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Exam Code: 350-801

Exam Name: Implementing Cisco Collaboration Core Technologies (CLCOR)



Exam A

QUESTION 1

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

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

Correct Answer: C

Section:

QUESTION 2

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

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

Correct Answer: D

Section:



QUESTION 3

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

- A. Configure all forced code on all router.
- B. Block international dial patterns in the SIP trunk CSS.
- C. Configure a Forced Authorization Code on the international route pattern.
- D. Set the call Classification to OnNet for the international route pattern.
- E. Set call forward All CSS to restrict international dial patterns.

Correct Answer: C, E

Section:

QUESTION 4

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

- A. Confirm the phone load name in the phone configuration.
- B. Set the prepare cluster for rollback to pre-8.0 enterprise parameter to true.
- C. Assign a universal device template to the phone.
- D. Assign a phone security profile with secure SIP.

Correct Answer: A

Section:

QUESTION 5

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

- A. Expressway-C
- B. Cisco Unified Communications Manager
- C. Cisco Unified presence server
- D. Expressway-E

Correct Answer: B

Section:

QUESTION 6

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

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

Correct Answer: C

Section:

QUESTION 7

Which transport protocol does the application layer protocol SNMP use?

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

Correct Answer: C

Section:

QUESTION 8

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 individual to access enterprise collaboration services via Jabber with the use of a VPN connection
- 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 all enterprise collaboration services using a PC within the corporate environment

Correct Answer: C



Section:

QUESTION 9

What causes poor voice quality and video pixelization in a video call?

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

Correct Answer: C

Section:

QUESTION 10

A presence redundancy group is deployed, and an engineer with ID012345678 initiates a Manager fallback. Which statement about Cisco server Recovery Manager is true?

- A. restart the Cisco Presence Engine
- B. disconnects all user that had been failed over
- C. restarts critical services on the secondary node
- D. disconnects all users that had been failed over, and the users must log in again

Correct Answer: D

Section:

QUESTION 11

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

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

Correct Answer: D

Section:

QUESTION 12

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

- A. Call policy service
- B. Default zone access rules
- C. TOLLFRAUD_APP
- D. class of service

Correct Answer: D

Section:

QUESTION 13



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 phone configuration page in CUCM Administration
- B. The SIP Trunk security profile page in CUCM Administration
- C. The software Upgrades page in CUCM OS Administration
- D. The In-Room control Editor on the webpage of the MX800

Correct Answer: D

Section:

QUESTION 14

Which action prevent toll fraud in Cisco Unified Communications Manager?

- A. Configure ad hoc conference restriction.
- B. Implement toll fraud restriction in the Cisco IOS router.
- C. Allow off-net to off-net transfers.
- D. Implement route patterns in Cisco Unified CM.

Correct Answer: A

Section:

QUESTION 15

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

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

Correct Answer: A

Section:

Explanation:

There is no @ for a wildcard in SIP route patterns. Because no default SIP route patterns exist in Cisco Unified Communications Manager, you must set them up. Domain name examples are cisco.com, my-pc.cisco.com, *.com, rtp-ccm[1-5].cisco.com. Valid characters for domain names are [, -, ., 0-9, AZ, a-z, *, and]. IPv4 address examples 172.18.201.119 or 172.18.201.119/32 (explicit IP host address); 172.18.0.0/16 (IPsubnet); 172.18.201.18/21 (IP subnet). Valid characters for IP addresses: 0-9, ., and /

QUESTION 16

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 controller shut down.
- C. The clock source is incorrect.
- D. The framing is configured incorrectly

Correct Answer: D

Section:

QUESTION 17

Which issue cause slips on a PRI?

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

Correct Answer: C

Section:

QUESTION 18

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

Correct Answer: B

Section:

QUESTION 19

Which two DNS records must be created to configure Service Discovery for on-premises Jabber?

(Choose two.)



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

Correct Answer: A, B

Section:

QUESTION 20

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

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

Correct Answer: D

Section:

QUESTION 21

Which filed is configured to change the caller ID information on a SIP route pattern?

- A. Calling party Transformation Mask
- B. Route partition
- C. Called party Transformation Mask
- D. Connected Line ID Presentation

Correct Answer: A

Section:

QUESTION 22

An incoming off-net call to a user fails. An engineer notices that the off-net call is G.711, but the phone accepts only G.729. Which media resource on a Cisco Unified Border Element and Cisco Unified Communications Manager must the engineer configure to Manage the engineer configure to manage the codec negotiation?

- A. CFB
- B. transcoder
- C. MTP
- D. MOH

Correct Answer: B

Section:

QUESTION 23

Which filed id configured to change the caller ID information on a SIP route pattern?

- A. Calling party transformation Mask



- B. Route partition
- C. Called party Transformation Mask
- D. Connected Line ID Presentation

Correct Answer: A

Section:

QUESTION 24

Given this H.323 gateway configuration and using cisco best practices, how must the called party transformation pattern 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.

The screenshot shows a configuration window with two main sections:

- Pattern Definition:**
 - Pattern*: \+.1
 - 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: (empty)
 - Prefix Digits: 9011
 - Called Party Number Type*: Unknown
 - Called Party Numbering Plan*: Unknown

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* 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

D.

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

Correct Answer: D

Section:

QUESTION 25

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

Correct Answer: B

Section:

QUESTION 26

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. Media Termination point
- B. Conference Bridge
- C. Annunciator
- D. Trusted relay point

Correct Answer: A

Section:

QUESTION 27

Which protocol does prime collaboration Assurance use to poll the health status of different systems in the collaboration environment?

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

Correct Answer: D

Section:

QUESTION 28

Refer to the exhibit.



```
INVITE sip:1010.10.10.219: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:1010.10.10.219>
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 annex=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. PRACK
- B. UPDATE
- C. SUBSCRIBE
- D. NOTIFY
- E. REGISTER

Correct Answer: C, D

Section:

QUESTION 29

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

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

Correct Answer: C

Section:

QUESTION 30

Which two functionalities does Cisco Expressway provide in the Cisco collaboration architecture?

(Choose two.)



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

Correct Answer: B, D

Section:

QUESTION 31

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

- A. Block OffNet to OffNet Transfer
- B. Drop Ad Hoc Conference
- C. H.225 Block Setup destination
- D. Enterprise Feature Access code for conference

Correct Answer: B

Section:

QUESTION 32

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

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

Correct Answer: B

Section:

QUESTION 33

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

- A. 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 delivery faxes.
- B. They collect information from callers by playing a series of questions and recording the answers.
- C. They answer calls, take message, and provide menus of options.
- D. They provide access to a corporate directory by playing an audio list that users and outside callers users and leave messages.

Correct Answer: C

Section:

QUESTION 34

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

Correct Answer: D

Section:

QUESTION 35

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. Configure single route pattern for international calls
- B. Reduce the T302 timer to less than 4 seconds
- C. Create a second route pattern followed by the # wildcard
- D. Set up all international route pattern to 0.1

Correct Answer: C

Section:

QUESTION 36

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. gateway



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

Correct Answer: A

Section:

QUESTION 37

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

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

Correct Answer: A, E

Section:

QUESTION 38

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



Correct Answer: C

Section:

Explanation:

"Most devices, such as computers and servers, cannot mark their own packets and should not be trusted even if they can. Cisco phones, however, can mark their own packets and can be trusted with the QoS markings they provide."

QUESTION 39

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

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

Correct Answer: B

Section:

QUESTION 40

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

Correct Answer: B

Section:

QUESTION 41

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. The date/time group on all phones default to the time zone of the engineer.
- B. The cluster loses access to port 124 at the firewall.
- C. Unified CM sends an NTPv4 packet.
- D. Database replication is not synchronized on the Unified CM nodes.

Correct Answer: D

Section:

QUESTION 42

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



Correct Answer: D

Section:

QUESTION 43

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. The date/time group on all phones default to the time zone of the engineer.
- B. The cluster loses access to port 124 at the firewall.
- C. Unified CM sends an NTPv4 packet.
- D. Database replication is not synchronized on the Unified CM nodes.

Correct Answer: D

Section:

QUESTION 44

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

- A. Cisco Unified IP phones (all models)
- B. H.323 clients
- C. H.225 trunks

- D. Music on hold servers
- E. SIP trunks

Correct Answer: A, B

Section:

QUESTION 45

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
codec aacld
```

B.

```
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 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 mp4a-latm
```

