

Cisco

Exam Questions 300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)



NEW QUESTION 1

When you troubleshoot H.323 call setup, which message informs you that the called party is being notified about the call?

- A. ALERTING
- B. PROCEEDING
- C. CONNECT
- D. RINGING

Answer: C

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

An administrator is troubleshooting call failures on an H.323 gateway via the CLI. To see signaling for media and call setup, which debug must the Administrator turn on?

- A. debug H.323 messages
- B. debug H.225 asn1
- C. debug H.246 asn 1
- D. debug H.225 media
- E. debug H.323 asn 1

Answer: B

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

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 8

For s SIP to SIP call flow, when does Cisco Unified Border Element require transcoding resources for DTMF?

- A. interworking between an OOB method and RFC2833 for flow-around calls
- B. interworking between h245-signal and rtp-nte
- C. interworking between an OOB method and RFC2833 for flow-through calls
- D. interworking between h245-alpha numeric and sip-kpml

Answer: A

NEW QUESTION 9

```
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 10

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 10

Which two descriptions of the Standard Local Route Group deployment are true? (Choose two.)

- A. can be associated under the route group
- B. can be associated only under the route list
- C. chooses the route group that is configured under the device pool of the calling-party device
- D. chooses the route group that is configured under the device pool of the called-party device
- E. can be assigned directly to the route pattern

Answer: BD

NEW QUESTION 11

After configuring a Cisco CallManager Express with Cisco Unity Express, inbound calls from the PSTN SIP trunk receive a ring tone for 20 seconds and then a busy signal instead of voicemail. Which configuration fixes this problem?

- A. Router(config)# voice service voipRouter(conf-voi-serv)#allow-connections h323 to h323
- B. Router(config)#dial-peer voice 2 voipRouter(config-dial-peer)#no vad
- C. Router(config)# voice service voipRouter(conf-voi-serv)#allow-connections voice-mail mod
- D. Router(config)# voice service voipRouter(conf-voi-serv)#no supplementary-service sip moved-temporarily

Answer: A

NEW QUESTION 12

