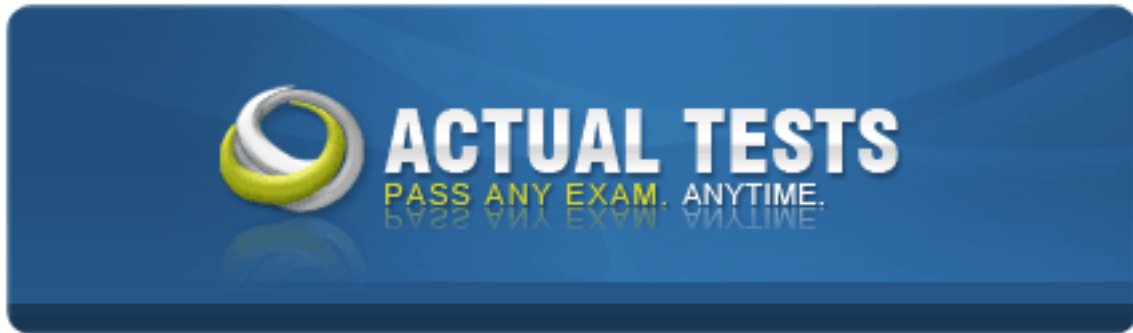


Cisco 642-437



**Implementing Cisco Unified Communications Voice
over IP and QoS v8.0 (CVOICE v8.0)**

Version: 33.0

Topic 1, Volume A**QUESTION NO: 1**

Which three Cisco IOS commands are required to configure a voice gateway as a DHCP server to support a data subnet with the IP address of 10.1.30.0/24 and a default gateway of 10.1.30.1/24? (Choose three.)

- A. ip dhcp pool
- B. subnet 10.1.30.1 255.255.255.0
- C. ip dhcp pool data
- D. network 10.1.30.1/24
- E. network 10.1.30.0 255.255.255.0
- F. default-gw 10.1.30.1/24
- G. default-router 10.1.30.1

Answer: C,E,G

Explanation:

1) To configure the DHCP address pool name and enter DHCP pool configuration mode, use the following command in global configuration mode.

Router(config)# ip dhcp pool name - Creates a name for the DHCP Server address pool and places you in DHCP pool configuration mode

2) To configure a subnet and mask for the newly created DHCP address pool, which contains the range of available IP addresses that the DHCP Server may assign to clients, use the following command in DHCP pool configuration mode.

Router(dhcp-config)# network network-number [mask | /prefix-length] - Specifies the subnet network number and mask of the DHCP address pool. The prefix length specifies the number of bits that comprise the address prefix. The prefix is an alternative way of specifying the network mask of the client. The prefix length must be preceded by a forward slash (/).

3) After a DHCP client has booted, the client begins sending packets to its default router. The IP address of the default router should be on the same subnet as the client. To specify a default router for a DHCP client, use the following command in DHCP pool configuration mode.

Router(dhcp-config)# default-router address [address2 ... address8] - Specifies the IP address of the default router for a DHCP client. One IP address is required; however, you can specify up to eight addresses in one command line.

http://www.cisco.com/en/US/docs/ios/12_2/ip/configuration/guide/1cfdhcp.html#wp1000999

QUESTION NO: 2

Which four Cisco IOS commands are required to configure a DHCP server on a voice gateway to support a voice subnet so that both IP addresses and the IP address of the TFTP server are provided? The voice subnet has an address of 10.1.130.0/24, the default gateway is 10.1.130.1/24, and the TFTP server is located at 10.1.5.2. (Choose four.)

- A. subnet 10.1.130.1/24
- B. ip dhcp pool voice
- C. default-router 10.1.130.1
- D. option 150 10.1.5.2
- E. network 10.1.130.0 255.255.255.0
- F. dhcp pool voice
- G. option 150 ip 10.1.5.2
- H. default-gw 10.1.130.1

Answer: B,C,E,G

Explanation:

QUESTION NO: 3

The router with the IP address of 10.1.120.1 needs to be configured to use the device 10.1.140.1 as the clock source. Which configuration command will accomplish this task?

- A. clock source 10.1.140.1
- B. ntp server 10.1.140.1
- C. clock set 10.1.140.1
- D. ntp source ip addr 10.1.140.1
- E. ntp client 10.1.120.1 server 10.1.140.1

Answer: B

Explanation: To configure your routers to use a NTP server for time synchronization, the command ntp server, followed by the IP address or hostname of the NTP server, is used. To specify additional timeservers for redundancy, simply repeat the ntp server command with the IP address of each additional server.

http://www.cisco.com/en/US/products/hw/switches/ps700/products_tech_note09186a008010e97e.shtml

QUESTION NO: 4

Which four types of ephone-dns are supported by SCCP in Cisco Unified Communications Manager Express? (Choose four.)

- A. single-line
- B. dual-line
- C. shared-line, nonexclusive
- D. two directory numbers with one telephone number
- E. dual-number
- F. octo-line

Answer: A,B,E,F

Explanation:

QUESTION NO: 5

In which situation would an administrator configure telephony services, but not configure any individual ephones?

- A. Phones that are controlled by Cisco Unified Communications Manager Express
- B. Cisco Unified Communications Manager SRST fallback
- C. Cisco Unified Communications Manager Express with HSRP
- D. Remotely located phones that are controlled by a third-party PBX
- E. This is not a valid scenario. Ephones are always required.