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
```

Correct Answer: B

Section:

QUESTION 46

When a remote office location is set up with limited bandwidth resources, which codec carries the most voice calls?

- A. G.722
- B. G.711
- C. G.723
- D. G.729

Correct Answer: D

Section:

QUESTION 47

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

Correct Answer: A, C

Section:

QUESTION 48

Which command is used in a Cisco IOS XE TDM gateway to configure the voice T1/E1 controller to provide clocking?

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

Correct Answer: A

Section:

QUESTION 49

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 Calling Party Transformation Patterns section, configure the Pattern as 9.011841234567
configure the Discard Digits as Predot 10-10-Dialing
- C. in the Called Party Transformation Pattern Configuration section, configure the Pattern as 9.011841234587
configure the Discard Digits as Predot 10-10-Dialing
- D. in the Calling Party Transformation Patterns section, configure the Pattern as a 9.011841234587
configure the Discard Digits as Predot

Correct Answer: A

Section:

QUESTION 50

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

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

Correct Answer: B

Section:

QUESTION 51

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

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

Correct Answer: A

Section:

QUESTION 52

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 Expressway
- B. Cisco prime collaboration provisioning server
- C. Cisco Unified Border Element
- D. Cisco IM and Presence node

Correct Answer: A

Section:

QUESTION 53

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

Correct Answer: D

Section:

QUESTION 54

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

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

Correct Answer: D

Section:

QUESTION 55

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 HDMI Output
- B. xStatus video Output
- C. xconfiguration video Output
- D. xcommand video status

Correct Answer: B

Section:

QUESTION 56

Which two protocols does Cisco IM and Presence use to authenticate Jabber? (Choose two.)

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

Correct Answer: B, D

Section:

QUESTION 57

Which Cisco Unified communications manager configuration is required for SIP MWI integration?

- A. Select "Redirecting Diversion Header Delivery – Inbound" on the SIP trunk.
- B. Enable "Accept presence subscription" on the SIP trunk security profile.
- C. Select "Redirecting Diversion Header Delivery – outbound" on the SIP trunk.
- D. Enable "Accept unsolicited notification" on the SIP Trunk security profile.



Correct Answer: D

Section:

QUESTION 58

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

Correct Answer: C

Section:

QUESTION 59

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

- A. Software conference bridge on Cisco Unified Communications Manager
- B. Cisco PVDM4-128

- C. Cisco Webex meetings server
- D. Cisco meeting server

Correct Answer: D

Section:

QUESTION 60

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

Correct Answer: D

Section:

QUESTION 61

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



Correct Answer: D

Section:

QUESTION 62

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

Correct Answer: C

Section:

QUESTION 63

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

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

Correct Answer: A

Section:

QUESTION 64

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

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

Correct Answer: B

Section:

QUESTION 65

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

A.

Voice Cos 6 (IP Precedence 4, PHB AF41, or DSCP 24)
Video Cos 5 (IP Precedence 4, PHB EF, or DSCP 34)

B.

Voice Cos 5 (IP Precedence 5, PHB EF, or DSCP 46)
Video Cos 4 (IP Precedence 4, PHB AF41, 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 6, PHB AF41, or DSCP 16)
Video Cos 4 (IP Precedence 5, PHB EF, or DSCP 32)

Correct Answer: B

Section:

QUESTION 66

Refer 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. Noting, the PRI is sending keepalives.
- D. Layer 2 is down on the controller.

Correct Answer: D

Section:

QUESTION 67

Where is the default for maximum session Bit Rate for a region configured?

- A. Region configuration
- B. Enterprise Phone configuration
- C. Service parameter configuration
- D. Enterprise parameters configuration

Correct Answer: C

Section:

QUESTION 68

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

Correct Answer: B

Section:

QUESTION 69

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.
- B. Call ingress must be normalized before the call being routed.
- C. Normalization is not required.
- D. Normalization is achieved by setting up calling search and partitions at the SIP trunk for PSTN connection.

Correct Answer: A

Section:

QUESTION 70



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

Correct Answer: D

Section:

QUESTION 71

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

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

Correct Answer: B

Section:

QUESTION 72

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?

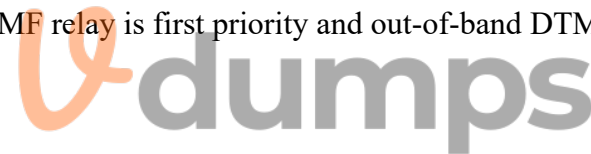
- A. dtmf-relay cisco-rtp
- B. sip-notify dtmf-relay rtp-nte
- C. dtmf-relay rtp-nte sip-notify
- D. dtmf-relay sip-kmpl cisco-rtp

Correct Answer: C

Section:

QUESTION 73

Refer to the exhibit.



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

Correct Answer: B

Section:

QUESTION 75

A customer has Cisco Unity Connection 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 IM and presence
- C. Cisco Unified Communications Manager end user
- D. Active directory

Correct Answer: D

Section:

QUESTION 76

An engineer is configuring a BOT device for a Jabber user in Cisco Unified Communication Manager. Which phone type must be selected?

- A. Cisco Dual mode for Android
- B. Cisco Unified Client services Framework
- C. third-party SIP device
- D. Cisco Dual Mode for iPhone



Correct Answer: A

Section:

QUESTION 77

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

- A. ccm-manager redundant-host
- B. Mgcpcall-agent
- C. Mgcpcapp
- D. ccm-manager fallback-mgcp

Correct Answer: A

Section:

QUESTION 78

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

- A. Standard CTI Enabled
- B. Standard CTI Allow Reception of SRTP key Material
- C. Allow control of device from CTI

D. Standard CTI Secure Connection

Correct Answer: A

Section:

QUESTION 79

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 rate=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 answer with this SDP. Which SDP parameter causes this issue?

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

Correct Answer: A

Section:

QUESTION 80

An engineer is designing a high availability and failover solution for two Cisco Unified Border Element routers. The first router (cube1.abc.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 1 60 cube1.abc.com
- B. _sip_udp.abc.com 60 IN SRV 1 40 cube2.abc.com
- C. _sip_udp.abc.com 60 IN SRV 60 1 cube1.abc.com
- D. _sip_udp.abc.com 60 IN SRV 2 60 cube1.abc.com
- E. _sip_udp.abc.com 60 IN SRV 3 60 cube2.abc.com

Correct Answer: A, B

Section:

QUESTION 81

An engineer is configuring a BOT device for a Jabber user in Cisco Unified Communication Manager. Which phone type must be selected?

- A. Cisco Dual Mode for iPhone
- B. Cisco Dual Mode for Android
- C. third-party SIP device
- D. Cisco Unified Client Services Framework

Correct Answer: B

Section:

QUESTION 82

Which statement about Cisco Unified Communications Manager and Cisco IM and Presence backups is true?

- A. Backups are saved as unencrypted.tar files.
- B. Backups should be scheduled during off-peak hours to avoid system performance issues.
- C. Backups are saved as .tar files and encrypted using the web administrator account.
- D. Backups are not needed for subscriber Cisco Unified Communications Manager and Cisco IM and Presence servers.

Correct Answer: B

Section:

QUESTION 83

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. CUC
- B. CUBE
- C. CMS
- D. Edge



Correct Answer: D

Section:

QUESTION 84

Which DiffServ marking is the most likely to drop packets?

- A. AF32
- B. AF12
- C. AF11
- D. AF13

Correct Answer: D

Section:

QUESTION 85

Endpoint A is attempting to call endpoint B. Endpoint A only supports G.711ulaw with a packetization rate of 20 ms, and endpoint B supports packetization rate of 30 ms for G.711ulaw. Which two media resources are allocated to normalize packetization rates through transrating?

- A. Hardware MTP on Cisco IOS Software
- B. Software MTP on Cisco Unified Communication Manager
- C. Software MTP on Cisco IOS Software
- D. Software transcoder on Cisco unified Communications manager
- E. Hardware transcoder on Cisco IOS Software

Correct Answer: A, B

Section:



QUESTION 86

When configuring Cisco Unified Communications Manager, which configuration enables phones to automatically register to a Cisco Unified Communications publisher when the connection to the subscriber is lost?

- A. Cisco Unified CM Group
- B. Device Pool
- C. SRST
- D. Route Group

Correct Answer: A

Section:

QUESTION 87

Which two elements of a dial plan define the domains that are accessible and are assigned to an endpoint? (Choose two.)

- A. Call Admissions Control
- B. Route patterns
- C. Calling Search Spaces
- D. Translation patterns
- E. partitions

Correct Answer: C, E

Section:

QUESTION 88

An engineer with ID378163512 is designing a new dial plan for a customer that has offices in several countries on four continents around the world. This client also want to integrate with a Microsoft Lync backend, Which dial plan type does the engineer recommend?

- A. SIP URI
- B. TEHO
- C. H.323
- D. E.164

Correct Answer: D

Section:

QUESTION 89

Which version is used to provide encryption for SNMP management traffic in collaboration deployments?

- A. SNMPv1
- B. SNMPv3
- C. SNMPv2
- D. SNMPv2c

Correct Answer: B

Section:



QUESTION 90

Which field of a Real-Time Transport Protocol packet allows receiving devices to detect lost packets?

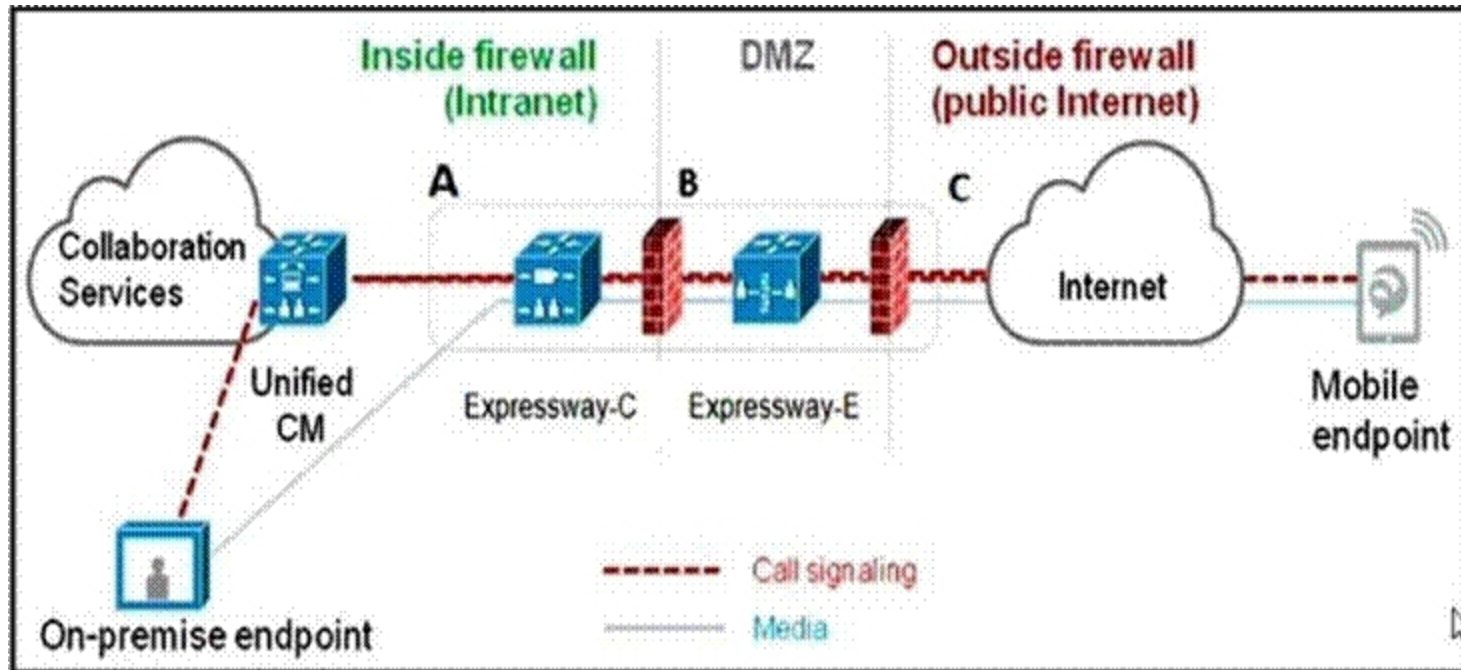
- A. CSRC (Contributing Source ID)
- B. Timestamp
- C. Sequence number
- D. SSRC (Synchronization identifier)

Correct Answer: C

Section:

QUESTION 91

Refer to the exhibit.



When making a call to a MRA client, what are the combinations of protocol on each of the different sections A-B-C?

- A. IP TCP/TLS(A) +SIP TCP/TLS (B) +TLS (C)
- B. SIP TLS (A) +SIP TLS (B) +SIP TLS (C)
- C. SIP TCP/TLS (A) + SIP TLS (B) + SIP TLS (C)
- D. SIP TCP/TLS +SIP TCP/TLS (B) + SIP TCP/TLS (C)

Correct Answer: C

Section:



QUESTION 92

An engineer is designing a load balancing solution for two Cisco Unified Border routers. The first router (cube1.abc.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 1 40 5060 cube2.abc.com
- B. sip_udp.abc.com 60 IN SRV 3 60 5060 cube2.abc.com
- C. sip_udp.abc.com 60 IN SRV 60 1 5060 cube1.abc.com
- D. sip_udp.abc.com 60 IN SRV 2 60 5060 cube1.abc.com
- E. sip_udp.abc.com 60 IN SRV 1 60 5060 cube1.abc.com

Correct Answer: A, E

Section:

QUESTION 93

Refer to the exhibit.

```

23031952: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Applying typeplan for sw-type 0x0 is 0x2 0x1, Calling num 4085556100
23031953: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Sending SETUP callref = 0x12BE callID = 0xA3F5 switch = primary-ni interface = User
23031954: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: TX -> SETUP pd = 8 callref = 0x12BE
Bearer Capability i = 0x8090A2
Standard = CCITT
Transfer Capability = Speech
Transfer Mode = Circuit
Transfer Rate = 64 kbit/s
Channel ID i = 0xA98393
Exclusive, Channel 19
Progress Ind i = 0x8183 - Origination address is non-ISDN
Calling Party Number i = 0x2181, '4085556100'
Plan:ISDN, Type:National
Called Party Number i = 0x91, '011443075552222'
Plan:ISDN, Type:International
23031956: Apr  9 17:43:21.279 EDT: ISDN Se0/1/0:23 Q931: RX <- CALL_PROC pd = 8 callref = 0x92BE
Channel ID i = 0xA98393
Exclusive, Channel 19
23031957: Apr  9 17:43:21.283 EDT: ISDN Se0/1/0:23 Q931: RX <- PROGRESS pd = 8 callref = 0x92BE
Cause i = 0x829F - Normal, unspecified
Progress Ind i = 0x8486 - In-band info or appropriate now available
23031981: Apr  9 17:43:46.802 EDT: ISDN Se0/1/0:23 Q931: TX -> DISCONNECT pd = 8 callref = 0x12BE
Cause i = 0x8090 - Normal call clearing
23031982: Apr  9 17:43:46.822 EDT: ISDN Se0/1/0:23 Q931: RX <- RELEASE pd = 8 callref = 0x92BE
23031983: Apr  9 17:43:46.822 EDT: ISDN Se0/1/0:23 Q931: TX -> RELEASE_COMP pd = 8 callref = 0x12BE

```

A call to an international number has failed. Which action corrects this problem?

- A. Assign a transcoder to the MRGL of the gateway.
- B. Strip the leading 011 from the called party number
- C. Add the bearer-cap speech command to the voice port.
- D. Add the isdn switch-type primart-dms100 command to the serial interface.

Correct Answer: B

Section:

QUESTION 94

Which user group is targeted by MRA services?

- A. Call center agents who dial out to remote customers
- B. Mobile workers in a host desk environment at HQ who log in every morning at possibility a different phone
- C. Production floor users who need wireless mobility in remote areas of the factory
- D. On-the-go mobile workforce who connect to corporate phone services using their own mobile device

Correct Answer: D

Section:

QUESTION 95

Which Cisco Unified Communications Manager feature is to determine the maximum bit rate for a call between two video endpoints?

- A. Partitions
- B. Locations
- C. Regions
- D. transformations

Correct Answer: C

Section:

QUESTION 96



When a 8800 series phone is registered over MRA, where does it register?

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

Correct Answer: A

Section:

QUESTION 97

An engineer must manually provision a Cisco IP Phone 8845 using SIP. Which two fields must be configured for a successful provision? (Choose two.)

- A. Media resources group list
- B. SIP profile
- C. CSS
- D. Location
- E. Device security profile

Correct Answer: B, E

Section:

QUESTION 98

What is the correct statement about CUCM and Cisco IM&P backups?

- A. Backups should be scheduled during hours to avoid system performance issues.
- B. Backups are saved as .tar files and encrypted using the web administrator account.
- C. Backups are saved as unencrypted tar files.
- D. Backups are not needed for subscriber Cisco Unified Communication Manager and Cisco IM and Presence servers.

Correct Answer: A

Section:

QUESTION 99

Due to service provider restriction, Cisco Unified Communications Manager cannot send video in the SDP. Which two options on Cisco Unified CM are configured to suppress video in the SDP is outgoing invites? (Choose two.)

- A. Check the Send send-receive SDP in mid-call INVITE check box on the SIP trunk SIP profile.
- B. Add the audio forced command to voice service voip on the Unified Border Elements.
- C. Check the Retry Call as Audio on the SIP trunk
- D. Set Video bandwidth in the Region settings to 0.
- E. Change the Video Capabilities dropdown on the endpoint to Disabled.

Correct Answer: D, E

Section:

QUESTION 100



There is a saturated link that has traffic shaping configured. How is incoming traffic processed?

- A. Excess traffic is queued for later transmission.
- B. Excess traffic is dropped.
- C. Traffic is compressed so that the traffic fits within the bandwidth of the link.
- D. Excess traffic is queued, and then dropped after the timer expires.

Correct Answer: D

Section:

QUESTION 101

Refer to the exhibit.

```
05:50:14.102: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=21653 ai=127
05:50:14.134: ISDN BR0/1/1 Q921: User RX <- SABMEp sapi=0 tei=0
05:50:14.150: ISDN BR0/1/1 Q921: User TX -> IDREQ ri=19004 ai=127
05:50:14.165: ISDN BR0/1/1 Q921: User RX <- SABMEp sapi=0 tei=0
```

A customer submits this output, captured on a Cisco IOS router. Assuming that an MGCP gateway is configured with a ISDN BRI interface, which BRI changes resolve the issue?

A.

```
interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

B.

```
interface BRI0/1/0
no ip address
isdn switch-type basic-net3
isdn point-to-multipoint-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

C.

```
interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn point-to-point-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

D.

```
interface BRI0/1/1
no ip address
isdn switch-type basic-net3
isdn point-to-multipoint-setup
isdn incoming-voice voice
isdn send-alerting
isdn static-tei 0
```

Correct Answer: C

Section:



QUESTION 102

Which SNMP service must be activated manually on the Cisco Unified Communications Manager after installation?

- A. Cisco CallManager SNMP
- B. SNMP Master Agent
- C. Connection SNMP Agent
- D. Host Resources Agent

Correct Answer: A

Section:

QUESTION 103

Refer to the exhibit.

```
hostname GATEWAY
ccm-manager config
ccm-manager config server 192.168.1.100
ccm-manager mgcp
mgcp call-agent CCMSub1.domain.com 2427 service-type mgcp version 0.1
```

An engineer verifies the configured of an MGCP gateway. The commands are already configured. Which command is necessary to enable MGCP?

