Cisco

CCNA Voice

640-460 IIUC 640-460 IIUC 640-460 IIUC 640-460 IIUC

PRINTABLES

PRINTABLE PRACTICE QUESTIONS

QUESTIONS, ANSWERS, AND DETAILED EXPLANATIONS IN AN EASY-TO-USE PRINTABLE FORMAT

LearnSmart™

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CCNA Voice (640-460 IIUC) Printables

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Chapter 1
Components of the Cisco Unified Communications Architecture

1. All CME phone features are built on top of which two foundation elements?Choose two:

❏ A. The Cisco IOS router engine
❏ B. Survivable Remote Site Telephony (SRST)
❏ C. H.323
❏ D. VoIP Infrastructure

Find the Answer p. 47

2. What is the Cisco proprietary protocol used to control local IP phones?Select the best answer.

❍ A. SIP
❍ B. H.323
❍ C. SCCP
❍ D. VoIP

Find the Answer p. 47

3. The CME acts as a gateway for what two VoIP areas?Select the best answer.

❏ A. PSTN
❏ B. IP phones
❏ C. Frame Relay
❏ D. FXO
❏ E. Firewall

Find the Answer p. 47

4. Which layer of the OSI model deals with the connectionless VoIP protocol?Select the best answer.

❍ A. Layer 3
❍ B. Layer 4
❍ C. Layer 1
❍ D. Layer 6

Find the Answer p. 47

5. What are the three voice packet technologies?Choose three:

❑ A. SIP
❑ B. Voice over ATM
❑ C. SCCP
❑ D. Voice over Frame Relay
❑ E. Voice over IP

Find the Answer p. 47

6. In regards to voice solutions, which solution operates at OSI layer 3?Select the best answer.

❍ A. Voice over ATM
❍ B. Voice over IP
❍ C. Voice over Frame Relay
❍ D. G.711

Find the Answer p. 47

7. What is the major difference between client/server and direct peer-to-peer VoIP signaling protocols?Select the best answer.

- ❍ A. A client/server setup can make calls without a call agent
- ❍ B. Peer-to-Peer protocol setup allows the client to speak to an agent to place calls
- ❍ C. Client/server end stations must use a call agent for calls to be made.
- ❍ D. Peer-to-Peer protocols can use either call agents or contact a VoIP end station directly.

Find the Answer p. 47

8. Which deployment model would you choose if you want voice communications from each site to traverse the IP WAN?Select the best answer.

- ❍ A. Partial-mesh processing model
- ❍ B. Single-site processing model
- ❍ C. Star processing model
- ❍ D. Multisite processing model

Find the Answer p. 47

9. What types of communication devices can the Cisco Unified Communications Express handle?Choose three:

- ❑ A. Software phones
- ❑ B. Video devices
- ❑ C. Instant Messaging
- ❑ D. Hardware Phones
- ❑ E. Cellular Phones

Find the Answer p. 47

10. What type of QOS is not supported on the Cisco Unified IPT network solution?Select the best answer.

- ❍ A. Traffic shaping
- ❍ B. Compressed Real-Time Transport Protocol (cRTP)
- ❍ C. Low-latency queuing
- ❍ D. Beaconing

Find the Answer p. 47

11. What are the two main functions of the Cisco Unity Express module?Choose two:

- ❑ A. Voice mail
- ❑ B. Auto Attendant
- ❑ C. Call Processing
- ❑ D. Web services

Find the Answer p. 47

12. What is not a benefit of using the distributed call processing design model?Select the best answer.

- ❍ A. PSTN call savings when using the IP WAN for calls between sites
- ❍ B. Scalability
- ❍ C. Increased utilization of WAN bandwidth
- ❍ D. Increased number of PSTN lines at each site

Find the Answer p. 47

13. What is the maximum number of phones Cisco recommends for the Cisco Unified CallManager Express?Select the best answer.

- ❍ A. Up to 30,000 phones
- ❍ B. Up to 240 phones
- ❍ C. Up to 50 phones
- ❍ D. Up to 250 phones

Find the Answer p. 47

14. What Cisco Unified Communications application provides integrated voice/video and web collaboration?Select the best answer.

- ❍ A. Cisco Unified MeetingPlace
- ❍ B. Cisco TAC
- ❍ C. Cisco IP Interactive Voice Response
- ❍ D. Cisco IP Communicator

Find the Answer p. 47

15. Which of the following is NOT a reason to choose Cisco Unified Communications Express solutions at remote sites over a centralized Unified Communications solution?Select the best answer.

- ❍ A. If WAN links cannot provide QOS
- ❍ B. If calls are typically placed over the PSTN
- ❍ C. If the site requires sophisticated call center applications
- ❍ D. If you need to deploy a minimum level of voice services on a single platform

Find the Answer p. 47

16. Which layer of the Cisco Unified Communications Architecture features elements such as voicemail, call-center applications and billing systems?Choose the best answer:

- ❍ A. Infrastructure Layer
- ❍ B. Call-Processing Layer
- ❍ C. Applications Layer
- ❍ D. Endpoint Layer

Find the Answer p. 47

17. What Unified Communications Application layer product gives users advanced call distribution, supervision, escalation and logging?Select the best answer.

- ❍ A. Cisco Unified Presence
- ❍ B. Cisco Unified Contact Center
- ❍ C. Cisco Unified Meeting Place
- ❍ D. Cisco Unified TelePresence

Find the Answer p. 47

18. What Unified Communications Application operates on the Cisco ISR's to provide communications failover and redundancy for remote sites that connect to a central Cisco Unified CallManager system?Select the best answer:

- ❍ A. Cisco RSVP agent
- ❍ B. Cisco Emergency Responder
- ❍ C. Survivable Remote Site Telephony
- ❍ D. Unified Presence Server

Find the Answer p. 47

Chapter 2
PSTN components and technologies

1. In dealing with time division multiplexing on a T1 circuit, a communication timeslot is read every 1/8000 of a second. How many channel(s) will be read every 1/8000 second?Select the best answer.

   ❍ A. 23 channels
   ❍ B. 1 channel
   ❍ C. 8000
   ❍ D. 24 channels

   Find the Answer p. 48

2. What type of multiplexing uses arbitrary number of variable bit-rate digital channels that is often used in packet-oriented communication?Select the best answer.

   ❍ A. Time division multiplexing
   ❍ B. Statistical multiplexing
   ❍ C. Frequency division multiplexing
   ❍ D. PRI

   Find the Answer p. 48

3. What type of PSTN signaling are you referring to when telephone receives a phone number string and attempts to establish a path between the calling and called party?Select the best answer.

   ❍ A. Alert signaling
   ❍ B. Address signaling
   ❍ C. Supervisory signaling
   ❍ D. Statistical Signaling

   Find the Answer p. 48

4. Supervisory signaling is can be broken into three distinct categories. What are they?Choose three:

- ❑ A. DTMF tones
- ❑ B. Call supervision
- ❑ C. Call process tones
- ❑ D. Alert signaling

Find the Answer p. 48

5. What channel is used for signaling on an E1 PRI?Select the best answer.

- ❍ A. Channel 31
- ❍ B. Channel 0
- ❍ C. Channel 16
- ❍ D. Channel 17

Find the Answer p. 48

6. On traditional POTS network, what information does the telephone network use to route calls to a specific destination?Select the best answer.

- ❍ A. MAC address
- ❍ B. Destination IP Address
- ❍ C. Telephone numbering plan
- ❍ D. FECN

Find the Answer p. 48

7. What are the two most common traditional business phone systems?Choose two:

- ❑ A. PBX
- ❑ B. Key System
- ❑ C. Central Office
- ❑ D. POTS

Find the Answer p. 48

8. What is signaling system 7 (SS7)?Select the best answer.

- ❍ A. Key system
- ❍ B. A global telephony standard to route calls between CO's and phone companies around the world. It is responsible for the call setup and teardown of calls.
- ❍ C. E.164
- ❍ D. E&M

Find the Answer p. 48

9. Looking at a North American numbering plan: NXX-NXX-XXXX What does the letter "N" represent?Select the best answer.

- ❍ A. Any number 0-9
- ❍ B. The number 3 or 4 only
- ❍ C. Any number 2-9
- ❍ D. N stands for North America

Find the Answer p. 48

10. What port type is used to connect an analog circuit from the PSTN?Select the best answer.

- ❍ A. FXS port
- ❍ B. FXO port
- ❍ C. E&M port
- ❍ D. Analogy DID

Find the Answer p. 48

11. Choose the components that make up the North American Numbering Plan local number.Choose three:

- ❑ A. Country code
- ❑ B. PBX extension
- ❑ C. Line number
- ❑ D. Central office code
- ❑ E. Area Code

Find the Answer p. 48

12. When connecting a PBX to a Cisco gateway using an E&M port, what is then name of the connection that the Cisco gateway is connected too?Select the best answer.

- ❍ A. Trunk side
- ❍ B. T1 PRI
- ❍ C. Tie-line side
- ❍ D. Signaling unit side

Find the Answer p. 48

13. What are characteristics of a basic numbering plan?Choose two:

- ❑ A. All plans are base on the North American Numbering Plan
- ❑ B. Established by the local PSTN to assign endpoint numbers
- ❑ C. All numbers are based on international standards
- ❑ D. Regulated the distribution of numbers and codes in the territory it covers

Find the Answer p. 48

Chapter 3
VoIP components and technologies

1. Besides digitizing voice, what three functions do digital signal processors do in a VoIP environment?Choose three

- ❑ A. Conferencing
- ❑ B. Digitize Video
- ❑ C. Transcoding and Media Termination points
- ❑ D. Echo Cancellation
- ❑ E. Provides dial tone

Find the Answer p. 49

2. What is the byte size for a voice packet with RTP, UDP and IP headers without using compression?Select the best answer.

- ❍ A. 2 bytes
- ❍ B. 4 bytes
- ❍ C. 20 bytes
- ❍ D. 40 bytes

Find the Answer p. 49

3. What are the voice payload sizes for the G.711 and G.729 codecs?Select the best answer.

❍ A. G.711 = 80 bytes
G.729 = 10 bytes

❍ B. G.711 = 160 bytes
G.729 = 20 bytes

❍ C. G.711 = 4 bytes
G.729 = 4 bytes

❍ D. G.711 = 2 bytes
G.729 = 2 bytes

Find the Answer p. 49

4. Which of the following statements is true?Select the best answer.

❍ A. VoIP signaling protocols are responsible for call setup, maintenance and teardown. The signaling protocols are contained within the RTP voice data packets.

❍ B. VoIP signaling protocols are responsible for call setup, maintenance and teardown. The signaling protocols are an entirely separate packet stream from RTP.

❍ C. VoIP signaling protocols are responsible for call setup and maintenance. The signaling protocols are an entirely separate packet stream from RTP. RTP handles call teardown at the conclusion of a call

❍ D. VoIP signaling protocols are responsible for the transport of voice traffic

Find the Answer p. 49

5. What are the two most commonly used voice codecs in a Unified Communications environment?Choose two:

- ❑ A. G.711
- ❑ B. G.726
- ❑ C. G.729
- ❑ D. H.323

Find the Answer p. 49

6. What protocol is designed to provide end-to-end transport for real-time data?Select the best answer.

- ❍ A. SIP
- ❍ B. MGCP
- ❍ C. RTP
- ❍ D. VoIP

Find the Answer p. 49

7. What master/slave plain text protocol allows a Cisco CallManager talk to the voice gateway which provides for centralized gateway administration?Select the best answer.

- ❍ A. SIP
- ❍ B. SCCP
- ❍ C. MGCP
- ❍ D. RTP

Find the Answer p. 49

8. What VoIP signaling protocol is not supported by the Cisco CallManager Express?Select the best answer.

❍ A. SCCP
❍ B. SIP
❍ C. MGCP
❍ D. H.323

Find the Answer p. 49

9. What is the name of the protocol that provides out-of-band management for RTP data?Select the best answer.

❍ A. RTCP
❍ B. SIP
❍ C. G.711
❍ D. SCCP

Find the Answer p. 49

10. When would you want to setup DTFM relay?Choose three

❑ A. To access any kind of DTFM interactive voice response (IVR) menu system
❑ B. To operate a remotely connected voice mail system
❑ C. To access the CallManager GUI
❑ D. To utilize intercom features on the IP phone
❑ E. For calling card access for PSTN calls placed through a remote VoIP PSTN gateway.

Find the Answer p. 49

11. What is a trunk access code used for?Select the best answer:

- ❍ A. To communicate with a non-Cisco PBX
- ❍ B. A way of informing the CallManager that an internal extension is being dialed.
- ❍ C. A code used to communicate to the CallManager that a phone is to be password protected.
- ❍ D. A number used to specify that a call should be routed to the PSTN as opposed to another extension.

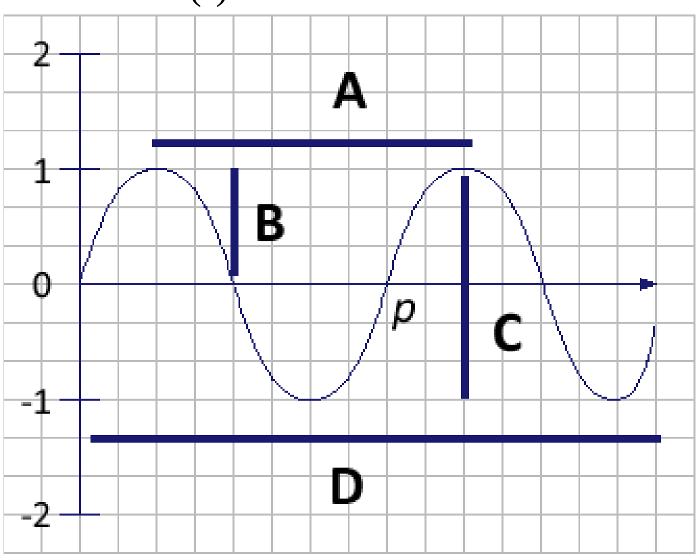
Find the Answer p. 49

12. Using the given analog sound wave diagram, which part of the sound wave depicts the amplitude?Select the best answer:

- ❍ A. Letter A
- ❍ B. Letter B
- ❍ C. Letter C
- ❍ D. Letter D

Find the Answer p. 49

Exhibit(s):



13. What protocol does a Cisco switch utilize to tell the IP phone how it should send voice traffic to the switch?Select the best answer:

- ❍ A. An IP Broadcast
- ❍ B. Cisco Discovery Protocol (CDP)
- ❍ C. An IP Multicast
- ❍ D. OSPF protocol

Find the Answer p. 49

14. What type of business environment is best suited for a key system as opposed to a PBX?Select the best answer:

- ❍ A. A very large organization when a large amount of calls are internal.
- ❍ B. A small organization where the majority of calls are external and all extensions can be answered on any phone.
- ❍ C. A large business with a separate voice mail system.
- ❍ D. A small business with a separate voice mail system.

