

➤ **Vendor: Cisco**

➤ **Exam Code: 350-801**

➤ **Exam Name: Implementing and Operating Cisco Collaboration Core Technologies (CLCOR)**

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QUESTION 28

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

Answer: B

Explanation:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_0_1/ccmcf/CUCM_BK_C95ABA82_00_admin-guide-100/CUCM_BK_C95ABA82_00_admin-guide-100_chapter_0100111.pdf

QUESTION 29

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

- A. Voice Cos 5 (IP Precedence 6, PHB AF41, or DSCP 16)
Video Cos 4 (IP Precedence 5, PHB EF, or DSCP 32)
- B. Voice Cos 6 (IP Precedence 4, PHB AF41, or DSCP 24)
Video Cos 5 (IP Precedence 4, PHB EF, 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 5, PHB EF, or DSCP 46)
Video Cos 4 (IP Precedence 4, PHB AF41, or DSCP 34)

Answer: D

Explanation:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/netstruc.html

Application	Layer-3 Classification			Layer-2 Classification
	Type of Service (ToS) IP Precedence (IPP)	Per-Hop Behavior (PHB)	Differentiated Services Code Point (DSCP)	Class of Service (CoS)
Routing	6	CS6	48	6
Voice Real-Time Transport Protocol (RTP)	5	EF	46	5
Videoconferencing	4	AF41	34	4

QUESTION 30

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

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```
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
```

- 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.

Answer: D

QUESTION 31

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

Answer: A

Explanation:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/9_1_1/ccmcf/CUCM_BK_A34970C5_00_admin-guide-91/CUCM_BK_A34970C5_00_admin-guide-91_chapter_0111.html

QUESTION 32

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.711ualw
- D. G.711ulaw and G.711alaw

Answer: B

QUESTION 33

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.

Answer: A

QUESTION 34

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 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

Answer: D

QUESTION 35

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

Answer: A

Explanation:

https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/jabber/11_5/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115/CJAB_BK_D00D8CBD_00_deployment-installation-guide-cisco-jabber115_chapter_01000.html

QUESTION 36

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

- A. ccm-manager redundant-host
- B. Mgcp call-agent
- C. Mgcpapp
- D. ccm-manager fallback-mgcp

Answer: A

QUESTION 37

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

Answer: A

QUESTION 38

Refer to the exhibit. A call is failing to establish between two SIP Devices. The Called device answer with this SDP. Which SDP parameter causes this issue?

```
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=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. 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.

Answer: A

QUESTION 39

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`

Answer: AB