- A. Device(config)# mgcp enable
- B. Device(config)# ccm-manager enable
- C. Device (config) # com-manager active
- D. Device (config)# mgcp



Correct Answer: D

Section:

QUESTION 104

What are two key features of the Expressway series? (Choose two.)

- A. VPN connect toward the internal UC resources
- B. B2B calls
- C. IP to PSTN call connectivity
- D. Device registration over the internet
- E. SIP header modification

Correct Answer: B, D

Section:

QUESTION 105

An engineer deploys a Cisco Expressway edge server for a customer who wants to utilize all feature on the server. Which feature does the engineer configure on the Expressway edge?

- A. VTC bridge

- B. H.323 endpoint registration
- C. SIP gateway for PSTN providers
- D. MRA

Correct Answer: D

Section:

QUESTION 106

An engineer is configuring a Cisco Unified Border Element to allow the video endpoints to negotiate without the Cisco Unified Border Element interfering in the process. What should the engineer configure on the Cisco Unified Border Element to support this process?

- A. Configure path-thru content sdp on the voice service.
- B. Configure a hardcoded codec on the dial peers.
- C. Configure a transcoder for video protocols.
- D. Configure codec transparent on the dial peers.

Correct Answer: D

Section:

QUESTION 107

Refer to the exhibit.

The exhibit shows two screenshots from the Cisco Unified CM Administration interface. The top screenshot displays the 'Region Relationships' table, and the bottom screenshot displays the 'Audio Codec Preference List Configuration' for the 'CCNP-COLLAB' list.

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
REGION1	CCNP-COLLAB	60 kbps	384 kbps	2147483647 kbps
REGION2	CCNP-COLLAB	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Audio Codec Preference List Information	
Name*	CCNP-COLLAB
Description*	CCNP-COLLAB
Codecs in List*	G.722 48k G.711 U-Law 64k G.729 8k G.711 A-Law 56k

An engineer is troubleshooting a codec negotiation issue where both endpoints that are involved in the call support the codecs listed in the exhibit. Which audio codec is selected if a call between two endpoints in Region 1 is placed?

- A. G.711u
- B. G.722
- C. G.729
- D. G.711a

Correct Answer: C

Section:

QUESTION 108

Which protocol is used between Cisco Jabber clients for instant messaging and presence?

- A. Jabber
- B. P2P
- C. SIP/SIMPLE
- D. XMPP

Correct Answer: C

Section:

QUESTION 109

What is the function of the Cisco Unity Connection Call Handler?

- A. routes calls to a user based on caller input
- B. queues calls
- C. searches a list of extensions until the call is answered
- D. allows customized scripts for IVR capabilities

Correct Answer: A

Section:

QUESTION 110

A Cisco IP Phone 7841 that is registered to a Cisco Unified Communications Manager with default configuration receives a call setup message. Which codec is negotiated when the SDP offer includes this line of text?
M=audio 498181 RTP/AVP 0 8 97

- A. G.711ulaw
- B. iLBC
- C. G.711alaw
- D. G.722

Correct Answer: A

Section:

QUESTION 111

Which type of greeting in the Call Handler configuration in Cisco Unity Connection overtimes all other greetings?

- A. holidays
- B. supervisory
- C. alternate
- D. priority

Correct Answer: C

Section:



QUESTION 112

A company deploys centralized cisco UCM architecture for a hub location and two remote sites.

The company has only one ITSP connection at the hub connection, and ITSP supports only G.711 calls Remote site A has a 1-Gbps fiber connection to the hub connection and calls to and from remote side A use G.711 codec Remote site B has a 1 T1 connection to the hub location and calls to and from remote site B use

- A. 729 codec
Based on the provided guidance, a Cisco voice engineer must design media resource management for the customer. What is the method that needs to be followed?
- B. configure the hardware transcoder on the site B router.
- C. configure the hardware transcoder on the site A router.
- D. configure the hardware transcoder on the hub location router.
- E. configure the software transcoder on Cisco UCM to support voice calls to and from both remote sites.

Correct Answer: C

Section:

QUESTION 113

When setting a new primary DNS server in the Cisco UCM CLI what is required for the change to take affect?

- A. restart of CallManager service
- B. restart of DirSync service
- C. restart of the network service
- D. restart of TFTP service

Correct Answer: C

Section:

**QUESTION 114**

What happens to voice packets from a Cisco 8845 IP phone in the QoS trust boundary?

- A. The voice packets are not trusted, and the access layer switch reclassifies the packets.
- B. The voice packets are classified by the phone, and the classification is accepted.
- C. The phone and access layer switch negotiate the classification of packets.
- D. Cisco UCM determines how the voice packets are classified.

Correct Answer: D

Section:

QUESTION 115

How is bandwidth allocated to traffic flows in a flow-based WFQ solution?

- A. Each type of traffic flow has equal bandwidth.
- B. Bandwidth is divided among traffic flows. Voice has priority.
- C. Voice has priority and the other types of traffic share the remaining bandwidth
- D. All the bandwidth is divided based on the QoS marking of the packets.

Correct Answer: A

Section:

QUESTION 116

Which application traffic does the DiffServ AF41 class according to the Cisco Collaboration System Solution Reference Network Design?

- A. audio call
- B. messaging
- C. video call
- D. signalling

Correct Answer: C

Section:

QUESTION 117

Which task is required when configuring self-provisioning for an end user in Cisco UCM?

- A. Enable Auto-Registration.
- B. Associate the end user to the Standard CCM Super Users group.
- C. Associate the end user to a SIP Profile.
- D. Disable Auto-Registration.

Correct Answer: A

Section:

QUESTION 118

Which DTMF relay method configured on a SIP dial-peer will ensure that a media resource is not invoked by Unified CM for calls to UCCX IVRs?

- A. dtmf-relay sip-kpml
- B. dtmf-relay rtp-nte
- C. sdtmf-relay h245-signal
- D. dtmf-relay sip-notify

Correct Answer: D

Section:

Explanation:

https://www.cisco.com/c/dam/en/us/td/docs/iosxml/ios/voice/cube_uccx/cube_uccx_interop_bestprac_guide.pdf

QUESTION 119

An engineer is configuring IP telephony. The network relies on DHCP to provide TFTP server addresses to the endpoints. Policy requires the endpoints to receive two server addresses. Which DHCP option must be configured?

- A. 66
- B. 143
- C. 150
- D. 166

Correct Answer: C

Section:

QUESTION 120

Which action enables Cisco MRA?

- A. Cisco UCC Express clients can obtain VPN connectivity to Cisco UCC Enterprise.
- B. VPN connectivity can be established to Cisco UCM.
- C. Clients such as Cisco Jabber can use call control on Cisco UCM.
- D. Internal SIP clients registered to Cisco UCM can call external companies

Correct Answer: C

Section:

QUESTION 121

End users report bad video quality and voice choppiness on Cisco Collaboration endpoints. The engineer changed the device pool the users were in but did not correct the problem. Which action should be taken to troubleshoot this issue?

- A. Set the service parameter Use Video Bandwidth Pool for Immersive Video Calls to "false".
- B. Check for duplex/speed mismatches between the network port settings of the system and network switch.
- C. Restart the Cisco Location Bandwidth Manager service on the Cisco UCM publisher.
- D. Use direct IP address calls between two endpoints to troubleshoot call quality issues.

Correct Answer: A

Section:

QUESTION 122

Cisco UCM delays routing of a call during digit analysis with an overlapping dial plan. How long is the default wait time?

- A. 5 seconds
- B. 10 seconds
- C. 15 seconds
- D. 20 seconds

Correct Answer: C

Section:

QUESTION 123

Refer to the exhibit.

```
voice translation-rule 1
rule 1 /^[2-9].....$/ /\0/ type any subscriber
rule 2 /^[2-9]..[2-9].....$/ /\0/ type any subscriber
```

What is the result of applying these two rules to a voice translation-profile for use with an ISDN PRI on a Cisco Voice gateway?

- A. The ISDN Type is modified to the administrators defined value.
- B. The ISDN Plan is modified to the administrators defined value.
- C. The leading Plus is stripped from the numeric phone number.
- D. Any Zero is stripped from the numeric phone number.

Correct Answer: A

Section:

QUESTION 124

An administrator would like to set several Cisco Jabber configuration parameters to only apply to mobile clients (iOS and Android). How does the administrator accomplish this with Cisco Jabber 12.9 and Cisco UCM 12.5?

- A. Assign the desired configuration file to "Mobile" Jabber Client Configuration in the Service Profile.
- B. Deploy jabber-config-user.xml on iOS and Android devices.
- C. Upload the jabber-config.enc file to TFTP
- D. Create a user profile in Jabber Policies.

Correct Answer: A

Section:

QUESTION 125

Refer to the exhibit.

```
voice class dpq 2000
dial-peer 2001 preference 1
dial-peer 2002 preference 2
dial-peer 2003 preference 3

dial-peer voice 1001 voip
description INBOUND
session protocol sipv2
session target ipv4:10.0.0.1
destination dpq 2000
incoming called-number 5T

dial-peer voice 2001 voip
destination-pattern 5506
session protocol sipv2
session target ipv4:10.0.0.2

dial-peer voice 2002 voip
destination-pattern 5507
session protocol sipv2
session target ipv4:10.0.0.3

dial-peer voice 2003 voip
destination-pattern 5507
session protocol sipv2
session target ipv4:10.0.0.4
```



A Cisco UCM user with directory number 4401 dials 5507, and the call is routed to a Cisco Unified Border Element. Which IP address will the call be sent to?

- A. 10.0.0.3
- B. 10.0.0.1
- C. 10.0.0.2

D. 10.0.0.4

Correct Answer: C

Section:

QUESTION 126

What is the maximum DNS SRV entries that should be defined in the SIP Trunk destination address field in Cisco UCM?

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

Correct Answer: C

Section:

QUESTION 127

Configuration of DNS is required to achieve a fully functional Cisco UCM system. Cisco UCM uses DNS to resolve fully qualified domain names to IP addresses for which destinations?

- A. H.323
- B. trunk
- C. AAR
- D. MRA

Correct Answer: B

Section:



QUESTION 128

Refer to the exhibit.

Time	Source	Destination	Info
18.683437	10.117.34.222	10.0.101.10	50310 → 5060 [SYN] Seq=0 Win=64240 Len=0 MSS=1460 WS=256 SACK
18.938881	10.117.34.222	10.0.101.10	50314 → 5060 [SYN] Seq=0 Win=64240 Len=0 MSS=1460 WS=256 SACK
21.686680	10.117.34.222	10.0.101.10	[TCP Retransmission] 50310 → 5060 [SYN] Seq=0 Win=64240 Len=0
21.941993	10.117.34.222	10.0.101.10	[TCP Retransmission] 50314 → 5060 [SYN] Seq=0 Win=64240 Len=0
21.687008	10.117.34.222	10.0.101.10	[TCP Retransmission] 50310 → 5060 [SYN] Seq=0 Win=64240 Len=0
21.942784	10.117.34.222	10.0.101.10	[TCP Retransmission] 50314 → 5060 [SYN] Seq=0 Win=64240 Len=0

An administrator is attempting to register a SIP phone to a Cisco UCM but the registration is failing.

The IP address of the SIP Phone is 10.117.34.222 and the IP 1 address of the Cisco UCM is 10.0.101.10. Pings from the SIP phone to the Cisco UCM are successful. What is the cause of this issue and how should it be resolved?

- A. The certificates on the SIP phone are not trusted by the Cisco UCM. The SIP phone must generate new certificates.
- B. DNS lookup for the Cisco UCM FQDN is failing. The SIP phone must be reconfigured with the proper DNS server.
- C. An NTP mismatch is preventing connection of the TCP session between the SIP phone and the Cisco UCM. The SIP phone and Cisco UCM must be set with identical NTP sources.
- D. An network device is blocking TCP port 5060 from the SIP phone to the Cisco UCM. This device must be reconfigured to allow traffic from the IP phone.

Correct Answer: D

Section:

QUESTION 129

Which two approaches are taken to implement CAC within Cisco UCM in full-mesh topology?
(Choose two.)

- A. Device Pool
- B. RSVP-enabled locations
- C. location-based CAC
- D. Audio Codec Preference List
- E. physical location

Correct Answer: B, C

Section:

QUESTION 130

Which two steps should be taken to provision after the Self-Provisioning feature was configured for end users?

- A. Dial the self-provisioning IVR extension and associate the phone to an end user.
- B. Plug the phone into the network.
- C. Ask the Cisco UCM administrator to associate the phone to an end user.
- D. Enter settings menu on the phone and press *,*,# (star, star, pound).
- E. Dial the hunt pilot extension and associate the phone to an end user.

Correct Answer: A, B

Section:

QUESTION 131

On a cisco catalyst switch which command is required to send CDP packets on a switch port that configures a cisco IP phone to transmit voice traffic in 802.1q frames tagged with the voice VLAN ID 221?

- A. Device(config-if)# switchport access vlan 221
- B. Device(config-if)# switchport trunk allowed vlan 221
- C. Device(config-if)# switchport vlan voice 221
- D. Device(config-if)# switchport voice vlan 221

Correct Answer: D

Section:

QUESTION 132

An end user at a remote site is trying to initiate an ad hoc conference call to an end user at the main site. The conference bridge is configured to support G.711 remote user phone only support G.729. The remote end user receives an error message on the phone "cannot complete conference call what is the cause of the issue?"

- A. The remote phone does not have the conference feature assigned.
- B. A Media Termination Point is missing
- C. The transcoder resource is missing.
- D. A software conference bridge is not assigned

Correct Answer: C

Section:



QUESTION 133

An administrator is in the process of moving cisco unity connection mailboxes between mailbox stores the administrator notices that some mailboxes have active message waiting indicators what happens to these mailboxes when they are moved?

- A. The move will fail If MWI status Is active.
- B. Moving the mailboxes from one store to another fails If MWI Is turned on.
- C. If source and target mailbox store are not disabled. MWI status is not retained.
- D. The MWI status is retained after a mailbox is moved from one store to another.