Find the Answer p. 49

Chapter 4

Describe and config gateways, voice ports and dial peers

1. Which of the following functions does a dial plan NOT perform?Select the best answer.

   ❍ A. Path selection
   ❍ B. Call setup
   ❍ C. Digit Manipulation
   ❍ D. Endpoint addressing
   ❍ E. Calling privileges

   Find the Answer p. 50

2. Which dial string answer fits the destination string given below?55[4-6,9]25.6Select the best answer.

   ❍ A. 5532516
   ❍ B. 554256
   ❍ C. 5572516
   ❍ D. 5592586

   Find the Answer p. 50

3. What digits in the following configuration are forwarded to the PSTN network?dial-peer voice 1 potsdestination-pattern 9911Choose the best answer.

   ❍ A. No digits are forwarded
   ❍ B. 9
   ❍ C. 9911
   ❍ D. 911

   Find the Answer p. 50

4. Which dial peer configuration will work to allow an access code of 9 and allows the user to be able to dial all local, long distance and international calls?Select the best answer.

❍ A. dial-peer voice 1 pots
destination-pattern 9.
port 1/0:1

❍ B. dial-peer voice 1 pots
destination-pattern 9#
port 1/0:1

❍ C. dial-peer voice 1 pots
destination-pattern 9$
port 1/0:1

❍ D. dial-peer voice 1 pots
destination-pattern 9T
port 1/0:1

❍ E. dial-peer voice 1 pots
destination-pattern 9%
port 1/0:1

Find the Answer p. 50

5. What is a voice gateway responsible for doing?Choose two:

❑ A. Translates voice communication between dissimilar networks

❑ B. Functions exclusively within the Unified CallManager Express solution

❑ C. Translates analog-to-digital and digital-to-analog communication

❑ D. Maintains the phone directory database

Find the Answer p. 50

6. Given the following dial string, what command needs to be added to forward all digits within the striing to the PSTN?dial-peer voice 1 potsdestination-pattern 555...port 1/0:1Select the best answer.

❍ A. no digit-strip
❍ B. No command is needed. The gateway will send all 7 digits to the PSTN.
❍ C. prefix 7
❍ D. forward-digits 3

Find the Answer p. 50

7. What is the definition of a hunt group?Select the best answer.

❍ A. A configuration option on a voice gateway that forwards a call to another voice port when one voice port is busy. The hunt groups are ordered using the "dial-peer" command.
❍ B. A configuration option on a voice gateway that forwards a call to the Unity voice mail when the phone is busy. The hunt groups are ordered using the preference command.
❍ C. A configuration option on a voice gateway that forwards a call to another voice port when one voice port is busy. The hunt groups are ordered using the "preference" command.
❍ D. A configuration option on a voice gateway that forwards a call to another voice port when one voice port is busy. The hunt groups are ordered using the "destination pattern" command.

Find the Answer p. 50

8. What is the name of a connection between two gateways/routers or between a gateway/router and an IPT device?Choose the correct answer:

❍ A. Hair-pinning
❍ B. Dial peer
❍ C. Quality of Service
❍ D. Call leg

Find the Answer p. 50

9. What two reasons might a company choose an Internet Telephony Service Provider (ITSP) over a PSTN provider?Choose two:

- ❑ A. PSTN connections provide no QOS and calls can have jitter
- ❑ B. ITSP contracts are typically cheaper in cost when compared to comparable PSTN connections.
- ❑ C. ITSP configuration on the voice gateway is typically less complex when compared to PSTN configurations
- ❑ D. Eliminates the need of a voice gateway router.

Find the Answer p. 50

10. What do private-line automatic ringdown (PLAR) configurations do?Select the best answer:

- ❍ A. PLAR creates a hunt-group line configuration that dials the next number in a hunt-group list if the first phone is busy.
- ❍ B. PLAR statically configures endpoints that do not require the end user to dial digits. The number is automatically dialed when the phone goes off-hook
- ❍ C. PLAR dynamically configures endpoints that do not require the end user to dial digits. The number is automatically dialed when the phone goes off-hook
- ❍ D. PLAR statically configures endpoints that requires the end user to dial the number manually.

Find the Answer p. 50

11. Given the configuration below, what is the purpose of the "session target" command?dial-peer voice 1 voipdestination-pattern 555session target ipv4:10.10.1.1Select the best answer.

- ❑ A. The session target points the dial-peer to the tftp-server
- ❑ B. The session target points the dial-peer to the Unity server.
- ❑ C. The session target codec for the dial peer

Find the Answer p. 50

12. Using the given dial peer configurations, which dial peer would match a dial string of 5551212?Select the best answer:

- ❍ A. dial-peer 1
- ❍ B. dial-peer 2
- ❍ C. All three dial-peers
- ❍ D. dial peer 3

Find the Answer p. 50

Exhibit(s):

```
dial-peer voice 1 voip
 destination-pattern 555
 session target ipv4:10.10.1.1

dial-peer voice 2 voip
 destination-pattern 5551212
 session target ipv4:10.10.1.2

dial-peer voice 3 voip
 destination-pattern 55512
 session target ipv4:10.10.1.3
```

13. What is a global configuration command that enables you to define a set of digits for the router to prepend to the beginning of a dialed string before passing it to the remote telephony device?Select the best answer.

- ❍ A. router(config)#num-exp <extension number> <expanded number>
- ❍ B. router(config)#number-expansion <extension number> <expanded number>
- ❍ C. router(config-dial-peer)#destination-pattern <string>
- ❍ D. router(config-dial-peer)#forward-digits <num-digit>

Find the Answer p. 50

14. Given the following MGCP configuration command, what Cisco device is 172.16.10.100?mgcp call-agent 172.16.10.100 service-type mgcp version 0.1Select the best answer

❍ A. The router loopback IP address
❍ B. The Cisco Unity IP address
❍ C. The Cisco CallManager

Find the Answer p. 50

15. Where are DSP's configured in a Cisco Unified Communications Architecture?Select the best answer.

❍ A. On the CallManager
❍ B. On the Cisco Unity Express
❍ C. On the Autonomous Access Point
❍ D. On the Voice Gateway

Find the Answer p. 50

16. When voice gateways transcode voice calls, when would it not use a DSP and perform the transcoding instead?Select the best answer:

❍ A. Only when all of the DSPs are in use
❍ B. All transcoding must be performed on hardware DSPs.
❍ C. If the two voice endpoints use the same codec and packetization time
❍ D. If the two voice endpoints use different codec's

Find the Answer p. 50

Chapter 5

Describe and configure a Cisco Network to support VoIP

1. What three configuration files are needed for a Cisco IP Phone to function properly?Choose three:

- ❑ A. SEPAAABBBCCCDDD.cnf.xml file where AAABBBCCCDDD is the IP address of each phone
- ❑ B. Firmware File
- ❑ C. SEPAAAABBBBCCCC.cnf.xml file where AAAABBBBCCCC is the MAC address of each phone.
- ❑ D. DefaultAAAABBBBCCCC.cnf.xml file where AAAABBBBCCCC is the MAC address of each phone.
- ❑ E. XMLDefault.cnf.xml file

Find the Answer p. 51

2. What is the IP dhcp excluded-address command used for?Select the best answer.

- ❍ A. It excludes IP phones from receiving certain DHCP addresses. Workstations are able to receive them however.
- ❍ B. Only the IP addresses contained in the ip dhcp excluded-address command are handed out to DHCP clients.
- ❍ C. The IP addresses contained within the ip dhcp excluded-addresses are reserved for the voice gateway only.
- ❍ D. It specifies IP addresses that should not be handed out to DHCP clients

Find the Answer p. 51

3. What protocol is used to synchronize the system clocks on all Cisco Unified Communications equipment?Select the best answer:

❍ A. DHCP
❍ B. IP
❍ C. NTP
❍ D. CDP

Find the Answer p. 51

4. A customer is interested in deploying a new Cisco Unified Communications system on their current network. They already have a network that is using public address space. Unfortunately, there are not enough public IP addresses to support both the voice and data network. What benefits would be achieved for deploying a separate voice VLAN on the network?Choose two:

❑ A. With a separate voice VLAN, you could deploy private (RFC 1918) address space on the phones to free up scarce public IP addresses
❑ B. Adding a voice VLAN makes QOS easier to configure.
❑ C. Each IP phone and PC will need a separate Ethernet connection.
❑ D. Having a separate voice VLAN reduced Spanning-tree overhead

Find the Answer p. 51

5. A VoIP call travels across the path depicted in the diagram. Where is the bottleneck?Select the best answer:

❍ A. Segment 1
❍ B. Segment 2
❍ C. Segment 3
❍ D. Segment 4

Find the Answer p. 51

6. You have configured DHCP for your IP phones on a subnet. The phones are properly receiving the IP address, subnet mask and gateway information. They are not getting the configuration file from the CallManager. What is likely the problem?Choose the correct answer:

❍ A. The IP phones are not connected to the network

❍ B. The DHCP option 43 command is missing

❍ C. The DHCP option 150 command is missing

❍ D. The IP phones must be on the same IP subnet as the CallManager

Find the Answer p. 51

7. When issuing the auto qos voip [trust] [fr-atm] command to enable auto-qos, if the "trust" command Is issued, what does the switch do to classify traffic?Choose two:

❑ A. It uses the DSCP value from the phone

❑ B. It uses the DSCP value on the switchport.

❑ C. It does not classify traffic at all unless the "trust" command is issued.

❑ D. It uses NBAR to classify traffic

❑ E. Area Code

Find the Answer p. 51

8. Configuring QOS on a network allows for traffic to be classified at both layer 2 and layer 3. What are the names of these classification values at the network and data-link layer?Choose 2

❑ A. Layer 3 - Class of Service (CoS)

❑ B. Layer 3 - Differentiated Services Code Point (DSCP)

❑ C. Layer 2 - Differentiated Services Code Point (DSCP)

❑ D. Layer 2 - Class of Service (CoS)

❑ E. Layer 2 - IP Precedence

Find the Answer p. 51

9. Quality of Service manages four parameters of traffic in order to provide guaranteed service for time sensitive traffic. What are they?Choose four:

- ❏ A. Delay
- ❏ B. Jitter
- ❏ C. DSCP
- ❏ D. CoS
- ❏ E. Bandwidth
- ❏ F. Packet loss

Find the Answer p. 51

10. Looking at the show mls qos command for port fa0/1, what can you determine about how the trust state is configured on this switchport?Select the best answer.

- ❍ A. Port fa0/1 will trust the DSCP priority sent from an IP phone.
- ❍ B. Port fa0/1 will trust the CoS priority sent from an IP phone.
- ❍ C. CoS override is enabled and will set all traffic to a CoS of 5
- ❍ D. The trust boundary is pushed to the switchport level. The phone's CoS is not trusted.

Find the Answer p. 51

Exhibit(s):

```
switch#show mls qos interface fastEthernet 0/1

FastEthernet0/1
 trust state: trust cos
 trust mode: trust cos
 COS override: dis
 default COS: 0
 DSCP Mutation Map: Default DSCP Mutation Map
 trust device: none
```

Chapter 6
Implement UC500 using Cisco Configuration Assistant

1. The Cisco Configuration Assistant provides the following functionality except.Select the best answer:

   - ❍ A. Simplified network reporting
   - ❍ B. Drag and drop software updates
   - ❍ C. Multiple network views
   - ❍ D. Load balancing
   - ❍ E. Troubleshooting
   - ❍ F. Simplified configuration for voice, data, security and wireless networks.

   Find the Answer p. 52

2. How does the Cisco Configuration Assistant notify the administrator when they have either incorrectly configured a section or left a necessary component blank?Select the best answer:

   - ❍ A. The tab is highlighted in read and explains the improper configuration.
   - ❍ B. The system reboots.
   - ❍ C. An error message is generated and sent to the log.
   - ❍ D. An audible beep will sound.

   Find the Answer p. 52

3. The Cisco Smart Business Communication System by default provides built-in SIP trunk support for what two service providers?Choose two:

- ❑ A. Broadvoice
- ❑ B. AT&T
- ❑ C. CBeyond Communications
- ❑ D. Verizon

Find the Answer p. 52

4. How does the Cisco Configuration Assistant discover the SBCS equipment on the network?Select the best answer:

- ❍ A. The CCA sends out a multicast request to 224.0.0.8
- ❍ B. The CCA sends out a broadcast to the network. The SBCS equipment replies to this broadcast.
- ❍ C. The CCA uses CDP (Cisco Discovery Protocol) to discover the network.
- ❍ D. You must manually configure the IP addresses of the SBCS equipment that you want to configure with CCA.

Find the Answer p. 52

5. What are the requirements to be able to use the Cisco Configuration Assistant (CCA)?Select the best answer:

- ❍ A. CCA is a Java based application. You can use any java enabled web browser.
- ❍ B. CCA is a web based application. You must be using Internet Explorer version 6 or greater.
- ❍ C. CCA is a Windows application. You must meet a set of minimum hardware requirements to use it.
- ❍ D. The CCA is a command line tool. You must telnet/ssh to the UC500 to access the assistant.

Find the Answer p. 52

6. Which tab of the Cisco Configuration Assistant would you go to if you want to configure voice features such as: MOH, paging, hunt groups can call park?Choose the best answer:

❍ A. The AA and Voicemail tab
❍ B. The Voice Features Tab
❍ C. The SIP Trunk tab
❍ D. The System tab

Find the Answer p. 52

7. What menu of the CCA would you be able to backup and/or restore a SBCS configuration file?Choose the best answer:

❍ A. The Maintenance menu
❍ B. The Telephony menu
❍ C. The Security menu
❍ D. The Configure menu

Find the Answer p. 52

8. What would happen if you plugged a non-Cisco IP phone (802.1af compatible) into a PoE port of a Cisco Unified Communications 500 sytem?Select the best answer:

❍ A. The phone would work on the network but it would require the need of an external power source.
❍ B. The phone would power up properly
❍ C. The UC 500 would detect a non-Cisco IP phone on the Ethernet port and put the port into Error-disable.
❍ D. The phone would only partially boot due to lack of power that it requires.

