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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 41 - NEW QUESTION 50: 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 41Which two statements are requirements regarding hunt group options for B-ACD implementation on Cisco Unified Communications Manager Express routers? (Choose two.) A. The ephone hunt group is mandatory.B. Either the ephone hunt group or the voice hunt group is acceptable.C. Hunt group members must be SCCP IP phones.D. Hunt group members can include both SCCP or SIP IP phones.E. Hunt group members must be SIP IP phones.F. The member hunting mechanism must be set to sequential. Answer: ACExplanation:The ephone hunt group is mandatory, and while ephone hunt groups only support Cisco Unified SCCP IP phones, a voice hunt group supports either a Cisco Unified SCCP IP phone or a Cisco Unified SIP IP phone http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/command/reference/cme_v1ht.html QUESTION 42Which call hunt mechanism is only supported by the voice hunt group in a Cisco Unified Communications Manager Express router? A. sequential B. peerC. longest idleD. parallelE. overlay Answer: DExplanation:Parallel Hunt-Group, allows a user to dial a pilot number that rings 2-10 different extensions simultaneously. The first extension to answer gets connected to the caller while all other extensions will stop ringing. A timeout value can be set whereas if none of the extensions answer before the timer expires, all the extensions will stop ringing and one final destination number will ring indefinitely instead. The final number could be another voice hunt-group pilot number or mailbox The following features are supported for Voice Hunt-Group:Calls can be forwarded to Voice Hunt-GroupCalls can be transferred to Voice Hunt-GroupMember of Voice Hunt-Group can be SCCP, ds0-group, pri-group, FXS or SIP phone/trunkMax member of Voice Hunt-Group will be 32 QUESTION 43Which Cisco Unified Communications Manager Express ephone button configuration separator enables overflow lines when the primary line for an overlay button is occupied by an active call? A. oB. cC. wD. xE. : Answer: DExplanation:x expansion/overflow, define additional expansion lines that are used when the primary line for an overlay button is occupied by an active call. QUESTION 44Which two statements describe characteristics of Cisco Unified Border Element high availability, prior to Cisco IOS release 15.2.3T, using a box-to-box redundancy configuration? (Choose two.) A. It leverages HSRP for router redundancy and GLBP for load sharing between a pair of routers.B. Cisco Unified Border Element session information is check-pointed across the active and standby router pair.C. It supports media and signal preservation when a switchover occurs.D. Only media streams are preserved when a switchover occurs.E. It can leverage either HSRP or VRRP for router redundancy.F. The SIP media signal must be bound to the loopback interface. Answer: BD QUESTION 45Refer to the exhibit. From this NFAS-enabled T1 PRI configuration on a Cisco IOS router, how many bearer channels are available to carry voice traffic? A. 91B. 92C. 93D. 94E. 95 Answer: DExplanation:In NFAS one channel is used for signaling so according to this we will have 94 channel for with bearer capability. QUESTION 46Refer to the exhibit. Assuming this NFAS-enabled T1 PRI configuration on a Cisco IOS router is fully functional, what will the controller T1 1/1 D-channel status be in the output of the show isdn status command? A. MULTIPLE_FRAME_ESTABLISHEDB. TEI_ASSIGNEDC. AWAITING_ESTABLISHMENTD. STANDBYE. INITIALIZED Answer: BExplanation: TEI_ASSIGNED, which indicates that the PRI does not exchange Layer 2 frames with the switch. Use the show controller t1 x command to first check the controller t1 circuit, and verify whether it is clean (that is, it has no errors) before you troubleshoot ISDN Layer 2 problem with the debug isdn q921. QUESTION 47Refer to the exhibit. In an effort to troubleshoot a caller ID delivery problem, a customer emailed you the voice port configuration on a Cisco IOS router. Which type of voice port is it? A. FXSB. E&MC. BRID. FXOE. DID Answer: D QUESTION 48The iLBC codec operates at 38 bytes per sample per 20-millisecond interval. What is its codec bit rate in kilobits per second? A. 6.3B. 13.3C. 15.2D. 16E. 24 Answer: CExplanation:The internet Low Bit Rate Codec (iLBC) is designed for narrow band speech and results in a payload bit rate of 13.33 kbits per second for 30-millisecond (ms) frames and 15.20 kbits per second for 20 ms frames. When the codec operates at block lengths of 20 ms, it produces 304 bits per block, which is packetized as defined in RFC 3952. Similarly, for block lengths of 30 ms it produces 400 bits per block, which is packetized as defined in RFC 3952.The iLBC has built-in error correction functionality to provide better performance in networks with higher packet loss QUESTION 49Assume 6 bytes for the Layer 2 header, 1 byte for the end-of-frame flag, and a 40-millisecond voice payload, how much bandwidth should be allocated to the strict priority queue for five VoIP calls that use a G.729 codec over a multilink PPP link? A. 87 kb/sB. 134 kb/sC. 102.6 kb/sD. 77.6 kb/sE. 71.3 kb/s Answer: A

Explanation:Voice payloads are encapsulated by RTP, then by UDP, then by IP. A Layer 2 header of the correct format is applied; the type obviously depends on the link technology in use by each router interface: A single voice call generates two one-way RTP/UDP/IP packet streams. UDP provides multiplexing and checksum capability; RTP provides payload identification, timestamps, and sequence numbering. QUESTION 50Assume 20 bytes of voice payload, 6 bytes for the Layer 2 header, 1 byte for the end-of-frame flag, and the IP, UDP, and RTP headers are compressed to 2 bytes, how much bandwidth should be allocated to the strict priority queue for six VoIP calls that use a G.729 codec over a multilink PPP link with cRTP enabled? A. 80.4 kb/sB. 91.2 kb/sC. 78.4 kb/sD. 69.6 kb/sE. 62.4 kb/s Answer: DExplanation:Voice payloads are encapsulated by RTP, then by UDP, then by IP. A Layer 2 header of the correct format is applied; the type obviously depends on the link technology in use by each router interface: A single voice call generates two one-way RTP/UDP/IP packet streams. UDP provides multiplexing and checksum capability; RTP provides payload identification, timestamps, and sequence numbering.