Correct Answer: D

Section:

QUESTION 134

Which command must be defined before an administrator changes the linecode value on an ISDN T1 PRI in slot 0/2 on an IOS-XE gateway?

- A. isdn incoming-voice voice
- B. pri-group timeslots 1-24
- C. card type t1 0 2
- D. voice-port 0/2/0:23

Correct Answer: C

Section:

QUESTION 135

Which DSCP class selector is necessary to mark scavenger traffic?

- A. CS1
- B. AF21
- C. AF11
- D. CS2

Correct Answer: A

Section:

QUESTION 136

Which service must be enabled when LDAP on cisco UCM is used?

- A. Cisco AXL Web Service
- B. Cisco CallManager SNMP Service
- C. Cisco DirSync
- D. Cisco Bulk provisioning Service

Correct Answer: C

Section:

QUESTION 137

Which two protocols should be configured for the cisco unity connection and cisco UCM integration?



- A. SIP
- B. H.323
- C. MGCP
- D. RTP
- E. SCCP

Correct Answer: A, E

Section:

QUESTION 138

An end user at a remote site is trying to initiate an Ad Hoc conference call to an end user at the main site. The conference receives an error message on the phone: Cannot complete conference call.

What is the cause of the issue?

- A. The transcoder resource is missing.
- B. The remote phone does not have the conference feature assigned
- C. A software conference bridge is not assigned.
- D. A Media Termination Point is missing.

Correct Answer: A

Section:

QUESTION 139

What dialed number match this cisco UCM route pattern?

1[23]XX

- A. 1200 through 1399 only
- B. 1230 through 1239 only
- C. 12300 through 12399 only
- D. 1200 through 1300 only

Correct Answer: A

Section:

QUESTION 140

Users want their mobile phones to be able to access their cisco unity connection mailboxes with only having to enter their voicemail pin at the login prompt calling pilot number where should an engineer configure this feature?

- A. message settings
- B. greetings
- C. alternate extensions
- D. transfer rules

Correct Answer: C

Section:

QUESTION 141

Refer to the exhibit.



```
Via: SIP/2.0/TCP
10.10.10.2:5060;branch=a8bH5bK7954A198F
From:
<sip:012345678@10.10.10.2>;tag=8D79AF62-DB2
To: <sip:90123456@10.10.4.14>;
tag=811681~ffa80926-5fac-4cc5-b802-
2dbde74ae7w2-
v=0
o=CiscoSystemsCCM-SIP 811681 1 IN IP4
10.10.4.14
s=SIP Call
c=IN IP4 10.5.4.3
t=0 0
m=audio 27839 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

Which Codec is negotiated?

- A. G.729
- B. ILBC
- C. G.711ulaw
- D. G.728

Correct Answer: D

Section:

QUESTION 142

An engineer is setting up a system with voice and video endpoints using auto-QoS policy on the switches. Which DSCP values are expected for interactive voice and video?

- A. EF AND AF41
- B. EF AND AF21
- C. EF AND CS3
- D. EF AND CS6

Correct Answer: A

Section:

QUESTION 143

During the Cisco IP Phone registration process the TFTP download fails. What are two reasons for this issue? (Choose two)

- A. The DNS server was not specified, which is needed to resolve a hostname in an Option 150 string.
- B. The Cisco IP Phone does not know the IP address of the TFTP server
- C. The Cisco IP Phone does not know the IP address of any of the Cisco UCM Subscriber nodes
- D. Option 100 string was not specified, or an incorrect Option 100 string was specified
- E. Option 150 string was not specified, or an incorrect Option 150 string was specified



Correct Answer: B, E

Section:

QUESTION 144

```
dial-peer voice 2 voip
destination-pattern 5555678
sessiontarget ipv4:10.5.6.7
codec g729r8
fax protocol t38 ls-redundancy 0 hs-redundancy 0 fallback cisco
fax rate voice
```

Refer to the exhibit. An administrator configures fax dial-peers on a Cisco IOS gateway and finds that faxes are not working correctly. Which change should be made to resolve this issue?

- A. codec g723ar63
- B. codec g729br81
- C. codec g726r32
- D. codec g711ulaw

Correct Answer: D

Section:

QUESTION 145

An administrator is integrating a Cisco Unity Express module to a Cisco UCME system. A test call is placed to the Cisco Unity Express pilot number, but the administrator receives a busy signal. The dialpeer is configured as follows:

```
dial-peer voice 4100 voip
destination-pattern 41..
session protocol sipv2
session target ipv4:10.1.10.1
dtmf-relay sip-notify no vad
```

How is this issue resolved?

- A. The dial-peer needs to be reconfigured to support the G.711 voice codec.
- B. The dial-peer needs to be reconfigured to support vad.
- C. The destination pattern needs to be changed to match the dialed number of 4100.
- D. The dial-peer needs to be reconfigured to support H.323 instead of SIP.

Correct Answer: A

Section:

QUESTION 146



What is the difference between Cisco Unified Border Element and a conventional Session Border Controller?

- A. SIP security
- B. DTMF interworking
- C. Voice policy
- D. Address hiding

Correct Answer: C

Section:

QUESTION 147

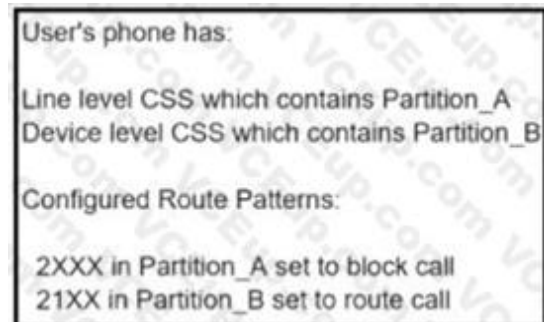
How does traffic policing respond to violations?

- A. AH traffic is treated equally.
- B. Excess traffic is retransmitted.
- C. Excess traffic is dropped.
- D. Excess traffic is queued.

Correct Answer: C

Section:

QUESTION 148



Refer to the exhibit. An engineer is confining class of control for a user in Cisco UCM. Which change will ensure that the user is unable to call 2143?

- A. Change line partition to Partition_A
- B. Change line CSS to only contain Partition_B
- C. Set the user's line CSS to <None>
- D. Set the users device CSS to <None>

Correct Answer: D

Section:

QUESTION 149

Users dial a 9 before a 10-digit phone number to make an off-net call. All 11 digits are sent to the Cisco Unified Border Element before going out to the PSTN. The PSTN provider accepts only 10 digits. Which configuration is needed on the Cisco Unified Border Element for calls to be successful?

- voice translation-rule 1 rule 1 /^9/ //
- voice translation-rule 1 rule 1 /^9(.....)/ //
- voice translation-rule 1 rule 1 /^9.+/ //
- voice translation-rule 1 rule 1 /^9...../ //

- A. Option A
- B. Option B
- C. Option C
- D. Option D

Correct Answer: A

Section:

QUESTION 150

A customer enters no IP domain lookup on the Cisco IOS XE gateway to suppress the interpreting of invalid commands as hostnames Which two commands are needed to restore DNS SRV or A record resolutions? (Choose two.)

- A. ip dhcp excluded-address
- B. ip dhcp-sip
- C. ip dhcp pool
- D. transport preferred none
- E. ip domain lookup

Correct Answer: D, E

Section:

QUESTION 151

An administrator is asked to implement toll fraud prevention in Cisco UCM, specifically to restrict offnet to off-net call transfers How is this implemented?

- A. Implement time-of-day routing.
- B. Use the correct route filters.
- C. Set the appropriate service parameter
- D. Enforce ad-hoc conference restrictions

Correct Answer: C

Section:

QUESTION 152

A network administrator with ID392116981 has determined that a WAN link between two Cisco UCM clusters supports only 1 Mbps of bandwidth for voice traffic How many calls does this link support if G.711 as the audio codec is used?

- A. 15
- B. 16



- C. 13
- D. 12

Correct Answer: D

Section:

QUESTION 153

An engineer is asked to implement on-net/off-net call classification in Cisco UCM Which two components are required to implement this configuration? (Choose two)

- A. SIP trunk
- B. CTI route point
- C. route group
- D. route pattern
- E. SIP route patterns

Correct Answer: A, B

Section:

QUESTION 154

An administrator installed a Cisco Unified IP 8831 Conference Phone that is failing to register Which two actions are taken to troubleshoot the problem? (Choose two)

- A. Disable HSRP on the access layer switch.
- B. Verify that the switch port of the phone is enabled
- C. Check the RJ-65 cable.
- D. Verify that the RJ-11 cable is plugged into the PC port.
- E. Verify that the phone's network can access the option 150 server.



Correct Answer: B, E

Section:

QUESTION 155

An administrator has been asked to implement toll fraud prevention in Cisco UCM Which tool is used to begin this process?

- A. Cisco UCM class of service
- B. Cisco Unified Mobility
- C. Cisco UCM Access Control Group restrictions
- D. Cisco Unified Real-Time Monitoring Tool

Correct Answer: A

Section:

QUESTION 156

What is the most efficient way to configure a speed dial to number 1111 on multiple phones using the Self-Provisioning feature in Cisco UCM?

- A. Configure each phone with the same DN and then configure a speed dial of 1111 on one of the phones manually by navigating to Devices>Phones and modifying the phone. This will automatically replicate to all the other phones.
- B. Configure the phones with a phone button template that has speed dials then specify 1111 as a speed dial via BAT>Phones>Add/Update phones web page to specify the speed dial number after the phone is in Cisco UCM.

- C. Configure a universal device template that has speed dials in the phone button configuration field, then allow the user to specify 1111 as a speed dial via the Cisco Unified Communications Self Care Portal.
- D. Configure a universal device template that has speed dials in the phone button configuration field, then specify 1111 as a speed dial by editing the speed dial within the universal device template.

Correct Answer: D

Section:

QUESTION 157

An administrator with ID392116981 is receiving complaints of pixilation smearing, and pulsing of video calls between two offices that are connected by a WAN. Assuming that QoS is implemented on the WAN connection, which classification is used to mark the video traffic, according to the Cisco QoS baseline?

- A. AF31
- B. CS3
- C. EF
- D. AF41

Correct Answer: D

Section:

QUESTION 158

What is required for Cisco UCM to accept SIP calls with a URI in the format of "sip 2001@cucmpub.cisco.com"?

- A. Define Cluster Fully Qualified Domain Name under Servers in Cisco UCM.
- B. Change the Destination Address to a Fully Qualified Domain Name on the SIP trunk.
- C. Set the SIP URI Handling to True in CallManager Service Parameters.
- D. Define Cluster Fully Qualified Domain Name in Enterprise Parameters.



Correct Answer: D

Section:

QUESTION 159

Configuration of DNS is required to achieve a fully functional Cisco UCM system Cisco UCM uses DNS to resolve fully qualified domain names to IP addresses for which destinations?

- A. Application server name
- B. Primary TFTP Server for option 150
- C. Cisco Unified Communications Manager Name
- D. Sip trunk

Correct Answer: D

Section:

QUESTION 160

Which feature is enabled by Cisco Mobile and Remote Access?

- A. Internal SIP clients registered to Cisco UCM can call external companies.
- B. Clients such as Cisco Jabber can use call control on Cisco UCM.
- C. VPN connectivity can be established to Cisco UCM.
- D. Cisco UCC Express clients can obtain VPN connectivity to Cisco UCC Enterprise.

Correct Answer: B

Section:

QUESTION 161

What is required when deploying co-resident VMs by using Cisco UCM?

- A. Provide a guaranteed bandwidth of 10 Mbps.
- B. Ensure that applications will perform QoS.
- C. Avoid hardware oversubscription.
- D. Deploy the VMs to a server running Cisco UCM.

Correct Answer: C

Section:

QUESTION 162

A customer asked to integrate Unity Connection with Cisco UCM using SIP protocol Which two features must be enabled on SIP security profiles? (Choose two)

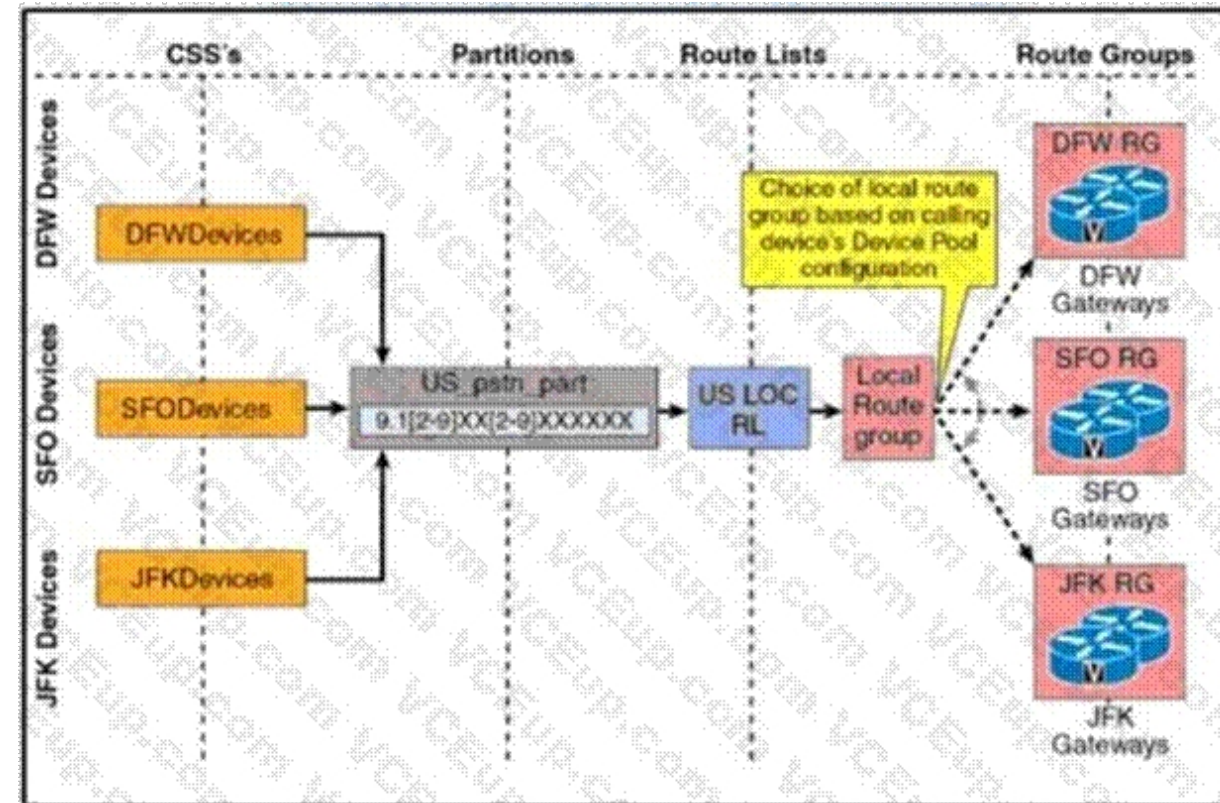
- A. Accept presence subscription
- B. Enable application level authorization.
- C. Allow charging header
- D. Accept replaces header
- E. Accept unsolicited notification

Correct Answer: D, E

Section:



QUESTION 163



Refer to the exhibit A user takes a phone from San Francisco to New York for a short reassignment.



The phone was set up to use the San Francisco device pool, and device mobility is enabled on the Cisco UCM. The user makes a call that matches a route pattern in a route list that contains the Standard Local Route Group. To where does the call retreat?

- A. The call egresses in New York because the device automatically is assigned a New York device pool and uses the local gateway
- B. The call egresses in San Francisco because the user uses device mobility and is allowed to roam while still keeping the number and resources assigned in San Francisco
- C. The call fails because the Standard Local Route Group is being used only if no configuration is set for the device pools
- D. The call fails because device mobility is turned on, and the phone is not configured in New York. The engineer must configure which sites the device should be roaming to

Correct Answer: B

Section:

QUESTION 164

A user reports that when receiving an inbound call on their IP Phone from the PSTN they are unable to transfer this call to another PSTN number. What is the reason for this failure?

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

Correct Answer: C

Section:

QUESTION 165

An administrator is designing a new Cisco UCM for a company with many departments and firm structure on their communications policies. The administrator must make sure that these communication policies are reflected in the phone system setup. Certain departments cannot be accessed directly, even if they have dedicated DID numbers. Some phones, especially public phones, must not be able to dial international numbers. Which type of function is configured to control which device is allowed to call another device in Cisco UCM?

- A. partitions and calling search spaces
- B. calling patterns and route patterns
- C. regions and device pools
- D. links and pipes

Correct Answer: A

Section:

QUESTION 166

What are two functions of Cisco Expressway in the Collaboration Edge? (Choose two)

- A. Expressway-E provides a VPN entry point for Cisco IP phones with a Cisco AnyConnect client using authentication based on certificates.
- B. Expressway-C provides encryption for Mobile and Remote Access but not for business-to-business communications.
- C. The Expressway-C and Expressway-E pair can interconnect H 323-to-SIP calls for voice.
- D. The Expressway-C and Expressway-E pair can enable connectivity from the corporate network to the PSTN via a T1/E1 trunk.
- E. Expressway-E provides a perimeter network that separates the enterprise network from the Internet.

Correct Answer: C, E

Section:

QUESTION 167

An administrator configured SIP route patterns cisco com and uc(5-7] cisco 1 com in Cisco UCM A route pattern must be added for ucce2.cisco1.com. Which two patterns complete the configuration*? (Choose two.)

- uc[c]e[12]*.cisco1.com
- ucce^[1-2].*
- succ[ce2]*.cisco1.com
- uc[c]e[^0-12]*.cisco1.com
- uc.*2.cisco1.com

- A. Option A
- B. Option B
- C. Option C
- D. Option D
- E. Option E

Correct Answer: A, E

Section:

QUESTION 168



Which open-standard protocol is the basis for the Extensible Messaging and Presence Protocol?

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

Correct Answer: C

Section:

QUESTION 169

An employee of company ABC just quit. The IT administrator deleted the employee's userid from the active directory at 10 a.m. on March 4th. The nightly sync occurs at 10 p.m. daily. The IT administrator wants to troubleshoot and find a way to delete the userid as soon as possible. How is this issue resolved?

- A. Wait until 10 pm on March 4th when the user is automatically removed from Cisco UCM.
- B. Wait until 10 pm on March 5th when the user is automatically removed from Cisco UCM.
- C. Wait until 3:15 a.m. on March 6th for garbage collection to remove the user from Cisco UCM.
- D. Wait until 3:15 a.m. on March 5th for garbage collection to remove the user from Cisco UCM.

Correct Answer: C

Section:

QUESTION 170

An administrator is configuring LDAP for Cisco UCM with Active Directory integration. A customer has requested to use 'ipphone' instead of 'telephoneNumber' as the phone number attribute. Where does the administrator specify this attribute mapping in Cisco UCM?

- A. LDAP directory custom user fields
- B. LDAP Authentication
- C. LDAP directory user fields
- D. LDAP custom Filter

Correct Answer: C

Section:

QUESTION 171

An administrator needs to stop the leading 9 from outbound calls on an IOS Voice Gateway and ensure that the system handles 911 emergency calls. Which configuration is needed to accomplish this task?

A.