Find the Answer p. 52

Chapter 7
Cisco Unified Communications Express

1. Localization on a CME allows administrators to specify country-specific tones and cadences; as well as 12 different languages. This allows an organization to modify the Cisco IP phones to fit the location they reside. What commands would change the network and user location to German?Select the best answer:

❍ A. Router(config)# telephony-service
Router(config-telephony-service)# network-locale GE
Router(config-telephony-service)# user-locale GE

❍ B. Router(config)# telephony-service
Router(config-telephony-service)# network-locale DE
Router(config-telephony-service)# user-locale DE

❍ C. Router(config)# network-locale DE
Router(config)# telephony-service
Router(config-telephony-service)# user-locale DE

❍ D. Router(config)# network-locale DE
Router(config)# user-locale DE

Find the Answer p. 53

2. Given the ephone and ephone-dn partial configuration given, if three calls are placed to extension 5001 simultaneously, what happens to the third call?Select the best answer:

❍ A. The call will ring on the second line of ephone 1.

❍ B. The call will ring on the second line of ephone 2.

❍ C. The call will ring on the second line of ephone 1 and 2.

❍ D. The call will receive a busy tone

Find the Answer p. 53

Exhibit(s):

```
ephone 1 dual-line
 button 1:1
 mac-address 0030.12c3.8434

ephone-dn 1
 number 5001
 preference 0
 huntstop channel

ephone 2 dual-line
 button 1:2
 mac-address 0030.24a2.3325

ephone-dn 2
 number 5001
 preference 1
 huntstop channel
```

3. How do you configure an ephone 4 to apply ephone-dn 3 to button 5?Select the best answer:

❍ A. Router(config)# ephone 4
Router(config-ephone)# button 5:3

❍ B. Router(config)# ephone 5
Router(config-ephone)# button 4:3

❍ C. Router(config)# ephone 3
Router(config-ephone)# button 4:5

❍ D. Router(config)# ephone 3
Router(config-ephone)# exit
Router(config)# ephone-dn 3

Find the Answer p. 53

4. Given the partial configuration provided, what does the 4s1 mean?Select the best answer:

❍ A. Button 1 is configured with ephone-dn 4. The ring will be silent

❍ B. Button 4 is configured with ephone-dn 1. The ring will be silent

❍ C. Button 4 is configured with ephone-dn 1. The button will hidden

❍ D. Button 4 is configured to ring when any of the ephone-dn extensions are dialed.

Find the Answer p. 53

Exhibit(s):

```
ephone-dn 1
 number 233

ephone-dn 4
 number 234

ephone-dn 16
 number 235

ephone-dn 19
 number 236

ephone 1
 button 1:19 2:4 3:16 4s1
```

5. You are adding a new Cisco 7961 phone to your CME environment. You have the partial configuration in the router. What command is needed to associate the new phone to the ephone 1 configuration if autoregistration is disabled?Select the best answer:

❍ A. Router(config)# ephone-dn1 dual-line

❍ B. Router(config)# ephone 1
Router(config-ephone)# mac-address xxxx.xxxx.xxxx.xxxx

❍ C. Router(config)# ephone 1
Router(config-ephone)# max-ephones 1

❍ D. Router(config)# ephone-dn 1
Router(config-ephone-dn)# ephone-dn-template 1

Find the Answer p. 53

Exhibit(s):

```
ephone-dn 1
 number 233

ephone-dn 4
 number 234

ephone-dn 16
 number 235

ephone-dn 19
 number 236

ephone 1
 button 1:19 2:4 3:16 4s1
```

6. What is the default SCCP protocol/port?Choose the best answer:

❍ A. UDP 2000
❍ B. Option 150
❍ C. TCP 2000
❍ D. Random TCP port over 1024

Find the Answer p. 53

7. What command is necessary for autoregistration to work properly on a CME?Select the best answer:

❍ A. Ip source-address <ip address>
❍ B. No auto-reg-ephone
❍ C. mac-address
❍ D. Show ephone attempted-registrations

Find the Answer p. 53

8. What function does the "auto-assign" command do when configured on a CME?Choose the best answer:

- ❍ A. A way to automatically assign an ephone to brand new phones. It helps speed the process of deploying new phone systems.
- ❍ B. A way to automatically assign a ephone-dn to brand new phones. It helps speed the process of deploying new phone systems.
- ❍ C. This command assigns the CME IP address to newly connected phones so they know where to send SCCP messages.
- ❍ D. This command sets up NTP

Find the Answer p. 53

9. What circumstances must you use the reset command to reboot the IP phone?Choose three:

- ❑ A. Changing an ephone-dn
- ❑ B. DHCP renew
- ❑ C. Changing URL's
- ❑ D. Changing user/network locales
- ❑ E. Updating the phone firmware

Find the Answer p. 53

10. What command is used to forward calls when a user does not answer the phone for a specified period of time?Select the best answer.

- ❍ A. call-forward all <directory number>
- ❍ B. call-forward busy <directory number>
- ❍ C. call-forward noan <directory number> timeout <seconds>
- ❍ D. call-forward max-length <length>

Find the Answer p. 53

11. Which Cisco configuration command will allow the CME to receive NTP messages to synchronize the system clock?Select the best answer:

❍ A. Router#clock set 08:00:00 january 1 2008
Router# (config) ntp source 10.1.1.1

❍ B. Router# (config) ntp server 10.1.1.1

❍ C. Router# (config) ntp source 10.1.1.1

❍ D. Router# (config) ntp master stratum 2

Find the Answer p. 53

12. You configured your router to support Cisco Unified Communications Manager Express but not all of the phones have registered. All of the 7941's and 7961's registered properly but the 7971's only received an IP address. When deploying the phones, the 7971's were brought up first. What is likely to be the problem?Select the best answer:

❍ A. The phones are in the wrong voice VLAN.

❍ B. The max-ephone level has been reached.

❍ C. The max-dn level has been reached.

❍ D. The 7971 firmware has not been configured.

Find the Answer p. 53

13. What command configures the source address on a Cisco CME?Select the best answer

❍ A. Router(config)# telephony-service
Router(config-telephony)# ip source-address 10.1.1.100 port 2000

❍ B. Router(config)# ip source-address 10.1.1.100 port 2000

❍ C. Router(config)# telephony-service
Router(config-telephony)# ip address 10.1.1.100 port 2000

❍ D. Router(config)# telephony-service ip source-address 10.1.1.100

Find the Answer p. 53

14. What configuration commands are used to reset the IP phones connected to a CME?Select the best answer.

❍ A. Router(config)# reset all

❍ B. Router(config)# telephony-service
Router(config-telephony)# reset all

❍ C. Router(config)# telephony-service reset all

❍ D. Router# reload

Find the Answer p. 53

15. Looking at the attached show ephone command, how many phones are properly connected to the Cisco CME?Select the best answer:

❍ A. All 4 phones are connected and working properly on the Cisco CME.

❍ B. None of the phones are properly connected to the CME

❍ C. Three of the four ephones are properly connected to the CME.

❍ D. None of the phones are properly connected because you can see that activeLine:0. This means that no calls are active.

Find the Answer p. 53

Exhibit(s):

```
Router# show ephone

ephone-1 Mac:0003.E3E7.F627 TCP socket:[2] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.2 51671 Telecaster 7940 keepalive 28 max_line 2
button 1: dn 1 number 4444
ephone-2 Mac:0030.94C3.F43A TCP socket:[1] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.0.0.3 50094 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 3 number 5555
ephone-3 Mac:0007.0E81.10F0 TCP socket:[-1] activeLine:0 UNREGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:0.0.0.0 0 Unknown 0  keepalive 0 max_line 0
ephone-4 Mac:0003.6B40.99DA TCP socket:[3] activeLine:0 REGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:10.2.168.200 51879 Telecaster 7960 keepalive 28 max_line 6
button 1: dn 2 number 3333
```

16. Before you can configure ephone and ephone-dn's, what configuration do you need to insure is on the router?Select the best answer.

❍ A. Router(config)telephony-service
Router(config-telephony)#max-ephones <number>
Router(config-telephony)#max-dn <number>

❍ B. Router(config)ephone 1

❍ C. Router(config)ephone-dn 1

❍ D. Router(config)#max-ephones <number>
Router(config)#max-dn <number>

Find the Answer p. 53

17. Which Unity Express capabilities are available with the telephone user interface (TUI)?Choose three:

❑ A. Recording personal greetings

❑ B. Modification of extension numbers.

❑ C. Vacation or emergency notifications

❑ D. Digit manipulation

❑ E. Remote directory lookup

Find the Answer p. 53

Chapter 8
Cisco Unity Configuration and Troubleshooting

1. What Unified Communications Manager configuration component deals with MWI?Select the best answer:

   ❍ A. ephone configuration
   ❍ B. Ephone template configuration
   ❍ C. h323 gatekeeper
   ❍ D. ephone-dn

   Find the Answer p. 54

2. What methods are available to create new auto-attendant greeting, prompt and announcement files?Choose two:

   ❑ A. Create your own voice announcements with a .mp3 extension and upload it to the CME.
   ❑ B. Use the Administration via telephone application via TUI to record announcements
   ❑ C. Create your own voice file with a .wav extension and upload it to the CME.
   ❑ D. Use the microphone input jack on the CME chassis to record audio.
   ❑ E. Create your own voice announcements with a .aac extension and upload it to the CME.

   Find the Answer p. 54

3. Given the active call display show command on a CUE, what command can you use to terminate the active call?Select the best answer.

❍ A. ccn call terminate port 4086502231
❍ B. ccn call terminate call 1566/1
❍ C. ccn call terminate call 4086502231
❍ D. ccn call terminate port 4086502231
or
ccn call terminate call 1566/1
❍ E. ccn call terminate port 1566/1

Find the Answer p. 54

4. What methods are used to upgrade the software on a Unity Express system?Select the best answer.

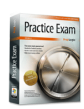
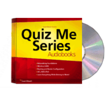
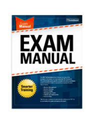
❍ A. Graphical user interface (GUI)
❍ B. Command line (CLI)
❍ C. Command line (CLI) or Graphical user interface (GUI)
❍ D. Graphical user interface (GUI) or Telephone user interface (TUI)

Find the Answer p. 54

5. What type of system reports does the Cisco Unity Express not provide?Select the best answer:

❍ A. Mailbox and message statistics
❍ B. Backup and restore history
❍ C. Memory and CPU usage
❍ D. Interface staticstics
❍ E. Call history

Find the Answer p. 54


6. What are the minimum router configuration steps needed to get Cisco Unity Express up and running?Choose three:

- ❑ A. Configure a static route pointing to Unity
- ❑ B. Assign an IP address to Unity
- ❑ C. Configure DHCP
- ❑ D. Assign an IP address on the router and configure routing.
- ❑ E. Configure EIGRP on Unity

Find the Answer p. 54

7. You are having problems with your Cisco Unity Express and wan to view the log files to determine the cause of the problem. How do you access the log files?Select the best answer.

- ❍ A. Connect to the Unity Express GUI and access the logs under the "Log Files" tab
- ❍ B. Connect to the CallManager console port to view the logs
- ❍ C. Access the CLI using either the console port or Telnet/SSH access to view the log files.
- ❍ D. Access either the CLI or GUI to access the log files.

Find the Answer p. 54

8. Is it possible to reboot the Cisco Unity Express independent of the router?Select the best answer.

- ❍ A. No. The CUE is a module and the only way it can be rebooted is if the entire router is rebooted.
- ❍ B. Yes. The CUE module can be rebooted without the need to reboot the entire router.
- ❍ C. No. The CUE will lose it's configuration if rebooted
- ❍ D. Yes, but the router will also need to be rebooted within a 24 hour time period.

Find the Answer p. 54

9. If you want to configure an external Unity Express solution from your CME router, what configuration must be completed to insure the two devices properly communicate?Select the best answer.

- ❍ A. Configure the SIP proxy server on the Cisco Unity Express device to point to the CallManager Express
- ❍ B. Configure the SIP proxy server on the CallManager Express device to point to the Unity Express device
- ❍ C. Configure the H.323 proxy server on the Cisco Unity Express device to point to the CallManager Express
- ❍ D. Configure the H.323 proxy server on the CallManager Express device to point to the Unity Express device

Find the Answer p. 54

10. What version of Cisco Unity Express (CUE) does not use an interface slot on a router?Select the best answer:

- ❍ A. NM-CUE
- ❍ B. Both the NM-CUE and AIM-CUE
- ❍ C. AIM-CUE
- ❍ D. Both the NM-CUE and AIM-CUE use a network module slot.

Find the Answer p. 54

11. Looking t the show voicemail command given, what does the "Zero Out Number" setting do?Select the best answer.

- ❍ A. When a caller is directed to a PrepLogic's Voicemail box, they can verbally say the number "0" to be redirected to extension 1234
- ❍ B. When a caller is directed to a PrepLogic's Voicemail box, they can press the 0 digit on their phone to end the call.
- ❍ C. When a caller is directed to a PrepLogic's Voicemail box, they can press the 0 digit on their phone to access the voicemail greeting.
- ❍ D. When a caller is directed to a PrepLogic's Voicemail box, they can press the 0 digit on their phone to be redirected to extension 1234

Find the Answer p. 54

Exhibit(s):

```
se-10-0-0-0# show voicemail detail mailbox preplogic
Owner:                          /sw/local/users/user3
Type:                           Personal
Description:                    PrepLogic mailbox
Busy state:                     idle
Enabled:                        true
Mailbox Size (seconds):         480
Message Size (seconds):         180
Play Tutorial:                  true
Space Used (seconds):           0
Total Message Count:            0
New Message Count:              0
Saved Message Count:            0
Future Message Count:           0
Deleted Message Count:          0
Expiration (days):              10
Greeting:                       alternate
Zero Out Number:                1234
Created/Last Accessed:          Aug 15 2008 18:31:15 PST
```

12. You have just configured the Cisco Unified Communications Manager Express to support Unity Express. When you attempt to access Unity, you receive the following message: "IP address needs to be configured on the interface Service-Engine 1/0". What command needs to be added to interface Service-Engine 1/0 to complete the configuration?interface Service-Engine1/0service-module ip address 10.10.10.2 255.255.255.0service-module ip default-gateway 10.10.10.1interface loopback 0ip add 10.10.10.3 255.255.255.0Select the best answer.

