# [2017 New 400-051 New Questions Free Download In Lead2pass (41-60)

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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 phonehttp://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. sequentialB. 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 44 Which 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: D Explanation: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: AExplanation: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. QUESTION 51To which QoS tool category does compressed RTP belong? A. classificationB. markingC. link efficiencyD. queuingE. prioritization Answer: CExplanation:LLQ is a feature that provides a strict PQ to CBWFQ. LLQ enables a single strict PQ within CBWFQ at the class level. With LLQ, delay-sensitive data (in the PQ) is dequeued and sent first. In a VoIP with LLQ implementation, voice traffic is placed in the strict PQ. QUESTION 52How are queues serviced in Cisco IOS routers with the CBWFQ algorithm? A. first-in, first-outB. weighted round robin based on assigned bandwidthC. strict priority based on assigned priorityD. last-in, first-outE. weighted round robin based on assigned priority Answer: BExplanation:Class Based Weighted Fair queuing is an advanced form of WFQ that supports user defined traffic classes I.e. one can define traffic classes based on match criteria like protocols, access control lists (ACLs), and input interfaces. A flow satisfying the match criteria for a class contributes the traffic for that particular defined class. A queue is allocated for each class, and the traffic belonging to that class is directed to the queue for that class. QUESTION 53In Cisco IOS routers that use low latency queuing, which algorithm is used to presort traffic going into the default queue? A. first-in, first-outB. last-in, first-outC. weighted round robinD. fair queuingE. random processing Answer: DExplanation:WFQ is a flow-based queuing algorithm used in Quality of Service (QoS) that does two things simultaneously: It schedules interactive traffic to the front of the queue to reduce response time, and it fairly shares the remaining bandwidth between high bandwidth flows. A stream of packets within a single session of a single application is known as flow or converstion. WFQ is a flow-based method that sends packets over the network and ensures packet transmission efficiency which is critical to the interactive traffic. This method automatically stabilizes network congestion between individual packet transmission flows. QUESTION 54Which statement describes the Cisco best practice recommendation about priority queue bandwidth allocation in relationship to the total link bandwidth when multiple strict priority LLQs are configured on the same router interface? A. Each LLQ should be limited to one-third of the link bandwidth capacity.B. The sum of all LLQs should be limited to two-thirds of the link bandwidth capacity.C. The sum of all LLQs should be limited to one-half of the link bandwidth capacity. D. The sum of all LLQs should be limited to one-third of the link bandwidth capacity.E. Cisco does not recommend more than one strict priority LLQ per interface. Answer: DExplanation:Cisco Technical Marketing testing has shown a significant decrease in data application response times when Real-Time traffic exceeds one-third of a link's bandwidth capacity. Cisco IOS Software allows the abstraction (and, thus, configuration) of multiple LLQs. Extensive testing and production-network customer deployments have shown that limiting the sum of all LLQs to 33 percent is a conservative and safe design ratio for merging real-time applications with data applications. QUESTION 55To which Cisco enterprise medianet application class does Cisco TelePresence belong? A. VoIP TelephonyB. Real-time InteractiveC. Multimedia ConferencingD. Broadcast VideoE. Low Latency Data Answer: B Explanation:Telepresence is used for video conferencing which can be done in Real-time so it is Real-time Interactive. QUESTION 56Refer to the exhibit. Assume that the serial interface link bandwidth is full T1.What is the maximum amount of bandwidth allowed for priority queuing of RTP packets with a DSCP value of EF? A. 33% of 1.544 Mb/sB. 5% of 1.544 Mb/sC. 38% of 1.544 Mb/sD. 62% of 1.544 Mb/sE. 0% of 1.544 Mb/s Answer: E QUESTION 57Which statement describes the key security service that is provided by the TLS Proxy function on a Cisco ASA appliance? A. It provides interworking to ensure that external IP phone traffic is encrypted, even if the rest of the system is unencrypted.B. It only applies to encrypted voice calls where both parties utilize encryption.C. It manipulates the call signaling to ensure that all media is routed via the adaptive security appliance. D. It enables internal phones to communicate with external phones without encryption.E. It protects Cisco Unified Communications Manager from rogue soft clients and attackers on the data VLAN. Answer: BExplanation:TLS Proxy is typically deployed in front of Cisco Unified Communications Manager and other unified communications application servers that utilize media encryption. TLS Proxy is not designed to provide remote-access encryption services for remote phones or client endpoints. Other solutions such as Cisco ASA Phone Proxy or IP Security/Secure Sockets Layer (IPsec/SSL) VPN services are more appropriate.TLS Proxy is not designed to provide a secure campus soft phone solution where the requirement is to provide secure data to phone VLAN traversal or for proxying connections to Cisco Unified Communications Manager. QUESTION 58Which two statements describe security services that are provided by the Phone Proxy function on a Cisco ASA appliance? (Choose two.) A. It is supported only on phones that use SCCP.B. It is supported on an adaptive security appliance that runs in transparent mode.C.

