Cisco

Exam Questions 300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

NEW QUESTION 1

```
SIP/2.0 200 OK
[..truncated..]
v=0
o=UAC 6107 7816 IN IP4 10.10.10.11
s=SIP Call
c=IN IP4 10.10.10.11
t=0 0
m=audio 8190 RTP/AVP 18 110
c=-IN IP4 10.10.10.11
a=rtpmap: 18 G729/8000
a=fmtp: 18 annexb=no
a=rtpmap:110 telephone-event/8000
a=fmtp: 110 0-16
a=ptime: 20

ACK sip:+123456789@10.10.20.20:5060 SIP/2.0
[..truncated..]
v=0
o=UAS 4692 9609 IN IP4 10.10.10.10
s=SIP Call
c=IN IP4 10.10.10.10
t=0 0
m=audio 8056 RTP/AVP 18
c=IN IP4 10.10.10.10
a=rtpmap: 18 G729/8000
a=fmtp: 18 annexb=no
a=ptime:20
```

Refer to the exhibit. Users report that when they dial to Cisco Unity Connection from an external network, they cannot enter any digits. Assuming only in-band DTMF is supported, what is a reason for this malfunction?

A. The negotiated RTP port is outside of the range described by RFC, so inband DTMFs do not work.
B. There is SIP Delayed Offe
C. DTMF is supported only in Early Offer.
D. The rtpmap:0 value for the negotiated codec is marking DTMF as inactive.
E. No DTMF is negotiated.

Answer: D

NEW QUESTION 2
The administrator of ABC company is troubleshooting a one-way audio issue for a call that uses H.323 protocol (slow-start mode). The administrator requests that you provide the IP and port information of the Real-Time Transport Protocol traffic that had the one-way audio call.
You gather the H.225 and H.245 messages for one of the one-way audio calls. Where can you find the RTP IP and port information for both sides? (Note: This call flow has not invoked any media resources like MTP or transcoders).

A. H.245 Terminal Capability Set
B. H.245 Open Logical Channel
C. H.225 Connect
D. H.245 Open Logical Channel Ack

Answer: B

NEW QUESTION 3
Which two extended capabilities must be configured on dial peers for fast start-to-early media scenarios (H.323 to SIP interworking)? (Choose two.)

A. DTMF
B. BFCP
C. VIDEO
D. FAX
E. AUDIO

Answer: AB

NEW QUESTION 4
Cisco SIP IP telephony is implemented on two floors of your company. Afterward, users report intermittent voice issues in calls established between floors. All calls are established, and sometimes they work well, but sometimes there is oneway audio or no audio. You determine that there is a firewall between the floors, and the administrator reports that it is allowing SIP signaling and UDP ports from 20000 to 22000 bidirectionally. What are two possible solutions? (Choose two.)

A. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 16384-32767
B. Ask the firewall administrator to change the ports to TCP.
C. Ask the firewall administrator to change the range of UDP ports to 16384-32767.
D. Go to the SIP profile assigned to these IP phones in Cisco Unified CM and change the range of media ports to 20000-22000.
E. Go to System Parameters in Cisco Unified Communications Manager and change the range of media ports to 20000-22000.

Answer: AC

NEW QUESTION 5
Which description of RTP timestamps or sequence numbers is true?

A. The sequence number is used to detect losses.
B. Timestamps increase by the time “carrying” by a packet.
C. Sequence numbers increase by four for each RTP packet transmitted.
D. The timestamp is used to place the incoming audio and video packets in the correct timing order (playout delay compensation).

Answer: D

NEW QUESTION 6
Which IOS command creates a SIP- enabled dial peer?

A. voice dial-peer 20 sip
B. dial-peer voice 20 voip
C. dial-peer voice 20 pots
D. dial peer voice 20 sip

Answer: B

NEW QUESTION 7
A user in location X dials an extension at location Y. The call travels through a QoS-enabled WAN network, but the user experiences choppy or clipped audio. What is the cause of this issue?

A. missing Call Admission Control
B. codec mismatch
C. ptime mismatch
D. phone class of service issue

Answer: B

NEW QUESTION 8
An engineer must route all SIP calls in the form of <user>@example.com to the SIP trunk gateway corporate local. Which two SIP route patterns can be used to accomplish this task? (Choose two.)