Answer: B

Explanation: When a phone registers for SRST service with a Cisco Router and the router discovers that the phone was configured with a specific extension number, the router searches for an existing prebuilt ephone-dn with that extension number and then assigns that ephone-dn number to the phone. If there is no prebuilt ephone-dn with that extension number, the system automatically creates one. In this way, extensions without prebuilt configurations are automatically populated with extension numbers and features as the numbers and features are "learned" by the Cisco router in SRST mode when the phone registers to the router after a WAN link fails.

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmestrst.html

QUESTION NO: 6

Refer to the exhibit. Which type of ephone-dn is configured for the two ephones that are shown?

```
ephone 1
 mac-address 0023.5322.98BA
 button 101,10,11,12
!
ephone 2
 mac-address 0023.5332.985C
 button 102,10,11,12
```

- A. single-line-octo
- B. hunt line
- C. shared-line, nonexclusive
- D. two directory numbers with one telephone number
- E. shared-line, overlay
- F. octo-line

Answer: E

Explanation: The above exhibit shows the configuration for a simple shared-line overlay set. The primary ephone-dn that is configured for each phone is unique while the remaining ephone-dns 10, 11, and 12 are shared in the overlay set on both phones. The primary ephone-dn in a shared-line overlay set is configured unique to the phone to guarantee that the phone has a line available for outgoing calls, and to ensure that the phone user can obtain dial-tone even when there are no idle lines available in the rest of the shared-line overlay set. Using a unique ephone-dn also provides a unique calling party identity on outbound calls made by the phone so that the called user can see which specific phone is calling.

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmecover.htm#wp1099687

QUESTION NO: 7

Refer to the exhibit.

```
Jun 4 18:12:50.259: TFTP: Looking for CTLSEP2893FE139B0F.tlv
Jun 4 18:12:50.311: TFTP: Looking for ITLSEP2893FE139B0F.tlv
Jun 4 18:12:50.367: TFTP: Looking for ITLFile.tlv
Jun 4 18:12:50.555: TFTP: Looking for SEP2893FE139B0F.cnf.xml.sgn
Jun 4 18:12:50.675: TFTP: Looking for XMLDefault.cnf.xml.sgn
Jun 4 18:12:51.247: TFTP: Looking for SK72f64050-7ad5-4b47-9bfa-5e9ad9cd4aa9.xml.sgn
Jun 4 18:13:00.299: TFTP: Looking for term65.default.loads
Jun 4 18:13:00.783: TFTP: Looking for term65.default.loads
```

A new Cisco Unified Communications Manager Express system has been deployed and the technician is trying to add the first new IP phone to the system. The phone powers up, but it does not register with the system. The technician has verified that the phone is getting the proper VLAN information from Cisco Discovery Protocol. The phone is also getting the correct IP address and TFTP server address from DHCP. The phone has been assigned to an ephone and the correct MAC address is configured. With the information provided, which two of the following does the administrator need to verify to resolve this situation? (Choose two.)

- A. Verify that the ip helper-address is correctly configured.
- B. Verify that telephony-service has been configured.
- C. Verify that the ephone has a button assigned.
- D. Verify that the tftp-server path has been configured.
- E. Verify that the Cisco Unified Communications Manager Express service is running.
- F. Verify that the correct phone type files are in the tftp-server path.

Answer: D,F

Explanation: Since the phone is getting the correct TFTP address, the next thing that needs to be verified is the TFTP Server path and IP Reachability for the IP Phone to the TFTP Server. Once the TFTP settings has been verified, check if the files mentioned in the termxx.defaults.loads file is available in the TFTP Server for the phone to download.

http://www.cisco.com/en/US/docs/voice_ip_comm/cuipph/7960g_7940g/7_0/sip/english/administration/guide/7960trbS.html

QUESTION NO: 8

The administrator has added a new ephone-dn and a new ephone to the Cisco Unified Communications Manager Express system, but the new phone will not register with the system. If other phones are operating properly, which of the following should the administrator do first to try to resolve the issue?

- A. Reboot the router.
- B. Remove the ephone, then re-add the ephone.
- C. Verify that the url authentication is configured for the correct authentication URL.
- D. Verify that the url services is configured to the correct URL for services.
- E. Enter the command no telephony-service, then enter telephony service in global configuration mode.
- F. Enter the command no create cnf-files, then enter create cnf-files under the telephony-service configuration.

Answer: F

Explanation:

QUESTION NO: 9

Refer to the exhibit.

```
!
telephony-service
max-ephones 6
max-dn 12
max-conferences 8 gain -6
transfer-system full-consult
!
```

Cisco Unified Communications Manager Express has been partially configured to support 6 IP phones and 12 directory numbers. The Cisco Unified Communications Manager Express will use the IP address 10.1.130.1/24. Which two elements of the configuration are missing from the command output and need to be added so that phones do not auto-register, but can manually register with Cisco Unified Communications Manager Express? (Choose two.)

- A. ip address 10.1.130.1
- B. no reg-ephone
- C. create profile
- D. ip source-address 10.1.130.1
- E. create cnf-files
- F. no auto-reg-ephone

Answer: D,F

Explanation: To identify the IP address and port through which IP phones communicate with a CiscoUnifiedCME router, use the ip source-address command in telephony-service or group configuration mode. This command enables a router to receive messages from CiscoUnifiedIPphones through the specified IP address and port. The CiscoUnifiedCME router cannot communicate with CiscoUnifiedCME phones if the IP address of the port to which they are attached is not configured.

