

350-801^{Q&As}

Implementing and Operating Cisco Collaboration Core Technologies
(CLCOR)

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QUESTION 1

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

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

Correct Answer: D

QUESTION 2

Which IP Precedence value is used to classify a call signalling packet?

- A. 6
- B. 5
- C. 4
- D. 3

Correct Answer: D

The IP precedence is a 3-bit field in TOS that treats high priority packets as more important than other packets. In contrast, DSCP is 6 bits of the Differentiated Services Field (DS Field) in the IP header for packet classification purposes.

Value Description 000 (0) Routine or Best Effort 001 (1) Priority 010 (2) Immediate 011 (3) Flash - mainly used for Voice Signaling or for Video. 100 (4) Flash Override 101 (5) Critical -mainly used for Voice RTP. 110 (6) Internet 111 (7) Network https://www.cisco.com/c/en/us/td/docs/switches/datacenter/nexus1000/sw/4_0/qos/configuration/guide/nexus1000v_qos/qos_6dscp_val.pdf

QUESTION 3

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

MRA isn't part of a fully functioning UCM system... Unified CM can use DNS to:

-

Resolve service (SRV) records to host names and then to IP addresses for SIP trunk destinations

-

Resolve service (SRV) records to host names and then to IP addresses for SIP trunk destinations

<https://community.cisco.com/t5/ip-telephony-and-phones/2-part-question-regarding-dns-on-cucm/td-p/2256179>

QUESTION 4

Refer to the exhibit.

```
INVITE sip:1@10.10.10.219;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.10.10.84:50083;branch=z9hG4bK471df613
From: "1234 - My Phone" <sip:1234@10.10.10.219>;tag=381c1aba7a78002c558eda31-12b8af63
To: <sip:1@10.10.10.219>
Call-ID: 381c1aba-7a78000d-4ca6894a-41dd3e0f@10.10.10.84
Max-Forwards: 70
CSeq: 101 INVITE
Contact: <sip:1234@10.10.10.84:50083;transport=tcp>
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER, UPDATE, SUBSCRIBE, INFO
Allow-Events: kpml, dialog
Content-Type: application/sdp
Content-Length: 658

v=0
o=Cisco-SIPUA 26529 0 IN IP4 10.10.10.84
s=SIP Call
b=AS:4064
t=0 0
m=audio 32136 RTP/AVP 114 9 124 113 115 0 8 116 18
c=IN IP4 10.10.10.84
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114
maxplaybackrate=16000;sprop-maxcapture=16000;maxaveragebitrate=64000;stereo=0;sprop-
stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0,mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1,mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=sendrecv
```

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

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

Correct Answer: DE

The DTMF Events through SIP Signaling feature allows telephone event notifications to be sent through SIP NOTIFY messages, using the SIP SUBSCRIBE/NOTIFY method as defined in the Internet Engineering Task Force (IETF) draft, SIP-Specific Event Notification.

QUESTION 5

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

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

Correct Answer: D

An annunciator is a software function of the Cisco IP Voice Media Streaming Application that provides the ability to stream spoken messages or various call progress tones from the system to a user.

Reference: https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab11/collab11/media.html#32774
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab09/clb09/media.html

QUESTION 6

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

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

Correct Answer: C

LLDP (Link Layer Discovery Protocol) is a vendor-neutral network discovery protocol that is used to discover the topology of a network. LLDP is similar to CDP (Cisco Discovery Protocol), but it is not proprietary to Cisco. LLDP is

supported by a wide range of network devices, including switches, routers, and firewalls. To configure LLDP on a network, you must enable LLDP on the devices that you want to discover. You can then use a network management tool, such as Cisco Network Assistant, to view the topology of the network. The other options are incorrect. TFTP (Trivial File Transfer Protocol) is a network protocol that is used to transfer files between devices. LACP (Link Aggregation Control Protocol) is a network protocol that is used to aggregate multiple network links into a single logical link. SNMP (Simple Network Management Protocol) is a network protocol that is used to manage network devices.

