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QUESTION 421A Call is made between two desk phones enabled with single number reach that are registered to a Cisco Unified CM cluster. The device pool for each device has a local route group defined. When the call is placed to exit the system, which device pool control the destination gateway? A. Destination RDP B. Source Phone C. Source RDP D. Destination phone Answer: B

QUESTION 422Refer to the exhibit. Which three facts can be determined about the audio parameters of this call from this session description protocol? (Choose Three) A. The DTMF relay will be RFC2833 B. The codec will be G711 C. The codec will be G729 D. VAD will be disabled for this call E. VAD will be enabled for this call F. The call will be a T38 fax call Answer: ACD

QUESTION 423Which two SCCP call states support the Meet Me soft key? (Choose two.) A. On Hook B. Connected C. On Hold D. Off Hook E. Ring Out F. Connected Conference Answer: AD

QUESTION 424A Jabber for Windows user is on a call with Cisco Telepresence EX90 endpoint at the same location. During the call, the video on the Jabber for Windows application was high quality but the video on the EX90 was choppy and slow. When the administrator checked the service rate on the EX90 it showed 2048 Kbps. Which two configuration changes can fix this problem? A. Lower the bit rate in the region configuration in Communication Manager between the endpoints B. Increase the location bandwidth for immersive video between the endpoints C. Enable BFCP in the SIP profile for the Jabber client D. Enable H.263 on the EX90 E. Replace the camera for the Jabber user with the precision HD USB camera F. Increase the bandwidth between the Jabber video client and the EX90 Answer: EF

QUESTION 425Refer to the exhibit. An engineer mapped the enhanced location call admission control configuration to match the physical links bandwidth allowances. Assuming no other calls are consuming any bandwidth, how many G722 calls are allowed between site A and site G? A. 5 B. 7 C. 12 D. 25 E. 37 Answer: B

QUESTION 426Refer to the exhibit. Which SIP message will trigger the calling device to open channels for early media reception? A. 180 Ringing B. ACK C. INVITED. 183 session-progress E. 200 OK Answer: D

QUESTION 427Refer to the exhibit. A network engineer is troubleshooting a NTP synchronization issue in CUCM. Why is NTP unsynchronized? A. The NTP server used is a Windows based NTP server B. The IOS command NTP server 172.25.140.151 version 3 is advertising NTPv3 C. The NTP server stratum is higher than four D. A firewall is blocking NTP port 123 Answer: A

QUESTION 428Refer to the exhibit. A Cisco Collaboration engineer is writing a report to summarize the call distribution characteristics in a Cisco Unified Contact Centre Express queue. Which three characteristics can be reported about the call distribution? (Choose three.) A. This queue will not work because no prompt has been selected B. Calls to this queue can be distributed in a round-robin manner between agents C. Agents that are answering calls for this queue can answer calls to other queues if available D. Agents in this queue are expected to finish (wrap-up) a call within 60 seconds E. Calls to this queue are handled in the order they were received unless prioritized by the script F. Changing the queue name from SupportQueue to Support01 requires updates to the script G. Agents logged in to this queue automatically receive calls without the need to do anything else (Automatic work) Answer: CEF

QUESTION 429A Cisco Unified CM engineer configured a phone VPN for remote users but the users cannot register the phones to the VPN. Which configuration change fixes this problem? A. Configure enable outside in the webVPN configuration on the Cisco ASA B. Configure the split-tunnel-policy tunnel all attribute on the Cisco ASA C. Configure the ssl trust-point SSL outside on the Cisco ASA D. Remove the Cisco ASA IP address from the VPN load-balancing configuration Answer: D

QUESTION 430Refer to the exhibit. An engineer is troubleshooting transcoding issue in a remote branch office after a WAN outage. All IP phones can register to a CME in SRST 2900 ISR Router however, the users reported that calls disconnect after pressing the answer soft key. Which three configuration are necessary for successful media resource failover? (Choose three) A. SCCP ccm group 1 Associate ccm 3 priority 1 B. SCCP ccm 10.1.1.2 identifier 3 version 7.0 C. Telephony-service Sdspfarm units 1 Sdspfarm tag 1 XD-REMOTED. Sccp ccm 10.1.1.2 identifier 1 version 7.0 E. Associate profile 3 register XD-REMOTE2 F. Sccp ccm group 1 Associate ccm 3 priority 3 Answer: BCF