- ❍ A. A second "service-module ip address" needs to be configured.
- ❍ B. Add "ip unnumbered loopback 0"
- ❍ C. Add "service-module ip active"
- ❍ D. Add "no shutdown" to the interface Service-Engine 1/0

Find the Answer p. 54

13. Looking t the show voicemail command given, what does the "Play Tutorial" setting do?Select the best answer.

- ❍ A. The voicemail tutorial will start up for mailbox owners logging in to their mailbox for the first time. Once the user has gone through the tutorial, it will not play again. This command is enabled by default.
- ❍ B. The voicemail tutorial will start up for mailbox owners logging in to their mailbox for the first time. Once the user has gone through the tutorial, it will not play again. This command is diabled by default.
- ❍ C. The voicemail tutorial will start up for mailbox owners logging in to their mailbox for the first time. The tutorial will continue to play until it is disabled on the CallManager This command is diabled by default.
- ❍ D. The voicemail tutorial is for system administrators only. A password must be entered to disable the tutorial on mailboxes.

Find the Answer p. 54

Exhibit(s):

```
se-10-0-0-0# show voicemail detail mailbox preplogic
Owner:                          /sw/local/users/user3
Type:                           Personal
Description:                    PrepLogic mailbox
Busy state:                     idle
Enabled:                        true
Mailbox Size (seconds):         480
Message Size (seconds):         180
Play Tutorial:                  true
Space Used (seconds):           0
Total Message Count:            0
New Message Count:              0
Saved Message Count:            0
Future Message Count:           0
Deleted Message Count:          0
Expiration (days):              10
Greeting:                       alternate
Zero Out Number:                1234
Created/Last Accessed:          Aug 15 2008 18:31:15 PST
```

14. Configure a Cisco Unity Express system to allow read-only access to an SNMP server at 172.16.160.224. The password string of the SNMP server is "ccnavoice". The password string on the CUE is "preplogic1". SNMP traps should also be enabled.Select the best answer.

❍ A. cisco(config)# snmp-server community preplogic1 rw
cisco(config)# snmp-server enable traps
cisco(config)# snmp-server host 172.16.160.224 ccnavoice

❍ B. cisco(config)# snmp-server community preplogic1 ro
cisco(config)# snmp-server traps enable
cisco(config)# snmp-server host 172.16.160.224 ccnavoice

❍ C. cisco(config)# snmp-server community preplogic1 ro
cisco(config)# snmp-server enable traps
cisco(config)# snmp-server host 172.16.160.224 ccnavoice

❍ D. cisco(config)# snmp-server community preplogic1 ro
cisco(config)# snmp-server enable traps
cisco(config)# snmp-server host ccnavoice

Find the Answer p. 54

Answers: Chapter 1

1. **A, D**	Review Question p. 2	Detailed Explanation p. 56
2. **C**	Review Question p. 2	Detailed Explanation p. 56
3. **A, B**	Review Question p. 2	Detailed Explanation p. 57
4. **B**	Review Question p. 3	Detailed Explanation p. 57
5. **B, D, E**	Review Question p. 3	Detailed Explanation p. 57
6. **B**	Review Question p. 3	Detailed Explanation p. 58
7. **C**	Review Question p. 4	Detailed Explanation p. 58
8. **D**	Review Question p. 4	Detailed Explanation p. 59
9. **A, B, D**	Review Question p. 4	Detailed Explanation p. 59
10. **D**	Review Question p. 5	Detailed Explanation p. 59
11. **A, B**	Review Question p. 5	Detailed Explanation p. 60
12. **D**	Review Question p. 5	Detailed Explanation p. 60
13. **B**	Review Question p. 6	Detailed Explanation p. 61
14. **A**	Review Question p. 6	Detailed Explanation p. 61
15. **C**	Review Question p. 6	Detailed Explanation p. 61
16. **C**	Review Question p. 7	Detailed Explanation p. 62
17. **B**	Review Question p. 7	Detailed Explanation p. 62
18. **C**	Review Question p. 7	Detailed Explanation p. 63

Answers: Chapter 2

1. **D**	Review Question p. 8	Detailed Explanation p. 64
2. **B**	Review Question p. 8	Detailed Explanation p. 64
3. **B**	Review Question p. 8	Detailed Explanation p. 64
4. **B, C, D**	Review Question p. 9	Detailed Explanation p. 65
5. **C**	Review Question p. 9	Detailed Explanation p. 65
6. **C**	Review Question p. 9	Detailed Explanation p. 65
7. **A, B**	Review Question p. 9	Detailed Explanation p. 66
8. **B**	Review Question p. 10	Detailed Explanation p. 66
9. **C**	Review Question p. 10	Detailed Explanation p. 67
10. **B**	Review Question p. 10	Detailed Explanation p. 67
11. **C, D, E**	Review Question p. 11	Detailed Explanation p. 68
12. **D**	Review Question p. 11	Detailed Explanation p. 68
13. **B, D**	Review Question p. 11	Detailed Explanation p. 68

Answers: Chapter 3

1. **A, C, D**	Review Question p. 12	Detailed Explanation p. 70
2. **D**	Review Question p. 12	Detailed Explanation p. 70
3. **B**	Review Question p. 13	Detailed Explanation p. 70
4. **B**	Review Question p. 13	Detailed Explanation p. 71
5. **A, C**	Review Question p. 14	Detailed Explanation p. 71
6. **C**	Review Question p. 14	Detailed Explanation p. 72
7. **C**	Review Question p. 14	Detailed Explanation p. 72
8. **C**	Review Question p. 15	Detailed Explanation p. 72
9. **A**	Review Question p. 15	Detailed Explanation p. 73
10. **A, B, E**	Review Question p. 15	Detailed Explanation p. 73
11. **D**	Review Question p. 16	Detailed Explanation p. 74
12. **B**	Review Question p. 16	Detailed Explanation p. 74
13. **B**	Review Question p. 17	Detailed Explanation p. 74
14. **B**	Review Question p. 17	Detailed Explanation p. 75

Answers: Chapter 4

1. **B**	Review Question p. 18	Detailed Explanation p. 76
2. **D**	Review Question p. 18	Detailed Explanation p. 76
3. **A**	Review Question p. 18	Detailed Explanation p. 76
4. **D**	Review Question p. 19	Detailed Explanation p. 77
5. **A, C**	Review Question p. 19	Detailed Explanation p. 77
6. **A**	Review Question p. 20	Detailed Explanation p. 78
7. **C**	Review Question p. 20	Detailed Explanation p. 78
8. **D**	Review Question p. 20	Detailed Explanation p. 78
9. **B, C**	Review Question p. 21	Detailed Explanation p. 79
10. **B**	Review Question p. 21	Detailed Explanation p. 79
11.	Review Question p. 21	Detailed Explanation p. 80
12. **C**	Review Question p. 22	Detailed Explanation p. 80
13. **A**	Review Question p. 22	Detailed Explanation p. 81
14. **C**	Review Question p. 23	Detailed Explanation p. 81
15. **D**	Review Question p. 23	Detailed Explanation p. 81
16. **C**	Review Question p. 23	Detailed Explanation p. 82

Answers: Chapter 5

1. **B, C, E**	Review Question p. 24	Detailed Explanation p. 83
2. **D**	Review Question p. 24	Detailed Explanation p. 83
3. **C**	Review Question p. 25	Detailed Explanation p. 83
4. **A, B**	Review Question p. 25	Detailed Explanation p. 84
5. **C**	Review Question p. 25	Detailed Explanation p. 84
6. **C**	Review Question p. 26	Detailed Explanation p. 85
7. **D, E**	Review Question p. 26	Detailed Explanation p. 85
8. **B, D**	Review Question p. 26	Detailed Explanation p. 85
9. **A, B, E, F**	Review Question p. 27	Detailed Explanation p. 86
10. **B**	Review Question p. 27	Detailed Explanation p. 86

Answers: Chapter 6

1. **D**	Review Question p. 28	Detailed Explanation p. 88
2. **A**	Review Question p. 28	Detailed Explanation p. 88
3. **B, C**	Review Question p. 29	Detailed Explanation p. 88
4. **C**	Review Question p. 29	Detailed Explanation p. 89
5. **C**	Review Question p. 29	Detailed Explanation p. 89
6. **B**	Review Question p. 30	Detailed Explanation p. 90
7. **A**	Review Question p. 30	Detailed Explanation p. 90
8. **B**	Review Question p. 30	Detailed Explanation p. 91

Answers: Chapter 7

1. **B**	Review Question p. 31	Detailed Explanation p. 92
2. **D**	Review Question p. 32	Detailed Explanation p. 92
3. **A**	Review Question p. 32	Detailed Explanation p. 92
4. **B**	Review Question p. 33	Detailed Explanation p. 93
5. **B**	Review Question p. 34	Detailed Explanation p. 93
6. **C**	Review Question p. 34	Detailed Explanation p. 94
7. **A**	Review Question p. 34	Detailed Explanation p. 94
8. **B**	Review Question p. 35	Detailed Explanation p. 94
9. **C, D, E**	Review Question p. 35	Detailed Explanation p. 95
10. **C**	Review Question p. 35	Detailed Explanation p. 95
11. **B**	Review Question p. 36	Detailed Explanation p. 96
12. **D**	Review Question p. 36	Detailed Explanation p. 96
13. **A**	Review Question p. 36	Detailed Explanation p. 96
14. **B**	Review Question p. 37	Detailed Explanation p. 97
15. **C**	Review Question p. 37	Detailed Explanation p. 97
16. **A**	Review Question p. 38	Detailed Explanation p. 98
17. **A, C, E**	Review Question p. 38	Detailed Explanation p. 98

Answers: Chapter 8

1. **D**	Review Question p. 39	Detailed Explanation p. 99
2. **B, C**	Review Question p. 39	Detailed Explanation p. 99
3. **D**	Review Question p. 40	Detailed Explanation p. 99
4. **B**	Review Question p. 40	Detailed Explanation p. 100
5. **D**	Review Question p. 40	Detailed Explanation p. 100
6. **A, B, D**	Review Question p. 41	Detailed Explanation p. 101
7. **C**	Review Question p. 41	Detailed Explanation p. 101
8. **B**	Review Question p. 41	Detailed Explanation p. 101
9. **B**	Review Question p. 42	Detailed Explanation p. 102
10. **C**	Review Question p. 42	Detailed Explanation p. 102
11. **D**	Review Question p. 43	Detailed Explanation p. 103
12. **B**	Review Question p. 43	Detailed Explanation p. 103
13. **A**	Review Question p. 44	Detailed Explanation p. 103
14. **C**	Review Question p. 45	Detailed Explanation p. 104

Explanations: Chapter 1

1. Review Question p. 2

   Answers: A, D

   Explanation A. Correct - Connecting standard PBX systems with routers was the first evolution. The Cisco CME now integrates their IOS router with the functionality of a PBX

   Explanation B. Incorrect - SRST is a feature of CME but not a foundation element

   Explanation C. Incorrect - H.323 is an ITU standard to transport multimedia communication over IP networks. While the CME can handle H.323 protocol, it is not one of the two foundation elements.

   Explanation D. Correct - The VoIP infrastructure was engineered to provide PSTN and PBX gateway connectivity on a router.

   More Information:

   Cisco IP Communications Express - Page 59

   PrepLogic Question: 11418-100

2. Review Question p. 2

   Answers: C

   Explanation A. Incorrect - Session Initialization Protocol (SIP) is an IETF protocol used for call setup.

   Explanation B. Incorrect - H.323 is an ITU standard that manages call control, multimedia management and bandwidth.

   Explanation C. Correct - A Cisco proprietary used for VoIP signaling to send messages between the call agent and the IP phone

   Explanation D. Incorrect - VoIP is a technology that allows voice data to be sent over an IP network.

   More Information:

   Cisco IP Communications Express - Page 61

   PrepLogic Question: 11418-101

3. Review Question p. 2

Answers: A, B

Explanation A. Correct - The CME provides protocol conversion between IP and non-IP based voice protocols.

Explanation B. Correct - The CME provides protocol conversion between IP and non-IP based voice protocols.

Explanation C. Incorrect - Frame Relay is a communications protocol. The CME is not considered a gateway for this technology

Explanation D. Incorrect - An FXO is a physical interface that connects PSTN analog phones on a PBX.

Explanation E. Incorrect - a filter that helps to protect one segment of a network from another. While a Cisco ISR can be a CME and firewall simultaneously, the CME is not a gateway for the firewall.

More Information:

Cisco IP Communications Express - Page 69

PrepLogic Question: 11418-102

4. Review Question p. 3

Answers: B

Explanation A. Incorrect - The network layer deals with IP based communication.

Explanation B. Correct - Layer 4 is where UDP is handled.

Explanation C. Incorrect - They physical layer deals with the physical connections and signaling.

Explanation D. Incorrect - The presentation layer deals with the context of the application layer

PrepLogic Question: 11418-103

5. Review Question p. 3

Answers: B, D, E

Explanation A. Incorrect - Session Initialization Protocol (SIP) is an IETF protocol used for call setup.

Explanation B. Correct - Voice over ATM provides for calls to be placed over an ATM network

Explanation C. Incorrect - SCCP is a Cisco proprietary protocol for call setup and control

Explanation D. Correct - Voice over Frame Relay provides for calls to be placed over a Frame Relay network

Explanation E. Correct - Voice over IP allows voice to travel over IP native networks.

PrepLogic Question: 11418-104

6. Review Question p. 3

Answers: B

Explanation A. Incorrect - VoATM operates at layer 2

Explanation B. Correct - VoIP operates at layer 3. This feature allows VoIP to operate over Frame Relay and ATM L2 networks that are configured to understand IP. Also, VoIP operates over typical LANs to go all the way to the desktop.