```
voice translation-rule 1
rule 1 /9?911/ /911/
rule 2 /^9\(.*\)/ /\0/
```

B.


```
voice translation-rule 1
rule 1 /9?911/ /911/
rule 2 /^9\(.*\)/ /\1/
```

C.

```
voice translation-rule 1
rule 1 /^9\(.*\)/ /\1/
rule 2 /9?911/ /911/
```

D.

```
voice translation-rule 1
rule 1 /9?911/ /911/
rule 2 /^9\(.*\)/ /&/
```



Correct Answer: B

Section:

QUESTION 172

What are two differences between media flow-around and media flow-through on cisco unified Border element? (Choose two.)

- A. When using media flow-through, the call signaling and media are passed through the Cisco Unified Border Element
- B. When using media flow-through, the call signaling goes through the Cisco Unified Border Element, but media is not passed through it.
- C. When using media flow-around, the call signaling goes through the Cisco Unified Border Element, but media is not passed through it
- D. When using media flow-around, both call signaling and media do not go through the Cisco Unified Border Element
- E. When using media flow-through, call signaling goes through the Cisco Unified Border Element but media does not

Correct Answer: A, C

Section:

QUESTION 173

An administrator must configure the local route group feature on cisco UCM. Which step will enable this feature?

- A. For each route list configure a route group to use as a Local Route Group.
- B. For each route pattern, select the Local Route Group as the destination.
- C. For each route group, check the box for the Local Route Group feature.
- D. For each device pool, configure a route group to use as a Local Route Group for that device pool

Correct Answer: D

Section:

QUESTION 174

In the cisco expressway solution, which two features does mobile and Remote access provide?
(Choose two)

- A. VPN-based enterprise access for a subset of Cisco Unified IP Phone models
- B. Secure reverse proxy firewall travelsal connectivity
- C. the ability to register third-party SIP or H.323 devices on Cisco UCM without requiring VPN
- D. the ability of Cisco IP Phones to access the enterprise through VPN connection
- E. the ability for remote users and their devices to access and consume enterprise collaboration applications and services

Correct Answer: B, E

Section:

QUESTION 175

AN engineer is implementing toll fraud prevention for incoming calls cluster-wide on cisco UCM.
What is the first step to configure this feature?

- A. Set service parameter 'Block OffNet to OffNet Transfer' to False
- B. Configure blocking for inbound calls based on caller ID.
- C. Set service parameter "System Remote Access Blocked Numbers' to True
- D. Set service parameter 'Block OffNet To OffNet Transfer' to True

Correct Answer: A

Section:

QUESTION 176

User A Calls user. The call gets connected, but the quality is bed. What are two reasons for this issue?
(Choose two)

- A. Incorrect partition
- B. No region relationship
- C. Network congestion
- D. Incorrect QoS
- E. Incompatible codec

Correct Answer: C, D

Section:

QUESTION 177



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 SOP 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. Payload type 110 was negotiated rather than type 101.
- C. G.729 rather than G.711ulaw was negotiated.
- D. DTMF was not negotiated on the call.

Correct Answer: D

Section:

QUESTION 178

What dialled numbers match this Cisco UCM route pattern?

1[23]XX

- A. 1200 through 1300 on1ly
- B. 1200 through 1399 only
- C. 12300 through 12399 only
- D. 1230 through 1239 only

Correct Answer: A

Section:

QUESTION 179

```
Server: Cisco-SIPGW/100-10.4.3.04
CSeq: 101 OPTIONS
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
INFO, REGISTER
Allow-Events: telephone-event
Accept: application/sdp
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Content-Type: application/sdp
Content-Length: 369

v=0
o=CiscoSystemsSIP-GW-UserAgent 6414 4717 IN IP4 10.8.140.23
s=SIP Call
c=IN IP4 10.8.140.23
t=0 0
m=audio 0 RTP/AVP 18 0 8 4 15
c=IN IP4 10.8.140.23
m=image 0 udptl t38
c=IN IP4 10.8.140.23
a=T38FaxVersion:0
a=T38FaxBitRate:9600
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:320
a=T38FaxUdpEC:t38UDPRedundancy
```



Refer to the exhibit. A customer wants the SIP 200 OK shown to advertise codecs in the following order:



After correcting the codec preferences. What should the audio payload show in the SIP Traces?



- m=audio 0 RTP/AVP 0 18 4 15
- m=audio 0 RTP/AVP 4 0 18 15
- m=audio 0 RTP/AVP 0 8 18 4 15
- m=audio 0 RTP/AVP 18 0 8 4 15

- A. Option A
- B. Option B
- C. Option C
- D. Option D

Correct Answer: D
Section:

QUESTION 180

Where is urgent priority enabled to bypass the T302 timer?

- A. route partition
- B. CTI port
- C. directory number
- D. transformation pattern

Correct Answer: A
Section:

QUESTION 181

Refer to the exhibit.



NAME	TTL	CLASS	TYPE	Priority	Weight	Port	Target Address
_sip_tcp.sample.com	86400	IN	SRV	10	60	5060	server1.sample.com
_sip_tcp.sample.com	86400	IN	SRV	10	30	5060	server2.sample.com
_sip_tcp.sample.com	86400	IN	SRV	5	20	5060	server3.sample.com

An administrator must fix the SRV records to ensure that server1.sample.com is always contacted first from the three servers. Which solution should apply to resolve this issue?

- A. Priority = 10, Weight = 5
- B. Priority = 100, Weight = 90
- C. Priority = 10, Weight = 10
- D. Priority = 5, Weight = 70

Correct Answer: D
Section:

QUESTION 182

An engineer must configure a route pattern that can route all + E.164 globalized international numbers for the dial plans of all countries. Which cisco UCM configuration accomplishes this task?

A.

Pattern Definition	
Route Pattern*	\+!
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	--Not Selected--
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

B.

Pattern Definition	
Route Pattern*	011.!
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	--Not Selected--
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

C.

Pattern Definition	
Route Pattern*	\+1.!
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	--Not Selected--
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

D.

Pattern Definition	
Route Pattern*	\+011!
Route Partition	International_PT
Description	Globalized Routing
Numbering Plan	--Not Selected--
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	PSTN_RL

Correct Answer: A

Section:

QUESTION 183

Which cisco collaboration Edge architecture product allows remote end points to leverage corporate on-premises cisco Unified communications infrastructure?

- A. Cisco Webex
- B. Cisco VPN Client
- C. Cisco Unified Communications Mobile and Remote Access
- D. Cisco Umbrella



Correct Answer: C

Section:

QUESTION 184

Refer to the exhibit.

```

ROUTER-1(config)# policy-map LLQ_POLICY
ROUTER-1(config-pmap)# class VOICE
ROUTER-1(config-pmap-c)# bandwidth 170
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# class VIDEO
ROUTER-1(config-pmap-c)# bandwidth remaining percent 30
ROUTER-1(config-pmap-c)# exit
ROUTER-1(config-pmap)# exit

```

An engineer must modify the existing QoS policy-map statement to implement LLQ for voice traffic. Which change must the engineer make in the configuration?

- A. bandwidth 170 to LLQ 170
- B. bandwidth 170 to priority 170
- C. bandwidth 170 to reserve 170
- D. bandwidth 170 to percent 170

Correct Answer: B

Section:

QUESTION 185

Refer to the exhibit.

