QUESTION 12
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
Explanation:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/admin/10_5_2/ccmfeat/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_1052/CUCM_BK_C3A84B33_00_cucm-feature-configuration-guide_chapter_011101.html#CUCM_TK_A337E035_00

QUESTION 13
How many Cisco Unified Mobility destinations can be configured per user?

A. 1
B. 10
C. 4
D. 6

Answer: B

QUESTION 14
Which type of search message appears in the Cisco TelePresence Video Communication Server search history page when it receives a H.323 call from a RAS-enabled endpoint that originates from an external zone?
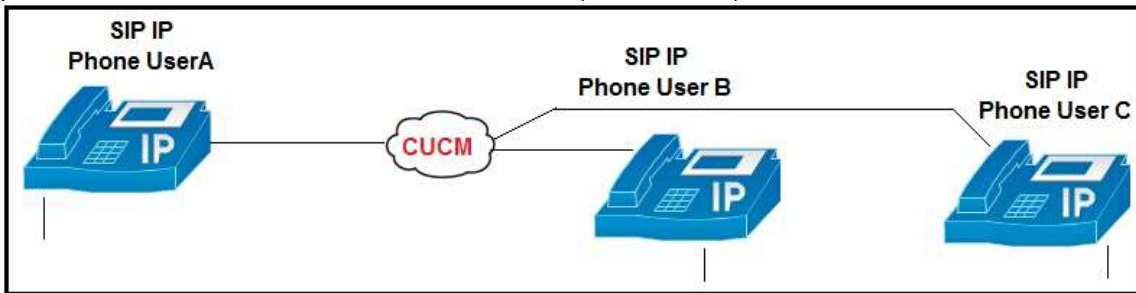
A. ARQ
B. SETUP
C. LRQ
D. INVITE
E. OPTIONS

Answer: C

QUESTION 15

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

QUESTION 16
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?

```
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
```

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. DTMF is supported only in Early Offer.
C. The rtpmap:0 value for the negotiated codec is marking DTMF as inactive.
D. No DTMF is negotiated.

Answer: D

QUESTION 17
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
Explanation:
http://ccievoicehopeful.blogspot.com/2012/09/h323-notes.html

QUESTION 18
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

QUESTION 19
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: D

QUESTION 20
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

QUESTION 21
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: A

QUESTION 22
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