Explanation C. Incorrect - VoFR operates at layer 2

Explanation D. Incorrect - G.711 is a voice codec standard.

PrepLogic Question: 11418-105

7. Review Question p. 4

Answers: C

Explanation A. Incorrect - A client/server protocol setup (such as SCCP) requires talking to a call agent.

Explanation B. Incorrect - A peer-to-peer only allows calls to be placed from one VoIP end station to another (such as SIP).

Explanation C. Correct - In a client/server configuration, a call agent is used in the middle to assist in making calls.

Explanation D. Incorrect - Peer-to-Peer protocols only allow for end station to end station communication and do not rely on call agents.

PrepLogic Question: 11418-106

8. Review Question p. 4

Answers: D

Explanation A. Incorect - There are only two call processing models, single site and multisite

Explanation B. Incorrect - Each site has it's own CME but site-to-site calls and all external calls are made over the PSTN.

Explanation C. Incorrect - There are only two call processing models, single site and multisite

Explanation D. Correct - Each site has it's own CME and intra-site communication is done across the IP WAN. The PSTN is used only when outbound calls are made.

PrepLogic Question: 11418-107

9. Review Question p. 4

Answers: A, B, D

Explanation A. Correct - Soft Phones are fully supported

Explanation B. Correct - Video Conferencing is supported

Explanation C. Incorrect - IM is not supported on the Unified Communications Express platform

Explanation D. Correct - Hard phones are fully supported.

Explanation E. Incorrect - Cell phones are not supported by the Cisco Unified Communications Express platform

More Information:

Cisco Unified CallManager Express Solution Reference Network Design Guide - Page 1-3

PrepLogic Question: 11418-108

10. Review Question p. 5

Answers: D

Explanation A. Incorrect - Traffic shaping is a supported QOS mechanism.

Explanation B. Incorrect - cRTP is supported and works very well over low-bandwidth links.

Explanation C. Incorrect - LLQ is a popular method of prioritizing voice traffic.

Explanation D. Correct - Beaconing refers to token-ring networks and has nothing to do with QoS.

More Information:

Cisco Unified CallManager Express Solution Reference Network Design Guide - Page 1-4

PrepLogic Question: 11418-109

11. Review Question p. 5

Answers: A, B

Explanation A. Correct - Cisco Unity Express is a distributed voice mail application.

Explanation B. Correct - The auto attendant provides businesses with way to transfer calls to various extensions without human interaction.

Explanation C. Incorrect - The CallManager Express handles call processing

Explanation D. Incorrect - the CME can handle simple web services.

More Information:

Cisco Unified CallManager Express Solution Reference Network Design Guide - Page 1-5

PrepLogic Question: 11418-110

12. Review Question p. 5

Answers: D

Explanation A. Incorrect - This is one of the main benefits of a distributed model. Being able to utilize the IP WAN can dramatically cut down on PSTN costs.

Explanation B. Incorrect - The distributed model is far more scalable than the single-site model.

Explanation C. Incorrect - The WAN bandwidth a site has may be underutilized, adding voice traffic can take advantage of the wasted bandwidth without additional costs.

Explanation D. Correct - Increasing the number of PSTN lines raises the monthly costs to each site. The goal is to reduce the amount of calls placed on the PSTN and to utilize the IP WAN links instead.

More Information:

Cisco Unified CallManager Express Solution Reference Network Design Guide - Page 2-4

PrepLogic Question: 11418-111

13. Review Question p. 6

Answers: B

Explanation A. Incorrect - The full-blown Cisco Unified CallManager scales up to 30,000 phones.

Explanation B. Correct - This is the maximum number of phones for a single CME site.

Explanation C. Incorrect - 50 phones is the minimum number of phones where a designer might want to consider a Unified CallManager. But the CME can also handle many more phones than this.

Explanation D. Incorrect - The CME is rated up to 240 IP phones.

More Information:

Cisco Unified CallManager Express Solution Reference Network Design Guide - Page 2-6

PrepLogic Question: 11418-112

14. Review Question p. 6

Answers: A

Explanation A. Correct - UnifiedMeetingPlace provides a single enterprise-class solution and user environment for voice, Web, and video conferencing.

Explanation B. Incorrect - Cisco TAC is where you would go for Cisco support either online or over the phone. It does not have anything to do with unified communications

Explanation C. Incorrect - Cisco IP IVR is a digital interactive voice system that can be configured in the UC environment. It does not have video or web cababilities.

Explanation D. Incorrect - The IP Communicator is a software phone that users use instead of a hardware based phone. This product does not deal with web or video components.

PrepLogic Question: 11418-113

15. Review Question p. 6

Answers: C

Explanation A. Incorrect - If the WAN cannot have limited bandwidth and cannot fully provide QOS, it's better to place CallManager Express boxes at each remote site.

Explanation B. Incorrect - If most calls are placed over the PSTN then it makes more sense to use an Express solution at the remote sites to provide local PSTN connections.

Explanation C. Correct - While the Cisco Unified Communications Express solution does provide most services that it's big brother offers, it does not have the ability to incorporate the more sophisticated call center application such as the Unified Customer Interaction Analyzer or the Unified Contact Center Management Portal

Explanation D. Incorrect - The beauty of the Unified Communications Express solution is that it can be deployed on a single platform and is easy to deploy. The solution requires no Linux/Windows servers to be maintained.

More Information:

Cisco IP Communications Express - Page 28

PrepLogic Question: 11418-114

16. Review Question p. 7

Answers: C

Explanation A. Incorrect - This layer refers to the network components such as routers/switches.

Explanation B. Incorrect - This layer manages the signaling of voice calls.

Explanation C. Correct - This layer deals with voice applications that enhance the phone call exerience.

Explanation D. Incorrect - This layer deals with voice endpoints such as IP phones.

More Information:

CCNA Voice Quick Reference - Page 7

PrepLogic Question: 11418-182

17. Review Question p. 7

Answers: B

Explanation A. Incorrect - Unified Presence gives the users the ability to better manage presence information on their phones that's typically found in Internet-based chat applications

Explanation B. Correct - This call center application helps to monitor and distribute calls in a complex call center environment.

Explanation C. Incorrect - Unified Meeting Place is a web/video conferencing application.

Explanation D. Incorrect - Unified TelePresence is an advanced high-definition voice/video conference application.

More Information:

CCNA Voice Quick Reference - Page 12

PrepLogic Question: 11418-183

18. Review Question p. 7

Answers: C

Explanation A. Incorrect - The Cisco RSVP agent provides a way to properly deliver voice traffic in timely manner.

Explanation B. Incorrect - This application identifies the location of 911 callers in an emergency

Explanation C. Correct - SRST is a tool that makes it possible for remote sites to continue to function using their phones for most functions when they no longer have connectivity to the central CallManager system.

Explanation D. Incorrect - This application delivers chat-like functionalities to the phone such as: presence-enabled directories, click-to-communicate, and personal communications preferences

PrepLogic Question: 11418-184

Explanations: Chapter 2

1. Review Question p. 8

Answers: D

Explanation A. Incorrect - There are 24 total channels that can be used on a T1 circuit. Students may think that there are only 23 channels because with ISDN there are 23 B channels and 1 D channel for out-of-band signaling.

Explanation B. Incorrect - On a single T1 circuit, there are 24 channels. This means that 24 PSTN voice calls can be made simultaneously.

Explanation C. Incorrect - 24 channels are read 8000 times a second.

Explanation D. Correct - A T1 consists of 24 channels which are read 8,000 times per second.

PrepLogic Question: 11418-115

2. Review Question p. 8

Answers: B

Explanation A. Incorrect - TDM allocates the same timeslot to every channel which can waste bandwidth if the channel is not being used. TDM is often used in PSTN communication such as standard T1 PRI voice circuits.

Explanation B. Correct - Statistical muxing can provide for more efficient use of a circuit when sending packet based data such as TCP or UDP communication.

Explanation C. Incorrect - This type of muxing is to divide up the sound spectrum among various channels. It is most often used in radio communication.

Explanation D. Incorrect - A Primary Rate Interface (PRI) is a physical medium with which to transport 24 POTS circuits from one point to another.

PrepLogic Question: 11418-116

3. Review Question p. 8

Answers: B

Explanation A. Incorrect - Alert signaling refers to the audible ring that notifies the called party of a new incoming call

Explanation B. Correct - Address signaling is what the phone system uses to connect calls.

Explanation C. Incorrect - This type of signaling deals with the management and functionality of voice calls.

Explanation D. Incorrect - Statistical signaling has nothing to do with PSTN.

PrepLogic Question: 11418-117

4. Review Question p. 9

Answers: B, C, D

Explanation A. Incorrect - This is part of address signaling

Explanation B. Correct - Detects the circuit as being either on or off-hook

Explanation C. Correct - Audible tones that indicate the stage where voice calls are at that current time. Tones such as dial-tone and ring-tones are good examples.

Explanation D. Correct - Notifies the called phone of an incoming call in the form of a ring that is delivered by an increase in voltage on the line.

PrepLogic Question: 11418-118

5. Review Question p. 9

Answers: C

Explanation A. Incorrect - This channel carries voice/data traffic (time slot 32)

Explanation B. Incorrect - This channel carries framing info (time slot 1)

Explanation C. Correct - This channel carries signaling (time slot 17)

Explanation D. Incorrect - This channel carries voice/data traffic (timeslot 18)

PrepLogic Question: 11418-119

6. Review Question p. 9

Answers: C

Explanation A. Incorrect - The MAC address is used on Ethernet, TokenRing, ATM and other more advanced networks.

Explanation B. Incorrect - Traditional telephony networks do not use IP.

Explanation C. Correct - a unique telephone number is attached at the CO switch level that uniquely identifies each subscriber line. This number is used to efficiently route the call to a destination subscriber line.

Explanation D. Incorrect - FECN deals with Frame Relay congestion avoidance and has nothing to do with standard PSTN networks.

More Information:

CCNA Voice Quick Reference - Page 20

PrepLogic Question: 11418-120

7. Review Question p. 9

Answers: A, B

Explanation A. Correct - A Private Branch Exchange (PBX) is similar to a TELCO switch but is located on site and owned by the business. It contains telephone numbering plans for all of the phone numbers the business owns. All external calls are routed out the PSTN to the CO.

Explanation B. Correct - This is a smaller version of a PBX. One main difference is that key systems typically allow each phone to be able to use any number from any phone. When a call is placed into a key system, all phones on the system ring and can be answered from any of them.

Explanation C. Incorrect - The CO is where the phone company has their PSTN phone equipment.

Explanation D. Incorrect - POTS stands for "plain old telephone system". It refers to standard analog phone lines.

More Information:

CCNA Voice Quick Reference - Page 21-22

PrepLogic Question: 11418-121

8. Review Question p. 10

Answers: B

Explanation A. Incorrect - A key system is a private switch that routes internal business calls.

Explanation B. Correct

Explanation C. Incorrect - E.164 is an international numbering system standard developed by the ITU.

Explanation D. Incorrect - E&M signaling is a tie-line trunk that connects one PBX to another.

More Information:

CCNA Voice Quick Reference - Page 23

PrepLogic Question: 11418-122

9. Review Question p. 10

Answers: C

Explanation A. Incorrect - The letter "X" stands for any number 0-9

Explanation B. Incorrect - N can be a number 2-9 only

Explanation C. Correct - N is a number from 2 to 9 but it cannot be a 0 or 1

Explanation D. Incorrect - The "N" in the dial number is part of the 10 digit numbering code. It represents a number from 2 to 9.

More Information:

CCNA Voice Quick Reference - Page 25

PrepLogic Question: 11418-124

10. Review Question p. 10

Answers: B

Explanation A. Incorrect - The foreign exchange station (FXS) port is used to connect analog phone sets or fax machines.

Explanation B. Correct - The foreign exchange office (FXO) port is used to connect to an analog PSTN line

Explanation C. Incorrect - The ear and mouth (E&M) port is used to interconnect PBX systems.

Explanation D. Incorrect - The direct inward dial (DID) is used to connect to an analog PSTN service that has DID service on it.

More Information:

Cisco IP Communications Express - Page 172

PrepLogic Question: 11418-125

11. Review Question p. 11

Answers: C, D, E

Explanation A. Incorrect - The country code is not required when dialing a NANP local number

Explanation B. Incorrect - A PBX extension is not part of the NANP

Explanation C. Correct - the line number is the 4 digit number that is specific to the specific PSTN line.

Explanation D. Correct - This 3 digit number is unique for each CO

Explanation E. Correct - This 3 digit code is unique for a specific geographical region.

More Information:

CCNA Voice Quick Reference - Page 25

PrepLogic Question: 11418-126

12. Review Question p. 11

Answers: D

Explanation A. Incorrect - This is the name of the E&M side that connects to the PBX.

Explanation B. Incorrect - A t1 is a PSTN term and does not deal with E&M.

Explanation C. Incorrect - this is the name for the entire trunk link between the PBX and Cisco gateway

Explanation D. Correct - this is the name of the side that the gateway attaches to on an E&M link

More Information:

CCNA Voice Quick Reference - Page 30

PrepLogic Question: 11418-127

13. Review Question p. 11

Answers: B, D

Explanation A. Incorrect - Numbering plans can originate from other countries and even businesses that create their own local numbering plans

Explanation B. Correct - Everyone needs local PSTN numbers and these come from the local PSTN company.

Explanation C. Incorrect - local dial plans can be base on any standard they choose

Explanation D. Correct - The PSTN also regulates and distributes local dial plan numbers.

More Information:

CCNA Voice Quick Reference - Page 29

PrepLogic Question: 11418-128

Explanations: Chapter 3

1. Review Question p. 12

   Answers: A, C, D

   Explanation A. Correct - DSPs mix audio streams from each parties line and transmits the mix

   Explanation B. Incorrect - DSPs are only responsible for voice communication

   Explanation C. Correct - The DSPs change packetized audio from one codec to another

   Explanation D. Correct - Eliminates repetitive noises that are caused by echo

   Explanation E. Incorrect - DSPs are not responsible for tones

   More Information:

   CCNA Voice Quick Reference - Page 41

   PrepLogic Question: 11418-130

2. Review Question p. 12

   Answers: D

   Explanation A. Incorrect - This is the size of RTP/UDP/IP voice packets with compression

   Explanation B. Incorrect - This is the size of RTP/UDP/IP voice packets with compression and checksums

   Explanation C. Incorrect - This is too small to contain all of the information in the RTP, UDP and IP headers combined, without compression

   Explanation D. Correct - This is the correct size of the RTP, UDP and IP headers without using compression. IP = 20 bytes, UDP = 8 bytes and RTP = 12 bytes. 20 + 8 + 12 = 40.

