[2017 New Dumps For Exam 400-051 With New Updated Exam Questions (21-40)

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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 22 What 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: BExplanation: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 27Which 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. SecureE. 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: EExplanation: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 30 Refer 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. QUESTION 31Refer to the exhibit. Ephone 1 has three active calls. The first two calls were inbound calls, which the user put on hold to place a third call outbound. What will happen on ephone 1 when a fourth call arrives for extension 2001? A. The fourth call will be delivered to ephone 1 because it only received two inbound calls, one call less than the busy-trigger-per-button setting.B. The fourth call will be delivered to ephone 1 because the huntstop channel setting is not yet saturated.C. The fourth call will be delivered to ephone 1 because it can handle up to five calls on each button.D. The fourth call will be held temporarily by the IOS Software until ephone 1 disconnects one of the active calls. E. The fourth call will not be delivered and the caller will hear a user busy tone. Answer: EExplanation:Because on line maximum 4 calls can be placed when user put the call on hold is consume a channel and reach the maximum number of calls on line. QUESTION 32Refer to the exhibit. How many simultaneous inbound calls can be handled by these two IP phones? A. 2B. 4C. 6D. 9E. 10 Answer: AExplanation:The line is configured as shared line so it will support maximum two calls at a time. QUESTION 33Refer to the exhibit. IP phone 1 has the MAC address 11111.1111.1111, while IP phone 2 has the MAC address 2222.2222.2222. The first two incoming calls were answered by IP phone 1, while the third incoming call was answered by IP phone 2. What will happen to the fourth incoming call? A. Both phones will ring, but only IP phone 2 can answer the call.B. Both phones will ring and either phone can answer the call.C. Only IP phone 2 will ring and can answer the call.D. Neither phone will ring and the call will be forwarded to 2100.E. Neither phone will ring and the call will be forwarded to 2200. Answer: BExplanation:In shared line configuration phone share the same line so it is possible for any phone to answer the call. QUESTION 34Which statement describes the question mark wildcard character in a SIP trigger that is configured on Cisco Unity Express? A. It matches any single digit in the range 0 through 9.B. It matches one or more digits in the range 0 through 9.C. It matches zero or more occurrences of the preceding digit or wildcard value.D. It matches one or more occurrences of the preceding digit or wildcard value.E. It matches any single digit in the range 0 through 9, when used within square brackets. Answer: C QUESTION

35Assume the IP address of Cisco Unity Express is 10.1.1.1. Which URL provides Cisco Unity Express end users with a GUI interface to access and manage their messages and mailbox settings? A. http://10.1.1.1/Web/Common/Login.doB. http://10.1.1.1/ciscopcaC. http://10.1.1.1/userD. http://10.1.1.1/inboxE. http://10.1.1.1/ Answer: CExplanation:For user access cisco unity has predefined url and it is http://10.1.1.1/user QUESTION 36Refer to the exhibit. Your customer sent you this debug output, captured on a Cisco IOS router (router A), to troubleshoot a problem where all H.323 calls that originate from another Cisco IOS router (router B) are being dropped almost immediately after arriving at router A. What is the reason for these disconnected calls? A. Calls were unsuccessful because of internal, memory-related problems on router A.B. Calls were rejected because the called number was denied on a configured class of restriction list on router A.C. Calls were rejected because the VoIP dial peer 1002 was not operational.D. Calls were unsuccessful because the router B IP address was not found in the trusted source IP address list on router A.E. Calls were rejected by router A because it received an admission reject from its gatekeeper because of toll fraud suspicion. Answer: DExplanation:Trusted source IP address list on router is a list which secure the connectivity of router if it is enabled then we need to give the trusted ebtry for any route to reach. QUESTION 37Which type of mailbox on Cisco Unity Express can play a user greeting and disconnect the call, but cannot take or send messages? A. PIN-less mailboxB. announcement-only mailboxC. general delivery mailboxD. call-handling mailboxE. personal mailbox Answer: BExplanation: Announcement-only mailbox are set for those user who only want the caller to listen the announcement and leave his message according to the announcement. QUESTION 38Refer to the exhibit. Assume the B-ACD configuration on a Cisco IOS Cisco Unified Communications Manager Express router is operational. What will happen to a call in queue that was not answered by any member of the hunt group after the maximum amount of time allowed in the call queue expires? A. The call will be forwarded to extension 2120.B. The call will be forwarded to extension 2220.C. The call will be forwarded to extension 2003.D. The call will be disconnected with user busy.E. The call will be forwarded to 2100. Answer: BExplanation:As we can see in the configuration 2220 is configured as voice mail forwarding extension so the call will forward to voice mail. QUESTION 39Refer to the exhibit. Assume the B-ACD configuration on a Cisco IOS Cisco Unified Communications Manager Express router is operational. What will happen to a new call that enters the call queue when there are already two calls in queue? A. The call will be forwarded to extension 2120.B. The call will be forwarded to extension 2220.C. The call will be forwarded to extension 2003. D. The call will be disconnected with user busy.E. The call will be forwarded to 2100. Answer: CExplanation:That is because queue over flow is forwaded to 2003 and maximum number of calls in queuq is configured as two. QUESTION 40When multiple greetings are enabled on Cisco Unity Express, which greeting will take the highest precedence? A. standardB. meetingC. busy D. closedE. internal Answer: BExplanation:Meeting greeting has highest priority because it is set by user when he don't want to take the call and notice the caller he is online. Lead2pass Cisco 400-051 exam dumps are audited by our certified subject matter experts and published authors for development. Lead2pass Cisco 400-051 exam dumps are one of the highest quality Cisco 400-051 Q&As in the world. It covers nearly 96% real questions and answers, including the entire testing scope. Lead2pass guarantees you pass Cisco 400-051 exam at first attempt. 400-051 new questions on Google Drive: https://drive.google.com/open?id=0B3Syig5i8gpDQ1ZudWVBRHk3bDQ 2017 Cisco 400-051 exam dumps (All 542 Q&As) from Lead2pass: http://www.lead2pass.com/400-051.html [100% Exam Pass Guaranteed]