```
admin:utils ntp status
ntpd (pid 17428) is running...
```

remote	refid	st	t	when	poll	reach	delay	offset	jitter
+192.168.1.1	17.253.14.125	2	u	39	64	3	0.456	-0.236	0.116
+192.168.1.2	17.253.14.125	2	u	38	64	3	0.817	-0.695	0.395

A collaboration engineer needs to replace the original, single NTP server that was configured during the initial install of a Cisco UCM server. What is the first step to accomplish this task?

- A. Restart the NTP service on Cisco UCM.
- B. Enable NTP authentication for the new NTP server on Cisco UCM.
- C. Stop the NTP service on Cisco UCM.
- D. Delete the original NTP server from Cisco UCM.

Correct Answer: A

Section:

QUESTION 186

An administrator executes the debug isdn q931 command while debugging a failed call. After a test call is placed, the logs return a disconnect cause code of 1. What is the cause of this problem?

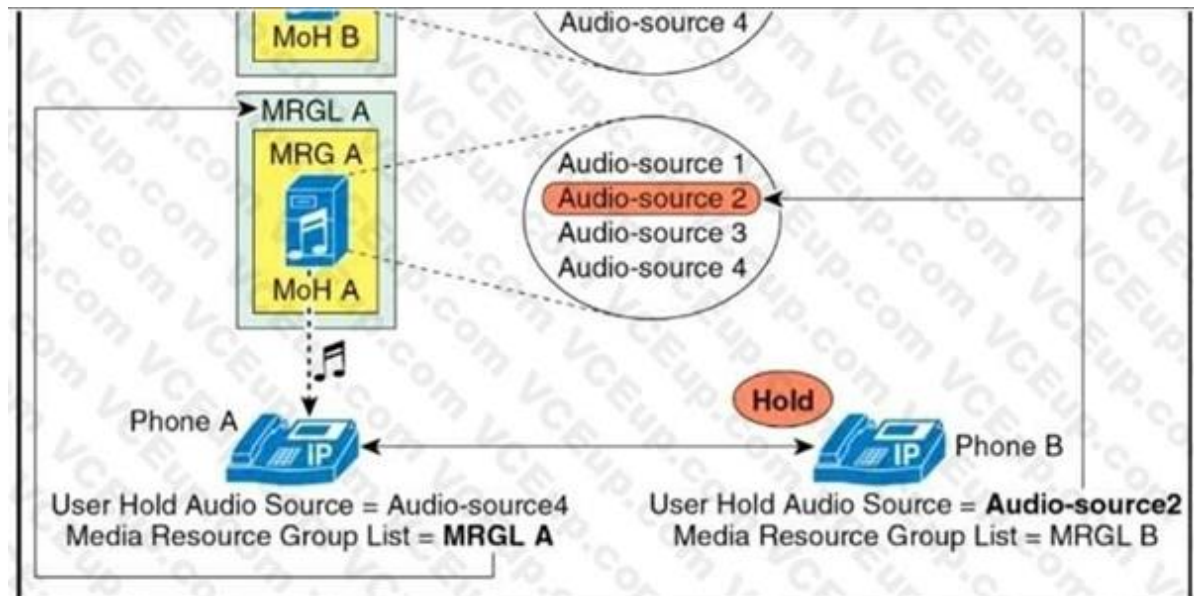
- A. The destination number rejects the call
- B. The destination number is busy
- C. The media resource is unavailable
- D. The dialed number is not assigned to an endpoint

Correct Answer: D

Section:

QUESTION 187

Refer to the exhibit.



There is a call flow between Phone A and Phone B. Phone B (holder) places Phone A (holder) on hold. Which MRGL and Audio Source are played to Phone A?

- A. MRGL A and Audio Source 4
- B. MRGL B and Audio Source 4
- C. MRGL B and Audio Source 2
- D. MRGL A and Audio Source 2

Correct Answer: D

Section:



QUESTION 188

A SIP phone has been configured in the system with MAC address 0030 96D2 D5CB. The phone retrieves the configuration file from the Cisco UCM. Which naming format is the file that is downloaded?

- A. SIPO03096D2D5CB.cnf.xml
- B. SEP003096D2D5CB.cnf.xml
- C. SEP0030406706705.cnf
- D. SIP0030406706705.cnf

Correct Answer: B

Section:

QUESTION 189

Where in Cisco UCM are codec negotiations configured for endpoints?

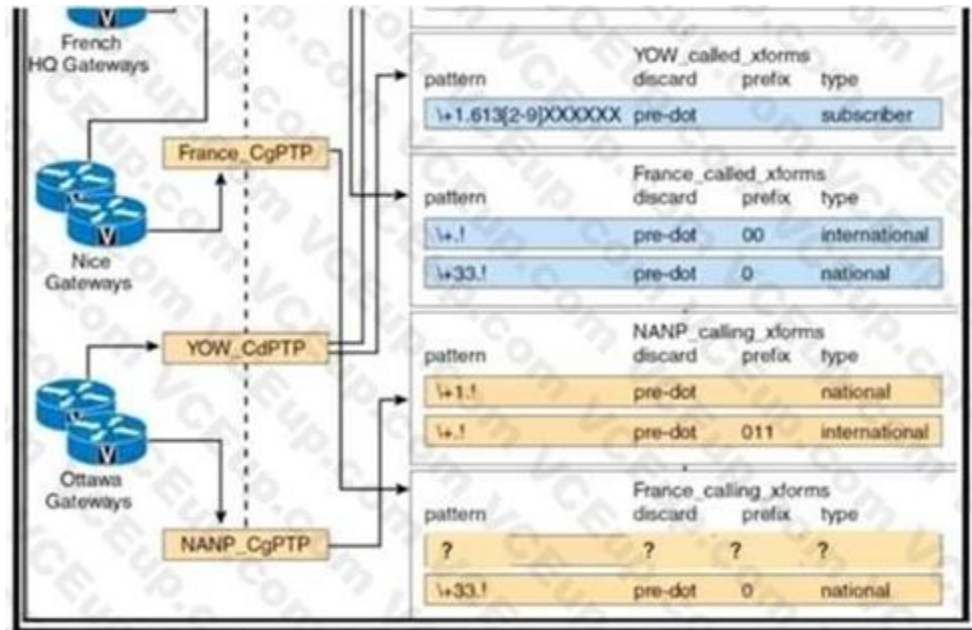
- A. in enterprise parameters
- B. under device profiles in device settings
- C. under regions using preference lists
- D. in in-service parameters

Correct Answer: C

Section:

QUESTION 190

Refer to the exhibit.



A call from +1 613 555 1234 that is sent out through the Nice Gateways should result in a calling party of 001 613 555 1234 with the numbering type "international" Which configuration accomplishes this goal?

- A. \+1 pre-dot 00 international
- B. \+ 0011 pre-dot 1 international
- C. \+1 ! none pre-dot 001 international
- D. 613XXXXXXX none +011 international

Correct Answer: C

Section:



QUESTION 191

Refer to the exhibit.

Use Device Pool Called Party Transformation CSS
 Calling Party Transformation CSS: < None >

Use Device Pool Calling Party Transformation CSS
 Calling Party Selection*: Originator

Calling Line ID Presentation*: Default

Calling Name Presentation*: Default

Calling and Connected Party Info Format*: Deliver DN only in connected party

Redirecting Diversion Header Delivery - Outbound
 Redirecting Party Transformation CSS: < None >

Use Device Pool Redirecting Party Transformation CSS

Caller Information
 Caller ID DN:
 Caller Name:
 Maintain Original Caller ID DN and Caller Name in Identity Headers

Unanswered calls do not reach the voicemail associated with the phones Instead, callers receive the default greeting Which action fixes the configuration?

- A. Reboot Cisco Unity Connection

- B. Review the conversation manager logs on Cisco Unity Connection
- C. Check the box "Redirecting Diversion Header Delivery - Outbound", then reset the trunk
- D. Check the box "Redirecting Diversion Header Delivery - Outbound"

Correct Answer: C

Section:

QUESTION 192

Callers from a branch report getting busy tones intermittently when trying to reach colleagues in other office branches during peak hours. An engineer collects Cisco CallManager service traces to examine the situation. The traces show:

```
50805567.000 |07:35:39.676 |Sd|Sig |StationOutputDisplayNotify |restart0 |StationD(1,100,63,6382) |StationCdpc(1,100,64,4725)
|1,100,40,6,706705^^^ |[[R:N-H:0,N:0,L:0,V:0,Z:0,D:0] TimeOutValue=10 Status=x807 UnicodeStatus= Locale=1 50805567.001 |07:35:39.676
|ApplInfo |StationD: (0006382) DisplayNotify timeOutValue=10 notify='x807' content='Not Enough Bandwidth' ver=85720014.
```

What should be fixed to resolve the issue?

- A. codec configuration
- B. region configuration
- C. class of service configuration
- D. geolocation configuration

Correct Answer: C

Section:

QUESTION 193

An engineer implements a new Cisco UCM based telephony system per these requirements:

- The local Ethernet bandwidth is sized based on the total bandwidth per call • A G.736 codec is used.
- The bit rate is 64 kbps
- The codec sample interval is 10 ms.
- The voice payload size is 160 bytes per 20 ms.

What should the size of the Ethernet bandwidth be per call?

- A. 31.2 kbps
- B. 38.4 kbps
- C. 55.2 kbps
- D. 87.2 kbps

Correct Answer: D

Section:

QUESTION 194

Refer to the exhibit.

The logo for Vdumps.com, featuring a stylized orange 'V' followed by the word 'dumps' in a grey, lowercase, sans-serif font.

Add successful

DHCP Server Information

Host Server*	192.168.10.240
Primary DNS IPv4 Address	192.168.99.1
Secondary DNS IPv4 Address	
Primary TFTP Server IPv4 Address (Option 150)	192.168.10.244
Secondary TFTP Server IPv4 Address (Option 150)	
Bootstrap Server IPv4 Address	
Domain Name	
TFTP Server Name (Option 66)	
ARP Cache Timeout (sec)*	0
IP Address Lease Time (sec)*	0
Renewal (T1) Time (sec)*	0
Rebinding (T2) Time (sec)*	0

Save Delete Copy Add New

A collaboration engineer configures Cisco UCM to act as a DHCP server. What must be done next to configure the DHCP server?

- A. Restart the TFTP service under Cisco Unified Serviceability
- B. Add a DHCP subnet to the DHCP server under Cisco UCM Administration
- C. Restart the Cisco DHCP Monitor Service under Cisco Unified Serviceability.
- D. Add the new DHCP server to the primary DNS server

Correct Answer: B
Section:



QUESTION 195

Which action is required for a firewall configuration on a Mobile and Remote Access through Cisco Expressway deployment?

- A. The traversal zone on Expressway-C points to Expressway-E through the peer address field on the traversal zone, which specifies the Expressway-E server address. For dual NIC deployments, set the Expressway-E address using an FQDN that resolves the IP address of the internal interface
- B. The internal firewall must allow these inbound and outbound connections between Expressway-C and Expressway-E. SIP: HTTPS (tunnelled over SSH between C and E): TCP 2222; TCP 7001; Traversal Media: UDP 2776 to 2777 (or 36000 to 36011 for large VM/appliance); XMPP: TCP 7400.
- C. The external firewall must allow these inbound connections to Expressway: SIP: TCP 5061; HTTPS: TCP 8443; XMPP: TCP 5222; Media: UDP 36002 to 59999.
- D. Do not use a shared address for Expressway-E and Expressway-C, as the firewall cannot distinguish between them. If static NAT for IP addressing on Expressway-E is used, ensure that any NAT operation on Expressway-C does not resolve the same traffic IP address. Shared NAT is not supported.

Correct Answer: D
Section:

QUESTION 196

Which characteristic of distributed class-based weighted fair queuing addresses jitter prevention?

- A. It provides additional granularity by allowing a user to create classes
- B. It minimizes jitter by implementing a priority queue for voice traffic
- C. It uses a priority queue for voice traffic to avoid jitter
- D. It provides additional granularity by allowing a user to define custom classes

Correct Answer: B

Section:

QUESTION 197

A company hosts a conference call with no local users How does the administrator stop the conference from continuing?

- A. modifies the Block OffNet to OffNet Transfer service parameter
- B. removes the transcoder
- C. changes the codecs that are supported on the conference resource
- D. modifies the Drop Ad Hoc Conference service parameter

Correct Answer: D

Section:

QUESTION 198

What happens to voice packets from a Cisco 8845 IP Phone in the QoS trust boundary?

- A. The phone and access layer switch negotiate the classification of packets
- B. The voice packets are classified by the phone, and the classification is accepted.
- C. Cisco UCM determines how the voice packets are classified
- D. The voice packets are not trusted, and the access layer switch reclassifies the packets.

Correct Answer: B

Section:

QUESTION 199

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

Correct Answer: B

Section:

QUESTION 200

A collaboration engineer adds a voice gateway to Cisco UCM The engineer creates a new gateway device in Cisco UCM, selects VG320 as the device type and selects MGCP as the protocol What must be done next to add the gateway to the Cisco UCM database?

- A. Configure the module in slot 0 of the new gateway
- B. Select the DTMF relay type for the gateway
- C. Select a device pool for the new gateway



D. Add the FQDN or hostname of the device

Correct Answer: D

Section:

QUESTION 201

An administrator configures Cisco UCM to use UDP for SIP signaling and finds that an endpoint cannot make calls Which action resolves this issue?

- A. Change the phone security profile.
- B. Change the SIP dial rules.
- C. Change the SIP profile
- D. Change the common phone profile.

Correct Answer: A

Section:

QUESTION 202

What is a capability of Cisco Expressway?

- A. It provides access to on-premises Cisco Unified Communications infrastructure for remote endpoints
- B. It has remote endpoint enrollment with Certificate Authority Proxy Function
- C. It functions as an analog telephony adapter
- D. It gives directory access for remote users via Cisco Directory Integration

Correct Answer: A

Section:



QUESTION 203

After an engineer implements the FAC and CMC features together, users report that calls take almost one minute to complete and that they occasionally hear the reorder tone Which two actions address this issue? (Choose two.)

- A. Advise the user to press the "#" button after dialling the FAC and CMC codes
- B. Change the code if the problem persists i/i-Advise the user to hang up and try again
- C. Adjust the T302 timer from the default of 15 seconds to 5 seconds to shorten the interdigit timer
- D. Do not wait for the tones immediately dial the FAC and CMC

Correct Answer: A, C

Section:

QUESTION 204

An administrator troubleshoots call flows and suspects that there are issues with the dial plan Which tool enables a quick analysis of the dial plan and provides call flows of dialed digits?

- A. Digit Analysis Analyzer
- B. Dialed Number Analyzer
- C. Cisco Dial Plan Analyzer
- D. Dial Plan Analyzer

Correct Answer: B

Section:

QUESTION 205

A company's employees have been complaining that they have been unable to select options on the internal IVR of the help desk IT support has been given Cisco UCM traces and below is the snippet of the SDP of the INVITE packet.

```
m=audio 25268 RTP/AVP 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=ptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

How is this issue resolved?

- A. Configure CODEC for G.729
- B. Configure DTMF for RFC 2833.
- C. Configure CODEC for G 722
- D. Configure DTMF for KPML

Correct Answer: D

Section:

QUESTION 206

Which Cisco Unity Connection handler plays a greeting that announces the option to dial a user extension by default?

