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Cisco

300-815

Implementing Cisco Advanced Call Control and Mobility Services (CLACCM) - CCNP

Question: 118

Refer to the exhibit.

```
voice translation-rule 84
        rule 1 /^\ ([2-9]..[2-9]……$\)/ /\2/
```

Users report that outbound PSTN calls from phones registered to Cisco Unified Communications Manager are not completing. The local service provider in North America has a requirement to receive calls in 10-digit format. The Cisco Unified CM sends the calls to the Cisco Unified Border Element router in a globalized E.164 format. There is an outbound dial peer on Cisco Unified Border Element configured to send the calls to the provider. The dial peer has a voice translation profile applied in the correct direction but an incorrect voice translation rule applied, which is shown in the exhibit.

Which rule modified DNIS in the format that the provider is expecting?
A . rule 1 /^/+([^1].*)//0111/
B . rule 1/^+1([2-9]..[2-9]……$)//1/
C . rule 1 /^([2-9]..[2-9]……$)//1/
D . rule 1 /^+1([2-9]..[2-9]……$)///

Answer: B

Question: 119

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).
B . 245 Terminal Capability Set
D . 245 Open Logical Channel
F . 225 Connect
H . 245 Open Logical Channel Ack

Answer: B

Explanation:

Reference: http://ccievoicehopeful.blogspot.com/2012/09/h323-notes.html

Question: 120

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:

Reference: https://www.cs.columbia.edu/~hgs/rtp/faq.html

Question: 121

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

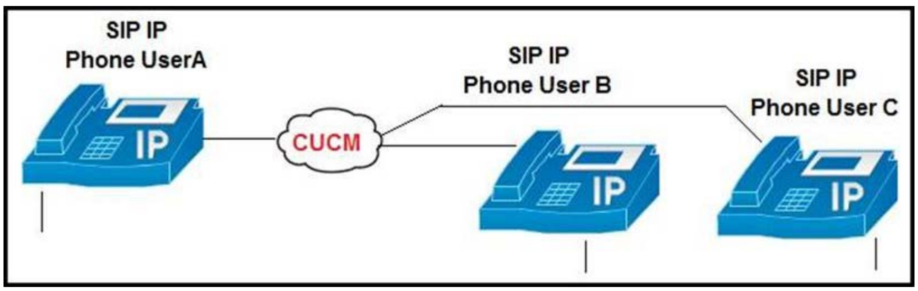
Answer: A

Explanation:

Reference: https://www.cisco.com/c/en/us/support/docs/unified-communications/unified-borderelement/200412-DTMF-Relay-and-Interworking-on-CUBE.html#anc35

Question: 122

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: 123

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:

Reference:
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

Question: 124

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
D . 38 fax relay

E . transcoding
F . SIP trunk

Answer: AC

Explanation:

Reference:
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: 125

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:

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

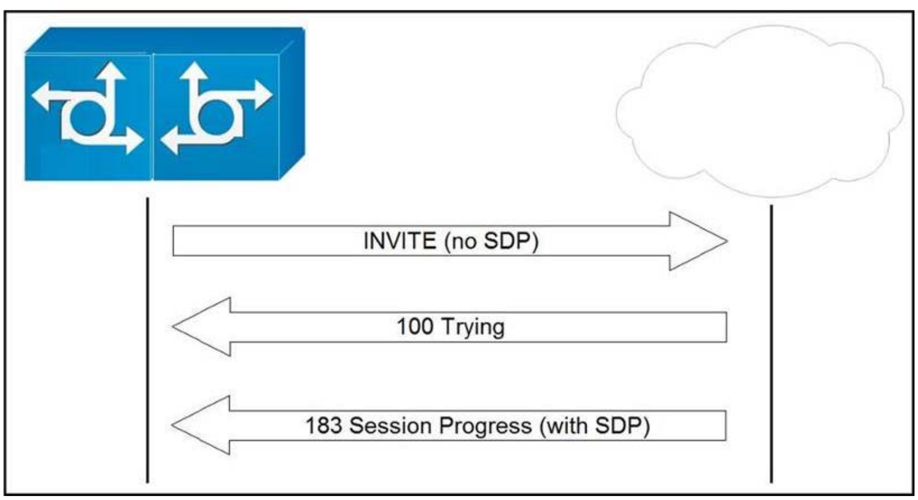
Question: 126

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

Question: 127

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 . Enanle PRAC
C . Enable Early Offer on the Cisco Unified Border Element.
D . Enable the supplementary-service media-renegotiate command.
E . Enable Media Flow Around
F . Enable Mid-Call Signaling Consumption.

Answer: AB

SAMPLE QUESTIONS

*These questions are for demo purpose only. **Full version** is up to date and contains actual questions and answers.*

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