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Exam : 640-460

Title : IIUC Implementing Cisco IOS
Unified Communications (IIUC)

Vendors : Cisco

Version : DEMO

1. You are CCNA VOICE associate in XXXX.com. You need configure a voice port that will allow the gateway to terminate 23 or 30 circuits from the PSTN. Which type of voice port you will configure?

- A. BRI
- B. FXS
- C. PRI T1/E1
- D. E&M

Answer: C

2. Which of the following is selected first for an incoming dial peer?

- A. Answer-address
- B. incoming called-number
- C. destination-pattern
- D. pots port

Answer: B

3. Power over Ethernet (PoE) is the ability for the LAN switching infrastructure to provide power over a copper Ethernet cable to an endpoint (Powered Device). This capability, once referred to as "inline power", was originally developed and first delivered by Cisco in 2000 to support the emerging IP Telephony deployments. IP Telephones, like desktop PBX phones, need power for their operation and Power over Ethernet enables scalable and manageable power delivery and simplifies deployments of IP Telephony. As Wireless networking emerged, Power over Ethernet was also used to power these devices to allow for deployments in locations where local power access did not exist. While IP telephones and wireless access points are the most intuitive uses for PoE, the advent of 802.3af standardization of PoE opens the door to a new generation of networked-attached devices such as video cameras, point-of-sale devices, security access control (card scanners), building automation and industrial automation just to name a few. When a Cisco switch supply PoE to an IP phone?

- A. whenever switch and IP phone are Cisco products.
- B. when Cisco switch supports prestandard PoE and the IP phone supports 802.3af.
- C. when Cisco switch and IP phone both support a common PoE method.
- D. Only when both the switch and IP phone use 802.3af PoE.

Answer: C

4. What is the default VTP mode on Cisco switches?

- A. Master
- B. Client
- C. Backup
- D. Server

Answer: D

5. Which Cisco Unified Communications layer is the call processing layer?

- A. Layer A including video,communications client, software-based phones, IP Phone
- B. Layer B including Voice mail,Presence,Contact Center
- C. Layer C including Cisco Unified Communications manger,Directory Server.
- D. Layer D including Gateway, Router, Switch.

Answer: C

6. What are the three VTP modes on Cisco switches?

- A. Master
- B. Client
- C. Server
- D. Transparent

Answer: BCD

7. A SIP Trunk is a logical connection between an IP PBX and a Service Provider's application servers that allows voice over IP traffic to be exchanged between the two. When a call is placed from an internal phone to an external number, the PBX sends the necessary information to the SIP Trunk provider who establishes the call to the dialed number and acts as an intermediary for the call. All signaling and voice traffic between the PBX and the provider is exchanged using SIP and RTP protocol packets over the IP network. Which two statements about SIP trunk are true?

- A. A SIP trunk configuration is mandatory for a UC500 device.
- B. A SIP trunk is needed for internet access
- C. A SIP trunk is needed only for voice if you are planning on using VoIP through a service provider.

D. A SIP trunk is not supported in a keyswitch configuration.

Answer: CD

8. You are CCNA VOICE associate in XXXX.com. One user from your company wants to use a signaling protocol on the voice gateways that require registration with the Cisco Unified Communications Manager. Which protocol should you suggest to him?

