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2016.07 Cisco Official: 400-051 Exam Questions New Updated! Braindump2go Offers Free 400-051 PDF & 400-051 VCE 454Q&As for Free Download Today! NEW QUESTION 21 - NEW QUESTION 30: 1.|2016.07 New Cisco 400-051 PDF & 400-051 VCE 454Q&As Dumps:http://www.braindump2go.com/400-051.html [100% Exam Pass Guaranteed!]2.|2016.07 Latest Cisco 400-051 Exam Questions PDF - Google Drive:https://drive.google.com/folderview?id=0B272WrTALRHcV0V4N2pvOUpFcUk&usp=sharing QUESTION 21Refer to the exhibit. A user is going through a series of dialing steps on a SIP Type B IP phone (for example, a Cisco 7975) to call an SCCP IP phone. Both phones are registered to the same Cisco Unified Communications Manager cluster. Assuming the calling SIP phone is associated with a SIP Dial Rule with a pattern value of 2001, which statement about the call setup process of this call is true? A.Each digit will arrive at Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event, and Cisco Unified Communications Manager will extend the call as soon as the collected digits match the extension of the SCCP IP phone, bypassing class of service configuration on both IP phones.B.Each digit will arrive at Cisco Unified Communications Manager in a SIP NOTIFY message as a KPML event. When the collected digits match the extension of the SCCP IP phone, Cisco Unified Communications Manager will extend the call only if the class of service configuration on both phones permits this action.C.As soon as the user selects the Dial softkey, the SIP IP phone will forward all digits to Cisco Unified Communications Manager in a SIP INVITE message. Cisco Unified Communications Manager will extend the call as soon as the collected digits match the extension of the SCCP IP phone, bypassing class of service configuration on both IP phones.D.As soon as the user selects the Dial softkey, the SIP IP phone will forward all digits to Cisco Unified Communications Manager in a SIP INVITE message. Cisco Unified Communications Manager will extend the call only if class of service configuration on both phones permits this action.E.The SIP IP phone will wait for the interdigit timer to expire, and then send all digits to Cisco Unified Communications Manager in a SIP INVITE message. Cisco Unified Communications Manager will extend the call as soon as the collected digits match the extension of the SCCP IP phone, bypassing class of service configuration on both IP phones. Answer: DExplanation:Cisco Type B SIP Phones offer functionality based SIP INVITE Message. Every key the end user presses triggers an individual SIP message. The first event is communicated with a SIP INVITE, but subsequent messages use SIP NOTIFY messages. The SIP NOTIFY messages send KPML events corresponding to any buttons or soft keys pressed by the user. Cisco Type B SIP IP Phones with SIP dial rules operate in the same manner as Cisco Type A phones with dial rules. QUESTION 22What does a comma accomplish when it is used in a SIP Dial Rule pattern that is associated with a Cisco 9971 IP Phone that is registered to Cisco Unified Communications Manager? A.It inserts a 500-millisecond pause between digits.B.It causes the phone to generate a secondary dial tone.C.It is a delimiter and has no significant dialing impact.D.It indicates a timeout value of 5000 milliseconds.E.It is an obsolete parameter and will be ignored. Answer: BExplanation:Comma is accepted in speed dial as delimiter and pause. -Comma used to delineate dial string, FAC, CMC, and post connect digits For post connect digits, commas insert a 2 second delay Commas may be duplicated to create longer delays QUESTION 23Which Call Admission Control mechanism is supported for the Cisco Extension Mobility Cross Cluster solution? A.Location CACB.RSVP CACC.H.323 gatekeeperD.intercluster Enhanced Location CACE.visiting cluster's LBM hub Answer: B Explanation:Configuring extension mobility cross cluster (EMCC) is nothing you should take lightly. EMCC requires a lot of configuration parameters including the exporting and importing of each neighbor cluster's X.509v3 digital certificates. EMCC is supported over SIP trunks only. Presence is another feature that's only supported over SIP trunks. If you want to be able to perform scalable Call Admission Control (CAC) in a distributed multi- cluster call processing model, you will need to point an H.225 or Gatekeeper controlled trunk to an H.323 Gatekeeper for CAC... but if you want to support presence and EMCC between clusters and maintain CAC. QUESTION 24Which two Cisco Unified Communications Manager SIP profile configuration parameters for a SIP intercluster trunk are