- A. the Goodbye call handler
- B. the Operator call handler
- C. the Directory handler
- D. the Interview handler

Correct Answer: C

Section:

QUESTION 207

Refer to the exhibit.

```
dspfarm profile 1 transcode universal
codec g729r8
  codec g729ar8
  codec g711ulaw
  codec g711alaw
maximum sessions 8
associate application SCCP
```

Which two codec permutations should be transcoded by this dspfarm? (Choose two)

- A. G.729ar8 to G.711alaw
- B. iLBC to G.711ulaw
- C. G.729r8 to G711ulaw
- D. G.722 to G 729r8
- E. G.729br8toG711alaw

Correct Answer: A, C

Section:

QUESTION 208

Which two steps are required for bulk configuration transactions on the Cisco UCM database utilizing BAT? (Choose two)

- A. A data file in comma-separated values format must be uploaded to Cisco UCM
- B. A server template must be created in Cisco UCM.
- C. A device template must be created in Cisco UCM.
- D. A data file in Abstract Syntax Notation One format must be uploaded to Cisco UCM
- E. A data file in Extensible Markup Language format must be uploaded to Cisco UCM

Correct Answer: A, E

Section:

QUESTION 209

A collaboration engineer troubleshoots issues with a Cisco IP Phone 7800 Series. The IPv4 address of the phone is reachable via ICMP and HTTP and the phone is registered to Cisco UCM. However, the engineer cannot reach the CLI of the phone. Which two actions in Cisco UCM resolve the issue? (Choose two)

- A. Enable FIPS Mode under Product Specific Configuration Layout in Cisco UCM
- B. Enable Settings Access under Product Specific Configuration Layout in Cisco UCM.
- C. Enable SSH Access under Product Specific Configuration Layout in Cisco UCM
- D. Disable Web Access under Product Specific Configuration Layout in Cisco UCM
- E. Set a username and password under Secure Shell Information in Cisco UCM



Correct Answer: C, D

Section:

QUESTION 210

What are two characteristics of jitter in voice and video over IP communications? (Choose two.)

- A. The packets arrive at varying time intervals
- B. The packets arrive out of sequence
- C. The packets arrive with frame errors
- D. The packets arrive at uniform time intervals.
- E. The packets never arrive due to tail drop

Correct Answer: A, B

Section:

QUESTION 211

Refer to the exhibit.

SIP Trunk Security Profile Information	
Name*	Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Encrypted
Incoming Transport Type*	TLS
Outgoing Transport Type	TLS
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	
Incoming Port*	5061

An administrator configures a secure SIP trunk on Cisco UCM. Which value is needed in the Secure Certificate Subject or Subject Alternate Name field to accomplish this task?

- A. the fully qualified domain name of the remote device that is configured on the SIP trunk
- B. the common name of the Cisco UCM CallManager certificate
- C. the common name of the remote device certificates
- D. the fully qualified domain name of all Cisco UCM nodes that run the CallManager service

Correct Answer: A

Section:

QUESTION 212

An administrator works with an ISDN PRI that is connected to a third-party PBX. The ISDN link does not come up; and the administrator finds that the third-party PBX uses the QSIG signalling method. Which command enables the Cisco IOS Gateway to use QSIG signalling on the ISDN link?

- A. isdn switch-type basic-qsig
- B. isdn switch-type basic-ni
- C. isdn switch-type primary-qsig
- D. isdn incoming-voice voice

Correct Answer: A

Section:

QUESTION 213

Which type of input is required when configuring a third-party SIP phone?

- A. digest user
- B. serial number
- C. manufacturer
- D. authorization code

Correct Answer: A

Section:

QUESTION 214

Refer to the exhibit.

```

admin:utils ntp status
ntpd (pid 17428) is running...

  remote           refid      st t when poll reach  delay  offset  jitter
-----
*192.168.1.1      17.253.14.125  2 u  36  64  377  0.435  0.039  0.047
192.168.1.2      .INIT.        16 u  -  64   0  0.000  0.000  0.000

```

A collaboration engineer adds a redundant NTP server to an existing Cisco Collaboration solution On the Cisco UCM OS Administration page the new NTP server shows as "Not Accessible" Which action resolves this issue?

- A. Restart NTPD on the Cisco UCM server
- B. Start the NTP service on the new NTP server
- C. Configure the "reach" value as "377" for the new NTP server
- D. Delete and re-add the new NTP server via the Cisco UCM command-line interface

Correct Answer: B

Section:

QUESTION 215

A company has an excessive number of call transfers to local and long-distance PSTN from Cisco Unity Connection voicemail Which action in the Cisco Unity Connection restriction table resolves this issue?

- A. Implement password complexity on voicemail boxes to prevent accounts from being compromised
- B. Block PSTN patterns on Default Transfer, Default Outdial, and Default System Transfer
- C. Create a custom restriction table ?????????? and block it
- D. Create a custom restriction table ***** and Wock it



Correct Answer: B

Section:

QUESTION 216

What are two features of Cisco Expressway that the customer gets if Expressway-C and Expressway-E are deployed? (Choose two.)

- A. highly secure firewall-traversal technology to extend organizational reach
- B. complete endpoint registration and monitoring capabilities for devices that are local and remote
- C. session-based access to comprehensive collaboration for remote workers, without the need for a separate VPN client
- D. additional visibility of the edge traffic in an organization
- E. utilization and adoption metrics of all remotely connected devices

Correct Answer: B, C

Section:

QUESTION 217

Which DiffServ marking is the most likely to drop packets?

- A. AF12
- B. AF32
- C. AF13

D. AF11

Correct Answer: C

Section:

QUESTION 218

An administrator needs to create a partial PRI consisting of the first seven timeslots available. Which configuration snippet configures the ISDN E1 PRI for this task?

A.

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#pri-group timeslots 1-7
```

B.

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#interface Serial0/0/0:15
2900(config-controller)#pri-group timeslots 1-7
```

C.

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-timeslots 1-7
```

D.

```
config t
2900(config)#isdn switch-type primary-ni
2900(config)#controller e1 0/0/0
2900(config-controller)#pri-group timeslots 1-7
```

Correct Answer: D

Section:



QUESTION 219

A collaboration engineer configures Global Dial Plan Replication for multiple Cisco UCM clusters. The local cluster acts as the hub cluster and the remaining clusters act as spoke clusters. Which service must the engineer configure on the local cluster?

- A. Location Conveyance on intercluster SIP trunks
- B. intra-Cluster Communication Signaling
- C. Mobility Cross Cluster
- D. Intercluster Lookup Service

Correct Answer: D

Section:

QUESTION 220

Which type of message must an administrator configure in the SIP Trunk Security Profile for a Message Waiting Indicator light to work with a SIP integration between Cisco UCM and Cisco Unity Connection?

- A. TCP port 5060
- B. SIP Register
- C. Unsolicited NOTIFY
- D. 200 OK

Correct Answer: C

Section:

QUESTION 221

Based on the provided guidance a cisco voice engineer must design media resource management for the customer what is the method that needs to be followed?

- A. configure the hardware transcoder on the hub location router
- B. configure the hardware transcoder on the site A router
- C. configure the hardware transcoder on the site B router.
- D. configure the software transcoder on Cisco UCM to support voice calls to and from both remote sites

Correct Answer: A

Section:

QUESTION 222



A Cisco UCM administrator wants to enable the Self-Provisioning feature for end users. Which two prerequisites must be met first? (Choose two.)

- A. Cisco Extended Functions service must be running
- B. End users must have a secondary extension
- C. End users must belong to Standard CCM Admin Users group, the Standard CCM End Users group, and the Standard CCM Self-Provisioning group
- D. End users must have a primary extension
- E. End users must be associated to a user profile or feature group template that includes a universal line template and universal device template

Correct Answer: D, E

Section:

QUESTION 223

The IP phones at a customer site do not pick an IP address from the DHCP. An engineer must temporarily disable LLDP on all ports of the switch to leave only CDP. Which two commands accomplish this task? (Choose two.)

- A. Switch(config)# no lldp transmit
- B. Switch# configure terminal
- C. Switch# copy running-config startup-config
- D. Switch(config)# no lldp run
- E. Switch(config)# interface GigabitEthernet1/0/1

Correct Answer: B, D

Section:

QUESTION 224

To provide high-quality voice and take advantage of the full voice feature set, which two access layer switches provide support? (Choose two.)

- A. Use multiple egress queues to provide priority queuing of RTP voice packet streams and the ability to classify or reclassify traffic and establish a network trust boundary.
- B. Deploy RSVP to improve VoIP QoS only where it can have a positive impact on quality and functionality where there is limited bandwidth and frequent network congestion.
- C. Map audio and video streams of video calls (AF41 and AF42) to a class-based queue with weighted random early detection.
- D. Use 802.1Q trunking and 802.1p for proper treatment of Layer 2 CoS packet marking on ports with phones connected.
- E. Implement IP RTP header compression on the serial interface to reduce the bandwidth required per voice call on point-to-point links.

Correct Answer: A, D

Section:

Explanation:

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/srnd/design/guide/cmesrnd/nstrct.html

QUESTION 225

A customer wants to conduct B2B video calls with a partner using an on-premises conferencing solution. Which two devices are needed to facilitate this request? (Choose two.)

- A. Expressway-C
- B. MGCP gateway
- C. Cisco Unified Border Element
- D. Cisco TelePresence Management Suite
- E. Expressway-E

Correct Answer: A, E

Section:

QUESTION 226

A company wants to provide remote users with access to its on-premises Cisco collaboration features. Which components are required to enable Cisco Mobile and Remote Access for the users?

- A. Cisco Unified Border Element, Cisco IM and Presence Server, and Cisco Video Communication Server
- B. Cisco Unified Border Element, Cisco UCM, and Cisco Video Communication Server
- C. Cisco Expressway-E, Cisco Expressway-C, and Cisco UCM
- D. Cisco Expressway-E, Cisco IM and Presence Server, and Cisco Video Communication Server

Correct Answer: C

Section:

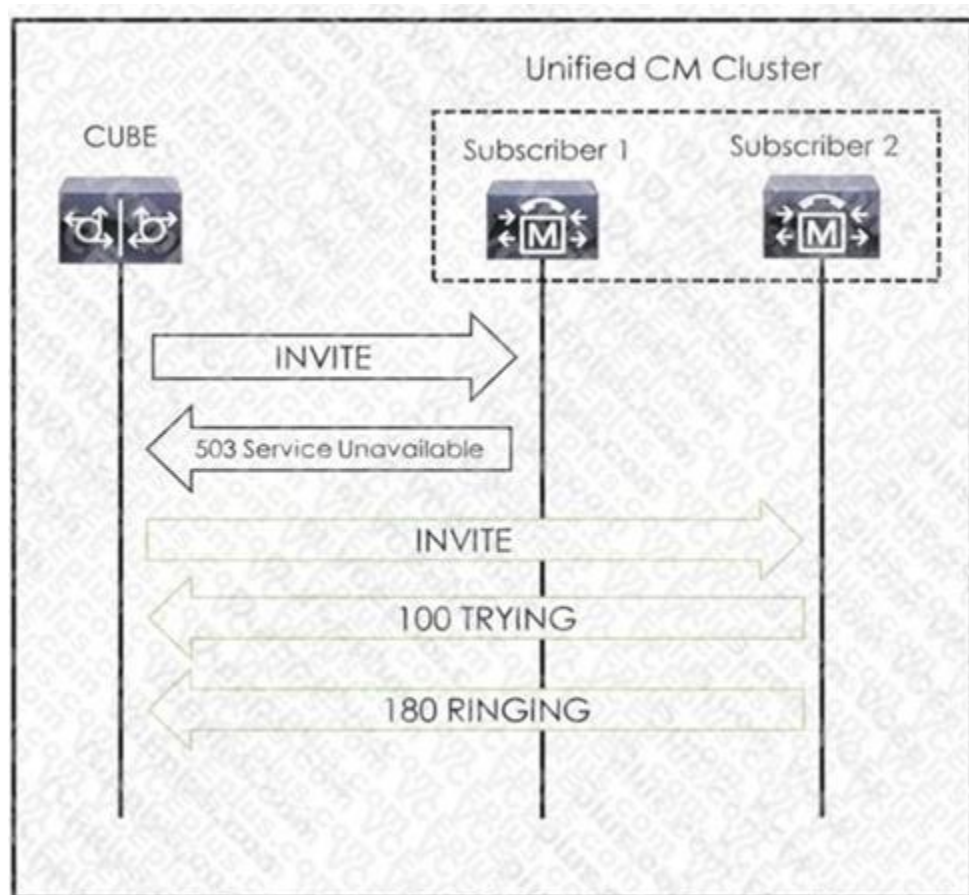
Explanation:

Reference:

<https://www.cisco.com/c/en/us/td/docs/solutions/CVD/Collaboration/enterprise/12x/120/collbcvd/edge.html>

QUESTION 227

Refer to the exhibit.



Vdumps

Cisco Unified Border Element is attempting to establish a call with Subscriber 1, but the call fails.

Cisco Unified Border Element then retries the same call with Subscriber 2, and the call proceeds normally. Which action resolves the issue?

- A. Verify that the Run On All Active Unified CM Nodes checkbox is enabled.
- B. Verify that the correct calling search space is selected for the Inbound Calls section.
- C. Verify that the Significant Digits field for Inbound Calls is set to All.

D. Verify that the PSTN Access checkbox is enabled.

Correct Answer: A

Section:

QUESTION 228

Refer to the exhibit.

SIP Trunk Security Profile Information

Name* CUP Non Secure SIP Profile

Description

Device Security Mode Non Secure

Incoming Transport Type* TCP+UDP

Outgoing Transport Type TCP

Enable Digest Authentication

Nonce Validity Time (mins)* 600

Secure Certificate Subject or Subject Alternate Name

Incoming Port* 5060

Enable Application level authorization

Accept presence subscription

Accept out-of-dialog refer**

Accept unsolicited notification

Accept replaces header

Transmit security status

Allow charging header

SIP V.150 Outbound SDP Offer Filtering* Use Default filter



A collaboration engineer is configuring the Cisco UCM IM and Presence Service. Which two steps complete the configuration of the SIP trunk security profile? (Choose two.)

- A. Check the box to accept replaces header.
- B. Check the box to allow charging header.
- C. Check the box to enable application-level authorization.
- D. Check the box to transmit security status.
- E. Check the box to accept unsolicited notification.

Correct Answer: A, E

Section:

Explanation:

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/im_presence/configAdminGuide/12_0_1/cup0_b_config-admin-guide-imp-1201/cup0_b_config-admin-guide-imp-1201_chapter_0100.html

QUESTION 229

Refer to the exhibit.

The top screenshot shows the 'Region configuration' page in Cisco Unified CM Administration. The 'Region Information' section shows 'Name' as 'REGION1'. The 'Region Relationships' table is as follows:

Region	Audio Codec Preference List Configuration	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
REGION1	CCNP COLLAB	60 kbps	384 kbps	2147483647 kbps
REGION2	CCNP COLLAB	64 kbps (G.722, 6.711)	Use System Default (384 kbps)	Use System Default (2900000000 kbps)
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

The bottom screenshot shows the 'Audio Codec Preference List Configuration' page for 'CCNP COLLAB'. The 'Status' is 'Ready'. The 'Audio Codec Preference Information' section shows 'Name' as 'CCNP COLLAB' and 'Description' as 'CCNP COLLAB'. The 'Codecs in List' section contains the following list:

- G.722 48k
- G.711 U-Law 64k
- G.729 8k
- G.711 A-Law 56k



An engineer is troubleshooting this video conference issue:

A video call between a Cisco 9971 in Region1 and another Cisco 9971 in Region1 works.

As soon as the Cisco 9971 in Region1 conferences in a Cisco 8945 in Region2, the Region1 endpoint cannot see the Region2 endpoint video.

What is the cause of this issue?

- A. Cisco 8945 does not have a camera connected.
- B. Maximum Audio Bit Rate must be increased.
- C. Maximum Session Bit Rate for Video Calls is too low.
- D. Maximum Session Bit Rate for Immersive Video Calls is too low.

Correct Answer: A

Section:

QUESTION 230

If a phone needs to register with cucm1.cisco.com, which network service assists with the phone registration process?

- A. SMTP
- B. ICMP

- C. DNS
- D. SNMP

Correct Answer: C

Section:

QUESTION 231

Which two protocols are proxied over an Expressway-E/C pair when a MRA login including phone services is performed? (Choose two.)

- A. SCCP
- B. HTTPS
- C. H.323
- D. SRTP
- E. SIP

Correct Answer: B, E

Section:

QUESTION 232

What is the purpose of Mobile and Remote Access (MRA) in the Cisco UCM architecture?

- A. MRA is used to access Webex cloud services only if authenticated with on-premises LDAP service.
- B. MRA is used to make secure PSTN calls by Cisco UCM only while on-premises authentication.
- C. MRA is used to make B2B calls through Expressway registration.
- D. MRA is used to access the collaboration services offered by Cisco UCM from off-premises network connections.

Correct Answer: C

Section:

QUESTION 233

Which external DNS SRV record must be present for Mobile and Remote Access?

- A. _cisco-uds._tcp.example.com
- B. _collab-edge._tls.example.com
- C. _collab-edge._tcp.example.com
- D. _cisco-uds._tls example.com

Correct Answer: B

Section:

Explanation:

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/expressway/config_guide/X14-0-1/mra/exwy_b_mra-deployment-guide-x1401/exwy_m_requirements-for-mra.html

QUESTION 234

When a call is delivered to a gateway, the calling and called party number must be adapted to the PSTN service requirements of the trunk group. If a call is destined locally, the + sign and the explicit country code must be replaced with a national prefix. For the same city or region, the local area code must be replaced by a local prefix as applicable. Assuming that a Cisco UCM has a SIP trunk to a New York gateway (area code 917), which two combinations of solutions localize the calling and called party for a New York phone user? (Choose two.)

- A. Configure the gateway to translate the calling number and apply it to the dial peer. Combine it with a translation profile for called numbers.

