

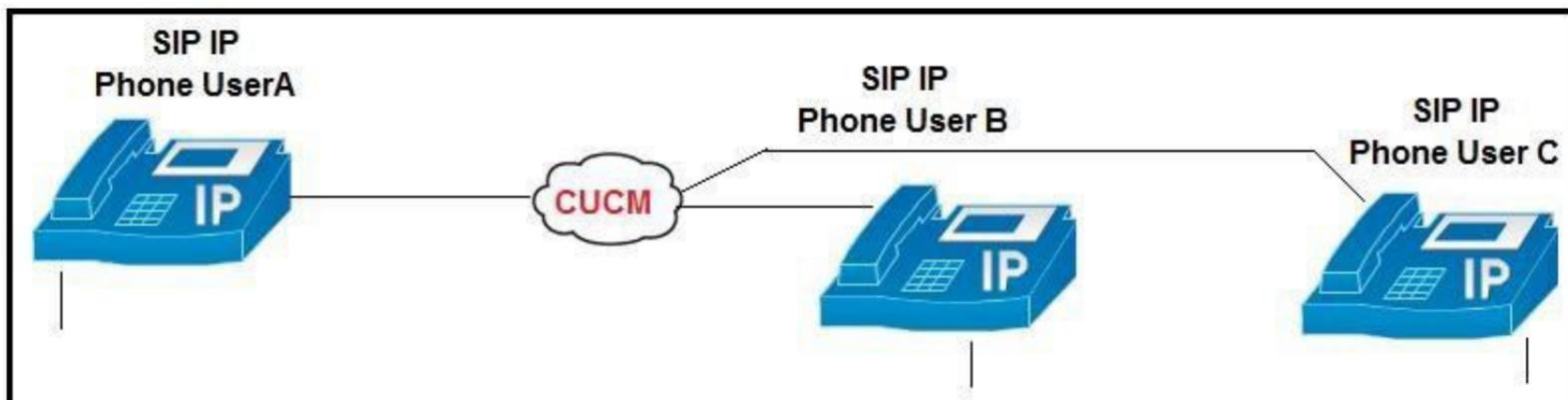


Cisco

Exam Questions 300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

NEW QUESTION 1



Refer to the exhibit. In an active SIP call between phone user A and phone user B, phone A initiates a call transfer to phone user C. Which two scenarios are correct? (Choose two.)

- A. Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.
- B. Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.
- C. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_B User Hold MOH Audio Source settings.
- D. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the music on hold and the MOH audio is chosen from Phone_A Network Hold MOH Audio Source settings.
- E. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_A User Hold MOH Audio Source settings.

Answer: AC

NEW QUESTION 2

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 3

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 4

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 if the 200 OK response
- C. o= line of SDP content
- D. c= line of SDP content

Answer: C

NEW QUESTION 5

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 6

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 7

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 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

An engineer must configure a secure SIP trunk with a remote provider, with a specific requirement to use port 5065 for inbound and otubound traffic. Which two items must be configured to complete this configuration? (Choose two.)

- A. Incoming Port in SIP Information section of the SIP Trunk configuration.
- B. Incoming Port in Security Information of the SIP Profile configuration.
- C. Destination Port in SIP Information section of the SIP Trunk configuration
- D. Incoming Port in SIP Trunk Security Profile configuration
- E. Destination Port in SIP Trunk Security Profile configuration