Normally when you configure basic telephony-service parameters, then phone can register with CME although no DN will be assigned to them. You can disable this by using the no auto-reg-ephone command. After this command the phone which will try to register will receive message "Registration Rejected. No configuration entry.....". When automatic registration is blocked, CiscoUnifiedCME records the MAC addresses of phones that attempt to register but cannot

because they are blocked.

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/command/reference/cme_a1ht.html#wp1031242

QUESTION NO: 10

Which three functions are associated with MGCP? (Choose three.)

- A.** Control is implemented by a series of plain-text commands that are sent over UDP port 2427 between Cisco Unified Communications Manager and the gateway.
- B.** A PRI backhaul channel forwards PRI Layer 2 (Q.921) signaling information via a TCP connection from the gateway to the call agent.
- C.** MGCP uses a separate channel for backhauling signaling information between the call agent and the gateway.
- D.** The gateway maintains a separate dial plan for redundancy in case the call agent fails.
- E.** Users query the call agent to determine the location of the call recipient.
- F.** A call agent uses control messages to direct its gateways and their operational behavior.

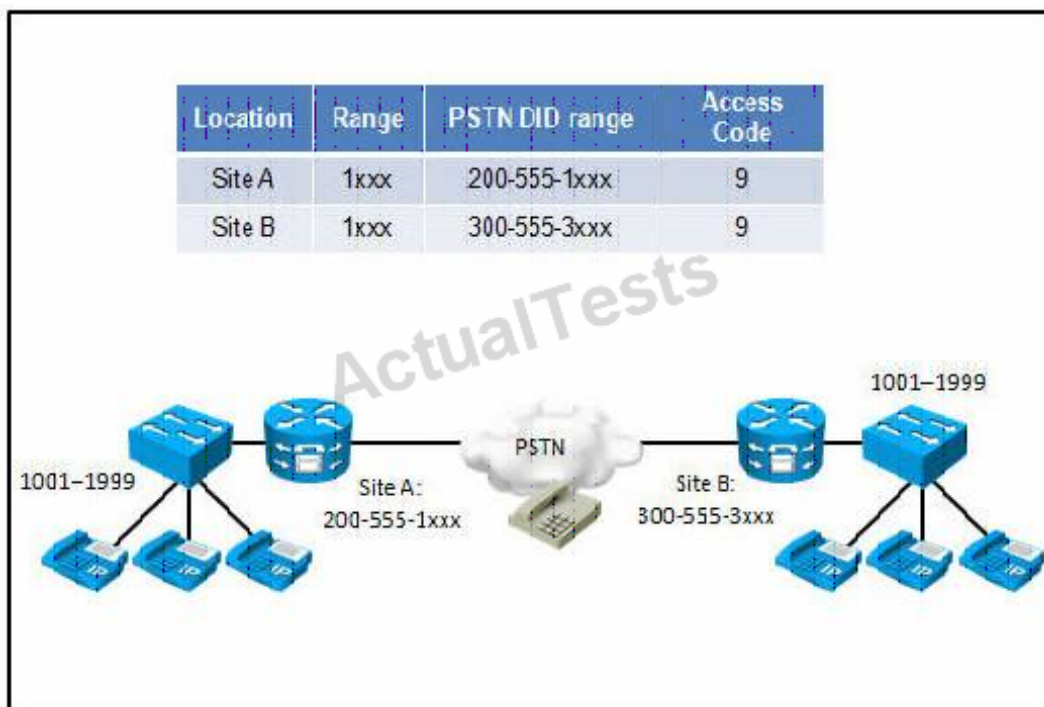
Answer: A,C,F

Explanation: MGCP is a plain-text protocol used by call-control devices to manage IP Telephony gateways. MGCP is a master/slave protocol that allows a call control device to take control of a specific port on a gateway. With this protocol, the Cisco CallManager knows and controls the state of each individual port on the gateway. It allows complete control of the dial plan from Cisco CallManager, and gives CallManager per-port control of connections to the PSTN, legacy PBX, voice mail systems, POTS phones and so forth. This is implemented with the use of a series of plain-text commands sent over User Datagram Protocol (UDP) port 2427 between the Cisco CallManager and the gateway.

Another concept relevant to the MGCP implementation with Cisco CallManager is PRI Backhaul. This occurs when Cisco CallManager takes control of the Q.931 signaling data used on an ISDN PRI. The one thing that distinguishes a PRI from other interfaces is the fact that the data that is received from the PSTN on the D-channel and needs to be carried in its raw form back to the Cisco CallManager to be processed. The gateway does not process or change this signaling data, it simply passes it onto the Cisco CallManager through TCP port 2428. The gateway is still responsible for the termination of the Layer 2 data. That means that all the Q.921 data-link layer connection protocols are terminated on the gateway, but everything above that (Q.931 network layer data and beyond) is passed onto the Cisco CallManager. This also means that the gateway does not bring up the D-channel unless it can communicate with Cisco CallManager to backhaul the Q.931 messages contained in the D-channel.

QUESTION NO: 11

Refer to the exhibit. An administrator is migrating a PBX telephony system to an IP Phone solution using a fixed numbering plan. The extension numbers and PSTN DIDs cannot be changed. Which of the following methods can be used in order to reach the individual extensions at Site B when called via the PSTN?