QUESTION 7

Refer to the exhibit.

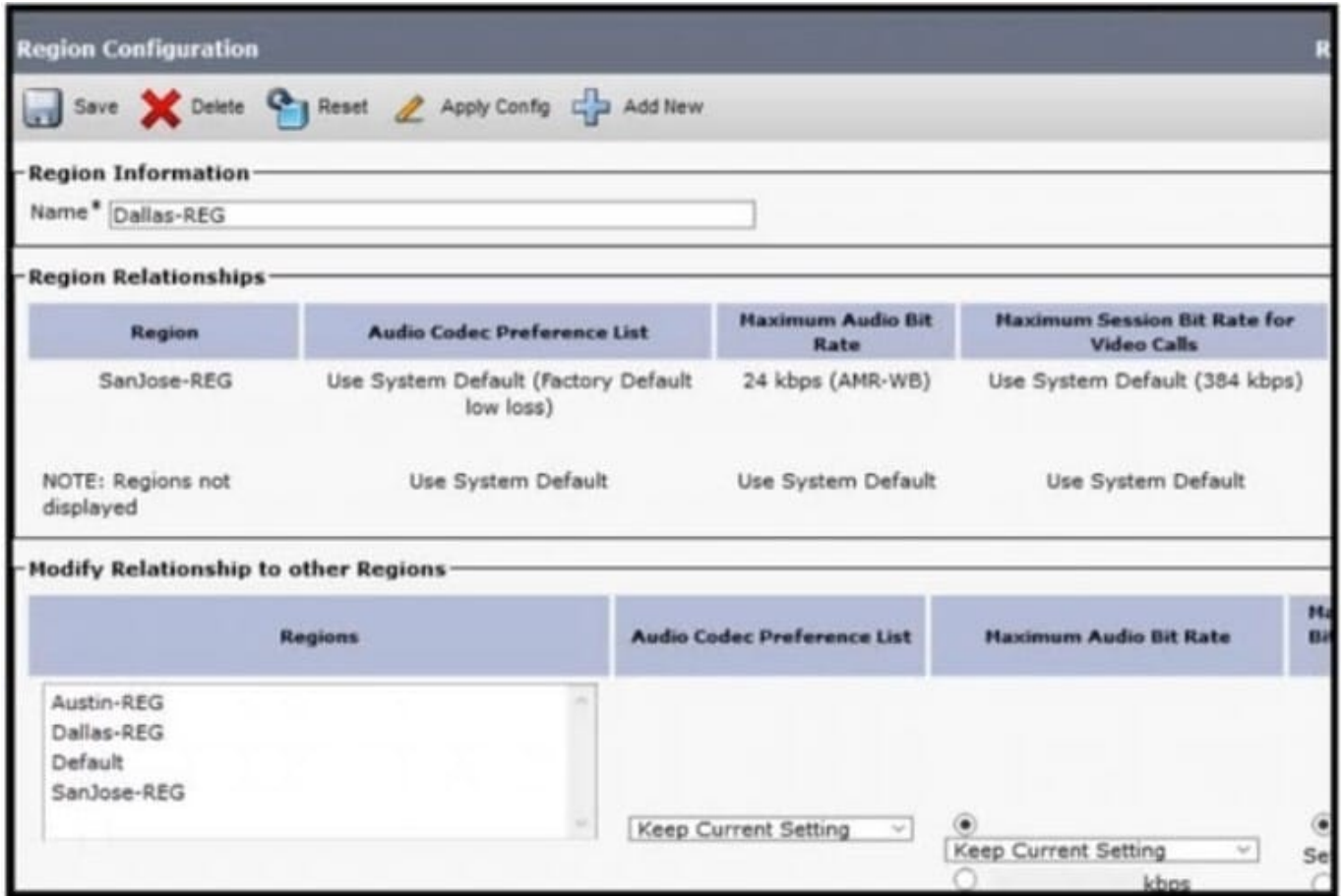
A.

B.

C.

D.