Answer: CD

NEW QUESTION 10

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 10

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 15



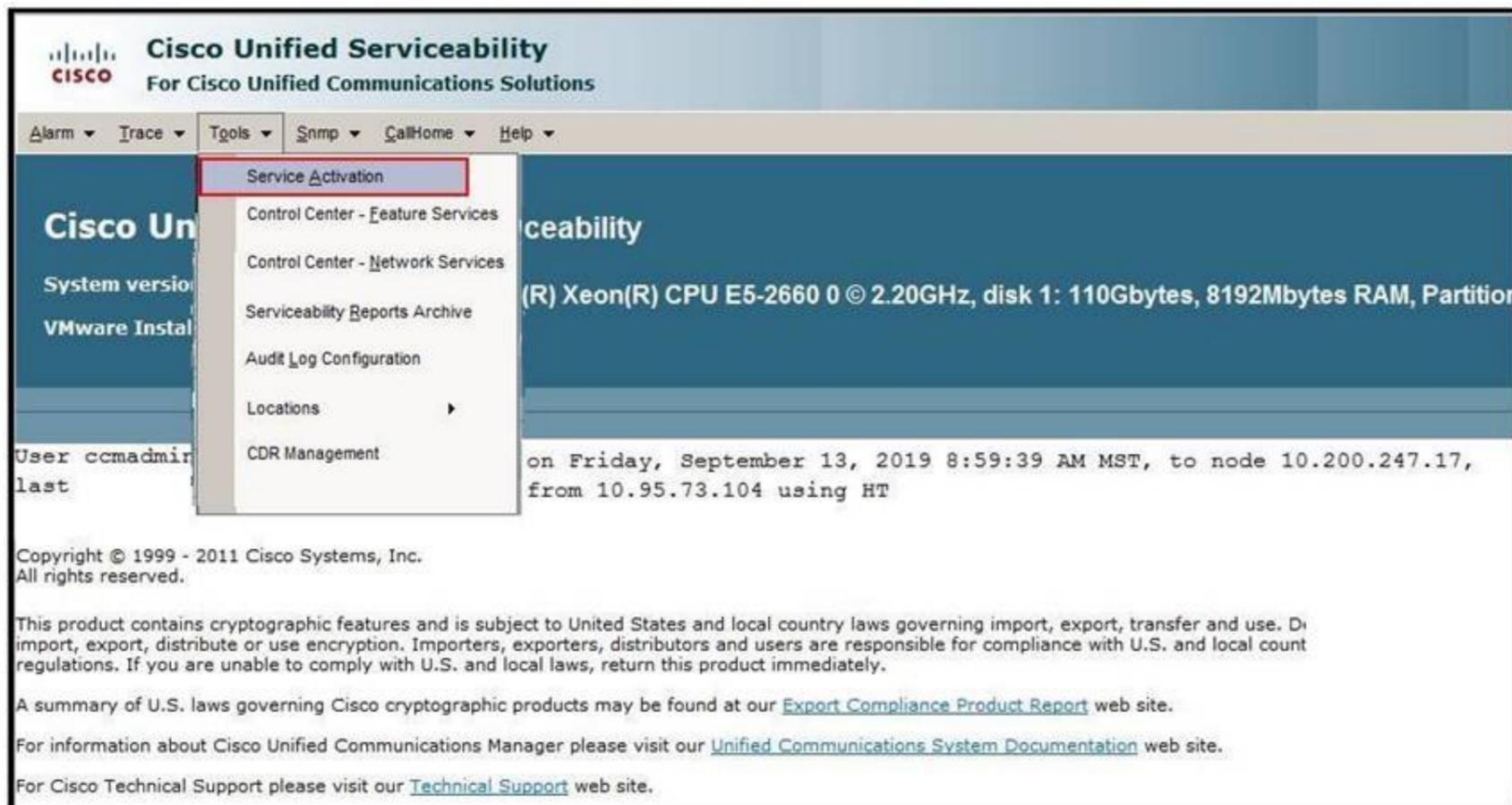
Pattern	Description	Partition	Route Filter	Associated Device
41XXXX	To AMER Cluster	Global-Internal		2-AMER-RL
55XX	Rendezvous meetings	Global-Internal		Rendezvous-Conductor
9.0XXXXXXXXXX	Local PSTN	Global-Internal		LocalDevice RL
9.911	Emergency PSTN	Global-Internal		LocalDevice RL
9.91[1-9]!	Emergency 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 timet 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 20