- A. The administrator can add a 1 to the DID for Site B to become 300-555-31xxx.
- B. The administrator needs to map the last four digits in the DID to the extension numbers and prefix a site code.
- C. The administrator needs to map the last four digits in the DID to the extension numbers and prefix an intersite code.
- D. The administrator needs to map the last four digits in the DID to the extension numbers using translation rules.
- E. No changes are necessary because PSTN calls are preceded with access code 9.

Answer: D

Explanation: Since the extension and PSTN DID is one and the same for the customer, no manipulation is required the Route Plan to reach individual extensions from PSTN DID

QUESTION NO: 12

Which of the following best describes the implementation challenges that are associated with variable-length numbering plans?

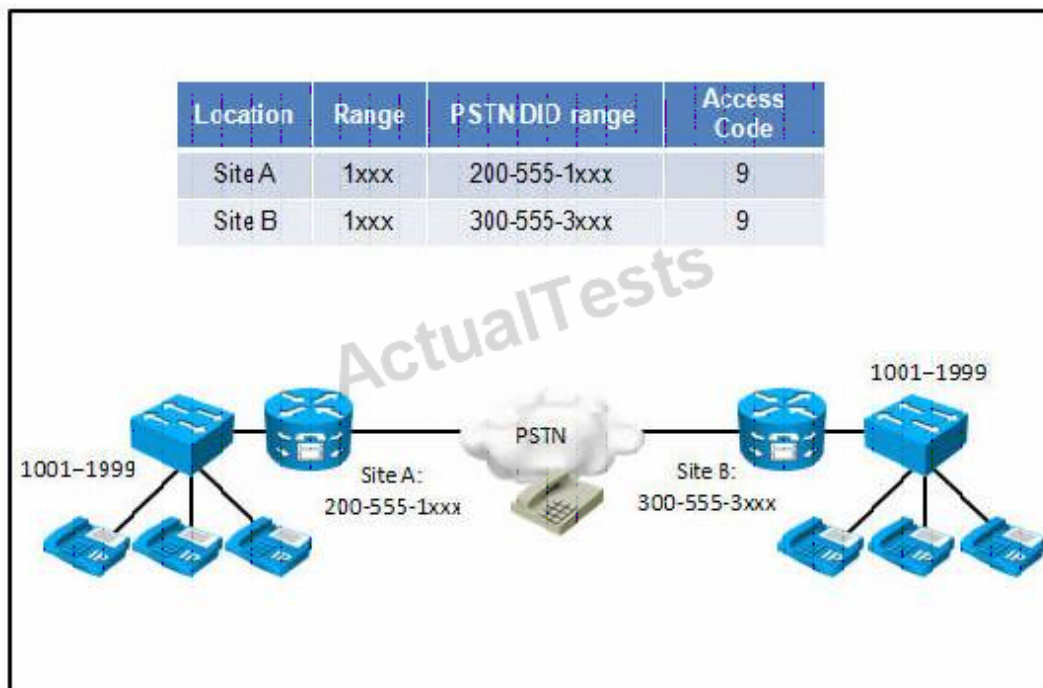
- A. the variable number of extensions that need to be implemented
- B. the number of trunks that need to be assigned
- C. the mapping between IP addresses and extension numbers
- D. the identification of the number of digits that need to be dialed before the call is routed
- E. the degree in which the dial plan varies

Answer: D

Explanation:

QUESTION NO: 13

Refer to the exhibit. An administrator is migrating a PBX telephony system to a VoIP solution using a fixed numbering plan. The extension numbers and PSTN DIDs cannot be changed. Which of the following methods can be used in order to reach the individual extensions at Site B when called via the PSTN?



- A. The administrator can replace the last three digits of the DID with xxx to cover the individual extensions.
- B. The administrator can replace the last three digits of the DID with xxx and use translation rules to map the individual extensions.
- C. The administrator needs to implement an auto-attendant solution where individual extensions

can be dialed.

D. The administrator needs to map the last four digits in the DID to the extension numbers using translation rules.

Answer: D

Explanation:

QUESTION NO: 14

Which two statements are true regarding SCCP? (Choose two.)

A. SCCP requires each endpoint or gateway event to be communicated to Cisco Unified Communications Manager.

B. Endpoints can operate autonomously if communication with Cisco Unified Communications Manager is lost.

C. SCCP may interoperate with H.323 endpoints if it is implemented with Cisco Unified Communications Manager.

D. Endpoints and gateways maintain the dial plan.

E. SCCP uses hex messages for communication.

Answer: A,C

Explanation: The Skinny client (i.e. an Ethernet Phone) uses TCP/IP to transmit and receive calls. Skinny messages are carried above TCP and use port 2000. Cisco IP Phones that use SCCP can coexist in an H.323 environment. When used with CUCM, the SCCP client can interoperate with H.323-compliant terminals. The client communicates with the CUCM using TCP/IP-based communication to establish a call with another H.323-compliant end station. Once the CUCM has established the call, the two H.323 end stations use connectionless UDP/IP-based communication for audio transmissions. The CUCM acts as a proxy by processing all H.323 and SIP transactions. This allows the IP Phone to process the VoIP RTP data stream.

http://www.cisco.com/en/US/docs/voice_ip_comm/cata/186_188/2_15_ms/english/administration/guide/sccp/sccpaaph.pdf