   More Information:

   CCNA Voice Quick Reference - Page 39-40

   PrepLogic Question: 11418-131

3. Review Question p. 13

   Answers: B

Explanation A. Incorrect - 80 bytes for the G.711 and 10 bytes for the G.729 codecs are the sample sizes and not payload size

Explanation B. Correct - Address signaling is what the phone system uses to connect calls.

Explanation C. Incorrect - 4 bytes is the size of the payload with cRTP header compression and checksums

Explanation D. Incorrect - 2 bytes is the size of the payload with cRTP header compression

PrepLogic Question: 11418-132

4. Review Question p. 13

Answers: B

Explanation A. Incorrect - Signaling protocols are a separate packet stream from RTP

Explanation B. Correct - Signaling occurs separate from the actual voice data. They are responsible for call setup, teardown and maintenance. VoIP signaling must succeed in order for a call to be made.

Explanation C. Incorrect - Call teardown is handled by the signaling protocol.

Explanation D. Incorrect - RTP handles the transport of voice traffic

More Information:

CCNA Voice Quick Reference - Page 42

PrepLogic Question: 11418-133

5. Review Question p. 14

Answers: A, C

Explanation A. Correct - The G.711 codec has a bit rate of 64 Kbps

Explanation B. Incorrect - While this is a supported codec, it is not often used. It is typically used on International trunks.

Explanation C. Correct - The G.711 codec has a bit rate of 8 Kbps

Explanation D. Incorrect - H.323 is a VoIP signaling protocol and not a codec

PrepLogic Question: 11418-135

6. Review Question p. 14

Answers: C

Explanation A. Incorrect - SIP is a signaling protocol. It does not transport real-time data such as voice or video.

Explanation B. Incorrect - MGCP is a signaling protocol. It does not transport real-time data such as voice or video.

Explanation C. Correct - Real-Time Transport Protocol (RTP) transports voice/video data on IP networks.

Explanation D. Incorrect - VoIP is a technology for providing voice calls over an IP network. VoIP is not a protocol

More Information:

Cisco IP Communications Express - Page 870

PrepLogic Question: 11418-136

7. Review Question p. 14

Answers: C

Explanation A. Incorrect - SIP is a peer-to-peer protocol

Explanation B. Incorrect - SCCP is a control protocol that communicates between a CallManager and an end-station.

Explanation C. Correct - MGCP is Client/Server plaintext protocol that is between a CallManager (or other call control device) and the voice gateway

Explanation D. Incorrect - RTP is a protocol that is responsible for the transport of voice over IP packets. It is not a signaling protocol

PrepLogic Question: 11418-137

8. Review Question p. 15

Answers: C

Explanation A. Incorrect - Cisco CME supports SCCP

Explanation B. Incorrect - Cisco CME supports SIP

Explanation C. Correct - Cisco CME uses H.323 and SIP to connect to MGCP networks.

Explanation D. Incorrect - Cisco CME supports H.323

More Information:

CCNA Voice Quick Reference - Page 43

PrepLogic Question: 11418-138

9. Review Question p. 15

Answers: A

Explanation A. Correct - RTCP provides feedback on the quality of service being provided by RTP.

Explanation B. Incorrect - SIP is a point-to-point control protocol. It uses RTP and RTCP for transmission of voice packets.

Explanation C. Incorrect - G.711 is a voice codec standard.

Explanation D. Incorrect - SCCP is a point-to-point control protocol. It uses RTP and RTCP for transmission of voice packets.

PrepLogic Question: 11418-139

10. Review Question p. 15

Answers: A, B, E

Explanation A. Correct - DTFM relay is needed to be configured when touch-tone digits are needed to be passed from the IP phone to a device that detects DTFM signals.

Explanation B. Correct - DTFM relay is needed to be configured when touch-tone digits are needed to be passed from the IP phone to a device that detects DTFM signals.

Explanation C. Incorrect - The CallManager GUI is a web based application that does not require DTFM relay

Explanation D. Incorrect - DTFM relay is not needed to configure intercom functionality.

Explanation E. Correct - DTFM relay is needed to be configured when touch-tone digits are needed to be passed from the IP phone to a device that detects DTFM signals.

PrepLogic Question: 11418-140

11. Review Question p. 16

Answers: D

Explanation A. Incorrect - An access code is a unique digit to inform the CallManager that an outside PSTN call is going to be made.

Explanation B. Incorrect - An access code is a unique digit to inform the CallManager that an outside PSTN call is going to be made.

Explanation C. Incorrect - An access code is a unique digit to inform the CallManager that an outside PSTN call is going to be made.

Explanation D. Correct - A typical access code is to used the 9 digit on a phone to get an outside line. This access code is used to inform the CallManager that a call will be going out the PSTN.

PrepLogic Question: 11418-141

12. Review Question p. 16

Answers: B

Explanation A. Incorrect - Letter A shows the wavelength

Explanation B. Correct - The amplitude is the height of the wave from point 0.

Explanation C. Incorrect - The amplitude is a measurement of a sound wave measured from the mean position to an extreme.

Explanation D. Incorrect - The amplitude is a measurement of a sound wave measured from the mean position to an extreme.

PrepLogic Question: 11418-142

13. Review Question p. 17

Answers: B

Explanation A. Incorrect - CDP is used to inform the IP phone what method it should send the voice traffic.

Explanation B. Correct - CDP packets are sent out the switchport to the phone to inform the switch on what method the switch expects voice traffic to be sent. The choices are:

Voice VLAN tagged with a Layer 2 CoS priority value
Access VLAN tagged with a Layer 2 CoS priority value
Access VLAN, untagged

Explanation C. Incorrect - CDP is used to inform the IP phone what method it should send the voice traffic.

Explanation D. Incorrect - CDP is used to inform the IP phone what method it should send the voice traffic.

PrepLogic Question: 11418-143

14. Review Question p. 17

Answers: B

Explanation A. Incorrect - Large organizations are better suited to use a full-blown PBX that assigns separate extensions to each phone.

Explanation B. Correct - Small businesses typically do not need to make calls internally as it is likely just as easy to walk over to another employee. They also likely want to be able to answer any PSTN line on any phone.

Explanation C. Incorrect - Key systems are used when personal extensions are not needed. It has nothing to do with voice mail.

Explanation D. Incorrect - Key systems are used when personal extensions are not needed. It has nothing to do with voice mail.

More Information:

Cisco IP Communications Express - Page 117

PrepLogic Question: 11418-144

Explanations: Chapter 4

1. Review Question p. 18

   Answers: B

   Explanation A. Incorrect - dial plans set paths for various dial number combinations.

   Explanation B. Correct - Dial plans do not deal with the setup of voice calls.

   Explanation C. Incorrect - In some cases, dial plans manipulate dialed digits before routing calls

   Explanation D. Incorrect - Dial plans assign directory numbers to all endpoints

   Explanation E. Incorrect - Dial plans can assign different groups of devices to different classes of services.

   More Information:

   Implementing Cisco Unified CallManager - Page 4-5

   PrepLogic Question: 11418-147

2. Review Question p. 18

   Answers: D

   Explanation A. Incorrect - The third digit must be either 4,5,6 or 9

   Explanation B. Incorrect - This number is only 6 digits in length

   Explanation C. Incorrect - The third digit must be either 4,5,6 or 9

   Explanation D. Correct - The third digit is a 4,5,6 or 9 and the 6th digit is a wildcard 0-9 number

   PrepLogic Question: 11418-148

3. Review Question p. 18

   Answers: A

   Explanation A. Correct - Because all digits are explicitly configured, no digits will be forwarded.

   Explanation B. Incorrect - Because all digits are explicitly configured, no digits will be forwarded.

Explanation C. Incorrect - Because all digits are explicitly configured, no digits will be forwarded.

Explanation D. Incorrect - Because all digits are explicitly configured, no digits will be forwarded.

PrepLogic Question: 11418-149

4. Review Question p. 19

Answers: D

Explanation A. Incorrect - The string 9. Specifies that the dial peer supports only a 2 digit string. The "." Is any number 0-9

Explanation B. Incorrect - The string 9# is not valid as the "#" wildcard is only for DTMF tones.

Explanation C. Incorrect - The string 9$ is not valid as the "$" wildcard is used in translation rules and not dial peers

Explanation D. Correct - The 9T string means that it accepts 9 as the first digit. The "T" wildcard matches any number of successive digits. This would allow a user to dial local, long distance and international numbers with a single dial-string.

Explanation E. Incorrect - The string 9% specifies that the dial peer supports the 9 digit first. The "%" wildcard means that it accepts the preceding digit any number of times. Because the only digit preceding the "%" wildcard is 9, then this string would not work.

PrepLogic Question: 11418-150

5. Review Question p. 19

Answers: A, C

Explanation A. Correct - The gateways convert communication from one type to another.

Explanation B. Incorrect - Voice gateways are needed within every voice solution.

Explanation C. Correct - The gateways can terminate POTs lines and converts the signals from digital systems to analog lines and visa versa.

Explanation D. Incorrect - The phone directory is stored within the CallManager system

More Information:

CCNA Voice Quick Reference - Page 44

PrepLogic Question: 11418-151

6. Review Question p. 20

Answers: A

Explanation A. Correct - This command after a dial string forces the gateway to send the static digits "555" to the PSTN instead of just the wildcard numbers.

Explanation B. Incorrect - By default, only the wildcard digits will be forwarded to the PSTN

Explanation C. Incorrect - The prefix command would be used to add specific digits to the front of the dialed string before it is forwarded to the telephony interface. In this case, the number 7.

Explanation D. Incorrect - This command tells you how many digits to forward to the PSTN. In this example, the last 3 digits of the entire string would be forwarded.

PrepLogic Question: 11418-152

7. Review Question p. 20

Answers: C

Explanation A. Incorrect - The hunt group order is configured using the "preference" command

Explanation B. Incorrect - The purpose of a hunt group is to forward the call to another extension if the first phone in the hunt group is busy.

Explanation C. Correct - The hunt group forwards a call to a different extension if the first number in the hunt group is busy. Hunt group extension numbers are ordered using the "preference" command.

Explanation D. Incorrect - The hunt group order is configured using the "preference" command

PrepLogic Question: 11418-153

8. Review Question p. 20

Answers: D

Explanation A. Incorrect - Hair-pinning is when a call originates and terminates on the SAME gateway.

Explanation B. Incorrect - A dial-peer is associated with each call leg but not the call leg itself.

Explanation C. Incorrect - QOS is a method to prioritize time sensitive traffic. It does not have anything to do with call legs.

Explanation D. Correct - Call legs are associated to dial peers. Dial peers direct either POTS to IPT or IPT to POTS conversion. This transfer is the call leg termination point.

PrepLogic Question: 11418-154

9. Review Question p. 21

Answers: B, C

Explanation A. Incorrect - QOS is only configured on IPT connections. PSTN networks do not require QOS

Explanation B. Correct - A ITSP link is usually a SIP or H.323 connection over an IP network. Once the voice packets get to the service provider network, that voice traffic is sent to the lowest cost route out PSTN lines. This helps to reduce toll charges.

Explanation C. Correct - PSTN gateway configurations are often more complex because of the fact that the administrator has to configure varying long distance and national dial-strings. With an ITSP configuration, that setup is handed on the service provider side.

Explanation D. Incorrect - A voice gateway is needed to connect the local Unified Communications system to the service provider. This is typically a SIP or H.323 connection.

More Information:

CCNA Voice Quick Reference - Page 48

PrepLogic Question: 11418-155

10. Review Question p. 21

Answers: B

Explanation A. Incorrect - a hunt group is a separate configuration option not related to PLAR

Explanation B. Correct - PLAR creates a "hotline" number. When an end user picks up a phone, a single number is automatically dialed.

Explanation C. Incorrect - The configuration of a number to an endpoint is statically configured on a gateway.

Explanation D. Incorrect - The purpose of PLAR is so the end user simply takes the phone off-hook and a specific number is automatically dialed.

More Information:

CCNA Voice Quick Reference - Page 55

PrepLogic Question: 11418-156

11. Review Question p. 21

Answers:

Explanation A. Incorrect - The session target points the dial peer to the call-leg target. In this example, the call-leg is an IPv4 voice gateway.

Explanation B. Incorrect - The session target points the dial peer to the call-leg target. In this example, the call-leg is an IPv4 voice gateway.

Explanation C. Incorrect - To define the codec of a dial peer, use the "codec" command.

PrepLogic Question: 11418-157

12. Review Question p. 22

Answers: C

Explanation A. Incorrect - The router defaults to variable-length matching, which means that as long as the left-justified digits in the dial string match the configured pattern in the dial peer, any digits beyond the configured pattern are ignored for the purposes of matching. So even though dial-peer 1 matches dial peer 5551212, it also matches dial peers 2 and 3 making answer 3 the correct choice.

Explanation B. Incorrect - The router defaults to variable-length matching, which means that as long as the left-justified digits in the dial string match the configured pattern in the dial peer, any digits beyond the configured pattern are ignored for the purposes of matching. So even though dial-peer 1 matches dial peer 5551212, it also matches dial peers 2 and 3 making answer 3 the correct choice.

Explanation C. Correct - By default, variable-length matching is used. The digits 555 are matched by all three dial peers. The digits past the first three are ignored.

Explanation D. Incorrect - The router defaults to variable-length matching, which means that as long as the left-justified digits in the dial string match the configured pattern in the dial peer, any digits beyond the configured pattern are ignored for the purposes of matching. So even though dial-peer 1 matches dial peer 5551212, it also matches dial peers 2 and 3 making answer 3 the correct choice.

PrepLogic Question: 11418-158

13. Review Question p. 22

Answers: A

Explanation A. Correct - The "num-exp" tells the router to use the expanded number in situations that require the full E.164 number.

