



Cisco

Exam Questions 300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

NEW QUESTION 1

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 2

End users at a new site report being unable to hear the remote party when calling or being called by users at headquarters. Calls to and from the PSTN work as expected. To investigate the SIP signaling to troubleshoot the problem, which field can provide a hint for troubleshooting?

- A. Contact: header of the 200 OK response
- B. Allow: header of the 200 OK response
- C. o= line of SDP content
- D. c= line of SDP content

Answer: C

NEW QUESTION 3

Why would RTP traffic that is sent from the originating endpoint fail to be received on the far endpoint?

- A. The far end connection data (c=) in the SDP was overwritten by deep packet inspection in the call signaling path.
- B. Cisco Unified Communications Manager invoked media termination point resources.
- C. The RTP traffic is arriving beyond the jitter buffer on the receiving end.
- D. A firewall in the media path is blocking TCP ports 16384-32768.

Answer: D

NEW QUESTION 4

What is first preference condition matched in a SIP-enabled incoming dial peer?

- A. incoming uri
- B. target carrier-id
- C. answer-address
- D. incoming called-number

Answer: A

NEW QUESTION 5

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 6

Which section under the Real-Time Monitoring Tool allows for reviewing the call flow and signaling for a SIP call in real time?

- A. Analysis Manager > Inventory > Trace File Repositories
- B. System > Tools > Trace and Log Central
- C. Voice/Video > Session Trace Log View > Real Time Data
- D. Voice/Video > Session Trace Log View > Open From Local Disk

Answer: C

NEW QUESTION 7

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 8

A support engineer is troubleshooting a voice network. When conducting a search for call setup details related to calling search space issues, which trace files should be investigated?

- A. CallManager traces
- B. CTI Manager traces
- C. Cisco IP Manager Assistant
- D. Call logs

Answer: A

NEW QUESTION 9

A company has an SRST gateway running an IOS XE image. The company plans to enable the IPv6 addressing companywide. To enable the IPv6 in a unified SRST gateway to support SIP phones, what are two supported supplementary features for an IPv6 fallback scenario? (Choose two.)

- A. three-way conference
- B. secure SIP lines
- C. T.38 fax relay
- D. transcoding
- E. SIP trunk

Answer: AC

NEW QUESTION 10

Which action is correct with respect to toll fraud prevention configuration in the Cisco Unified Communications Manager Express?

- A. Configure Direct Inward Dial for Incoming ISDN Calls with overlap dialing.
- B. Configure IP Address Trusted Authentication for Incoming VoIP Calls.
- C. Configure the command no ip address trusted authenticate under "voice service voip".
- D. Enable Secondary Dial tone on Analog and Digital FXO Ports.

Answer: B

NEW QUESTION 10

```
voice translation-profile incoming
  translate called 999
!
voice translation-rule 999
  rule 1 /\ (^([1-2] [1-2] [1-2]\ ) 333\ ([4-5] [4-5] .\ ) $ / / \2333\1/
!
dial-peer voice 999 voip
  translation-profile outgoing incoming
  session protocol sipv2
  incoming called-number
  dtmf-relay rtp-nte
  codec transparent
  destination dpg 888
  no vad
!
voice class dpg 888
  dial-peer 888
!
dial-peer voice 888 voip
  destination-pattern 888
  session protocol sipv2
  session target ipv4:192.168.0.1
  codec transparent
  dtmf-relay rtp-nte
  no vad
```

Refer to the exhibit. Calls incoming from the provider are not working through newly set up Cisco Unified Border Element. Provider engineers get the 404 Not Found SIP message. Incoming calls are coming from the provider with called number "222333444" and Cisco Unified Communications Manager is expecting the called number to be delivered as "444333222". The administrator already verified that the IP address of the Cisco Unified CM is set up correctly and there are no dial peers configured other than those shown in the exhibit. Which action must the administrator take to fix the issue?

- A. Change the destination-pattern on the outgoing dial peer to match "444333222".
- B. Set up translation-profile on the incoming dial peer to match incoming traffic.
- C. Create specific matching for "222333444" on the incoming dial peer.
- D. Fix the voice translation-rule to match specifically number "222333444" and change it to "444333222".

Answer: B

NEW QUESTION 13

```
voice translation-rule 84
  rule 1 /\ ([2-9]..[2-9].....$)/ \2/
```

