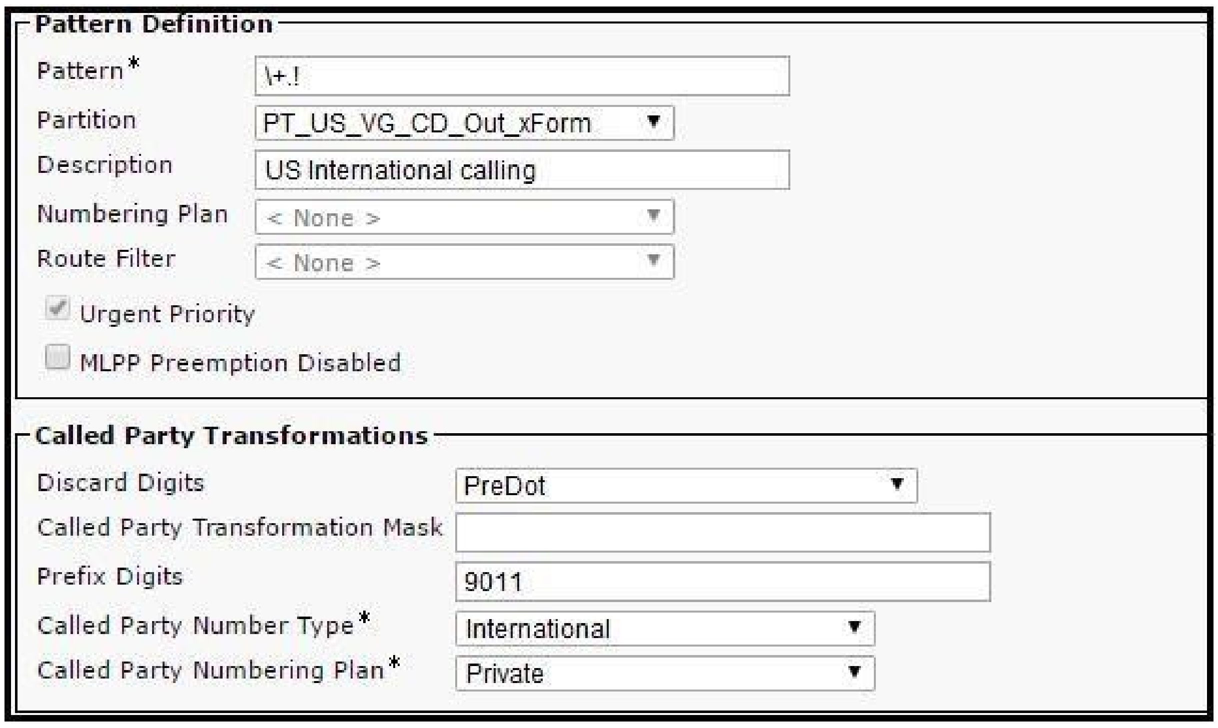
NEW QUESTION 30
Given the H.323 gateway configuration and using Cisco best practices, how must the called party transformation pattern be configured to ensure that a proper ISDN type of number is set?

```
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.


B.


C.


C.


Answer: C

NEW QUESTION 35
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 36
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 38

Refer to the exhibit

```
INVITE sip:2002@10.10.10.10:5060 SIP/2.0
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=IN IP4 10.10.10.11
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp:110 0-16
a=ptime:20

SIP/2.0 200 OK
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=ptime:20
```

The SDP offer/answer has been completed successfully but there is no DTMF when users press keys. What is the cause of the issue?

A. DTMF was negotiated properly in these messages.
B. G.729 rather than G.711ulaw was negotiated.
C. Payload type 110 was negotiated rather than type 101.
D. DTMF was not negotiated on the call.

Answer: D

NEW QUESTION 42
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 45
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 47
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 50
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 55
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 PVDM4-128
B. software conference bridge on Cisco Unified Communications Manager
C. Cisco Webex Meetings Server
D. Cisco Meeting Server

Answer: C

NEW QUESTION 58
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 62
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 67
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 68
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 70
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 71
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 76
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 78
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 79
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 84
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 85
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 89
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 93
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 97
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 98
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 102
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