Explanation B. Incorrect - The correct command is "num-exp"

Explanation C. Incorrect - The destination pattern associates a dialed string with a telephony device.

Explanation D. Incorrect - The forward-digits command controls the number of digits that are stripped before the dialed string is passed on.

PrepLogic Question: 11418-159

14. Review Question p. 23

Answers: C

Explanation A. Incorrect - This command is configured on the voice gateway. The mgcp call-agent IP address is the IP address of the Cisco CallManager.

Explanation B. Incorrect - The MGCP call-agent points to the Cisco CallManager.

Explanation C. Correct - The CallManager is the MGCP call-agent in a unified architecture.

PrepLogic Question: 11418-186

15. Review Question p. 23

Answers: D

Explanation A. Incorrect - DSPs help to transcode voice calls between analogue and digitial. They are configured on the voice gateway.

Explanation B. Incorrect - DSPs help to transcode voice calls between analogue and digitial and between codecs. They are configured on the voice gateway.

Explanation C. Incorrect - An aAP is a wireless device and has nothing to do with DSPs.

Explanation D. Correct - The DSPs assist in voice transcoding, which is done on the voice gateway.

PrepLogic Question: 11418-187

16. Review Question p. 23

Answers: C

Explanation A. Incorrect - Transcoding is performed in hardware if the two streams have the same codec and packetization time.

Explanation B. Incorrect - Transcoding is performed in hardware if the two streams have the same codec and packetization time.

Explanation C. Correct - This type of transcoding is done in software because it does not require the high speed processing power of the DSPs.

Explanation D. Incorrect - A DSP is needed to transcode between different codec's in order to get the voice packets sent to the end user in time.

PrepLogic Question: 11418-188

Explanations: Chapter 5

1. Review Question p. 24

Answers: B, C, E

Explanation A. Incorrect - The file specifies the end phones MAC address and not the IP address

Explanation B. Correct - This file is loaded into NVRAM

Explanation C. Correct - This file specifies the IP address, port, firmware version, locale, directory URL and other information that can be unique to each phone

Explanation D. Incorrect - The default configuration file does not contain the MAC address

Explanation E. Correct - The configuration information is used when there is not a more specific configuration in the SEPAAABBBBCCCC.cnf.xml file.

More Information:

CCNA Voice Quick Reference - Page 64

PrepLogic Question: 11418-161

2. Review Question p. 24

Answers: D

Explanation A. Incorrect - IP addresses contained within the ip dhcp excluded-address range are not handed out to ANY DHCP client.

Explanation B. Incorrect - IP addresses contained within the ip dhcp excluded-address range are not handed out to ANY DHCP client.

Explanation C. Incorrect - The excluded IP addresses can be used for any number of reasons, not just for the voice gateway.

Explanation D. Correct - The excluded-address command removes IP addresses from the DHCP scope so they will never be handed out to DHCP clients.

PrepLogic Question: 11418-162

3. Review Question p. 25

Answers: C

Explanation A. Incorrect - DHCP dynamically assigns IP addresses. It does not synchronize system clocks

Explanation B. Incorrect - IP is a network layer addressing protocol. It does not synchronize system clocks

Explanation C. Correct - Network Time Protocol (NTP) is designed to synchronize time on networked devices. NTP runs over UDP port 123

Explanation D. Incorrect - CDP is a Cisco proprietary protocol that identifies directly connected Cisco devices

PrepLogic Question: 11418-165

4. Review Question p. 25

Answers: A, B

Explanation A. Correct - This is a great way to get around the limited public IP addresses as the IP phones likely do not need a public IP address.

Explanation B. Correct - by separating the voice traffic from the data, it allows for easier QOS configuration and implementation.

Explanation C. Incorrect - Cisco IP phones allow both the voice and data VLANs to be trunked. A single Ethernet connection is needed from the switch. The PC can then plug into the IP phone.

Explanation D. Incorrect - Adding a second VLAN for voice does not impact STP.

PrepLogic Question: 11418-166

5. Review Question p. 25

Answers: C

Explanation A. Incorrect - This is not the lowest bandwidth along the path

Explanation B. Incorrect - This is not the lowest bandwidth along the path

Explanation C. Correct - Segment 3 is only 512 Kbps and is by far the lowest bandwidth from phone to phone. This is the potential bottleneck,

Explanation D. Incorrect - This is not the lowest bandwidth along the path

PrepLogic Question: 11418-168

6. Review Question p. 26

Answers: C

Explanation A. Incorrect - Because the phones are receiving IP addresses from the DHCP server, we know they are connected to the network.

Explanation B. Incorrect - DHCP option 43 is for vendor specific information and is not used with the Cisco CallManager

Explanation C. Correct - DHCP option 150 is used so the IP phone knows what IP address the CallManager is located. Once it knows the CallManager IP, it initiates a TFTP connection to download the configuration files.

Explanation D. Incorrect - The CallManager can reside on a separate subnet from the IP phones on a network.

PrepLogic Question: 11418-169

7. Review Question p. 26

Answers: D, E

Explanation A. Incorrect - The switch will not use the DSCP values unless the "trust" command is issued.

Explanation B. Incorrect - The switch will not use the DSCP values unless the "trust" command is issued.

Explanation C. Incorrect - The switch will classify traffic using NBAR if DSCP values are not trusted.

Explanation D. Correct - When the switch is not told to trust the DSCP value, NBAR is used to classify traffic.

Explanation E. Correct - This 3 digit code is unique for a specific geographical region.

More Information:

CCNA Voice Quick Reference - Page 73

PrepLogic Question: 11418-171

8. Review Question p. 26

Answers: B, D

Explanation A. Incorrect - Layer 3 classification is based on DSCP values

Explanation B. Correct - DSCP is used to classify Network layer packets. DSCP values range between 0 and 63.

Explanation C. Incorrect - Layer 2 classification is based on CoS values

Explanation D. Correct - CoS values range between zero for low priority and seven for high priority.

Explanation E. Incorrect - IP Precedence is a layer 3 way of classifying packets. IP precedence values range between 0 for low priority and 7 for high priority. Because the number of classification ranges is limited, most engineers choose to use DSCP over IP Precedence to classify traffic.

PrepLogic Question: 11418-172

9. Review Question p. 27

Answers: A, B, E, F

Explanation A. Correct - QoS reduces delay for time-sensitive traffic by prioritizing the traffic so it is sent in a timely mannar.

Explanation B. Correct - Jitter is variation in delay. QoS is configured to eliminate jitter.

Explanation C. Incorrect - DSCP is a way to classify network layer traffic. While it's part of QoS, it's not a component that QoS is attempting to regulate.

Explanation D. Incorrect - CoS is a way to classify data-link layer traffic. While it's part of QoS, it's not a component that QoS is attempting to regulate.

Explanation E. Correct - QoS is configured to set bandwidth requirements for traffic based on the importance.

Explanation F. Correct - QoS is configured to eliminate packet loss for traffic that is judged to be more important.

PrepLogic Question: 11418-173

10. Review Question p. 27

Answers: B

Explanation A. Incorrect - The port is a layer 2 connection and will understand and trust CoS values.

Explanation B. Correct - the command "mls qos trust cos" was configured on the interface.

Explanation C. Incorrect - COS override is disabled on the interface.

Explanation D. Incorrect - The trust state for the switchport is set to trust CoS coming from an IP phone.

PrepLogic Question: 11418-174

Explanations: Chapter 6

1. Review Question p. 28

   Answers: D

   Explanation A. Incorrect - The configuration assistant provides GUI base network reporting tools

   Explanation B. Incorrect - Drag and drop software updates simplify updates

   Explanation C. Incorrect - The configuration assistant supports multiple views.

   Explanation D. Correct - The Cisco Configuration assistant does not provide load balancing.

   Explanation E. Incorrect - The configuration assistant has several tools to assist in troubleshooting.

   Explanation F. Incorrect - This powerful GUI configuration tool assists in configuring all the mentioned network functions.

   More Information:

   CCNA Voice Quick Reference - Page 119

   PrepLogic Question: 11418-175

2. Review Question p. 28

   Answers: A

   Explanation A. Correct - A simple visual queue notifies the administrator of an error.

   Explanation B. Incorrect - The system notifies the administrator in red text.

   Explanation C. Incorrect - a mis-configuration will not generate a log message.

   Explanation D. Incorrect - The notification is visual and not audible.

   More Information:

   CCNA Voice Quick Reference - Page 130

   PrepLogic Question: 11418-176

3. Review Question p. 29

   Answers: B, C

Explanation A. Incorrect - Broadvoice is not a default service provider SIP template on the SBCS. Additional service providers are configured using the default SIP trunks.

Explanation B. Correct - AT&T is one of the two default SIP trunk providers on the SBCS.

Explanation C. Correct - CBeyond is one of the two default SIP trunk providers on the SBCS.

Explanation D. Incorrect - Verizon is not a default service provider SIP template on the SBCS. Additional service providers are configured using the default SIP trunks.

More Information:

CCNA Voice Quick Reference - Page 134

PrepLogic Question: 11418-177

4. Review Question p. 29

Answers: C

Explanation A. Incorrect - The CCA uses CDP to discover SBCS equipment.

Explanation B. Incorrect - The CCA uses CDP to discover SBCS equipment.

Explanation C. Correct - CDP is used to find all Cisco SBCS equipment that can be configured by the CCA.

Explanation D. Incorrect - The CCA uses CDP to discover SBCS equipment.

PrepLogic Question: 11418-178

5. Review Question p. 29

Answers: C

Explanation A. Incorrect - The CCA is a Windows executable application (fat client). You need to have the following system requirements:

Processor speed: 1 GHz
DRAM: 512 MB minimum; 1024 MB recommended for better performance
Hard-disk space: 150 MB for Cisco Configuration Assistant alone; 300 MB recommended

Explanation B. Incorrect - The CCA is a Windows executable application (fat client). You need to have the following system requirements:

Processor speed: 1 GHz
DRAM: 512 MB minimum; 1024 MB recommended for better performance
Hard-disk space: 150 MB for Cisco Configuration Assistant alone; 300 MB recommended

Explanation C. Correct - The CCA is a Windows executable application (fat client). You need to have the following system requirements:

Processor speed: 1 GHz
DRAM: 512 MB minimum; 1024 MB recommended for better performance
Hard-disk space: 150 MB for Cisco Configuration Assistant alone; 300 MB

Explanation D. Incorrect - The CCA is a Windows XP/Vista executable application (fat client). You need to have the following system requirements:

Processor speed: 1 GHz
DRAM: 512 MB minimum; 1024 MB recommended for better performance
Hard-disk space: 150 MB for Cisco Configuration Assistant alone; 300 MB

PrepLogic Question: 11418-179

6. Review Question p. 30

Answers: B

Explanation A. Incorrect - The Voice Features tab configures various voice features such as MOH, paging, hunt groups and others.

Explanation B. Correct - This screen is where you can configure various CME features.

Explanation C. Incorrect - The Voice Features tab configures various voice features such as MOH, paging, hunt groups and others.

Explanation D. Incorrect - The Voice Features tab configures various voice features such as MOH, paging, hunt groups and others.

More Information:

CCNA Voice Quick Reference - Page 135

PrepLogic Question: 11418-180

7. Review Question p. 30

Answers: A

Explanation A. Correct - The Maintenance tab contains the section to backup and restore configuration files.

Explanation B. Incorrect - The Telephony menu contains the configuration options specific to voice operations.

Explanation C. Incorrect - The Security menu contains the configuration options specific to security operations including firewalling.

Explanation D. Incorrect - The Configure menu contains the various options to configure SBCS equipment.

More Information:

CCNA Voice Quick Reference - Page 146

PrepLogic Question: 11418-181

8. Review Question p. 30

Answers: B

Explanation A. Incorrect - The UC 500 is capable of delivering Power over Ethernet using the 802.1af standard.

Explanation B. Correct - 802.1af is a universal PoE standard.

Explanation C. Incorrect - Non-Cisco IP phones are fully compatible with the UC 500 series hardware and applications.

Explanation D. Incorrect - The UC 500 is capable of delivering Power over Ethernet using the 802.1af standard.

PrepLogic Question: 11418-185

Explanations: Chapter 7

1. Review Question p. 31

   Answers: B

   Explanation A. Incorrect - The country code for Germany is DE.

   Explanation B. Correct - Both the network-locale and user-locale are under the telephony-service configuration. The country code for Germany is DE.

   Explanation C. Incorrect - Both the network-locale and user-locale are under the telephony-service configuration.

   Explanation D. Incorrect - Both the network-locale and user-locale are under the telephony-service configuration.

   PrepLogic Question: 11418-189

2. Review Question p. 32

   Answers: D

   Explanation A. Incorrect - The "huntstop channel" command allows only 1 call to go to the ephone-dn. The second channel on each phone is used for placing calls on hold or for outbound calls.

   Explanation B. Incorrect - The "huntstop channel" command allows only 1 call to go to the ephone-dn. The second channel on each phone is used for placing calls on hold or for outbound calls.

   Explanation C. Incorrect - The "huntstop channel" command allows only 1 call to go to the ephone-dn. The second channel on each phone is used for placing calls on hold or for outbound calls.

   Explanation D. Correct - The "huntstop channel" command will only allow a single call. The first call will be sent to ephone 1 because of the lower preference. The second call will be sent to ephone 2. The third call will receive a busy signal.

   PrepLogic Question: 11418-190

3. Review Question p. 32

   Answers: A

   Explanation A. Correct - Ephone 4 is the physical configuration setup of the phone. It must include the MAC address of the physical phone to work. The button 5:3 command applies ephone-dn 3 to button 5.

Explanation B. Incorrect - This would apply ephone-dn 3 to button 4 on phone 5.

Explanation C. Incorrect - This would apply ephone-dn 5 to button 4 on phone 3.

Explanation D. Incorrect - No ephone-dn is applied to a button with this example.

PrepLogic Question: 11418-191

4. Review Question p. 33

Answers: B

Explanation A. Incorrect - Button 4 is configured for ephone-dn 1

Explanation B. Correct - The s in the configuration means that the ring will be silent. There still will be a visual alert on the phone lamp.

Explanation C. Incorrect - The s in the configuration means that the ring will be silent.