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

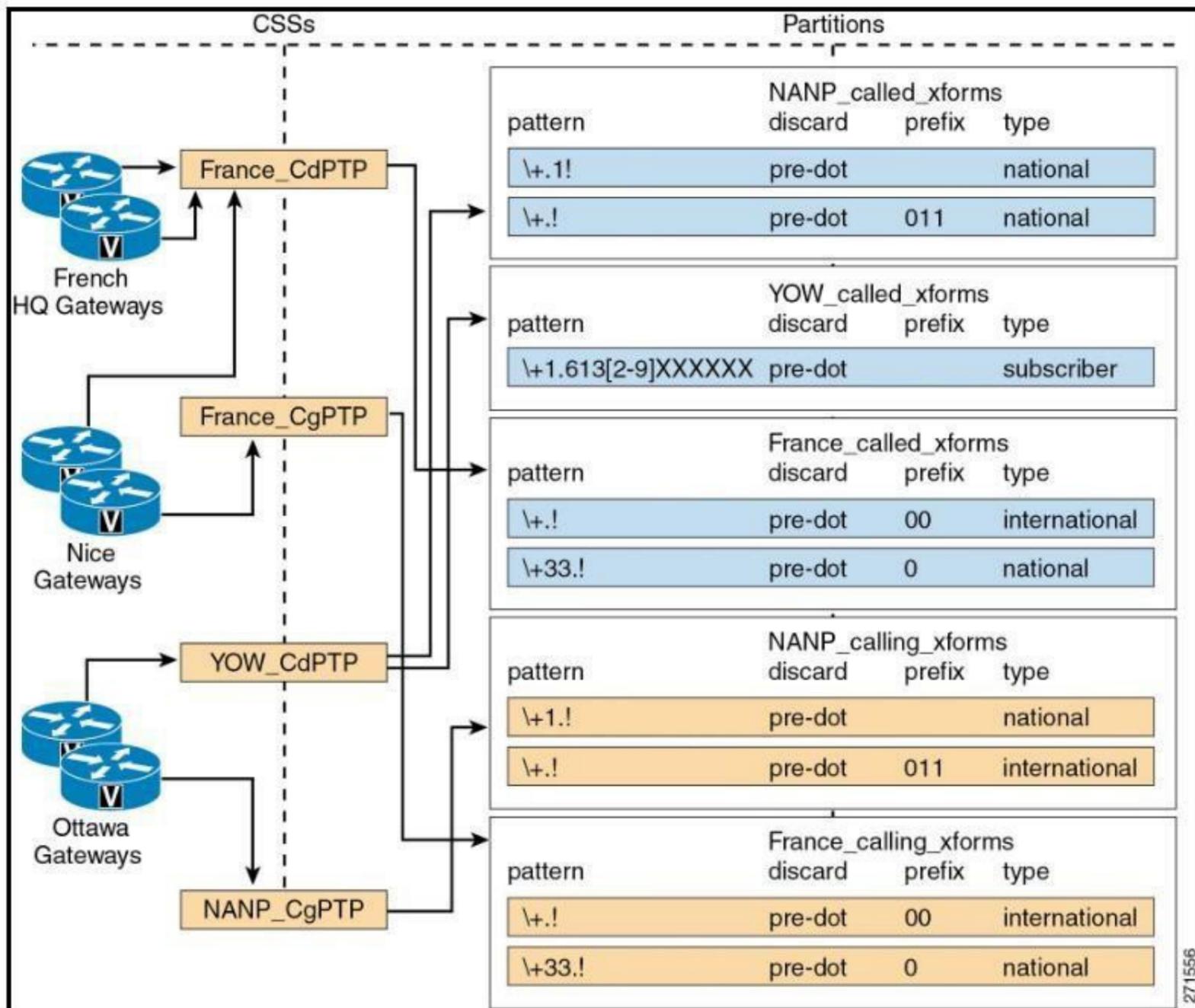
NEW QUESTION 25

In Cisco Unified Communications Manager globalized call routing is implemented and must confirm that it is correctly implemented without making a call. Which tool do you use for verification?

- A. Dialed Number Analyzer
- B. Real-Time Monitoring Tool
- C. SDI trace
- D. SDL trace

Answer: A

NEW QUESTION 30



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 31

How does an engineer globalize routing for ingress calls coming from the PSTN to internal DNs?

- A. At the PSTN gateway, put the calling number in PSTN format and the called number in DN format.
- B. At Cisco Unified CM, put the calling number in E.164 format and the called number in PSTN format.
- C. At the PSTN gateway, put the calling number in E.164 format and the called number in localized (DN) format.
- D. At Cisco Unified Communications Manager, put the calling number in E.164 format and the called number in E.164 format.