mandatory to enable end-to-end RSVP SIP Preconditions between clusters? (Choose two.) A.Set the RSVP over SIP parameter to Local RSVP.B.Set the RSVP over SIP parameter to E2E.C.Set the SIP Rel1XX Options parameter to Disabled.D.Set the SIP Rel1XX Options parameter to Send PRACK If 1xx Contains SDP.E.Set the SIP Rel1XX Options parameter to Send PRACK for All 1xx Messages.F.Check the Fall Back to Local RSVP check box. Answer: BDExplanation:Each Unified Communications Manager cluster and Unified CME should have the same configuration information. For example, Application ID should be the same on each Unified Communications Manager cluster and Unified CME. RSVP Service parameters should be the same on each Unified Communications Manager cluster. QUESTION 25What is the number of directory URIs with which a Cisco Unified Communications Manager directory number can be associated? A.1B.up to 2C.up to 3D.up to 4E.up to 5 Answer: E Explanation:Cisco Unified Communications Manager supports dialing using directory URIs for call addressing. Directory URIs look like email addresses and follow the username@host format where the host portion is an IPv4 address or a fully qualified domain name. A directory URI is a uniform resource identifier, a string of characters that can be used to identify a directory number. If that directory number is assigned to a phone, Cisco Unified Communications Manager can route calls to that phone using the directory URI. URI dialing is available for SIP and SCCP endpoints that support directory URIs. QUESTION 26Which Cisco Unified Communications Manager partition will be associated with a directory URI that is configured for an end user with a primary extension? A.nullB.noneC.directory URID.defaultE.any partition that the Cisco Unified Communications Manager administrator desires Answer: CExplanation:Cisco Unified Communications Manager supports dialing using directory URIs for call addressing. Directory URIs look like email addresses and follow the username@host format where the host portion is an IPv4 address or a fully qualified domain name. A directory URI is a uniform resource identifier, a string of characters that can be used to identify a directory number. If that directory number is assigned to a phone, Cisco Unified Communications Manager can route calls to that phone using the directory URI. URI dialing is available for SIP and SCCP endpoints that support directory URIs. QUESTION 27 Which Call Control Discovery function allows the local Cisco Unified Communications Manager to listen for advertisements from remote call-control entities that use the SAF network? A.CCD advertising serviceB.CCD requesting serviceC.SAF forwarderD.SAF enabled trunksE.CCD registration service Answer: BExplanation:SAF and CCD will allow large distributed multi-cluster deployments to have the directory number (DN) ranges of each call routing element advertised dynamically over SAF. Cisco routers act as SAF Forwarders (SAFF), while the call routing elements (e.g. CUCM) act as clients that register with the routers to advertise their DN ranges and listen to the advertisements of other routers. QUESTION 28Which codec complexity mode, when deployed on Cisco IOS routers with DSPs using the C5510 chipset, supports the most G.711 calls per DSP? A.LowB.MediumC.HighD.Secure E.Flex Answer: EExplanation:The flex parameter allows the complexity to automatically adjust to either medium or high complexity depending on the needs of a call. For example, if a call uses the G.711 codec, the C5510 chipset automatically adjusts to the medium-complexity mode. However, if the call uses G.729, the C5510 chipset uses the high complexity mode QUESTION 29When DSP oversubscription occurs on a Cisco IOS router using DSP modules that are based on the C5510 chipset, what will happen when an analog phone connected to a FXS port goes off-hook? A.A fast busy tone will be played.B.A slow busy tone will be played.C.A network busy tone will be played.D.A dial tone will be played, but digits will not be processed.E.No tone will be played. Answer: E Explanation:When DSP oversubscription occurs for both analog ports and digital ports, except PRI and BRI. FXO signaling and application controlled endpoints are not supported. This feature does not apply to insufficient DSP credits due to mid-call codec changes (while a call is already established). QUESTION 30Refer to the exhibit. How many simultaneous outbound calls are possible with this Cisco Unified Communications Manager Express configuration on these two phones? A.6B.7C.8D.9E.11 Answer: CExplanation:Ephone is configured as octo line so maximum call number is 8 and it will be devided between lines.