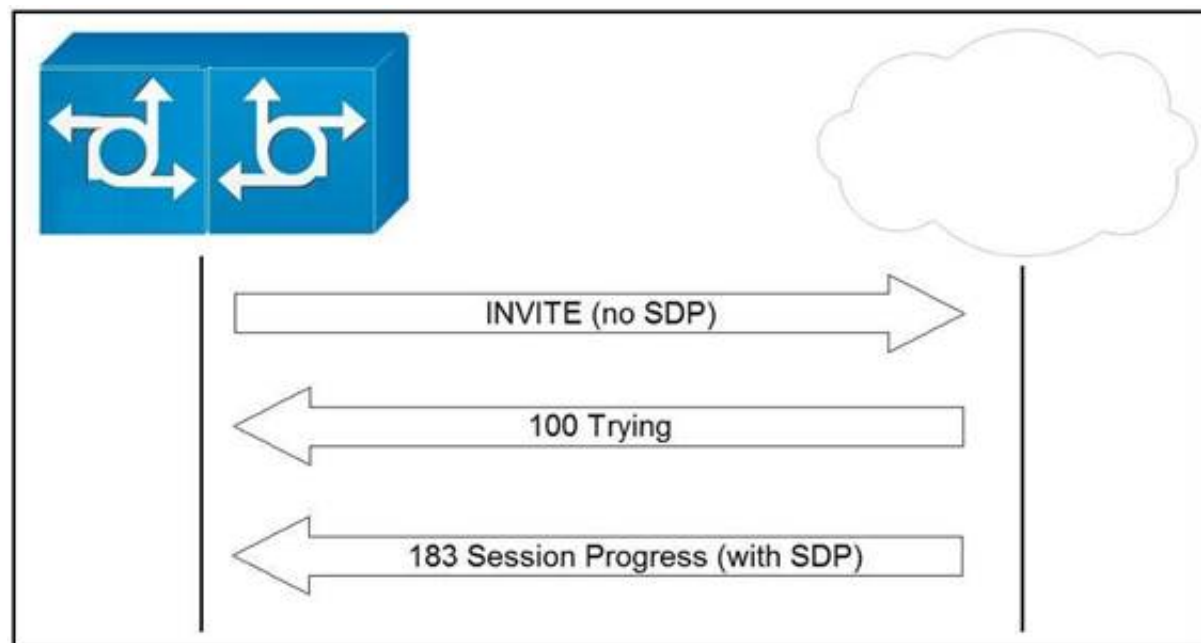
Refer to the exhibit. Users report that outbound PSTN calls from phones registered to Cisco Unified Communications Manager are not completing. The local service provider in North America has a requirement to receive calls in 10-digit format. The Cisco Unified CM sends the calls to the Cisco Unified Border Element

router in a globalized E.164 format. There is an outbound dial peer on Cisco Unified Border Element configured to send the calls to the provider. The dial peer has a voice translation profile applied in the correct direction but an incorrect voice translation rule applied, which is shown in the exhibit. Which rule modified DNIS in the format that the provider is expecting?

- A. rule 1 /^+([1].*)/ /011\1/
- B. rule 1 /^+1\([2-9]..[2-9].....\$\)/ \1/
- C. rule 1 /^([2-9]..[2-9].....\$\)/ \1/
- D. rule 1 /^+1\([2-9]..[2-9].....\$\)/ \0/

Answer: B

NEW QUESTION 15



Refer to the exhibit. An administrator is troubleshooting why users are not hearing audio when dialing long distance numbers across their Cisco Unified Border Element. The customer's carrier has a requirement that dialing long distance requires an access code to be entered. Looking at the exhibit, what two actions can be taken to correct signaling? (Choose two.)

- A. Enable PRACK.
- B. Enable Early Offer on the Cisco Unified Border Element.
- C. Enable the supplementary-service media-renegotiate command.
- D. Enable Media Flow Around
- E. Enable Mid-Call Signaling Consumption.

Answer: AB

NEW QUESTION 19

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 24

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 26

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 30

A network engineer designs a new dial plan and wants to block a certain range of numbers (8135100 through 8135105). What is the most specific route pattern that can be configured to block only the numbers in this range?

- A. 813510[012345]
- B. 813510[12345]
- C. 813510[^0-5]
- D. 81XXXXX

Answer: A

NEW QUESTION 32

If all patterns below are configured in Cisco Unified Communications Manager which would be used when dialing the pattern "123"?

- A. 12!
- B. 12X (urgent priority set)
- C. 1XX (urgent Priority Set)
- D. 12[2-5]