In Cisco Unified Communications Manager, which tool do you use to check SIP traces?

- A. MTP
- B. CCSIP
- C. RTMT
- D. OS Administration Page

Answer: C

NEW QUESTION 15

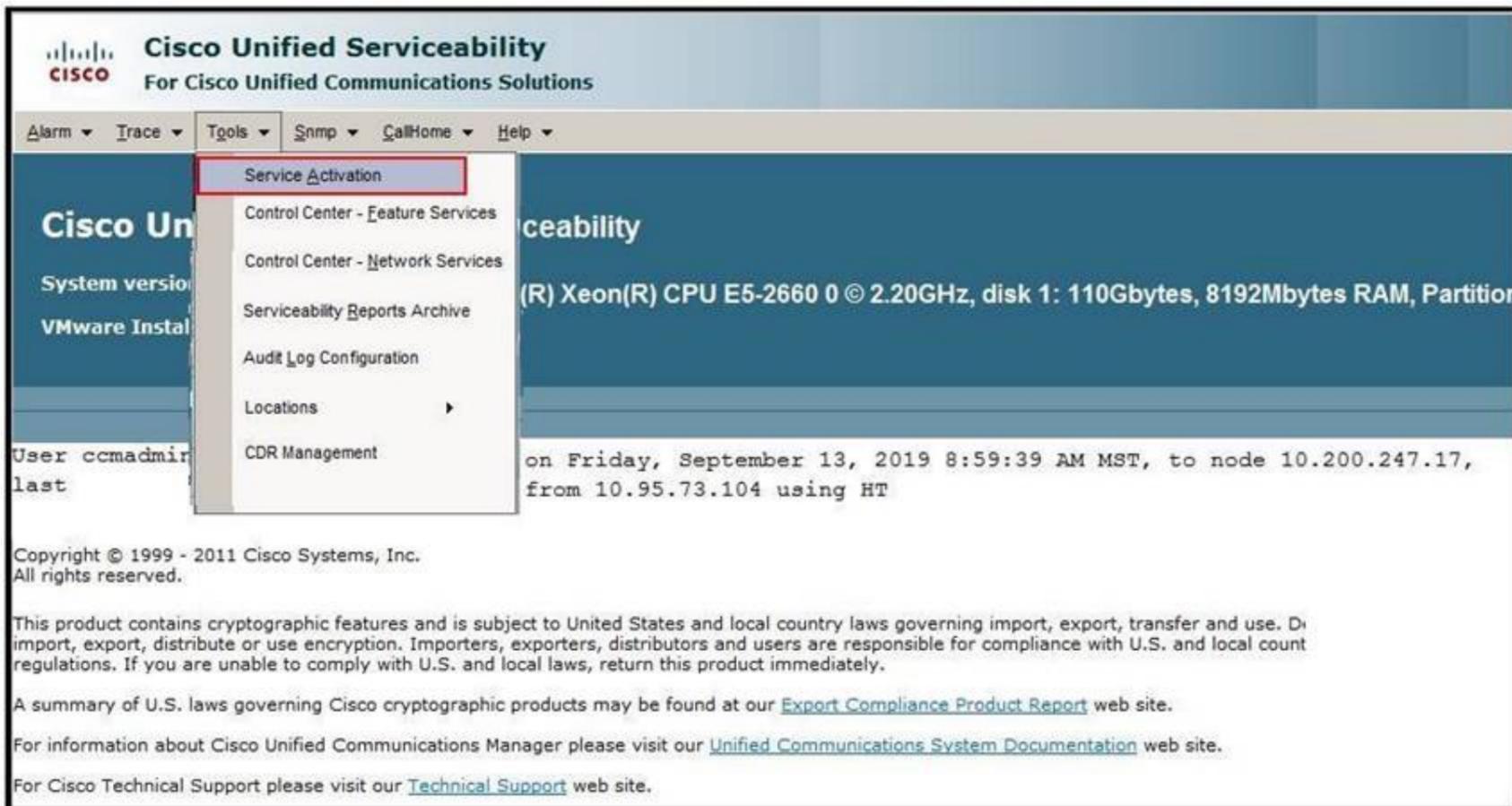
Route Patterns (1-5 of 5)					
Find	Route Patterns	where	Pattern	begins with	Find
<input type="checkbox"/>	41XXXX	To AMER Cluster	Global-Internal		2-AMER-RL
<input type="checkbox"/>	55XX	Rendezvous meetings	Global-Internal		Rendezvous-Conductor
<input type="checkbox"/>	9.0XXXXXXX	Local PSTN	Global-Internal		LocalDevice RL
<input type="checkbox"/>	9.911	Emergency PSTN	Global-Internal		LocalDevice RL
<input type="checkbox"/>	9.911[1-9]!	Emergency PSTN	Global-Internal		LocalDevice RL

Refer to the exhibit. Users report that when they dial the emergency number 9911 from any internal phone, it takes a long time to connect with the emergency operator. Which action resolves this issue?

- A. Adjust the service parameter T302 timer to the desired value.
- B. Adjust the service parameter T204 timer to the desired value.
- C. Check the Urgent Priority check box under 9.911 pattern.
- D. Point the emergency pattern directly to the PSTN gateway.

Answer: C

NEW QUESTION 18



Refer to the exhibit. An administrator is troubleshooting a situation where a call placed from a phone registered to Cisco Unified Communications Manager does not complete. The administrator wants to use the Dialed Number Analyzer on Cisco Unified CM to check which translation pattern the call is matching. However, when logging in to Cisco Unified Serviceability there is no option for Dialed Number Analyzer under the tool menu. Which two steps must be performed to resolve this issue? (Choose two.)

- A. Restart the subscriber
- B. Activate the Cisco Extended Functions service.
- C. Activate the Cisco CallManager service.
- D. Activate the Cisco Dialed Number Analyzer service.
- E. Activate the Cisco Dialed Number Analyzer Server service.

Answer: DE

NEW QUESTION 22

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 23

Which call pickup feature allows users to pick up incoming calls in a group that is associated with their own group?

- A. Other Group Pickup
- B. BLF Call Pickup
- C. Group Call Pickup
- D. Directed Call Pickup

Answer: A

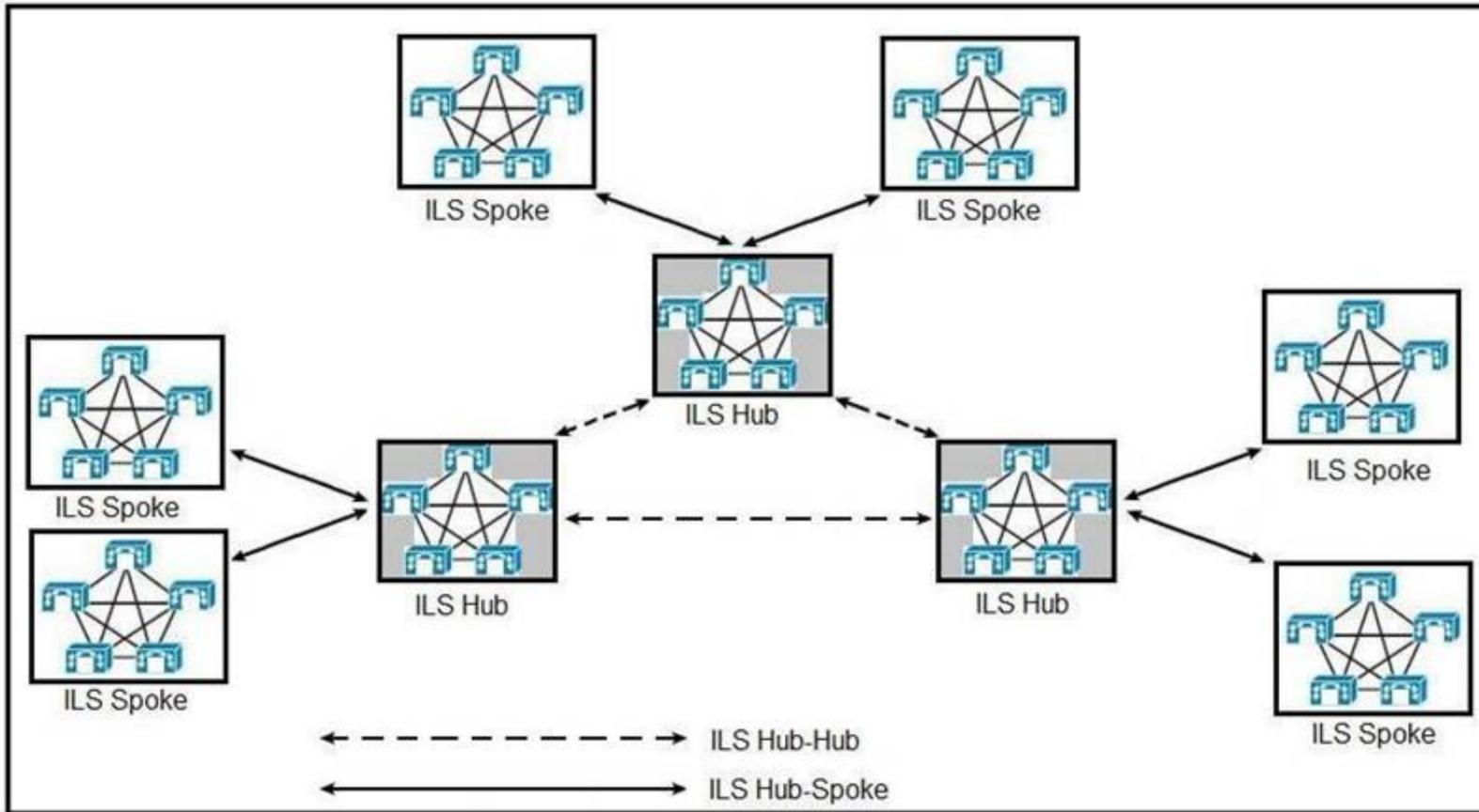
NEW QUESTION 25

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 30