Explanation D. Incorrect - Button 4 is only configured to ring when extension 233 is dialed. The ring will be silent.

PrepLogic Question: 11418-192

5. Review Question p. 34

Answers: B

Explanation A. Incorrect - The "dual-line" command allows for transfer of calls on the line. This will not help associate the ephone 1 configuration to the new 7961 phone

Explanation B. Correct - The MAC address is needed in the ephone 1 configuration to associate the 7961 phone to the config.

Explanation C. Incorrect - The "max-ehpones" configures the maximum number of Cisco IP phones that can be supported by a router.

Explanation D. Incorrect - The "ephone-dn-template" will import the template dn configuration of template 1. This will not help associate the ephone 1 configuration to the new 7961 phone.

PrepLogic Question: 11418-193

6. Review Question p. 34

Answers: C

Explanation A. Incorrect - SCCP communicates between the CallManager and the IP phones for call setup and teardown. It uses TCP 2000 by default.

Explanation B. Incorrect - Option 150 is a DHCP option to allow the IP phones to receive the IP address of the CallManager so they can download the configuration files via TFTP. It is not a port/protocol for SCCP.

Explanation C. Correct - SCCP runs along TCP 2000

Explanation D. Incorrect - SCCP communicates between the CallManager and the IP phones for call setup and teardown. It uses TCP 2000 by default.

More Information:

CCNA Voice Quick Reference - Page 81

PrepLogic Question: 11418-194

7. Review Question p. 34

Answers: A

Explanation A. Correct - The ip source-address command defines the IP address that will be used as the source for SCCP messages. Without this command the new phones do not know what CallManager to communicate with

Explanation B. Incorrect - This command actually disables autoregistration on a CME.

Explanation C. Incorrect - This command manually configures the MAC address of the IP phone onto the CME.

Explanation D. Incorrect - This command lets an engineer view the MAC addresses of the phones that attempted to connect to the CME but were blocked due to autoregistration being disabled.

More Information:

CCNA Voice Quick Reference - Page 81

PrepLogic Question: 11418-195

8. Review Question p. 35

Answers: B

Explanation A. Incorrect - This command assigns ephone-dn extensions to new phones.

Explanation B. Correct - This command is great for the deployment of phones in situations where you do not need to assign specific extensions to users.

Explanation C. Incorrect - The IP source-address command lets phones know where to send SCCP messages.

Explanation D. Incorrect - NTP is a way to synchronize clocks on network devices. It has nothing to do with auto-assign.

More Information:

CCNA Voice Quick Reference - Page 83

PrepLogic Question: 11418-196

9. Review Question p. 35

Answers: C, D, E

Explanation A. Incorrect - A restart can change the extension number on a phone without a full reset.

Explanation B. Incorrect - A DCHP renew automatically happens without a reset

Explanation C. Correct - When modifying a user URL's, a reset is required for changes to be seen on the IP phone.

Explanation D. Correct - A reset is necessary when changing user/network locals on the CME network.

Explanation E. Correct - Updating firmware requires the use of the reset command

More Information:

CCNA Voice Quick Reference - Page 87

PrepLogic Question: 11418-197

10. Review Question p. 35

Answers: C

Explanation A. Incorrect - This forwards all calls to the configured number

Explanation B. Incorrect - This forwards all calls only when the phone is off-hook

Explanation C. Correct - This command forwards calls after a predetermined number of seconds.

Explanation D. Incorrect - This configures the maximum number of digits a call forward number can have. It helps prevent users from entering long-distance and international numbers which can be costly.

More Information:

[CCNA Voice Quick Reference - Page 88](#)

PrepLogic Question: 11418-198

11. Review Question p. 36

Answers: B

Explanation A. Incorrect - The first command sets the system clock manually but does not setup NTP. The second command only sets the NTP interface.

Explanation B. Correct - This command properly configures the CME to synchronize time off the NTP server located at 10.1.1.1

Explanation C. Incorrect - This command only sets the NTP interface.

Explanation D. Incorrect - This command sets the CME router as the stratum 2 level NTP server.

PrepLogic Question: 11418-199

12. Review Question p. 36

Answers: D

Explanation A. Incorrect - If this were true, none of the phones would register.

Explanation B. Incorrect - Since the 7971 phones were brought up first, then they would have registered properly and the other phone models would not have been able to register.

Explanation C. Incorrect - The max-dn limitation has nothing to do with registering phones.

Explanation D. Correct - If firmware for a specific phone is not available, they cannot register.

PrepLogic Question: 11418-200

13. Review Question p. 36

Answers: A

Explanation A. Correct - The source address is typically on the same network subnet as your IP phones. You can change the port that the CME communicates on. TCP 2000 is the default.

Explanation B. Incorrect - The ip source-address command must be configured in the (config-telephony) command section.

Explanation C. Incorrect - The correct command is "ip source-address"

Explanation D. Incorrect - The ip source-address command is configured within the telephony service configuration.

PrepLogic Question: 11418-201

14. Review Question p. 37

Answers: B

Explanation A. Incorrect - The reset all command must be performed under telephony-service.

Explanation B. Correct - The reset all command must be performed under telephony-service.

Explanation C. Incorrect - In order to reset the phones, the administrator must be in config-telephony mode.

Explanation D. Incorrect - While this will reset the IP phones, it will also reboot the Cisco CME.

PrepLogic Question: 11418-202

15. Review Question p. 37

Answers: C

Explanation A. Incorrect - Ephone 3 is not registered.

Explanation B. Incorrect - Three of the four phones are properly connected. You can see this by the "REGISTERED" statement in the first line of ephone 1,2 and 4.

Explanation C. Correct - Three of the four phones are properly connected. You can see this by the "REGISTERED" statement in the first line of ephone 1,2 and 4.

Explanation D. Incorrect - While it's true that no calls are active, that does not mean that the phones are not properly registered. It simply means that no phone calls were made while the show ephone command was executed.

PrepLogic Question: 11418-203

16. Review Question p. 38

Answers: A

Explanation A. Correct - By default, a router sets a max-ehpone and max-dn to 0. If you want to configure ephones and dial-numbers, you must set these numbers manually.

Explanation B. Incorrect - This command will begin configuring ephone 1. Before you can do this, you must set the max-ephones to a number higher than 0. If you do not, you receive a message stating that the max-ephones limit has been reached.

Explanation C. Incorrect - This command will begin configuring ephone-dn 1. Before you can do this, you must set the max-dn to a number higher than 0. If you do not, you receive a message stating that the max-dn limit has been reached.

Explanation D. Incorrect - The max-ehpones and max-dn commands are found in the telephony-service sub command structure.

PrepLogic Question: 11418-204

17. Review Question p. 38

Answers: A, C, E

Explanation A. Correct - Users can use their handsets to record personal greetings using TUI.

Explanation B. Incorrect - TUI does not have the ability for users to modify extension numbers.

Explanation C. Correct - Special vacation and/or emergency voice notifications can be made with TUI.

Explanation D. Incorrect - Digit manipulation is not possible with TUI.

Explanation E. Correct - Remote directory lookup can be used with TUI although it is disabled by default.

PrepLogic Question: 11418-213

Explanations: Chapter 8

1. Review Question p. 39

Answers: D

Explanation A. Incorrect - The ephone configuration is the physical portion of the phone configuration and does not deal with the message waiting indicator.

Explanation B. Incorrect - The ephone template helps to standardize basic configuration options that are the same on every IP phone. It does not contain MWI configuration options.

Explanation C. Incorrect - The h323 gatekeeper command specifies the IP address of the h.323 gateway.

Explanation D. Correct - you can configure specific functions of the specific dial number such as voice-mail ports, and the message waiting indicator by entering ephone-dn mode.

PrepLogic Question: 11418-205

2. Review Question p. 39

Answers: B, C

Explanation A. Incorrect - The proper format for the files is .wav

Explanation B. Correct - The AvT is a telephone-based application that allows CME administrators to record or delete prompts and announcements without using a PC or sound-editing software.

Explanation C. Correct - Administrators can use either the GUI or CLI to upload the files.

Explanation D. Incorrect - There is no microphone input jack on the CME.

Explanation E. Incorrect - The proper format for the files is .wav

PrepLogic Question: 11418-206

3. Review Question p. 40

Answers: D

Explanation A. Incorrect - While this is a correct way to terminate the call, you can also use the "ccn call terminate call" command.

Explanation B. Incorrect - While this is a correct way to terminate the call, you can also use the "ccn call terminate port" command.

Explanation C. Incorrect - This is the correct command but the "Call Impl Id" number should be used.

Explanation D. Correct - Either of these methods will successfully terminate the call.

Explanation E. Incorrect - This is the correct command but the "Port Impl ID" number should be used.

PrepLogic Question: 11418-207

4. Review Question p. 40

Answers: B

Explanation A. Incorrect - While you can perform administration tasks using GUI, you cannot upgrade the software

Explanation B. Correct - This is the only way to upgrade the Unity Express software.

Explanation C. Incorrect - You cannot upgrade the Unity Express software using the GUI.

Explanation D. Incorrect - Neither of these methods can be used to upgrade the Unity Express system software.

PrepLogic Question: 11418-208

5. Review Question p. 40

Answers: D

Explanation A. Incorrect - The CUE provides mailbox and message stats.

Explanation B. Incorrect - The CUE provides backup and restore history

Explanation C. Incorrect - The CUE provides memory and CPU reporting via command line.

Explanation D. Correct - The Cisco CUE does not provide interface statistics system reporting.

Explanation E. Incorrect - The CUE provide call history reporting

PrepLogic Question: 11418-209

6. Review Question p. 41

Answers: A, B, D

Explanation A. Correct - A static route is necessary to get packets routed to Unity properly.

Explanation B. Correct - An IP address must be assigned to Unity so the router can send packets correctly to Unity.

Explanation C. Incorrect - While DHCP is often used, it is not one of the minimum router configuration requirements.

Explanation D. Correct - This is necessary to be able to communicate with Unity and other IP devices on the network.

Explanation E. Incorrect - Unity does not support EIGRP.

More Information:

Cisco IP Communications Express - Page 349

PrepLogic Question: 11418-210

7. Review Question p. 41

Answers: C

Explanation A. Incorrect - The only way to access log files is though the CLI.

Explanation B. Incorrect - While you can access the log files while connected to the console port, you can also see log files by using Telnet/SSH access to the CME.

Explanation C. Correct - There are several ways to access the CLI. Any of these methods will give you access to see the logs.

Explanation D. Incorrect - You cannot see log files on the GUI.

PrepLogic Question: 11418-211

8. Review Question p. 41

Answers: B

Explanation A. Incorrect - A reboot of the CUE module does not require a reboot of the router. The Cisco Unity Express module and the router can be rebooted independently of each other.

Explanation B. Correct - A reboot of the CUE module does not require a reboot of the router. The Cisco Unity Express module and the router can be rebooted independently of each other.

Explanation C. Incorrect - The CUE can be rebooted if necessary and will not lose it's current configuration.

Explanation D. Incorrect - The CUE module can be rebooted without the need to reboot the entire router.

PrepLogic Question: 11418-212

9. Review Question p. 42

Answers: B

Explanation A. Incorrect - The SIP proxy resides on the CME and must be configured to point to the CUE for proper setup.

Explanation B. Correct- The SIP proxy resides on the CME and must be configured to point to the CUE for proper setup.

Explanation C. Incorrect - Cisco Unity Express uses the SIP protocol for communication. The SIP proxy resides on the CME and must be configured to point to the CUE for proper setup.

Explanation D. Incorrect - Cisco Unity Express uses the SIP protocol for communication.

PrepLogic Question: 11418-214

10. Review Question p. 42

Answers: C

Explanation A. Incorrect - The NM-CUE uses 1 network module slot

Explanation B. Incorrect - Only the AIM-CUE does not use an interface slot

Explanation C. Correct - The AIM-CUE does not use a network module slot. Instead, it connects to the inside of the router.

Explanation D. Incorrect - Only the NM-CUE uses a network module slot.

More Information:

🌐 Cisco IP Communications Express - Page 16

PrepLogic Question: 11418-215

11. Review Question p. 43

Answers: D

Explanation A. Incorrect - The user must press the number 0 on their telephone keypad.

Explanation B. Incorrect - The zeronumberout command redirects users to extension 1234 when the "0" key is pressed on the telephone keypad.

Explanation C. Incorrect - The zeronumber out command is used when a caller does not want to leave a voicemail and instead would like to talk to an operator at extension 1234.

Explanation D. Correct - The zeronumberout command specifies the extension where a caller is routed when the caller presses "0' to reach an operator after being transferred to a subscriber's mailbox.

PrepLogic Question: 11418-216

12. Review Question p. 43

Answers: B

Explanation A. Incorrect - Having a single service-module ip address is correct.

Explanation B. Correct - Using the loopback 0 interface is the best option because it never goes down. Once you associate an interface to the service-engine, the interface comes up and you can access it properly.

Explanation C. Incorrect - This is not a valid IOS command.

Explanation D. Incorrect - The Service-Engine is already in a no-shut state. It just needs to have an interface assigned to it to bring it up.

PrepLogic Question: 11418-217

13. Review Question p. 44

Answers: A

Explanation A. Correct - The tutorial command enables the mailbox tutorial program when the telephone subscriber logs in to the voice-mail system for the first time. To disable this option, use the "no tutorial" command.

Explanation B. Incorrect - The tutorial is enabled on all mailboxes by default. To disable this option, use the "no tutorial" command.

Explanation C. Incorect - Once the mailbox owner goes through the tutorial for the first time, it will not play again. The tutorial is enabled on all mailboxes by default. To disable this option, use the "no tutorial" command.

Explanation D. Incorrect - The Voicemail tutorial is for general users that have a mailbox. The tutorial assists the user to setup various functions of their specific mailbox.

PrepLogic Question: 11418-218

14. Review Question p. 45

Answers: C

Explanation A. Incorrect - The instructions state that read-only access should be configured. This configuration allow for read-write access.

Explanation B. Incorrect - The "snmp-server traps enable" is out of order. The correct command should be "snmp-server enable traps".

Explanation C. Correct - This configuration allows RO access with traps to the SNMP server.

Explanation D. Incorrect - The "snmp server host" command is missing the SNMP server IP address.

PrepLogic Question: 11418-219