Answer: B

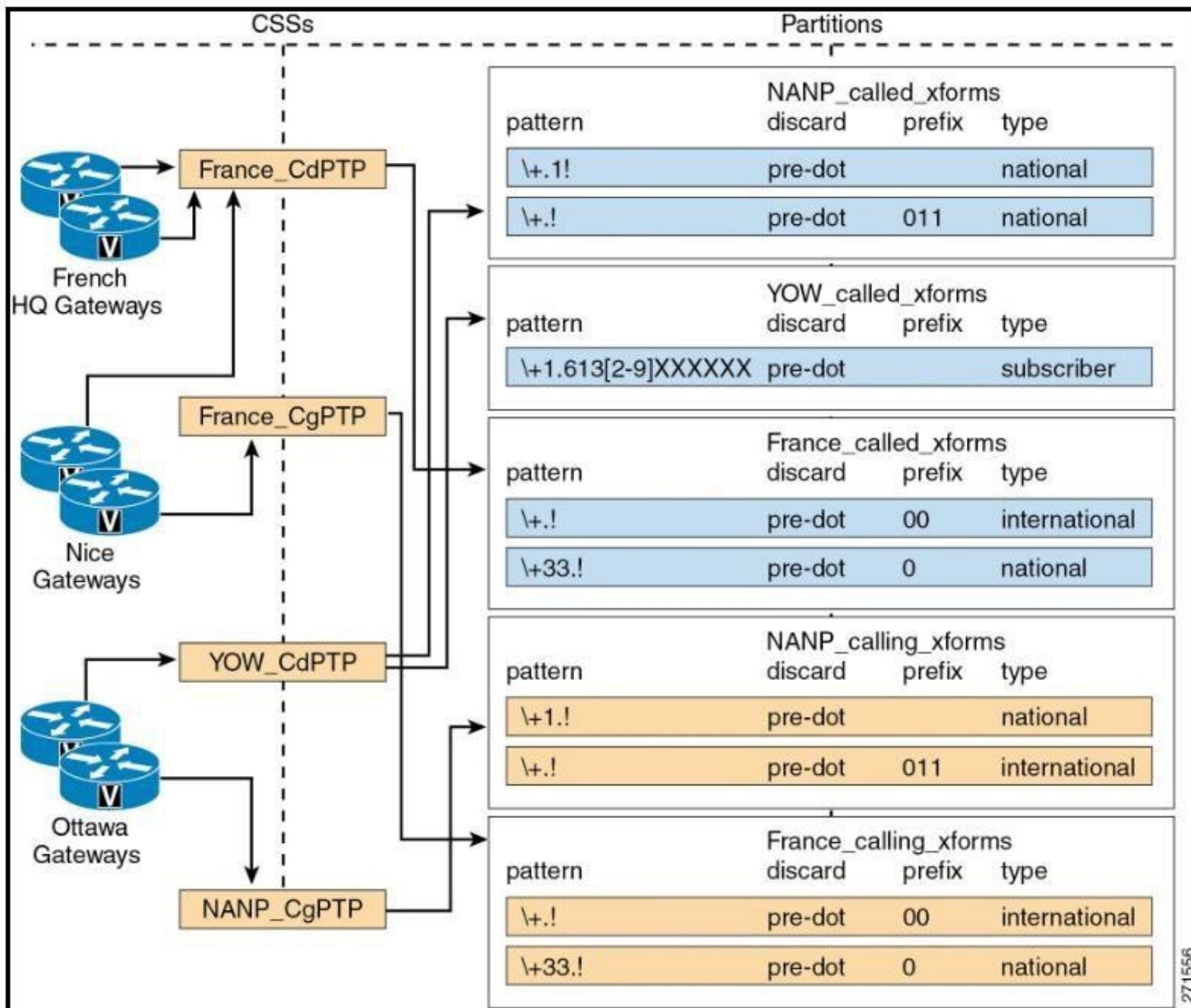
NEW QUESTION 34

The Cisco Unified Communications Manager Dialed Number Analyzer allows analysis of calls from which two devices? (Choose two.)

- A. translation patterns
- B. device pools
- C. CTI ports
- D. CTI route points
- E. IP phones

Answer: CE

NEW QUESTION 38



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 39

Where on Cisco Unified Communications Manager do you configure the standard local route group for a group of devices?

- A. System > Location Info
- B. Call Routing > Route/Hunt > Local Route Group Names
- C. System > Device Pool
- D. Call Routing > Emergency Location > Emergency Location (ELIN) Groups

Answer: B

NEW QUESTION 42

Which two types of distribution algorithm are within a line group? (Choose two.)

- A. random
- B. circular
- C. highest preference
- D. top down
- E. bottom up

Answer: BD

NEW QUESTION 47

An engineer is configuring a call park feature in Cisco Unified Communications Manager Express. Which command does the engineer use to ensure that the call is reverted to the user after 60 seconds?

- A. R2(config-ephone-dn)#park reservation-group 60
- B. R2(config-ephone-dn)#park-slot timeout 60 limit 2 recall alternate 3002
- C. R2(config-ephone-dn)#park reservation-group 1
- D. R2(config-ephone-dn)#park-slot timeout 30 limit 2 recall alternate 3002

Answer: B

NEW QUESTION 48

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. enterprise alternate number mask

Answer: AC

NEW QUESTION 51

An administrator is configuring a cluster for ILS and wants to limit the amount of entities that Cisco Unified Communications Manager can write to the database for data that is learned through ILS. Which service parameter is used to adjust this limit?

- A. ILS Max Number of Learned Objects in Database
- B. ILS Active Learned Object Upper Limit
- C. Global Data Service Parameter Limit
- D. Imported Dial Plan Replication Database Object Lower Limit

Answer: A

NEW QUESTION 54

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

Answer: A

NEW QUESTION 56

A user reports that when they attempt to log out from the Cisco Extension Mobility service by pressing the Services button, they cannot log out. What is the most likely cause of this issue?

- A. The Cisco Extension Mobility service has not been configured on the phone.
- B. There might be a significant delay between the button being pressed and the Cisco Extension Mobility service recognizing it.
- C. It would be best to check network latency.
- D. The user device profile has not been assigned to the user.
- E. The user device profile is not subscribed to the Cisco Extension Mobility service.

Answer: D

NEW QUESTION 57

What is a component of Cisco Unified Mobility?

- A. Unified IVR
- B. Mobile Connect
- C. Smart Client Support
- D. Single Number Connect

Answer: B

NEW QUESTION 59

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