Answer: B

NEW QUESTION 35

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 36

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 41

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 43

Configure Call Queuing in Cisco Unified Communications Manager. Where do you set the maximum number of callers in the queue?

- A. in the telephony service configuration
- B. in the queuing configuration
- C. in Cisco Unified CM Enterprise Parameters
- D. in Cisco Unified CM Service Parameters

Answer: B

NEW QUESTION 45

A user reports when they press the services key they do not receive a user ID and password prompt to assign the phone extension. Which action resolves the issue?

- A. Create the default device profiles for all phone models that are used.
- B. Subscribe the phone to the Cisco Extension Mobility service.
- C. Create the end user and associate it to the device profile.
- D. Assign the extension as a mobile extension.

Answer: B

NEW QUESTION 47

What are the elements for Device Mobility configuration?

- A. physical location, device pool, and Device Mobility group
- B. device pool, Device Mobility group, and region
- C. physical locatio
- D. Device Mobility group, and region
- E. device pool, Device Mobility group, and Cisco IP phone

Answer: A

NEW QUESTION 50

.....

About ExamBible

Your Partner of IT Exam

Found in 1998

ExamBible is a company specialized on providing high quality IT exam practice study materials, especially Cisco CCNA, CCDA, CCNP, CCIE, Checkpoint CCSE, CompTIA A+, Network+ certification practice exams and so on. We guarantee that the candidates will not only pass any IT exam at the first attempt but also get profound understanding about the certificates they have got. There are so many alike companies in this industry, however, ExamBible has its unique advantages that other companies could not achieve.

Our Advances

* 99.9% Uptime

All examinations will be up to date.

* 24/7 Quality Support

We will provide service round the clock.

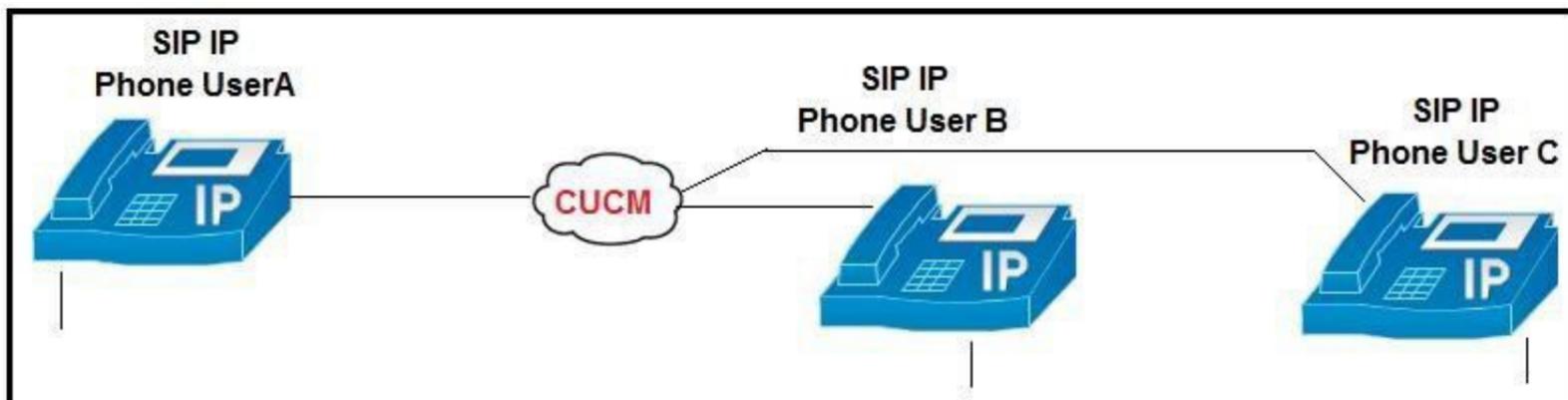
* 100% Pass Rate

Our guarantee that you will pass the exam.

* Unique Gurantee

If you do not pass the exam at the first time, we will not only arrange FULL REFUND for you, but also provide you another exam of your claim, ABSOLUTELY FREE!