Refer to the exhibit. How many maximum hops can an ILS update traverse?

- A. 3
- B. 6
- C. 9
- D. 12

Answer: A

NEW QUESTION 33

When configuring hunt groups, where do you add the individual directory numbers that will be part of the group?

- A. route group
- B. line group
- C. hunt list
- D. hunt pilot

Answer: B

NEW QUESTION 37

Which configuration element of a hunt group allows for changing Calling Party Transformations settings?

- A. line group
- B. hunt pilot
- C. route group
- D. hunt list

Answer: B

NEW QUESTION 40

Which two types of authentication are supported for the configuration of Intercluster Lookup Service? (Choose two.)

- A. TokenID
- B. username and secret key
- C. TLS certificates
- D. passwords
- E. FQDN of the servers defined in DNS

Answer: CD

NEW QUESTION 42

Which two configuration parameters are prerequisites to set Native Call Queuing on Cisco Unified Communications Manager? (Choose two.)

- A. Cisco IP Voice Media Streaming Service must be activated on at least one node in the cluster.
- B. A unicast music on hold audio source must be configured.
- C. Cisco RIS data collector service must be running on the same server as the Cisco CallManager service.
- D. The maximum number of callers allowed in queue must be 10.
- E. The phone button template must have the Queue Status Softkey configured.

Answer: AC

NEW QUESTION 44

When the services key is pressed Cisco Extension Mobility does not show up. What is the cause of the issue?

- A. The URL configured for Cisco Extension Mobility is not correct.
- B. Cisco Extension Mobility Service is not running.
- C. The phone is not subscribed to Cisco Extension Mobility Service.
- D. Cisco Extension Mobility is not enabled in the Phone Configuration Window (Device > Phone)

Answer: C

NEW QUESTION 45

.....

Thank You for Trying Our Product

We offer two products:

1st - We have Practice Tests Software with Actual Exam Questions

2nd - Questions and Answers in PDF Format

300-815 Practice Exam Features:

- * 300-815 Questions and Answers Updated Frequently
- * 300-815 Practice Questions Verified by Expert Senior Certified Staff
- * 300-815 Most Realistic Questions that Guarantee you a Pass on Your First Try
- * 300-815 Practice Test Questions in Multiple Choice Formats and Updates for 1 Year

100% Actual & Verified — Instant Download, Please Click
[Order The 300-815 Practice Test Here](#)