



Free Questions for 350-801
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Question 1

Question Type: MultipleChoice

An administrator is configuring a new Cisco UCM with PSTN capabilities. Due to bandwidth constraints, audio compression is used on the codec. DTMF must work as expected because the customer is calling many call centers where the users must select options in the call. Where is DTMF out-of-band in a CCM 12.5 with SIP-based gateway configured?



Options:

- A- in the DTMF setting under SIP profile on the Cisco Unified Border Element
- B- in the dial peer on the Cisco IOS router
- C- in regions on the Cisco UCM where the appropriate codec to use is set
- D- in DTMF settings in the audio codec preference list under regions in the Cisco UCM

Answer:

B

Question 2

Question Type: MultipleChoice

Refer to the Exhibit.

```
dspfarm profile 1 mtpl
  codec g711ulaw
  maximum sessions software 50
  associate application SCCP
```



Which command is required to allow this media resource to handle Video Media streams?

Options:

- A- maximum sessions hardware 50
- B- video codec h264
- C- codec pass-through
- D- associate application Cisco unified border element

Answer:

C

Question 3

Question Type: MultipleChoice

An engineer must configure codec on a Cisco Unified Border Element to prefer the G.711 ulaw and use G.711 codec as the

next The engineer logs in to the CUBE, enters the dial-peer configuration level, and runs the voice-class codec 100 command. Which set of commands completes the configuration?

Options:

A- voice class codec 100 codec g711ulaw preference 1 codec a7Hulaw preference 2

B- voice class codec 11j codec <?7iulaw preferred codec g7iialaw

C- vice class codec 100

codec preference 1 g711ulaw

codec preference 2 o711alaw

D- voice class codec ::: codec g711ulaw g711alaw

Answer:

C

Explanation:

The following commands are used to configure the codec on a Cisco Unified Border Element to prefer the G.711 ulaw and use G.711 alaw as the next codec:

Code snippet

```
voice class codec 100
```

```
codec preference 1 g711ulaw
```

```
codec preference 2 g711alaw
```

The voice class codec 100 command creates a new voice class with the ID of 100. The codec preference 1 g711ulaw command sets the preference for the G.711 ulaw codec to 1. The codec preference 2 g711alaw command sets the preference for the G.711 alaw codec to 2.

Question 4

Question Type: MultipleChoice

What are the predefined call handlers in Cisco Unity Connection?

Options:

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

Answer:

D

Question 5

Question Type: MultipleChoice

Refer to the exhibit.

<https://i.postimg.cc/C57TkczG/image.png>



```
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 call is failing to establish between two SIP Devices. The called device answers with these SIP parameters. Which SIP parameter causes the issue?

Options:

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

Answer:

D

Explanation:

The RTP port is used to send and receive media packets during a call. If the RTP port is set to 0, the called device will not be able to send or receive media packets, and the call will fail.

The other options are not correct because:

- A) The calling device did not offer a ptime value: The ptime value is used to specify the amount of time between each media packet. If the calling device does not offer a ptime value, the called device will use the default value of 20 milliseconds.
- B) The media stream is set to sendonly: The media stream is set to sendonly when the called device is only able to send media packets, and not receive them. This is not a problem, and the call will still succeed.
- C) The payload for G.711ulaw must be 18: The payload for G.711ulaw is the type of media packet that is used. The payload must be set to 18 for G.711ulaw, but this is not a problem, and the call will still succeed.



Question 6

Question Type: MultipleChoice

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

Options:

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

Answer:

B



Question 7

Question Type: MultipleChoice

Refer to the exhibit.

```
E:\Users\CISCO>nslookup
Default Server: dns.example.com
Address: 192.168.100.1

> set type=SRV
> _collab-edge._tcp.example.com
Server: dns.example.com
Address: 192.168.100.1