QUESTION NO: 15

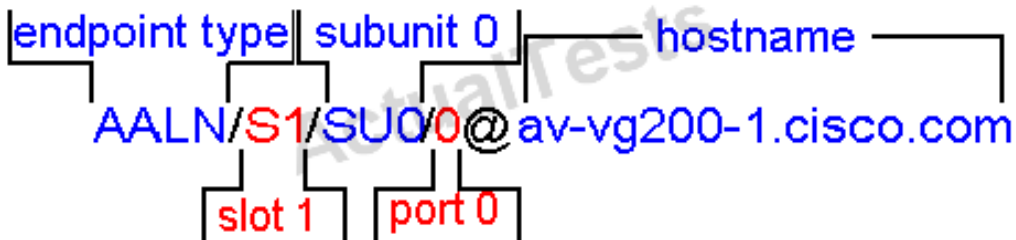
You are configuring a network to support voice to the PSTN. One important aspect to the configuration is to be able to determine the individual slot, subunit, and port number from the gateway endpoint identifier. Which signaling protocol is appropriate for this situation?

A. H.323

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

Answer: D

Explanation: Endpoints are any of the voice ports on the designated gateway. These voice ports provide connectivity to both analog ports and digital trunks to the PSTN. Ports on gateways are identified by endpoints in very specific ways. It is important to note that gateways can have multiple endpoints dependent on the number of ports it contains, and that the endpoints are case insensitive. A sample MGCP endpoint addressing scheme is provided below.



http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00801da84e.shtml

QUESTION NO: 16

Which two functions are associated with a voice gateway? (Choose two.)

- A. Switches voice channels between connected analog and digital voice circuits
- B. Provides voice-messaging services to connected analog and digital voice circuits
- C. Interconnects two logically separate VoIP networks
- D. Negotiates endpoint capabilities
- E. Controls opening and closing of logical channels that are used to carry media streams

Answer: A,E

Explanation: The basic function of a gateway is to translate between different types of networks. In a VoIP environment, voice gateways are the interface between a VoIP network and the public switched telephone network (PSTN), a private branch exchange (PBX), or analog devices such as fax machines. In its simplest form, a voice gateway has an IP interface and a legacy telephone interface, and it handles the many tasks involved in translating between transmission formats and protocols. The gateway allows communication between the two networks by performing tasks such as Interfacing with the IP network and the PSTN or PBX, Supporting IP call control protocols, Performing call setup and teardown for calls between the VoIP and PSTN networks by terminating and reoriginating the call media and signaling, Providing supplementary services, such as call hold and transfer, Relaying dual tone multifrequency (DTMF) tones, Supporting analog fax and

modems over the IP network.

http://www.cisco.com/en/US/prod/collateral/routers/ps5854/product_data_sheet0900aecd80169812.pdf

QUESTION NO: 17

Which type of voice port supports immediate-start, wink-start, and delay-start followed by pulse or DTMF tones?

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

Answer: D

Explanation:

QUESTION NO: 18

Which types of voice ports allow a small office to provide outbound DNIS and inbound DID?

- A. FXS and FXO
- B. FXO and E&M
- C. FXS and FXS-DID
- D. FXS and E&M
- E. FXS-DID and FXO

Answer: E

Explanation: An FXO trunk is one of the simplest analog trunks available. Because Dialed Number Information Service (DNIS) information can only be sent out to the PSTN, no direct inward dialing (DID) is possible. ANI is supported for inbound calls. Two signaling types exist, loopstart and groundstart, with groundstart being the preferred method. An FXS DID trunk can receive only inbound calls, thus a combination of FXS DID, and FXO ports is required for inbound and outbound calls

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/srnd/design/guide/gateway.html#wp1052323

QUESTION NO: 19

In a voice gateway, the configured codec complexity of the DSPs on a voice card can be changed. What is the impact on the DSPs if high codec complexity is configured?

- A.** The codec complexity affects call density, which is the number of calls that are reconciled on the DSPs. This results in lower call density when high complexity is configured.
- B.** With higher codec complexity, more calls can be processed.
- C.** Lower codec complexity supports the fewest number of voice channels, provided that the lower complexity is compatible with the particular codecs that are in use.
- D.** The DSP will process codecs that support high complexity transparently and shift to flex mode for those codecs that are not high complexity.

Answer: A

Explanation: The difference between medium and high complexity codecs is the amount of CPU utilization necessary to process the codec algorithm, and therefore, the number of voice channels that can be supported by a single DSP. For this reason, all the medium complexity codecs can also be run in high complexity mode, but fewer (usually half) of the channels are available per DSP.

http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00800b6710.shtml#code_com

QUESTION NO: 20

Which codec complexity type will offer the greatest number of voice channels, provided that the complexity type is compatible with the particular codecs that are in use?

- A.** low complexity
- B.** medium complexity
- C.** high complexity
- D.** flex complexity

Answer: A

Explanation: Codec complexity refers to the amount of processing required to perform voice compression. Codec complexity affects the call density—the number of calls reconciled on the DSPs. With higher codec complexity, fewer calls can be handled. Select a higher codec complexity if that is required to support a particular codec or combination of codecs. Select a lower

codec complexity to support the greatest number of voice channels, provided that the lower complexity is compatible with the particular codecs in use.

http://www.cisco.com/en/US/docs/ios/12_3t/voice/command/reference/vrht_c6_ps5207_TSD_Products_Command_Reference_Chapter.html

QUESTION NO: 21

Your PSTN carrier sends digits to your T1 PRI circuit in a digit-by-digit format. How must the T1 PRI circuit be configured to support this capability?