A. example.com@gateway.corporate.local
B. *@example.com
C. gateway.corporate.local
D. example.com
E. *.*

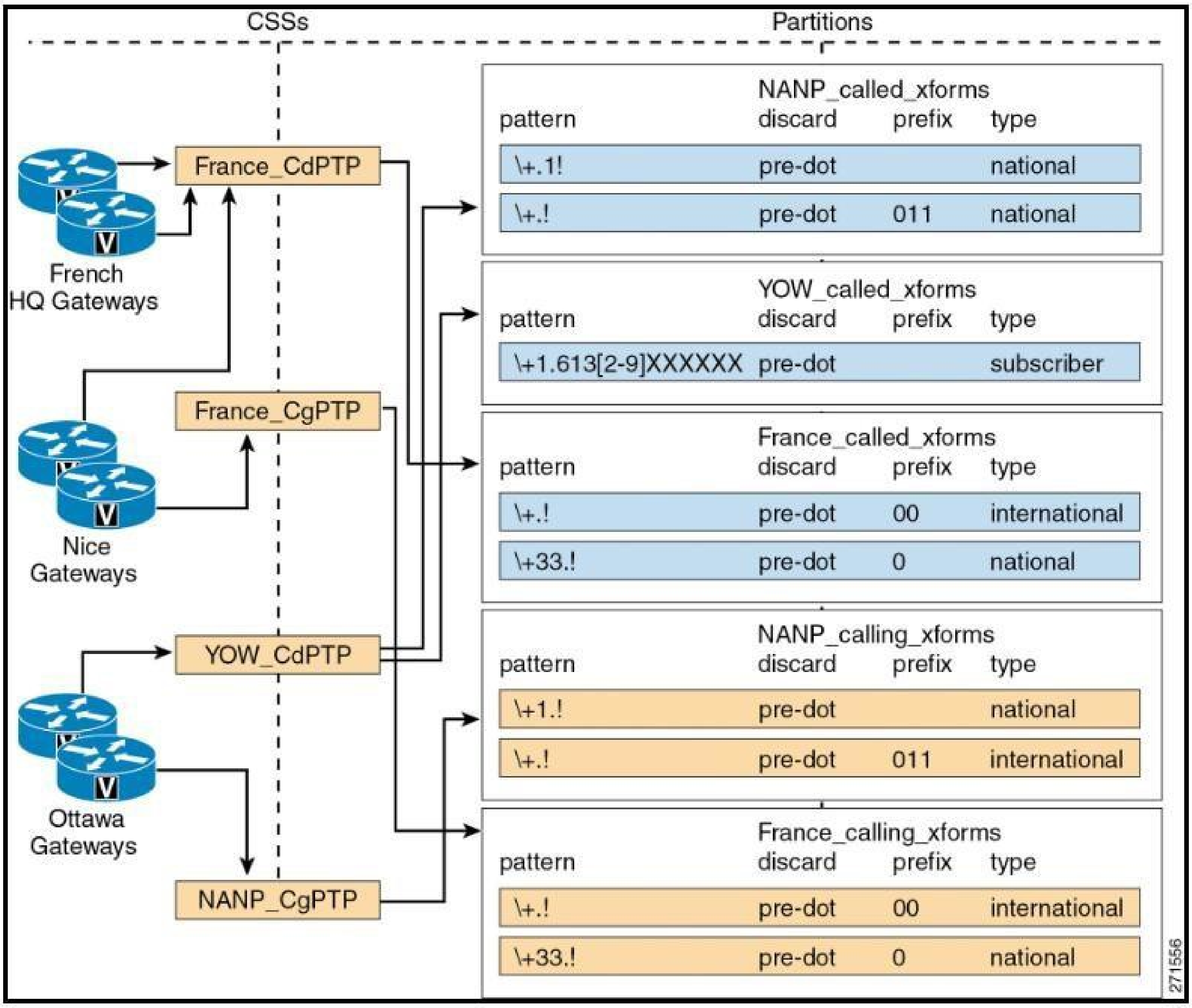
Answer: BE

NEW QUESTION 9
Which two statements are correct with respect to the Client Matter Code setting in the route pattern configuration? (Choose two.)

A. The Client Matter Code feature does not support overlap sending because the Cisco Unified CM cannot determine when to prompt the user for the code.
B. If you check the Allow Overlap Sending check box, the Require Client Matter Code check box becomes disabled.
C. If you check the Allow Overlap Sending check box, you can also check the Require Client Matter Code check box.
D. The Client Matter Code feature does support overlap sending because the Cisco Unified Communications Manager can determine when to prompt the user for the code.
E. The Client Matter Code has the option to configure Authorization Level such as in the Forced Authorization Code.

Answer: AB

NEW QUESTION 10



Refer to the exhibit. Within the North American Numbering Plan, gateways located in Ottawa, Canada and marked as “YOW” are assigned to the Calling Party Transformation CSS NANP_CgPTP, which contains partition NANP_calling_xforms. What is the calling-party number and the numbering type if the calling user +1613-555-1234 dials the number?

A. calling number 613-555-1234 and numbering type “subscriber”
B. calling number 011-1-613-555-1234 and numbering type “subscriber”
C. calling number 011613-555-1234 and numbering type “international”
D. calling number 613-555-1234 and numbering type “national”

Answer: D

NEW QUESTION 10
When locations-based Call Admission Control denies the call, which two masks can AAR apply when routing the call through the PSTN? (Choose two.)

A. AAR destination mask
B. called party transform mask
C. external phone number mask
D. +E.164 alternate number mask
E. enterrise alternate number mask

Answer: AC

NEW QUESTION 15
What is the relationship between partition, time schedule, and time period in Time-of-Day routing in Cisco Unified Communications Manager?

A. A partition can have multiple time schedules assigne
B. A time schedule contains one or more time periods.
C. A partition can have one time schedule assigne
D. A time schedule contains one or more time periods.
E. A partition can have multiple time schedules assigne
F. A time schedule contains only one time period.
G. A partition can have one time schedule assigne
H. A time schedule contains only one time period.

Answer: A

NEW QUESTION 18
Which services are needed to successfully implement Cisco Extension Mobility in a standalone Cisco Unified Communications Manager server?

A. Cisco Extended Functions, Cisco Extension Mobility, and Cisco AXL Web Service
B. Cisco CallManager, Cisco TFTP, and Cisco CallManager SNMP Service
C. Cisco CallManager, Cisco TFTP, and Cisco Extension Mobility
D. Cisco TAPS Service, Cisco TFTP, and Cisco Extension Mobility

Answer: C

NEW QUESTION 22
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