

Exam Questions 300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)

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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 Offer
- 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

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 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

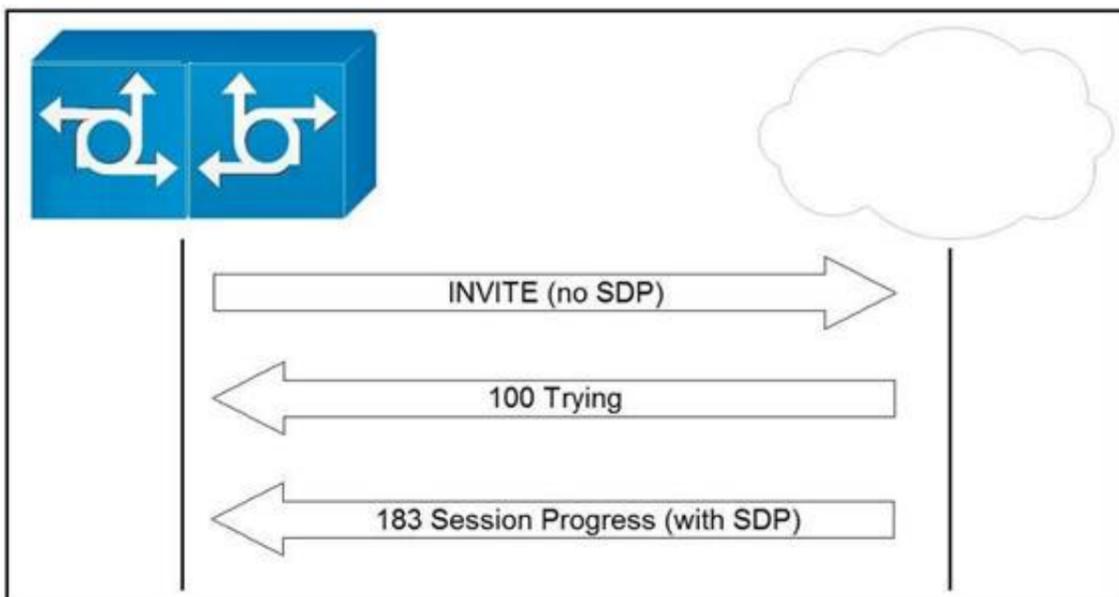
```
voice translation-profile incoming
  translate called 999
!
voice translation-rule 999
  rule 1/\ (^([1-2] [1-2] [1-2]\ ) 333\ ([4-5] [4-5] .\ ) $ / / \2333\1/
!
dial-peer voice 999 voip
  translation-profile outgoing incoming
  session protocol sipv2
  incoming called-number
  dtmf-relay rtp-nte
  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 6



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 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

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 9

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 10

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 10

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 14

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 15

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 18

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