- A. The T1 PRI controller supports either en-bloc or digit-by-digit formats natively.
- B. The serial interface that is associated with the T1 controller needs to include the isdn incoming-voice command.
- C. The T1 controller needs to include the isdn overlap-receiving command.
- D. The serial interface that is associated with the T1/E1 controller needs to include the isdn overlap-receiving command.

Answer: D

Explanation: Configuring Overlap-receiving on the D-channel changes the way routers behave when receiving ISDN calls. Overlap receiving allows the matching of dial peers as the digits are being received. The router responds to the setup message with a SETUP ACK. This informs the network that it is ready to receive further information messages containing additional call routing elements.

http://www.cisco.com/en/US/tech/tk801/tk133/technologies_tech_note09186a00800b48cb.shtml

QUESTION NO: 22

Refer to the exhibit. Callers dial 0 to reach an outside line. When they try to place calls to directory services (322) or services (422), they hear the reorder tone. What needs to be edited in the dial peer to allow these calls to complete successfully?

```
!
dial-peer voice 22 pots
destination-pattern [1-9]22
port 0/0/0:15
!
```

- A. The destination pattern is incorrect. It needs to start with a 9.
- B. A "prefix 1" statement needs to be added to the dial-peer configuration.
- C. The forward-digits all command needs to be applied to the dial peer.
- D. The destination pattern needs to be edited so that the first digit that is matched is a 0.
- E. The destination pattern needs to be edited so that the first digit that is matched is a 0 and the forward-digits all command needs to be added to the dial peer.
- F. The destination pattern needs to be edited so that the first digit that is matched is a 1 and the forward-digits all command needs to be added to the dial peer.
- G. The destination pattern needs to be edited so that the first digit that is matched is a 0 and the forward-digits 3 command needs to be added to the dial peer.

Answer: G

Explanation: Since the callers dial 0 before any actual number to go outside line, they should have a destination pattern starting with 0 to place a successful call to directory services or other services. The forward-digits command controls the number of digits that are stripped before the dialed string is passed to the telephony interface. On outbound POTS dial peers, the terminating router normally strips off all digits that explicitly match the destination pattern in the terminating POTS dial peer. Only digits matched by the wildcard pattern are forwarded. The forward-digits command can be used to forward a fixed number of dialed digits, or all dialed digits, regardless of the number of digits that explicitly match the destination pattern.

http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/dial_peer/dp_config.html#wp1067010

QUESTION NO: 23

What is the reason that an outgoing call succeeds when there is no COR list that is applied to the incoming dial peer and a COR list is applied to the outgoing dial peer?

- A. The COR list for incoming calls on the incoming dial peer is a superset of COR lists for outgoing calls on the outgoing dial peer.
- B. COR lists for incoming calls on the incoming dial peer are not a superset of COR lists for outgoing calls on the outgoing dial peer.
- C. The outgoing dial peer, by default, has the lowest priority.
- D. The incoming dial peer, by default, has the highest COR priority when no COR is applied.

Answer: D

Explanation: By default, an incoming call leg has the highest COR priority and the outgoing COR list has the lowest COR priority. This means that if there is no COR configuration for incoming calls on a dial-peer, then you can make a call from this dial-peer (a phone attached to this dial-peer) going out of any other dial-peer, irrespective of the COR configuration on that dial-peer.

http://www.cisco.com/en/US/tech/tk652/tk90/technologies_configuration_example09186a008019d649.shtml

QUESTION NO: 24

What is the reason that an outgoing call succeeds when COR is applied to the incoming dial peer, but no COR is applied to the outgoing dial peer?

- A.** The COR list for incoming calls on the incoming dial peer is a superset of COR lists for outgoing calls on the outgoing dial peer.
- B.** COR lists for incoming calls on the incoming dial peer are not a superset of COR lists for outgoing calls on the outgoing dial peer.
- C.** The outgoing dial peer, by default, has the lowest priority.
- D.** The incoming dial peer, by default, has the highest COR priority when no COR is applied.

Answer: C

Explanation:

QUESTION NO: 25

Calls are failing to egress the local PSTN gateway that uses an E1 PRI circuit. Which debug command would be most useful in determining which dialed digits are being sent to the PSTN?

- A.** debug voice dial-peer
- B.** debug isdn q921
- C.** debug isdn q931
- D.** ccapi inout

Answer: C

Explanation: Debug isdn q931 command to display information about call setup and teardown of ISDN network connections (Layer 3). In order to verify the layer 3 signaling we need to enable layer 3 signaling command. ISDN q921 is for layer2. Debug isdn q931 shows the calling number and called number. If the calls are failing, we can also see the ISDN cause codes from the debug isdn q931 command.

http://www.cisco.com/en/US/docs/ios/11_2/debug/command/reference/dipx.html#wp13263

QUESTION NO: 26

Refer to the exhibit. When 5551234 is being matched with the outgoing dial peer that is shown in the exhibit, which of the following called numbers will be sent to the PSTN?

```
dial-peer voice 1 pots
 destination-pattern 555....
 prefix 5
 port 0/0/0:23
```

- A. 5551234
- B. 1234
- C. 555
- D. Null
- E. 5
- F. 51234

Answer: F

Explanation: On outbound POTS dial peers, the terminating router normally strips off all digits that explicitly match the destination pattern in the terminating POTS dial peer. Only digits matched by the wildcard pattern are forwarded. The forward-digits command can be used to forward a fixed number of dialed digits, or all dialed digits, regardless of the number of digits that explicitly match the destination pattern.