NEW QUESTION 1



Refer to the exhibit. In an active SIP call between phone user A and phone user B, phone A initiates a call transfer to phone user C. Which two scenarios are correct? (Choose two.)

- A. Phone_A sends a SIP-REFER message to the Cisco Unified Communications Manager with Phone_C information in the Refer-To section.
- B. Phone_B sends a SIP-REFER message to the Cisco Unified CM with Phone_C information in the Refer-To section.
- C. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_B User Hold MOH Audio Source settings.
- D. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the music on hold and the MOH audio is chosen from Phone_A Network Hold MOH Audio Source settings.
- E. As soon as Phone_A presses the Transfer button for the first time, Phone_B hears the MOH and the MOH audio is chosen from Phone_A User Hold MOH Audio Source settings.

Answer: AC

NEW QUESTION 2

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 3

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 4

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 if the 200 OK response
- C. o= line of SDP content
- D. c= line of SDP content

Answer: C

NEW QUESTION 5

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 6

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 7

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 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

An engineer must configure a secure SIP trunk with a remote provider, with a specific requirement to use port 5065 for inbound and otubound traffic. Which two items must be configured to complete this configuration? (Choose two.)

- A. Incoming Port in SIP Information section of the SIP Trunk configuration.
- B. Incoming Port in Security Information of the SIP Profile configuration.
- C. Destination Port in SIP Information section of the SIP Trunk configuration
- D. Incoming Port in SIP Trunk Security Profile configuration
- E. Destination Port in SIP Trunk Security Profile configuration

Answer: CD

NEW QUESTION 10

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 10

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 15

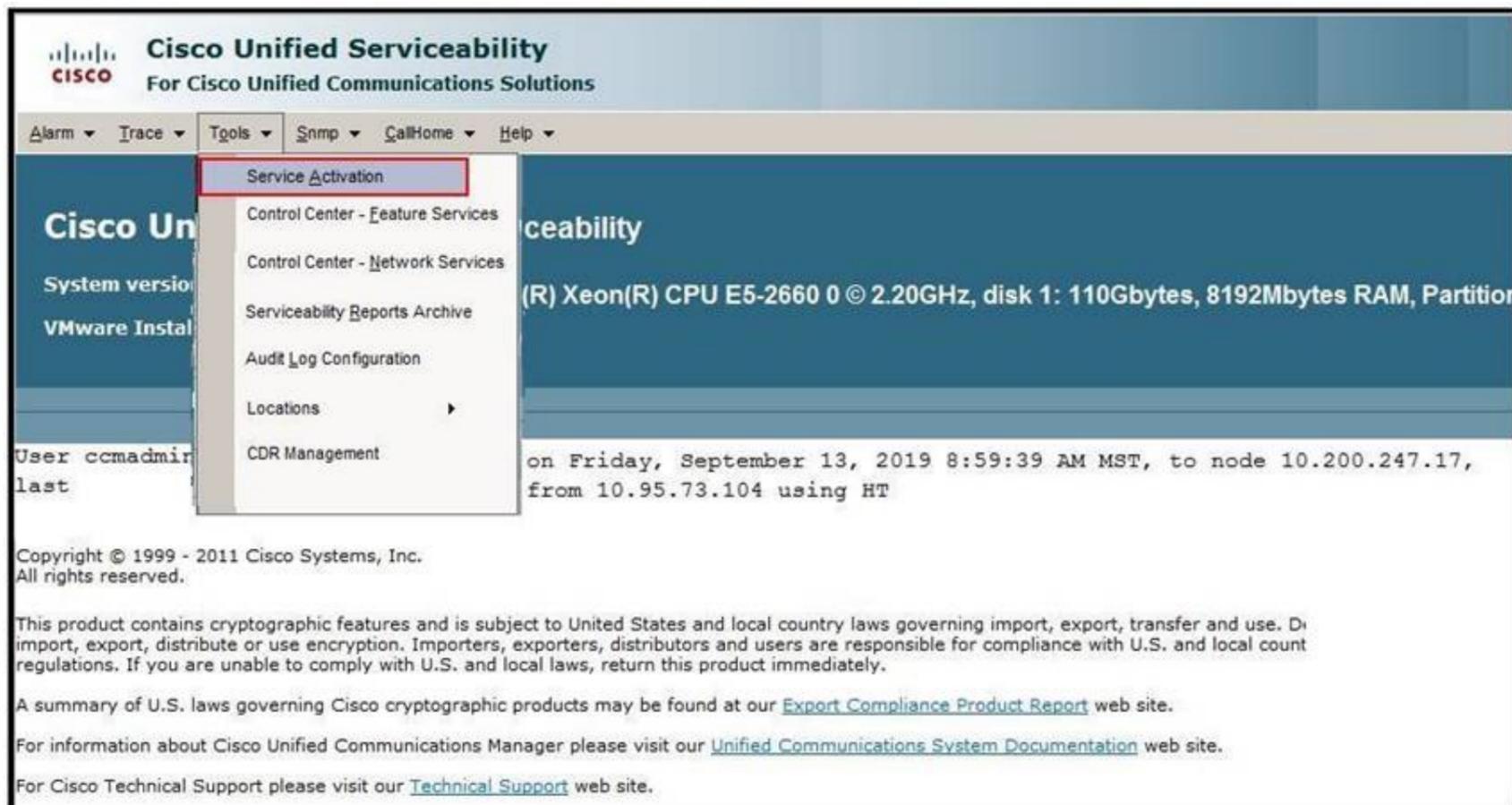
Route Patterns (1-5 of 5)								
Find	Route Patterns	where	Pattern	begins with	Find	Clear Filter		
	Pattern	Description	Partition	Route Filter	Associated Device			
<input type="checkbox"/>	41XXXX	To AMER Cluster	Global-Internal		2-AMER-RL			
<input type="checkbox"/>	55XX	Rendezvous meetings	Global-Internal		Rendezvous-Conductor			
<input type="checkbox"/>	9.0XXXXXXXXXX	Local PSTN	Global-Internal		LocalDevice RL			
<input type="checkbox"/>	9.911	Emergency PSTN	Global-Internal		LocalDevice RL			
<input type="checkbox"/>	9.91[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 timet 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 20