It provides interworking to ensure that the external IP phone traffic is encrypted, as long as the Cisco Unified Communications Manager cluster runs in secure mode.D. It provides a proxy of phone signaling, with optional use of NAT, to hide the Cisco Unified Communications Manager IP address from the public Internet.E. It proxies phone media so that internal phones are not directly exposed to the Internet.F. It supports IP phones that send phone proxy traffic through a VPN tunnel. Answer: DE Explanation:TLS Proxy is typically deployed in front of Cisco Unified Communications Manager and other unified communications application servers that utilize media encryption. TLS Proxy is not designed to provide remote-access encryption services for remote phones or client endpoints. Other solutions such as Cisco ASA Phone Proxy or IP Security/Secure Sockets Layer (IPsec/SSL) VPN services are more appropriate. TLS Proxy is not designed to provide a secure campus soft phone solution where the requirement is to provide secure data to phone VLAN traversal or for proxying connections to Cisco Unified Communications Manager. QUESTION 59Which entity signs a Cisco IP phone LSC? A. Godaddy.com Enrollment ServerB. Manufacturer Certificate AuthorityC. Registration AuthorityD. Certificate Authority Proxy FunctionE. Cisco Certificate Authority Answer: DExplanation:By default, LSC certificates are not installed on Cisco IP phones. Cisco IP phones that are required to use LSC certificates must be provisioned to allow TLS transactions before deployment in the field. LSC certificates can be provisioned to the Cisco IP phones through the Certificate Authority Proxy Function (CAPF) process. This process is completed using TLS and USB tokens coupled with the CTL client. Moreover, the Cisco ASA Phone Proxy feature can serve LSC certificates to the Cisco IP phones. Cisco IP phones will only work with the Cisco ASA Phone Proxy and will not establish secure connectivity with the Cisco Unified Communications Manager. QUESTION 60A Cisco Unity Connection administrator receives a name change request from a voice-mail user, whose Cisco Unity Connection user account was imported from Cisco Unified Communications Manager. What should the administrator do to execute this change? A. Change the user data in the Cisco Unity Connection administration page, then use the Synch User page in Cisco Unity Connection administration to push the change to Cisco Unified Communications Manager.B. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unity Connection administration to pull the changes from Cisco Unified CM.C. Change the user data in the Cisco Unified Communications Manager administration page, then use the Synch User page in Cisco Unified CM administration to push the change to Cisco Unity Connection.D. Change the user profile from Imported to Local on Cisco Unity Connection Administration, then edit the data locally on Cisco Unity Connection.E. Change the user data in Cisco Unity Connection and Cisco Unified Communications Manager separately. Answer: BExplanation:As we can see user are getting synch from call manager so we first have to change the details of user on call manager so that user will synch the changes from call manager. Lead2pass is now offering Lead2pass 400-051 PDF dumps with 100% passing guarantee. Use Lead2pass 400-051 PDF and pass your exam easily. Download Cisco 400-051 exam dumps and prepare for exam. 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]