http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/dial_peer/dp_config.html#wp1067737

QUESTION NO: 27

Refer to the exhibit. When 5551234 is being matched with the outgoing dial peer that is shown in the exhibit, which of the following called numbers will be sent to the VoIP network?

```
dial-peer voice 1 pots
 destination-pattern 555....
 prefix 5
 port 0/0/0:23
```

- A. 5551234
- B. 1234
- C. 555
- D. Null
- E. 5
- F. 51234

Answer: D

Explanation:

QUESTION NO: 28

Refer to the exhibit. When an inbound PSTN call to 4087071222 is received by the router that is shown in the exhibit, what is the resulting called number?

```
!  
voice translation-rule 1  
rule 1 /^.*\(...$)/ /\1/  
!  
voice translation-profile pstn-in  
translate called 1  
!  
voice-port 0/0/0:23  
translation-profile incoming pstn-in
```

- A. 14087071222
- B. 11222
- C. 14081222
- D. 1222
- E. 4087071222

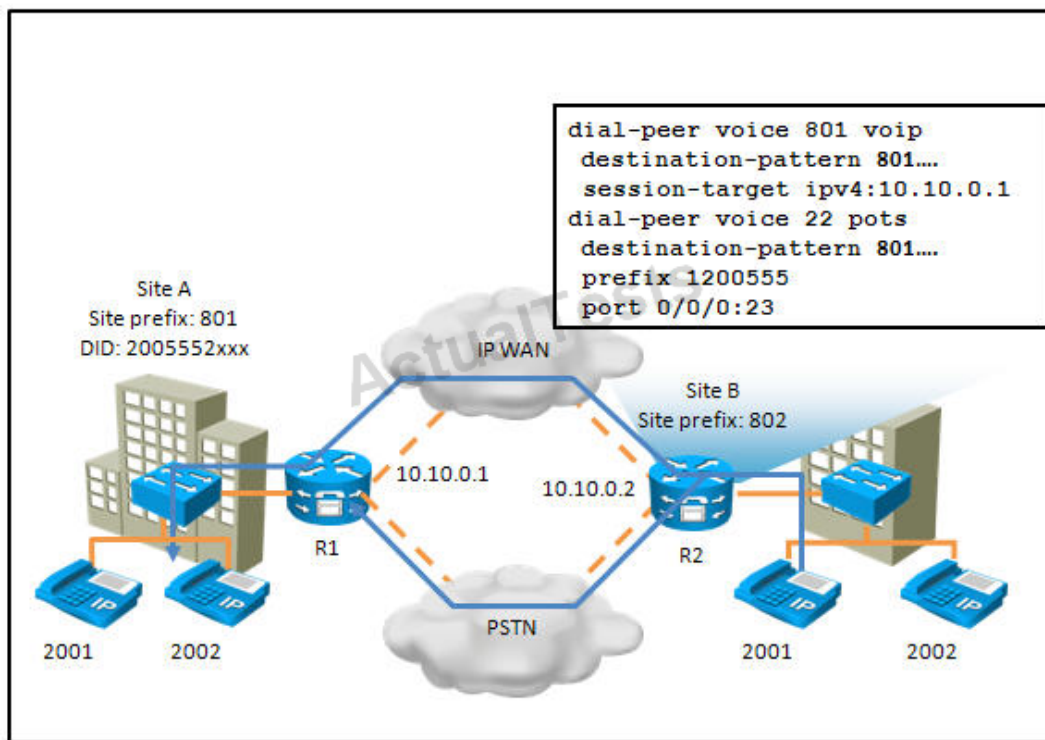
Answer: D

Explanation: /^.*\(...\$) – Truncates Numbers down to the last 4 digits.

http://www.cisco.com/en/US/tech/tk652/tk90/technologies_tech_note09186a0080325e8e.shtml

QUESTION NO: 29

Refer to the exhibit.



What happens when users at Site B place calls to Site A when the IP WAN is operational?

- A. The calls will always take the IP WAN route.
- B. The calls will always take the PSTN route.
- C. The calls will fail because the destination patterns are identical.
- D. The calls will use round-robin scheduling between the IP WAN and PSTN paths.
- E. The calls will use the IP WAN route unless there is a failure or congestion during which the calls will reroute via the PSTN.

Answer: D

Explanation:

QUESTION NO: 30

Refer to the exhibit. When an inbound PSTN call from 4087071222 arrives at the ISDN port that is shown in the exhibit, which dial peer will be matched for the inbound leg?

```
!
dial-peer voice 123 pots
destination-pattern 4087071222
direct-inward-dial
port 0/0/0:23
!
!
dial-peer voice 2123 pots
answer-address 4087071222
port 0/0/0:23
!
voice-port 0/0/0:23
```

- A. Dial-peer 123, because incoming called-number takes precedence over answer-address.
- B. Dial-peer 2123, because answer-address takes precedence over incoming called-number.
- C. The matching inbound dial peer will be selected at random.
- D. Although dial-peer 123 takes precedence, there is no direct-inward-dial that is configured, therefore 2123 will be selected.
- E. Although dial-peer 123 takes precedence, there is no port that is configured under dial-peer 123, therefore dial-peer 2123 will be selected.

