- **Vendor:** Cisco
- **Exam Code:** 300-815
- **Exam Name:** Implementing Cisco Advanced Call Control and Mobility Services (CLACCM)
- **New Updated Questions from Braindump2go (Updated in Sep/2020)**

QUESTION 23
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: A
Explanation:
https://www.cisco.com/c/en/us/support/docs/voice/ip-telephony-voice-over-ip-voip/211306-In-Depth--of-Cisco-IOS-and-IO.html#anc8

QUESTION 24
Which two Cisco Unified Communications Manager gateway types require manual dial-peer configuration to enable PSTN call routing? (Choose two.)

A. MGCP
B. H.323
C. SCCP
D. SIP
E. VG224

Answer: BD

QUESTION 25
When configuring Cisco Unified Mobility, which parameter defines the access control for a call that reaches out to a remote destination?

A. Calling Party Transformation Calling Search Space under Remote Destination Profile Information
B. User Local under Remote Destination Profile Information
C. Rerouting Calling Search Space under Remote Destination Profile Information
D. Rerouting Calling Search Space under Remote Destination information
E. Calling Search Space under Phone Configuration

Answer: C

QUESTION 26
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: AC
Explanation:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/port/9_1_1/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-91/CUCM_BK_T2CA6EDE_00_tcp-port-usage-guide-91_chapter_01.html

QUESTION 27
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
Explanation:
https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-communications-manager-callmanager/213583-procedure-to-analyse-call-flow-of-sip-ca.html

QUESTION 28
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 (playout delay compensation).

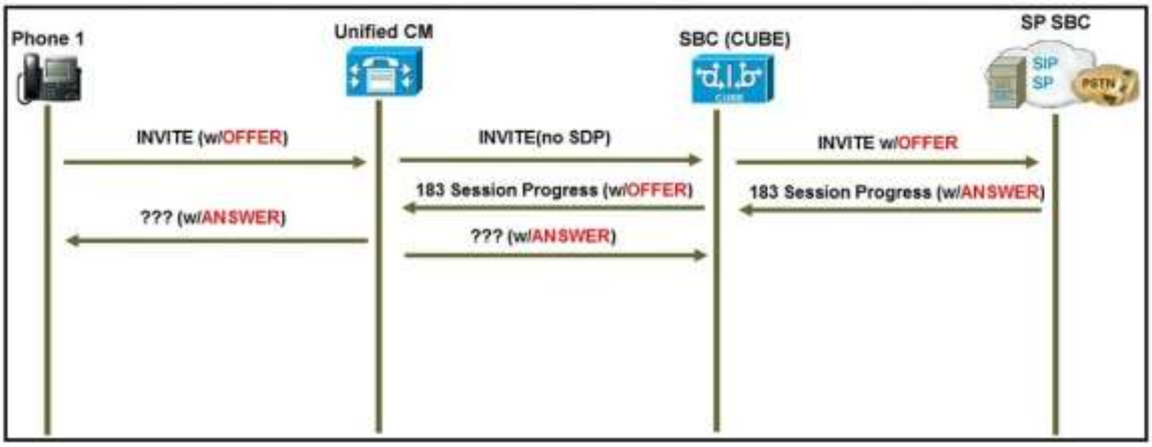
Answer: D
Explanation:
https://www.cs.columbia.edu/~hgs/rtp/faq.html

QUESTION 29
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: A

QUESTION 30
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: C

QUESTION 31
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
Explanation:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/ srst_sip_isr4000.html

QUESTION 32
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
Explanation:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/manual/cmeadm/
cmetoll.html#concept_ECC4F4E7ED0F45C594B703EEF34762F2

QUESTION 33
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: C
Explanation:
https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cusrst/admin/sccp_sip_srst/configuration/guide/SCCP_and_SIP_SRST_Admin_Guide/ srst_setting_up_using_sip.html