A. SIP

B. Frame relay

C. SRTP

D. MGCP

Answer: D

9. An analog telephone is connected to a _____ port on a router?

A. FXO

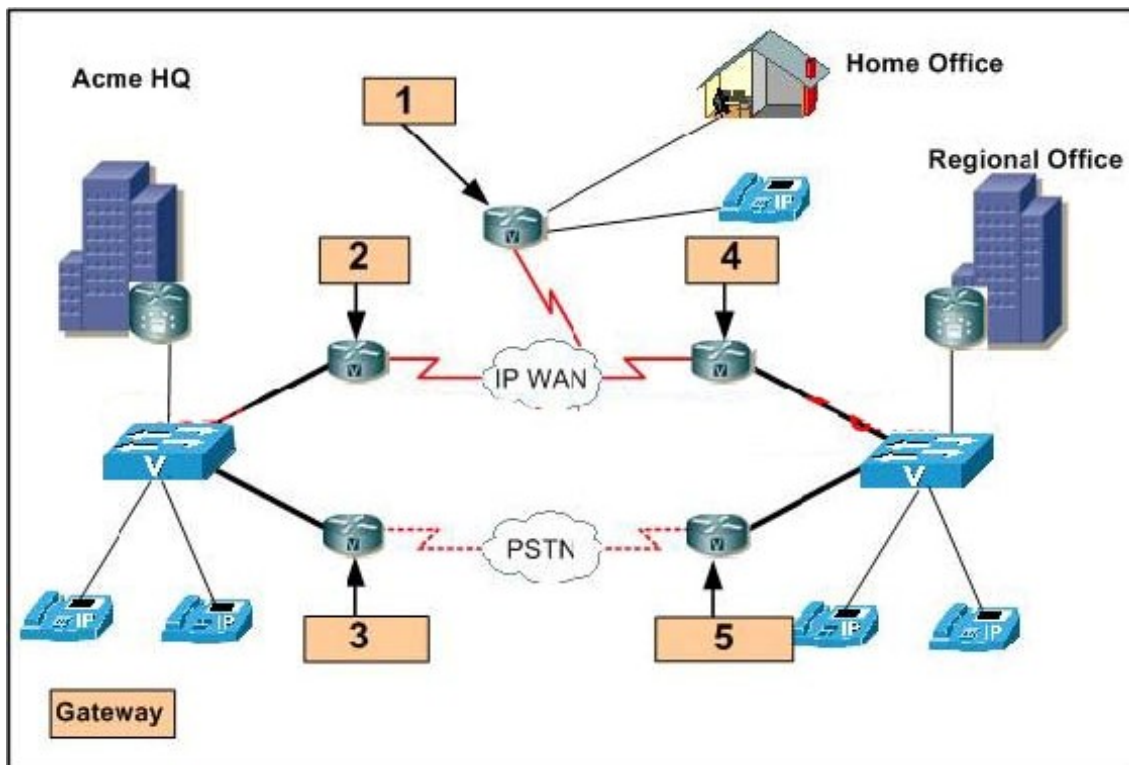
B. E1

C. T1

D. FXS

Answer: D

10. Choose the Gateway to the proper locations.



- A. 2,3
- B. 2,5
- C. 3,5
- D. 1,3

Answer: C

11. What is required to convert a G711ulaw call to G729?

- A. Voice Termination resources
- B. Conferencing resources
- C. Converter resources
- D. Transcoding resources

Answer: D

12. Which statement is true about the difference between voice VLAN and native VLAN?

- A. Voice VLAN uses untagged frames whereas native VLAN uses 802.1Q frames
- B. Voice VLAN uses tagged 802.1Q frames whereas native VLAN uses 802.1P frames

C. Voice VLAN uses tagged 802.1Q frames whereas native VLAN uses untagged frames

D. Voice VLAN uses untagged frames only when no PCs are connected behind the phones whereas native VLAN always uses untagged frames

Answer: C

13. What is the maximum number of phones are supported on Call Manager Express?

A. 48

B. 28

C. 240

D. 500

Answer: C

14. You are CCNA VOICE associate in XXXX.com. Which value should you assign to option 150 for DHCP to operate correctly in a Cisco Unified Communications Manager Express environment?

A. FTP server of the DNS server

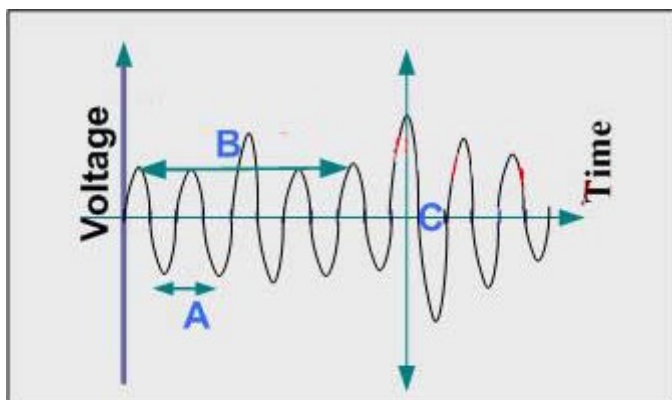
B. MAC address of the DHCP server

C. MAC address of the TFTP server

D. IP address of the TFTP server

Answer: D

15. You are CCNA VOICE associate in XXXX.com. Which identifies the amplitude of an analog signal stream?



A. $A \times C$

B. $A \times B$

C. A

D. C

Answer: D

16. What protocol needs to be enabled on an ATA if a fax machine is connected to the ATA?

A. MGCP

B. SCCP

C. H323

D. SIP

Answer: C

17. You are CCNA VOICE associate in XXXX.com. Which two are considered endpoints in a Cisco Unified Communications solution?

A. router

B. call agent

C. IP telephone

D. gateway

Answer: CD

18. You are CCNA VOICE associate in XXXX.com. You execute the show ephone command in the Cisco Unified Communications manager as below. What information will you get from the output?

UCME#show ephone

```
ephone-1 Mac:0030.94C2.8A44 TCP socket:[2] activeLine:0 REGISTERED in SCCP ver 1
1 and Server in ver 8
mediaActive:0 of fhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:8
IP:10.3.130.10 50374 Telecaster 7960 keepalive 4 max_line 6
button 1: dn 1 number 5001 CH1 IDLE CH2 IDLE mwi
button 2: dn 3 number 5010 CH1 IDLE CH2 IDLE

ephone-2 Mac:0003.E3C4.463C TCP socket:[-1] activeLine:0 DECEASED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0 caps:7
IP:10.3.130.12 49939 Telecaster 7960 keepalive 5162 max_line 6
button 1: dn 2 number 5002 CH1 DOWN CH2 DOWN
button 2: dn 3 number 5010 CH1 IDLE CH2 IDLE shared
```

- A. There is only one IP phone.
- B. There are two registered IP phones. Shared number 5010 on line 2. Message waiting on shared line.
- C. There are two IP phones. Phone 2 is unregistered. Shared number 5010 on line 2. Message waiting on shared line
- D. There are two IP phones. Phone 2 is unregistered. Shared number 5010 on line 2. Message waiting on line 1 of phone

Answer: D

19. What protocol needs to be enabled on an ATA if an analog telephone is connected to the ATA?

- A. MGCP
- B. SCCP
- C. H323
- D. SIP

Answer: B

20. Please match the CAS component to the office that it will apply.

(1)CAS E1 (2)CAS T1 (3)1.544 mb/s (4)2.048 mb/s

(5)RBS in-band (6)24 voice channels (7)30 voice channels (8)out-of-band signaling in time slot 17

(I)London, United Kingdom (II)Chicago, United States

- A. (I)-(1 4 5 8);(II)-(2 3 7 6)
- B. (I)-(2 4 7 8);(II)-(1 3 5 6)
- C. (I)-(1 4 7 8);(II)-(2 3 5 6)
- D. (I)-(1 3 7 8);(II)-(2 4 5 6)

Answer: C