```
! voice translation-rule 1
rule 1 /^1917/ //
rule 2 /^[+]1917/ //
!
```

voice translation-profile strip+1 translate calling 1 !

- B. Configure two calling party transformation patterns:
\+1917.CCCCCC, strip pre-dot, numbering type: subscriber
\+!, strip pre-dot, numbering type: national
- C. Configure two called party transformation patterns:
\+1917.XXXXXXX, strip pre-dot, numbering type: subscriber
\+1.!, strip pre-dot, numbering type: national
- D. Configure the gateway to translate called numbers and apply it to the dial peer. Combine it with a translation profile for calling numbers.
! voice translation-rule 1
rule 1 /^1917!/ //
rule 2 /^[+]1917!/ //
!
voice translation-profile strip+1 translate called 1 !
- E. Configure two calling party transformation patterns:
\+1917.XXXXXXX, strip pre-dot, numbering type: subscriber
\+1.!, strip pre-dot, numbering type: national

Correct Answer: D, E

Section:

QUESTION 235

A customer reports that the Cisco UCM toll-fraud prevention does not work correctly, and the customer is receiving charges for unapproved international calls as a result. Which two configuration changes resolve the issues? (Choose two.)

- A. Use Cisco Unified Border Element to debug the calls.
- B. Disable call forwarding on the phone.
- C. Make the calls route through a firewall.
- D. Mark patterns as off-net or on net.
- E. Modify the Block OffNet to OffNet Transfer service parameter.

Correct Answer: D, E

Section:

QUESTION 236

An administrator configures international calling on a Cisco UCM cluster and wants to minimize the number of route patterns that are needed. Which route pattern enables the administrator to match variable-length numbers?

- A. 9.011@
- B. 9.011#
- C. 9.011*
- D. 9.011!

Correct Answer: C

Section:

Explanation:

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab12/collab12/dialplan.html

QUESTION 237

A Cisco UCM administrator sets up new route patterns to support phones in four different locations, all with local gateways. The administrator wants to use the same route pattern for all four locations. How must the system be configured to achieve this goal?

- A. Use transforms in the route groups.
- B. Use standard local route groups.
- C. Add a CSS to each local gateway.
- D. Use CSS alternate routing rules.

Correct Answer: B

Section:

QUESTION 238

What are the last two bits of a DS field in DiffServe Byte used for?

- A. AFxy
- B. ECN
- C. INC
- D. RMI

Correct Answer: B

Section:

Explanation:

Reference:

https://www.cisco.com/c/en/us/td/docs/ios-xml/ios/qos_dfsrv/configuration/15-mt/qos-dfsrv-15-mt-book/qos-dfsrv.html



QUESTION 239

What are two QoS requirements for VoIP traffic? (Choose two.)

- A. Loss must be no more than 1 percent.
- B. Voice traffic must be marked to DSCP AF41.
- C. One-way latency must be no more than 200 ms.
- D. Voice traffic must be marked to DSCP EF.
- E. Average one-way jitter is greater than 50 ms.

Correct Answer: A, D

Section:

QUESTION 240

What is an indicator of network congestion in VoIP communications?

- A. jitter increase due to variable delay
- B. video loss due to video frame corruption
- C. gaps in the voice due to packet loss
- D. discards in the interface of routers and switches

Correct Answer: A

Section:

QUESTION 241

An engineer configures a SIP trunk for MWI between a Cisco UCM cluster and Cisco Unity Connection. The Cisco UCM cluster fails to receive the SIP notify messages. Which two SIP trunk settings resolve this issue? (Choose two.)

- A. transmit security status
- B. accept unsolicited notification
- C. allow charging header
- D. accept out-of-band notification
- E. accept out-of-dialog refer

Correct Answer: B, E

Section:

QUESTION 242

What are the predefined call handlers in Cisco Unity Connection?

- A. opening greeting, operator, and goodbye
- B. opening greeting, welcome, and default system
- C. greetings, operator, and closed
- D. caller input, greetings, and transfer

Correct Answer: A

Section:

Explanation:

Reference:

https://www.cisco.com/en/US/docs/voice_ip_comm/connection/1x/administration/guide/acm030.html

QUESTION 243

An administrator configures the voicemail feature in a Cisco collaboration deployment. The user mailboxes must be configured when the Cisco Unity Connection server is configured. Which action accomplishes this task?

- A. Configure an SCCP integration with Cisco UCM.
- B. Configure a SIP integration with Cisco UCM to sync users.
- C. Configure an AXL server to access the Cisco UCM users.
- D. Configure an active directory to sync the users who will have a voicemail box.

Correct Answer: B

Section:

Explanation:

Reference:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/unity_exp/rel3_1/administration/guide/voicemail/3VMpara.html

QUESTION 244

A company has an excessive number of call transfers to local and long-distance PSTN from Cisco Unity Connection voicemail. Which action in the Cisco Unity Connection restriction table resolves this issue?

- A. Block PSTN patterns on Default Transfer, Default Outdial, and Default System Transfer.



- B. Implement password complexity on voicemail boxes to prevent accounts from being compromised.
- C. Create a custom restriction table ***** and block it.
- D. Create a custom restriction table ?????????? and block it.

Correct Answer: A

Section:

QUESTION 245

What is required when deploying co-resident VMs by using Cisco UCM?

- A. Provide a guaranteed bandwidth of 10 Mbps.
- B. Deploy the VMs to a server running Cisco UCM.
- C. Avoid hardware oversubscription.
- D. Ensure that applications will perform QoS.

Correct Answer: C

Section:

Explanation:

When deploying co-resident VMs by using Cisco UCM, it is important to avoid hardware oversubscription. This means that you should not assign more resources to the VMs than the physical hardware can provide. For example, if you have a server with 16 CPU cores, you should not assign more than 16 CPU cores to the VMs.

If you oversubscribe the hardware, the VMs will not be able to get the resources they need to run properly. This can lead to performance problems and even outages.

To avoid hardware oversubscription, you should carefully plan your VM deployments. You should also monitor the performance of the VMs to make sure that they are not overusing the resources.

Here are some additional tips for deploying co-resident VMs by using Cisco UCM:

Use a virtualization platform that supports Cisco UCM.

Make sure that the VMs have the correct operating system and software installed.

Configure the VMs to use the correct network settings.

Monitor the performance of the VMs to make sure that they are running properly.



QUESTION 246

An engineer with ID012345678 must build an international dial plan in Cisco UCM. Which action is taken when building a variable-length route pattern?

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

Correct Answer: D

Section:

Explanation:

When building a variable-length route pattern, you need to create a second route pattern followed by the # wildcard. This will allow the user to indicate the end of the number by dialing #. For example, if you want to create a route pattern for international calls, you would create a route pattern like this:

```
9.011!#
```

This route pattern will match any number that starts with 9.011, followed by any number of digits, and then ends with #.

The other options are incorrect because:

Configuring a single route pattern for international calls will not allow the user to indicate the end of the number.

Setting up all international route patterns to 0.! will not allow the user to indicate the end of the number.

Reducing the T302 timer to less than 4 seconds will not allow the user to indicate the end of the number.

QUESTION 247

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

Correct Answer: C

Section:

Explanation:

Balanced mode provides user load balancing and user failover in the event of an outage.

Active/standby mode provides an always on standby node in the event of an outage, but it does not provide load balancing.

Here is a more detailed explanation of the two modes:

Balanced mode: In balanced mode, the IM and Presence Service nodes are configured to work together to provide high availability. The nodes are configured in a redundancy group, and the system automatically balances the load of users across the nodes in the group. If one of the nodes

fails, the system automatically fails over the users to the other nodes in the group.

Active/standby mode: In active/standby mode, one of the IM and Presence Service nodes is designated as the active node, and the other nodes are designated as standby nodes. The active node handles all of the user traffic, and the standby nodes are only used if the active node fails. If the active node fails, the system automatically fails over to one of the standby nodes.



QUESTION 248

What is the function of the Cisco Unity Connection Call Handler?

- A. routes calls to a user based on caller input
- B. queues calls
- C. allows customized scripts for IVR capabilities
- D. searches a list of extensions until the call is answered

Correct Answer: A

Section:

Explanation:

A Cisco Unity Connection Call Handler is a software application that answers calls, plays greetings, and routes calls to users based on caller input. Call handlers can be used to create automated attendants, voice menus, and other interactive voice response (IVR) applications.

Call handlers are created and managed using the Cisco Unity Connection Administration interface.

When creating a call handler, you can specify a variety of settings, including the greeting that is played, the caller input options that are available, and the destination that calls are routed to.

Call handlers are a powerful tool that can be used to create a variety of IVR applications. By using call handlers, you can improve the efficiency of your organization's communications and provide a better experience for your callers.

Here are some additional tips for using call handlers:

Use call handlers to create automated attendants that can answer calls and route them to the appropriate person or department.

Use call handlers to create voice menus that can provide callers with information or options.

Use call handlers to create interactive voice response (IVR) applications that can collect information from callers and process their requests.

QUESTION 249

If a phone needs to register with cucm1.cisco.com, which network service assists with the phone registration process?

- A. SNMP
- B. ICMP
- C. SMTP
- D. DNS

Correct Answer: D

Section:

Explanation:

According to the Cisco Community website¹, the phone uses DNS to resolve the hostname of the CUCM server (cucm1.cisco.com) to its IP address. DNS is a network service that translates domain names into IP addresses.

QUESTION 250

Where is urgent priority enabled to bypass the T302 timer?

- A. route partition
- B. transformation pattern
- C. directory number
- D. CTI port

Correct Answer: C

Section:

Explanation:

Urgent priority is enabled on the directory number configuration page. This allows the call to be routed at once to the fully qualified DN without any necessity to wait for inter-digit-timeout. If the Urgent Priority checkbox is disabled and you have overlap patterns configured, then CUCM waits for the user to dial further digits.

The other options are incorrect because:

Route partitions are used to group route patterns and route lists.

Transformation patterns are used to convert dialed digits into a different format.

CTI ports are used to connect Cisco Unified Communications Manager to third-party applications.

<https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communicationsmanager-callmanager/200477-Urgent-Priority-Configuration-on-Directo.html>