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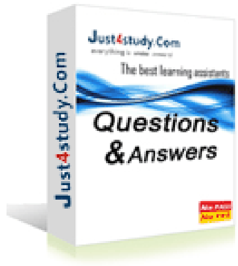
Cisco CCIE

350-030: CCIE Voice Written
Practice Exam: 350-030 Exams
Exam Number/Code: 350-030
Exam Name: CCIE Voice Written
Questions and Answers: 312 Q&As
(CCIE)



Exam : *350-030*

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Exam : Cisco 350-030
Title : CCIE Voice Exam

1. Two divisions in your company need to exchange Unity voice messages using VPIM. Calls from Division A to Division B are made using a site code of "919" followed by the recipient's 4 digit extension. The primary extension in Unity is the user's four digit extension. Which of the following configurations on Division A's Unity server will allow messages to be forwarded between Unity systems using the same seven digit dialing that is used to place direct calls? (Choose 3)
A. Add the seven digit number as an alternate extension to each VPIM subscriber
B. Configure the Extension on the VPIM subscriber to "919" plus the recipient's four digit extension
C. Configure the Remote Mailbox Number on the VPIM subscriber to the recipient's four digit extension
D. Configure the Remote Phone Prefix to "919"
E. Configure the Dial ID to "919"
Answer: BCE

2. An IP phone is connected to a Cisco inline power switch Port. Switch is running IOS image on it. The switch port is acting as a trunk and is running both Voice and Data VLAN configuration on it. We would like the IP Phone connected to switch port in voice VLAN to set layer 2 priority of all the packets coming from PC to default 0. Which IOS CLI in Interface Port configuration on Inline power switch can help us achieve our objective?
A. switchport access priority extend cos 0
B. switchport priority extend cos 0
C. switchport trunk priority cos 0
D. switchport mode access priority extend cos 0
E. mls qos priority extend cos 0
F. switchport access extend cos 0
Answer: B

3. Which of the following statements outline the way to implement a non-standard softkey template?
A. Select a softkey template; copy the template and rename it; insert it; modify the template and update the changes.
B. Select a softkey template; rename it; modify it and update the changes.
C. Select the default softkey template, rename it; insert it; modify it and update the changes.
D. Select add softkey template, name it; update it; modify the template and update the changes.
Answer: A

4. The IOS command "call rsvp-sync resv-timer 10" is used to set the timer on the:
A. Originating VoIP gateway for completing RSVP reservation setups within 10 seconds
B. Originating and terminating VoIP gateway for completing RSVP reservation setups within 10 seconds
C. Terminating VoIP gateway for completing RSVP reservation setups within 10 seconds
D. VoIP gatekeeper for completing RSVP reservation setups within 10 seconds
Answer: C

5. Which of the following are NOT true statements about Certificate Trust List (CTL) File? (Choose 2)
A. It is a list of devices and credentials that a phone should trust on the network.
B. The CTL file is signed by administrator workstation password.
C. It contains identity, public key and role information.
D. Phones need to trust all entries in the CTL file which could be CCM, TFTP, CAPF, etc.
E. The CTL file is loaded to the phone each time when authentication is required.
F. The CTL is created by CTL Client on administrator workstation.
Answer: BE

6. Which 3 functions are NOT performed by a Route Pattern? (Choose 3)
A. Points to the actual IP phone
B. Matches dialed number for external calls
C. Performs digit manipulation
D. Points to a route list for routing
E. Chooses path for call routing
F. Points to prioritized route groups
Answer: AEF

7. What occurs if the system clocks are not synchronized between the sender and receiver of an RTP stream?
A. Packets can be placed in sequence but jitter cannot be compensated for.
B. Packets cannot be reordered for sequence and jitter cannot be compensated for.
C. Jitter can be compensated for, but packets cannot be reordered if they arrive out of sequence.
D. Packets may be reordered and jitter may be compensated for as the timestamp is not related to the system time.
E. When the RTP stream is opened, the sender and receiver synchronize their clocks before the stream commences so that packet sequencing and dejitter will function correctly.
Answer: D

8. There are 2 remote sites and one main site. Each site has a CME router with many IP phones in an IPT deployment. The Network Administrator wants to provide all of the phones voicemail access using CUE. Which way can CUE be deployed?
A. Cisco Unity Express and the CME gateway at each site must be collocated in the same router chassis providing voicemail access to local IP phones registered to local CME.
B. One Cisco Unity Express can be used at the main site with CME router providing voicemail access to all the 3 sites.
C. One Cisco Unity Express with CME can be used at the main site to provide voicemail access to the IP phones at the main site. Another Cisco Unity Express with CME can be used at one of the remote sites to provide voicemail access for all of the IP phones at the two remote sites.
D. Cisco Unity Express and the CME gateway at each site may NOT be collocated in the same router chassis providing a voicemail access to local IP phones registered to local CME.
Answer: A