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

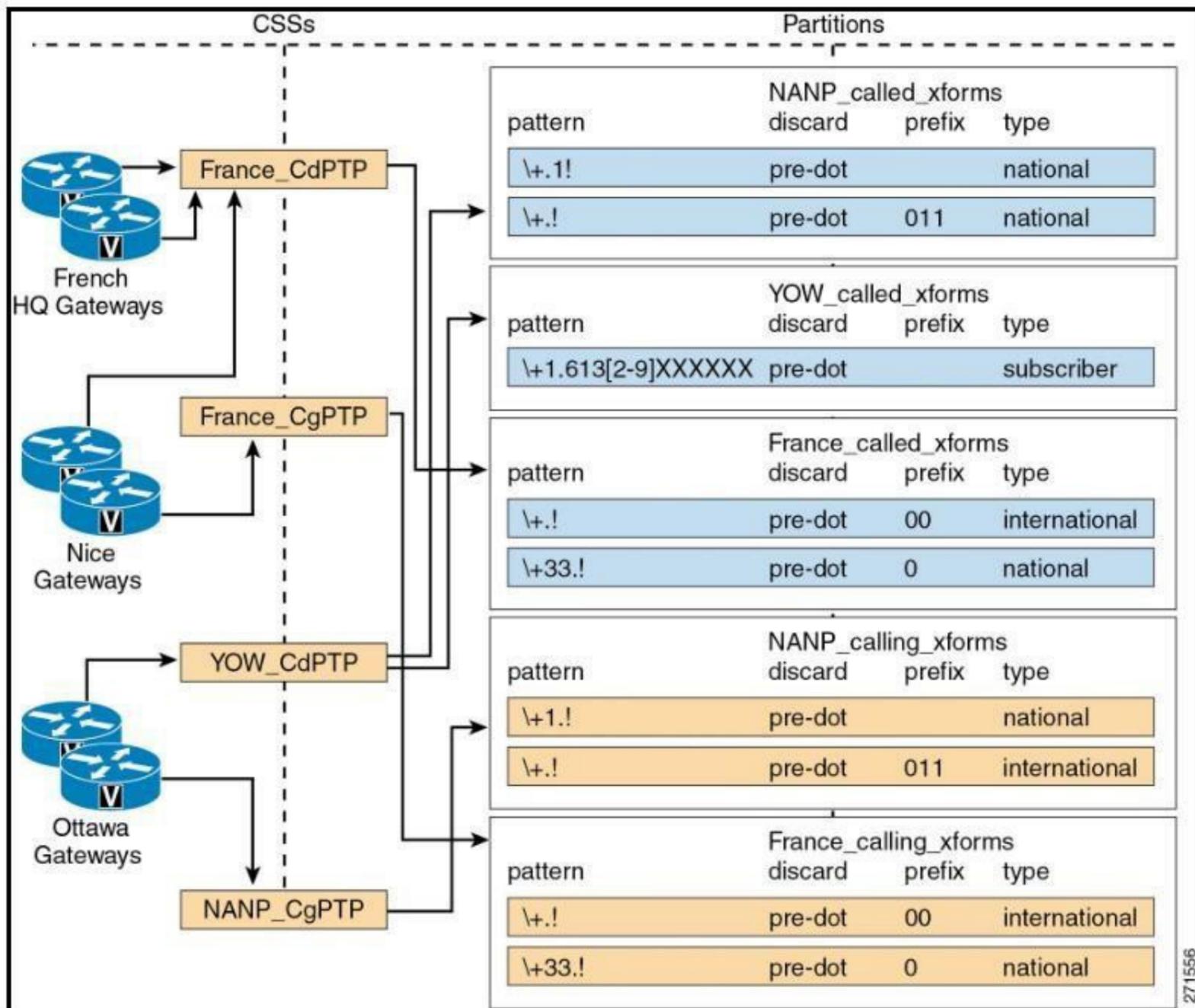
NEW QUESTION 25

In Cisco Unified Communications Manager globalized call routing is implemented and must confirm that it is correctly implemented without making a call. Which tool do you use for verification?

- A. Dialed Number Analyzer
- B. Real-Time Monitoring Tool
- C. SDI trace
- D. SDL trace

Answer: A

NEW QUESTION 30



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 31

How does an engineer globalize routing for ingress calls coming from the PSTN to internal DNs?

- A. At the PSTN gateway, put the calling number in PSTN format and the called number in DN format.
- B. At Cisco Unified CM, put the calling number in E.164 format and the called number in PSTN format.
- C. At the PSTN gateway, put the calling number in E.164 format and the called number in localized (DN) format.
- D. At Cisco Unified Communications Manager, put the calling number in E.164 format and the called number in E.164 format.

Answer: B

NEW QUESTION 35

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 36

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 41

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 43

Configure Call Queuing in Cisco Unified Communications Manager. Where do you set the maximum number of callers in the queue?

- A. in the telephony service configuration
- B. in the queuing configuration
- C. in Cisco Unified CM Enterprise Parameters
- D. in Cisco Unified CM Service Parameters

Answer: B

NEW QUESTION 45

A user reports when they press the services key they do not receive a user ID and password prompt to assign the phone extension. Which action resolves the issue?

- A. Create the default device profiles for all phone models that are used.
- B. Subscribe the phone to the Cisco Extension Mobility service.
- C. Create the end user and associate it to the device profile.
- D. Assign the extension as a mobile extension.

Answer: B

NEW QUESTION 47

What are the elements for Device Mobility configuration?

- A. physical location, device pool, and Device Mobility group
- B. device pool, Device Mobility group, and region
- C. physical locatio
- D. Device Mobility group, and region
- E. device pool, Device Mobility group, and Cisco IP phone

Answer: A

NEW QUESTION 50

.....

Relate Links

100% Pass Your 300-815 Exam with ExamBible Prep Materials

<https://www.exambible.com/300-815-exam/>

Contact us

We are proud of our high-quality customer service, which serves you around the clock 24/7.

Viste - <https://www.exambible.com/>