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The default codec does not matter if you have defined a hardware MTP in your Cisco Unified Communications Manager environment.D. To deploy a Cisco H.323 gatekeeper, you must configure MTP resources on the gatekeeper and only use G.711 between regions.Answer: BExplanation:The Cisco best practicse enforces G.729/24K to compress BW for regions. The hardware MTP only supports G.711 a-law and G.711 u-law. Also regions will need transcoders if multiple codecs are deployed, NOT hardware MTP. Talking to other friends of mine, it seems that there is a question on the exam now, where it asks for two answers, then, it would be B and C, but taking in consideration the below questions asks for one answer only, the most correct one is B.New QuestionWhen you use the Query wizard to configure the trace and log central feature to collect install logs, if you have servers in a cluster in a different time zone, which time is used?A. TLC adjusts the time change appropriately.B. TLC uses its local time for all systems.C. TLC queries for the time zone as part of configuration.D. TLC produces an error and must be run remotely.**Answer: A**New QuestionWhen a call is made from a video endpoint to a Cisco TelePresence EX90 that is registered to a Cisco VCS Control, which portion of the destination URI is the is the first match that is attempted?A. the full URI, including the domain portionB. the destination alias, without the domain portionC. the E.164 number that is assigned to the Cisco TelePresence EX90D. the directory number that is assigned to the Cisco TelePresence EX90 **Answer: B**New QuestionIn a distributed call processing network with locations-based CAC, calls are routed to and from intercluster trunks. Which trunk type is implemented in this network?A. intercluster trunk with gatekeeper controlB. intercluster trunk without gatekeeper controlC. SIP trunkD. h225 trunk**Answer: B**New QuestionRefer to the exhibit. The "DSCP for Video Calls" Cisco CallManager service parameter is set to 34. What is the correct DSCP value to use when configuring a class map in a Cisco IOS router? A. cs4B. efC. af23D. af41**Answer: D**New QuestionWhat is the default DSCP/PHB for TelePresence video conferencing packets in Cisco Unified Communications Manager?A. EF/46B. CS6/48C. AF41/34D. CS3/24E. CS4/32 **Answer: E**New QuestionHow is a SIP trunk in Cisco Unified Communications Manager configured for SIP precondition?A. The configuration is done by selecting a SIP precondition trunk for trunk type.B. The configuration is automatically selected when RSVP is enabled for the location assigned to the trunk.C. SIP precondition is configured by selecting E2E for RSVP over SIP on the default SIP profile assigned to the SIP trunk.D. SIP precondition is configured by configuring a new SIP profile and selecting E2E for RSVP over SIP. The new SIP profile must then be assigned to the SIP trunk.**Answer: D**New QuestionWhich statement about SIP precondition is most correct?A. When configuring SIP precondition, the SIP trunk must have access to an RSVP agent. B. When configuring SIP precondition, the IP phones must have access to an RSVP agent.C. When configuring SIP precondition, the IP phones and SIP trunk must have access to an RSVP agent.D. RSVP agents are only required for the IP phones. SIP trunks require RSVP agents only when fall back to local RSVP is configured.E. SIP trunk will always require RSVP agents regardless of what RSVP type is configured.**Answer: D**New QuestionRefer to the exhibit. Assume that NANP is being used and 9 is used for PSTN access code Long distance national calls are preceded with 1.How should the HQ Cisco Unified Communications Manager be configured for calls to 3XXX to be sent to the gatekeeper at 1 0 1 6 1 with PSTN backups? A. Configure a route pattern for 3XXX Assign this route pattern to a route list that points to two route groups. The first route group contains the H 225 trunk. The second route group contains the MGCP gateway with prefix digits 1 408555 for the outgoing called number.B. Configure a route pattern for 1#3XXX Assign this route pattern to a route list that points to a route group that lists the H 225 trunk as first choice and the MGCP gateway as a second choice.C. Configure a route pattern for 4085543XXX. Assign this route pattern to a route list that points to two route groups. The first route group contains the H 226 trunk. The second route group contains MGCP gateway.D. Configure a route pattern for 3XXX Assign this route pattern to a route list that points to two route groups. The first route group contains the H 225 trunk. The second route group contains MGCP gateway with prefix digits 91 408554 for the called number.**Answer: A**New QuestionRefer to the exhibit. IT shows an H.323 gateway configuration in a Cisco Unified

Communications Manager environment. An inbound PSTN call to this H.323 gateway fails to connect to IP phone extension 2001. The PSTN user hears a reorder tone. Debug isdn q931 on the H.323 gateway shows the correct called-party number as 5015552001. Which two configuration changes can correct this issue? (Choose two.)A.   Add port 1/0:23 to dial-peer voice 123 pots.B.   Ensure that the Significant Digits for inbound calls on the H.323 gateway configuration is 4.C.   Add a voice translation profile to truncate the number from 10 digits to 4 digits. Apply the voice translation profile to the Voice-port. The configuration field "Significant Digits for inbound calls" is left at default (All).D.   Add the command h323-gateway voip id on interface vlan120.E.   Change the destination-pattern on the dial-peer voice 23000 VoIP to 501501? and change the Significant Digits for inbound calls to 4.Answer: BEExplanation:Choose the number of significant digits to collect, from 0 to 32. Cisco Unified Communications Manager counts significant digits from the right (last digit) of the number that is called. !!!RECOMMEND!!!1.|2019 Latest 300-075 Exam Dumps (PDF & VCE) Instant Download:https://www.braindump2go.com/300-075.html2.|2019 Latest 300-075 Study Guide Video Instant Download: YouTube Video: YouTube.com/watch?v=S2wMYKmSgoc