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## Cisco Implementing and Operating Cisco Collaboration Core Technologies Sample Questions (Q379-Q384):

### NEW QUESTION # 379

An engineer configures a Cisco Unified Border Element and must ensure that the codecs negotiated meet the ITSP requirements. The ITSP supports G.711ulaw and G.729 for audio and H.264 for video. The preferred voice codec is G.711. Which configuration meets this requirement?

```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8

dial-peer voice 101 voip
  session protocol sipv2
  destination el64-pattern-map 1
  voice-class codec 10
```

• A.

```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  video codec h264
```

```
dial-peer voice 101 voip
  session protocol sipv2
  destination el64-pattern-map 1
  voice-class codec 100
```

• B.

```
voice class codec 10
  codec preference 1 g729r8
  codec preference 2 g711ulaw
  video codec h264
dial-peer voice 101 voip
  session protocol sipv2
  destination el64-pattern-map 1
  voice-class codec 10
```

• C.

```
voice class codec 10
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  video codec h264
dial-peer voice 101 voip
  session protocol sipv2
  destination el64-pattern-map 1
  voice-class codec 10
```

• D.

**Answer: D**

**NEW QUESTION # 380**

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 3 60 cube2.abc.com
- B. \_sip.\_udp.abc.com 60 IN SRV 60 1 cube1.abc.com
- C. **\_sip.\_udp.abc.com 60 IN SRV 1 60 cube1.abc.com**
- D. **\_sip.\_udp.abc.com 60 IN SRV 1 40 cube2.abc.com**
- E. \_sip.\_udp.abc.com 60 IN SRV 2 60 cube1.abc.com

**Answer: C,D**

**NEW QUESTION # 381**

What is a capability provided by the Cisco Expressway Series?

- A. **endpoint registration**
- B. voice and video transcoding
- C. intercluster extension mobility
- D. voice and video conferencing

**Answer: A**

Explanation:

The Cisco Expressway Series is primarily used for secure communication and collaboration services across networks. One of its key capabilities is endpoint registration, which allows remote Cisco Jabber clients and SIP-based endpoints to register with Cisco Unified Communications Manager (CUCM) via Mobile and Remote Access (MRA).

**NEW QUESTION # 382**

Refer to the exhibit. An engineer configures a VoIP dial peer on a Cisco gateway.

Which codec is used?

```
dial-peer voice 10 voip
  destination-pattern 1...
  session target ipv4:10.1.1.1
  no vad
```

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

**Answer: A**

Explanation:

Default codec for POTS is g.711

Default codec for VOIP is g.729

[https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucme/admin/configuration/manual/cmeadmin/cmebasic.html#concept\\_2A67E83FAC004CF5AC02FCFF49FCE6BB](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadmin/cmebasic.html#concept_2A67E83FAC004CF5AC02FCFF49FCE6BB)

**NEW QUESTION # 383**

Refer to the exhibit.

```

v=0
o=Cisco-SIPUA 13439 0 IN IP4 10.10.10.10
s=SIP Call
b=AS:4064
t=0 0
m=audio 0 RTP/AVP 114 9 124 113 115 0 8 116 18 101
c=IN IP4 10.10.10.10
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-
maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

```

```

b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-
maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-stereo=0;usedtx=0
a=rtpmap:9 G722/8000
a=rtpmap:124 ISAC/16000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

```

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

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

**Answer: B**

#### NEW QUESTION # 384

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