Non-authoritative answer:
_collab-edge._tcp.example.com      SRV service location:
  priority      = 10
  weight        = 10
  port          = 8443
  svr hostname  = expe.example.com
```

You deploy Mobile and Remote Access for Jabber and discover that Jabber for Windows does not register to Cisco Unified Communications Manager while outside of the office. What is a cause of this issue?



Options:

- A- The DNS record should be created for _cisco-uds._tcp example.com.
- 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.

Answer:

B

Question 8

Question Type: MultipleChoice

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



Options:

- A- the operator call handler
- B- the Interview handler
- C- the Goodbye call handler
- D- the Directory handler

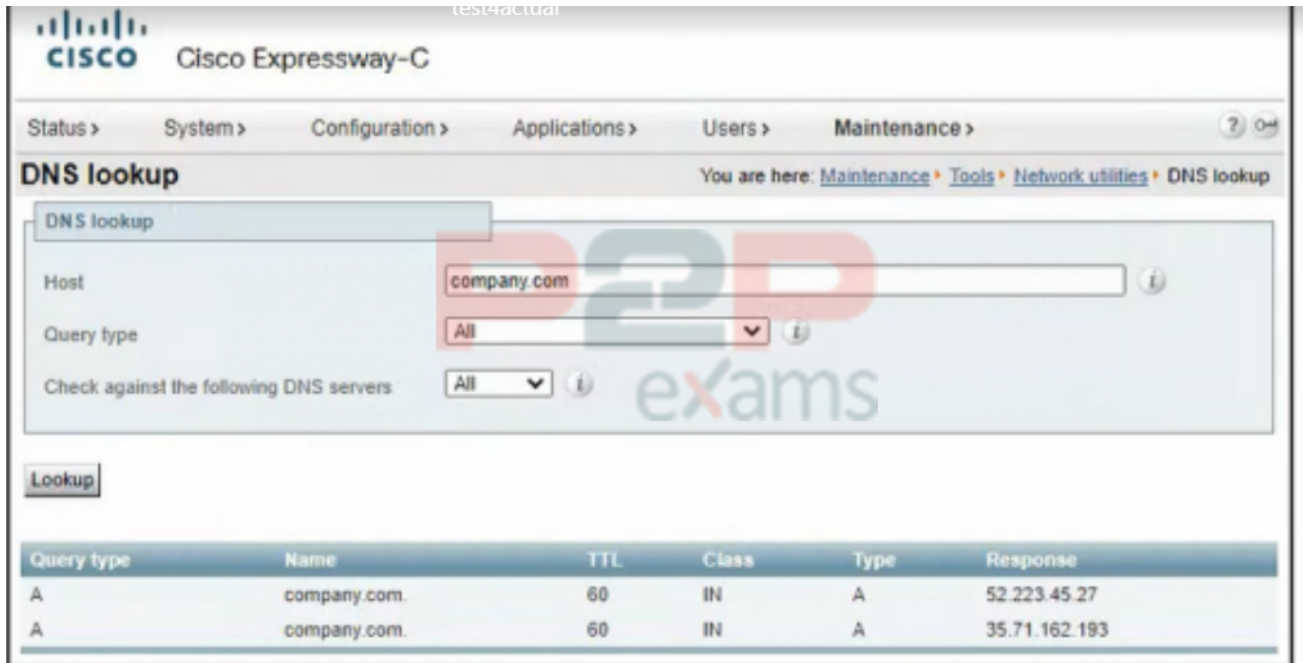
Answer:

A

Question 9

Question Type: MultipleChoice

Refer to exhibit.



The screenshot shows the Cisco Expressway-C interface with the DNS lookup tool. The host is set to 'company.com' and the query type is 'All'. The results table shows two A records for 'company.com' with a TTL of 60. The first record has a response of '52.223.45.27' and the second has '35.71.162.193'.

Query type	Name	TTL	Class	Type	Response
A	company.com.	60	IN	A	52.223.45.27
A	company.com.	60	IN	A	35.71.162.193

A company recently deployed CISCO Jabber Users log in to Jabber by using their email address in a domain named

company.com. The users report that they cannot register their telephony services when working from unless they use a VPN. An

engineer runs DNS lookup tool in Cisco Expressway-C to troubleshoot the issue. What is the cause of the issue?

Options:

- A- The company.com domain must be resolved only in Expressway-E
- B- There is a missing SRV record for the company.com domain.
- C- The TTL value for the company.com is too short.
- D- There must be only one response for the company.com domain

Answer:

B

Question 10

Question Type: MultipleChoice

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

Options:

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

Answer:

B

Explanation:

To restrict off-net to off-net call transfers, an administrator can set the 'Block Offnet to Offnet Transfer' service parameter to 'On'. This will prevent users from transferring calls from one external number to another external number.

The other options are not correct because:

- A) Enforce ad-hoc conference restrictions: This will prevent users from creating ad-hoc conferences, but it will not prevent them from transferring calls.
- C) Implement time-of-day routing: This will allow calls to be routed to different destinations based on the time of day, but it will not prevent users from transferring calls.
- D) Use the correct route filters: This will allow calls to be filtered based on the destination, but it will not prevent users from transferring calls.

Question 11

Question Type: MultipleChoice

A high-speed network is often configured with a five-class QoS model. Which classes are used in the model?

Options:

- A- real-time, call-signaling, critical data, best-effort, and scavenger
- B- real-time, signaling, critical data, best-effort and drop-class
- C- call-signaling, real-time, critical data, best-effort, and drop-class
- D- voice, video, signaling, critical data, and best-effort

Answer:

A



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