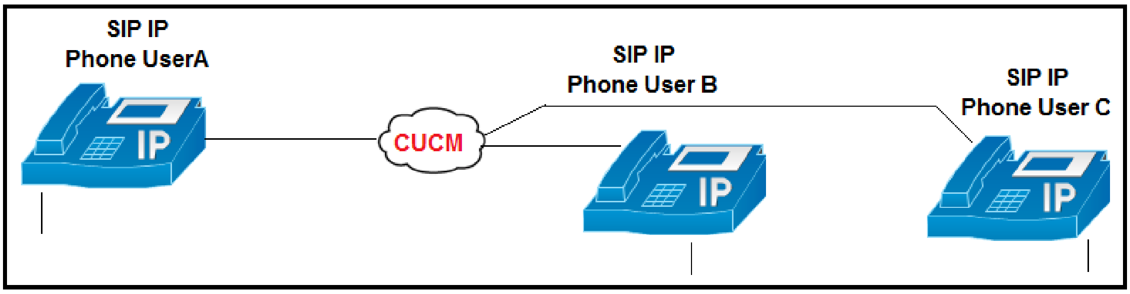
# Implementing Cisco Advanced Call Control and Mobility Services (CLASSM)

**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(s):** A D

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**2.** Refer to the exhibit.

```
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
```

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. 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(s):** D

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**3.** The administrator of ABC company is troubleshooting a one-way audio issue for a call that usesH.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(s):** D

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**4.** 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(s):** A B

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**5.** 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(s):** A

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**6.** 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(s):** D

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**7.** 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(s):** A

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**8.** 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(s):** B

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**9.** 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(s):** A

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**10.** 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 one-way 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(s):** C D

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**11.** 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(s):** C

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**12.** 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 (playoutdelay compensation).

**Answer(s):** D

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**13.** 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(s):** A

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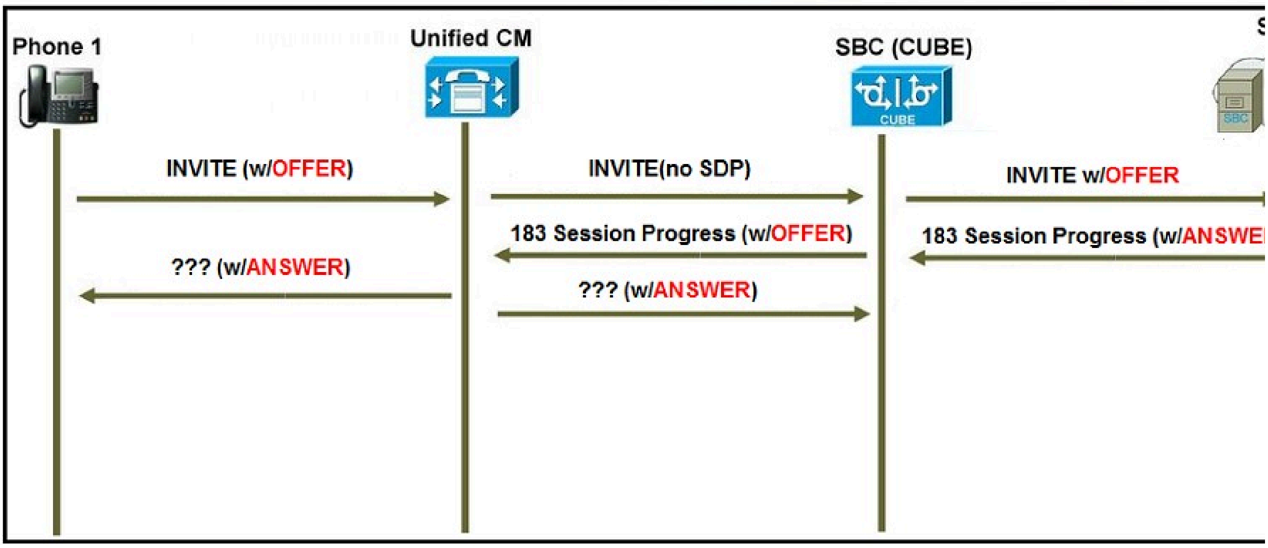
**14.** Refer to the exhibit.




A user reports that when they call a specific phone number, no one answers the call, but when they call from a mobile phone, the call is answered. The engineer troubleshooting the issue is expecting the far-end gateway to cut through audio on the 183 Session Progress SIP message. Which SIP Profile configuration element is necessary for the Cisco Unified Communications Manager to send acknowledgement of provisional responses?

A. Allow Passthrough of Configured Line Device Caller Information must be enabled.

B. Accept Audio Codec Preferences in Received Offer must be set to On.

C. On the SIP Profile, the configuration parameter SIP Rel1XX Options must be set to Send PRACK for all 1xx Messages. D. Early Offer for G Clear Calls must be enabled.

**Answer(s):** C

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**15.** 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(s):** A C

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**16.** 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(s):** B

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**17.** You see the voice register pool 1 command in your Cisco Unified Communications Manager Express configuration. Which configuration is occurring in this section?

A. configuration for a single SIP phone

B. configuration items common for all SIP phones

C. configuration for a pool of SIP phones (similar to device pool on Cisco Unified Communications Manager)

D. configuration for SIP registrar service

**Answer(s):** A

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**18.** Which top-level IOS command is needed to begin the configuration of a Cisco Unified Communications Manager Express gateway to enable phones to be registered via SIP?

A. allow-connections sip to sip

B. voice service voip

C. voice register global

D. voice register dn

**Answer(s):** C

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**19.** 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(s):** A

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**20.** Where is the dtmf-relay command configured on Cisco Unified Border Element?

A. in the voice-class VoIP configuration

B. in the VoIP dial peer

C. in global SIP configuration

D. in the VoIP or POTS dial peers

**Answer(s):** B

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