QUESTION 431Refer to the exhibit. A collaboration engineer is configuring device mobility. Which three events occur to phone A1 when it moves to physical location B? (Choose three) A. Phone A PSTN calls preserve home location dialling behaviour B. Phone A inherits CSS from roaming device pool DP_B2 C. Phone A PSTN calls adopts roaming location dialling behaviour D. Phone A retains CSS from Home device pool DP_A1 E. Phone A retains media resource group list from home device pool DP_A1 F. Phone A inherits media resource group list from roaming device pool DP_B2 Answer: ADE

QUESTION 432Refer to the exhibit. A network engineer is troubleshooting a call routing issue where failed

calls on the primary path (SiteA-RTR) was not sent to the secondary path (SiteC-RTR). Why is CUCM unable to extend the call setup through SiteC-RTR? A. Stop routing on Q.931 Disconnect cause code is set to 27B. Stop routing on Unallocated Number Flag is set to trueC. Stop Routing on User Busy Flag is set to trueD. Retry count for SIP Invite is set to 1E. Retry count for SIP response is set to 1 Answer: B QUESTION 433Which definition is included in a Cisco UC on UCS TRC? A. Required RAID configuration When TRC uses direct attached storageB. Configuration of virtual-to-physical network interface mappingC. Step-by-step procedures for hardware BIOS, firmware, drivers, and RAID setupD. Server model and local components (CPU, RAM, adapters, local storage) by name only. Part numbers are not included because they change over timeE. Configuration settings and patch recommendations for VMware software Answer: A QUESTION 434Refer to the exhibit. A phone VPN failed to establish a VPN with the Cisco ASA. The support engineer downloaded the console logs and analysed them. When two steps resolve this issue? (Choose two) A. Configure user and password authentication instead of certificate onlyB. Uncheck the enable Host ID check checkbox under the VPN profile in Cisco Unified CMC. Reset the Cisco Unified CM TFTP service to allow caching of the new certificateD. Delete the current certificate so the phone can download a new oneE. Register the phone internally to download the new configuration Answer: BE QUESTION 435Refer to the exhibit. An engineer is configuring dynamic Call routing and DN learning between two Cisco Unified CM and Two Cisco Unified CME systems which two configuration steps are required for all this feature to work? (Choose two) A. Configure routers A and B to use a different autonomous system number for DN routingB. Configure routers A and B to use EIGRP for IP RoutingC. Configure Cisco Unified CM A+B as service advertisement framework clientsD. Configure router A and B to use OSPF for IP RoutingE. Configure Cisco Unified CME A+B as service advertisements forwardersF. Configure routers A and B to use the same autonomous system number for DN Routing Answer: CF QUESTION 436Refer to the exhibit. A carrier delivers a SIP call to cisco Unified CM through a Cisco Unified border Elements with The Invite destination different than "To" field. The Unified CM Administration engineer sees that the calls go to the invite destination instead of the "To" field Unified CM. Which option shows how the engineer correct that problem in the Cisco Unified border Elements router? A. B. C. D. Answer: A QUESTION 437A Cisco collaboration architect is evaluating a list of codecs to use in a voice infrastructure. Which three facts are associated with iSAC and should be considered in the decision? (Choose three) A. The codec has better quality with less bandwidth for sideband applicationsB. The codec will not be supported in TDM voice gatewaysC. The codec will adjust its bandwidth consumption to the network conditionsD. The codec will not be available for H.323 and MGCP devicesE. The codec will not support low complexityF. The codec will not be supported by SCCP configured on DSPFARMS Answer: ACE QUESTION 438Refer to the Exhibit. An agent initiated a video call but was establish as audio only. The support engineer collected and analysed the Cisco Unified CM traces.Which two options caused this problem? (Choose two) A. A hardware MTP was assigned to the callB. SIP Notify DTMF was requested and negotiatedC. MTP required was checked on the SIP TrunksD. Use Trusted Relay Point is set on one of the phoneE. MRGL assigned to phones with Trusted Relay Point Answer: DE QUESTION 439Refer to the exhibit. A collaboration engineer is troubleshooting a cluster that has been configured to use RSVP. The calls are being rejected and the caller receives a busy tone. What is the root cause of this problem? A. The RSVP Agents are only using an Ipv6 address.B. The IP Addressing mode preference for signalling is set to use System Default.C. The RSVP relationship between Main and Remote is set to MandatoryD. The IP Addressing Mode is set to use Ipv4 and Ipv6 Answer: A QUESTION 440Refer to the exhibit. Which three pieces of information can be derived from this sip message? (Choose Three) A. The call will have no audioB. Only OOB DTMF will be supportedC. G722 codec will be chosen D. The B2BUA uses IP 172.16.100.50E. This is a flow-around configurationF. The call will last only 30 minutes Answer: BDF

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