



Free Questions for 350-801
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Question 1

Question Type: MultipleChoice

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

Options:

- 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.

Answer:

D

Question 2

Question Type: MultipleChoice

A collaboration engineer is configuring the QoS trust boundary for Cisco UCM voice and video conferencing. Which two trust boundary configurations are valid? (choose two)

Options:

- A- QoS trust boundaries include all the devices directly attached to the access switch ports
- B- QoS trust boundaries can be extended to Jabber running on a PC
- C- QoS trust boundaries exclude Jabber softphone running on a PC
- D- QoS trust boundaries can be extended to voice and video devices if the connected PCs are included
- E- QoS trust boundaries can be extended to voice and video devices exclusively

Answer:

C, D

Question 3

Question Type: MultipleChoice

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?

Options:

- A- SIP003096D2D5CB.cnf.xml
- B- SEP003096D2D5CB.cnf.xml
- C- SEP003096D2D5CB.cnf
- D- SIP003096D2D5CB.cnf



Answer:

B

Question 4

Question Type: MultipleChoice

What is the default TCP port for SIP OAuth mode in Cisco UCM?

Options:

- A- 5011
- B- 3174
- C- 8443
- D- 5090



Answer:

D

Explanation:

The Cisco Unified Communications Manager (CUCM) uses SIP Phone OAuth Port (5090) to listen for SIP line registration from Jabber OnPremise devices over TLS. However, CUCM uses SIP Mobile Remote Access Port (default 5091) to listen for SIP line registrations from Jabber over Expressway through mTLS. Both of these ports are configurable.

Question 5

Question Type: MultipleChoice

Which two functions are provided by Cisco Expressway Series? (Choose two.)

Options:

- A- voice and video transcoding
- B- voice and video conferencing
- C- interworking of SIP and H.323
- D- intercluster extension mobility
- E- endpoint registration



Answer:

A, C

Explanation:

The Cisco Expressway Series provides the following functions:

Voice and video transcoding

Interworking of SIP and H.323

Firewall traversal

Session border controller (SBC) functionality

Endpoint registration

Call admission control (CAC)

Quality of service (QoS)

Security

The Cisco Expressway Series does not provide voice and video conferencing or intercluster extension mobility.



Question 6

Question Type: MultipleChoice

Refer to the exhibit.

```
23031952: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Applying typeplan for sv-type Q93 in Q93 SMI, Calling num 4065504100
23031953: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: Sending RTTSM callref = 0x128f callID = 0x412f switch = primary-sd interface = Ser
23031954: Apr  9 17:43:21.203 EDT: ISDN Se0/1/0:23 Q931: TX -> SETUP pd = 8 callref = 0x128f
  Bearer Capability i = 0x8090a2
  Standard = OCITT
  Transfer Capability = Speech
  Transfer Mode = Circuit
  Transfer Rate = 64 kbit/s
  Channel ID i = 0x4b9293
  Exclusive, Channel 19
  Progress Ind i = 0x8183 - Origination address is non-ISDN
  Calling Party Number i = 0x2181, '4065504100'
  Plan:ISDN, Type:National
  Called Party Number i = 0x91, '01143078542222'
  Plan:ISDN, Type:International
23031955: Apr  9 17:43:21.379 EDT: ISDN Se0/1/0:23 Q931: RX <- CALL_PROC pd = 8 callref = 0x218f
  Channel ID i = 0x838292
  Exclusive, Channel 19
23031957: Apr  9 17:43:21.289 EDT: ISDN Se0/1/0:23 Q931: RX <- PROGRESS pd = 8 callref = 0x218f
  Cause i = 0x829f - Normal, unspecified
  Progress Ind i = 0x8188 - In-band info not appropriate now available
23031961: Apr  9 17:43:44.001 EDT: ISDN Se0/1/0:23 Q931: TX -> DISCONNECT pd = 8 callref = 0x218f
  Cause i = 0x8080 - Normal call clearing
23031962: Apr  9 17:43:44.022 EDT: ISDN Se0/1/0:23 Q931: RX <- RELEASE pd = 8 callref = 0x218f
23031963: Apr  9 17:43:44.022 EDT: ISDN Se0/1/0:23 Q931: TX -> RELEASE_COMPLETE pd = 8 callref = 0x218f
```

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

Options:

- 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.

Answer:

B



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