9. Why can't TCP be used for transferring audio and video over UDP? (Choose 5)
A. TCP does not have a mechanism for sufficiently long buffering and adequate average throughput.
B. Reliable transmission is inappropriate for delay-sensitive data such as real-time audio and video.
C. TCP cannot support multicast.
D. The TCP congestion control mechanisms decreases the congestion window when packet losses are detected ("slow start").
E. TCP headers are larger than a UDP header.
F. TCP does not contain the necessary timestamp and encoding information needed by the receiving application.
Answer: BCDEF

10. Which type of media resources would be required for a single site call processing model?

A. MTP
B. Locations
C. Regions
D. Transcoders
Answer: A

11. When configuring IP Manager Assistant (IPMA) in a shared line mode, how are the manager and assistant Directory Numbers (DN) configured?
A. The manager and assistant both share the same directory number (DN).
B. The manager and assistant have separate directory numbers (DN), but share an IPMA directory number.
C. The manager and assistant each have separate directory numbers (DN).
D. The manager and assistant share a directory number (DN) and an IP Manager Assistant (IPMA) directory number.
Answer: A

12. A CallManager Group can provide which two features to your call processing system? (Choose 2)
A. Support for SRST in remote offices
B. Enables you to distribute the control of devices across multiple Cisco CallManagers
C. Enables you to distribute voice mail support across multiple Unity servers
D. Support for redundancy by enabling you to designate a primary and backup Cisco CallManagers for each group
E. Support for control of IPMA across primary and backup Cisco CallManagers for each group
Answer: BD

13. Which one of the following does NOT state Multicast Technologies Advantages?
A. Enhanced Efficiency: controls network traffic and reduces server and CPU loads
B. Optimized Performance: eliminates traffic redundancy
C. Distributed Applications: makes multipoint applications possible
D. Bandwidth-conserving technology that reduces traffic by simultaneously delivering a single stream of information to thousands of corporate recipients and homes
E. Prevent Denial of service (DoS) attacks in the networks
Answer: E

14. Which 2 are NOT functions performed by Cisco Media Streaming App Service?
A. Provides SCCP stack for 4 software devices: ANN, CFB, MOH, and MTP
B. Supports DB change notification processing
C. Converts new MOH source files to separate WAV files for MOH codecs
D. Provides SDI trace, event logs, and Perfmon counters
E. Adjusts volume levels of MOH source files
F. Provides audio data from WAV files: ANN, MOH
Answer: CE

15. Calculate the percentage of overall bandwidth saved (at Layer 3) by cRTP for a G.729 VoIP call packetized at 50 pps.
A. Approximately 60%
B. Approximately 50%
C. Approximately 40%
D. Approximately 30%
E. Approximately 20%
Answer: A

16. Which method could be used to determine if there is a JTAPI memory leak in a CallManager server?
A. Look at the physical memory available of the server.
B. Review all CCM User logs
C. Check for changes to IP phone settings, like ring settings reverting to default values

D. Determine if dialing the voice mail pilot number fails to connect to voice mail
E. Check for an increasing number of fast busys when dialing to the PSTN
Answer: A

17. Which statement does NOT describe dialing domain functionality in Cisco Unity?
A. Dialing domains are multiple Unity servers that are handling subscribers that are on a single switch or networked switch
B. All users in the dialing domain should be able to pick up their phones and dial each other directly without having to dial trunk access codes or use outside lines.
C. Dialing domains are also necessary if the Unity servers involved don't have overlapping dial plans.
D. Dialing domain IDs are stored on the primary location object. Multiple primary location objects with the same value for this ID make up a dialing domain.
E. All delivery locations get the same dialing domain ID as the primary location of the box they are created on.
Answer: C

18. A company has a headquarters with a centralized CallManager and 5 remote offices. All the remote offices have extensions in the range of 1000-1150. To allow inter-office calls each office has been assigned a 3 digit site code. To call between sites, users will dial an access code followed by the 3 digit site code and the extension. Which of the following describes how these inter-office access codes should be configured?
A. A translation pattern is created for each office and placed in a partition available to all phones. The translation pattern strips the access code and site code and is assigned a Calling Search Space that includes only the phones located in the office.
B. A translation pattern is created for each office and is placed in a partition available to the phones at that office. The translation pattern strips the access code and site code and is assigned to Calling Search Space that includes all local phones.
C. A route pattern is created for each office and placed in a partition available to all phones. The route pattern strips the access code and the site code and routes the call to the remote office's gateway.
D. A route pattern is created for each office and placed in a partition available to phones at that office. The route pattern strips the access code and site code and routes the call to the remote office's gateway.
Answer: B

19. When comparing SIP, H.323, and MGCP, and SCCP in a VoIP deployment, which Protocol will satisfy the following requirements:
Requirement 1: It has a mechanism for a centralized dial-plan
Requirement 2: The endpoints are considered to be unintelligent
Requirement 3: The protocol must be text-based
A. SIP
B. H.323
C. MGCP
D. SCCP
Answer: C

20. Which port(s) must be opened on an IOS firewall to allow successful MGCP (Media Gateway Control Protocol) message exchanges between a CallManager and an IOS MGCP PRI gateway?
A. TCP 2000 and TCP 2002
B. TCP 2427 and UDP 2428
C. UDP 2427
D. UDP 2427 and UDP 2428
E. UDP 2427 and TCP 2428
Answer: E

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