An engineer must implement toll fraud prevention on a Cisco UCM cluster. Only the SIP protocol must be allowed for connections passing through Cisco Unified Border Element. What must be configured?



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

Correct Answer: B

QUESTION 8

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

- A. The external firewall must allow these inbound connections to Expressway: SIP: TCP 5061; HTTPS: TCP 8443; XMPP: TCP 5222; Media: UDP 36002 to 59999.
- B. The internal firewall must allow these inbound and outbound connections between Expressway- and Expressway-E: SIP: HTTPS (tunneled over SSH between and E): TCP 2222; TCP 7001; Traversal Media: UDP 2776 to 2777 (or 36000 to 36011 for large VM/appliance); XMPP: TCP 7400.
- C. Do not use a shared address for Expressway-E and Expressway-, 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- does not resolve the same traffic IP address. Shared NAT is not supported.

D. The traversal zone on Expressway- 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.

Correct Answer: A

Reference: https://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/expressway/config_guide/X8-7/Cisco-Expressway-IP-Port-Usage-for-Firewall-Traversal-Deployment-Guide-X8-7.pdf

QUESTION 9

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: AE

QUESTION 10

Refer to the code.

- A.

```
voice service voip
  allow-connections sip to h323
  ip address trusted list
  ipv4 192.168.10.10
```
- B.

```
voice service voip
  allow-connections sip to sip
  ip address trusted list
  ipv4 192.168.11.10
```
- C.

```
voice service voip
  allow-connections h323 to sip
  ip address trusted list
  ipv4 192.168.11.11
```
- D.

```
voice service voip
  allow-connections sip to h323
  ip address trusted list
  ipv4 192.168.10.11
```

Which two codec permutations should be transcoded by this dspfarm?

- A. G.722 to G.729r8
- B. G.729r8 to G.711ulaw
- C. G.729br8 to G.711alaw
- D. iLBC to G.711ulaw
- E. G729ar8 to G.711alaw

Correct Answer: BE

QUESTION 11

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

QUESTION 12

Which two media encryptions should be used when configuring SRTP between Cisco UCM and Cisco VCS Expressway? (Choose two.)

- A. optional
- B. pass-through
- C. priority
- D. mandatory
- E. none

Correct Answer: DE

Reference: <https://www.cisco.com/c/en/us/support/docs/unified-communications/telepresence-video-communication-server-vcs/118877-config-vcs-00.html>

QUESTION 13

Refer to the exhibit.

DHCP Server Configuration

Save Delete Copy Add New

Status

Add successful

DHCP Server Information

| | |
|---|---|
| Host Server* | <input type="text" value="192.168.10.240"/> |
| Primary DNS IPv4 Address | <input type="text" value="192.168.99.1"/> |
| Secondary DNS IPv4 Address | <input type="text"/> |
| Primary TFTP Server IPv4 Address (Option 150) | <input type="text" value="192.168.10.244"/> |
| Secondary TFTP Server IPv4 Address (Option 150) | <input type="text"/> |
| Bootstrap Server IPv4 Address | <input type="text"/> |
| Domain Name | <input type="text"/> |
| TFTP Server Name (Option 66) | <input type="text"/> |
| ARP Cache Timeout(sec)* | <input type="text" value="0"/> |
| IP Address Lease Time(sec)* | <input type="text" value="0"/> |
| Renewal(T1) Time(sec)* | <input type="text" value="0"/> |
| Rebinding(T2) Time(sec)* | <input type="text" value="0"/> |

An administrator configured a codec preference list with 0,122 and ILBC codecs. Which change must the administrator make in the dial-peer section of the configuration to use this list?

- A. add voice-codecs 20
- B. add session codec 20
- C. add codec preference 20
- D. add voice-class codec 20

Correct Answer: D

QUESTION 14

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

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

Correct Answer: D

The setting is under the Cisco CallManager service specifically within the Service Parameters and you can choose to completely disable it as well.

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

QUESTION 15

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

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

Correct Answer: D

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