Answer: B

Explanation:

QUESTION NO: 31

Refer to the exhibit.

```
!  
dial-peer voice 123 pots  
  destination-pattern 4087071222  
  direct-inward-dial  
  port 0/0/0:23  
!  
!  
dial-peer voice 2123 pots  
  answer-address 4087071222  
  port 0/0/0:23  
!  
voice-port 0/0/0:23
```

When an inbound PSTN call from 4087071222 arrives at the ISDN port that is shown in the exhibit, which dial peer will be matched for the inbound leg?

- A.** Dial-peer 123, because destination-pattern takes precedence over answer-address.
- B.** Dial-peer 2123, because answer-address takes precedence over destination-pattern.
- C.** The matching inbound dial peer will be selected at random.

- D.** Although dial-peer 2123 takes precedence, it will not be matched because the command direct-inward-dial is missing.
- E.** Dial-peer 123 will be matched because dial-peer 2123 will strip all the digits.

Answer: B

Explanation: The inbound call will first try to match the with the incoming called-number command. We can also use 'answer-address command' which is searched if 'incoming called-number' is not present. And if there is no 'incoming called-number command' and 'answer-address command', then the gateway will hunt for dialpeer with destination-pattern of calling party number.
http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/dial_peer/dp_config.html#wp1067989

QUESTION NO: 32

Which QoS methodology combines strict priority queuing with class-based weighted fair queuing?

- A.** IP RTP Priority
- B.** Multilink PPP
- C.** IP Frame Relay RTP Priority
- D.** RSVP
- E.** LLQ

Answer: E

Explanation:

QUESTION NO: 33

What are the three acceptable values for one-way delay, jitter, and packet loss in a VoIP network? (Choose three.)

- A.** 0-400 ms for delay
- B.** 1 packet loss
- C.** 20 ms for jitter
- D.** 0-150 ms for delay
- E.** 1 percent packet loss
- F.** 30 ms for jitter

Answer: D,E,F

Explanation:

(http://www.cisco.com/en/US/docs/solutions/Enterprise/WAN_and_MAN/QoS_SRND/QoSIntro.ht

ml#wp46447)

QUESTION NO: 34

What are the PHBs that DiffServ use?

- A. resource reservation and admission control
- B. default, AF, and EF PHBs
- C. AF, EF, and CS PHBs
- D. AF and EF PHBs
- E. default, AF, EF, and CS PHBs

Answer: E

Explanation: A Per Hop Behavior refers to the packet scheduling, queuing, policing, or shaping behavior of a node on any given packet belonging to a Behavior Aggregate, and as configured by a Service Level Agreement (SLA) or policy. To date, four standard PHBs are available to construct a DiffServ-enabled network and achieve coarse-grained, end-to-end CoS and QoS: The Default PHB, Class-Selector PHBs, Expedited Forwarding PHB and Assured Forwarding PHB.

http://www.cisco.com/en/US/technologies/tk543/tk766/technologies_white_paper09186a00800a3e2f_ps6610_Products_White_Paper.html

QUESTION NO: 35

What are two benefits of using the DiffServ model? (Choose two.)

- A. DiffServ is a flow-based architecture.
- B. DiffServ is highly scalable.
- C. DiffServ keeps flow state on each node in the network.
- D. DiffServ supports a large number of service classes.
- E. DiffServ uses repetitive signaling for each flow.

Answer: B,D

Explanation:

QUESTION NO: 36

What is the decimal equivalent of the DSCP value AF21?

- A. 16
- B. 17
- C. 18
- D. 21

Answer: C

Explanation: Assured Forwarding (AF) is a means to offer different levels of forwarding assurances for IP packets. Four AF classes are defined, where each AF class is in each DS node allocated a certain amount of forwarding resources(buffer space and bandwidth). Within each AF class IP packets are marked with one of three possible drop precedence values. A congested node tries to protect packets with a lower drop precedence value from being lost by preferably discarding packets with a higher drop precedence value. Classes 1 to 4 are referred to as AF classes. The following table illustrates the DSCP coding for specifying the AF class with the probability. Bits DS5, DS4 and DS3 define the class; bits DS2 and DS1 specify the drop probability; bit DS0 is always zero.

he following table illustrates the DSCP coding for specifying the AF class with the probability. Bits DS5, DS4 and DS3 define the class; bits DS2 and DS1 specify the drop probability; bit DS0 is always zero.

Drop	Class 1	Class 2	Class 3	Class 4
Low	001010 AF11 DSCP 10	010010 AF21 DSCP 18	011010 AF31 DSCP 26	100010 AF41 DSCP 34
Medium	001100 AF12 DSCP 12	010100 AF 22 DSCP 20	011100 AF32 DSCP 28	100100 AF42 DSCP 36
High	001110 AF13 DSCP 14	010110 AF23 DSCP 22	011110 AF33 DSCP 30	100110 AF43 DSCP 38

http://www.cisco.com/en/US/tech/tk543/tk757/technologies_tech_note09186a00800949f2.shtml

QUESTION NO: 37

If a packet is marked with an IP precedence value of 011, what is the corresponding binary DSCP class-selector value?

- A. 000011
- B. 011110
- C. 011000
- D. 011010
- E. 011100

Answer: C

Explanation:

QUESTION NO: 38

In which situation would the trust boundary be located at the access layer?

- A. if the endpoints, both IP phones and PCs, are incapable of marking traffic properly
- B. if PCs are switched through an IP phone and the IP phone traffic can be trusted to mark both traffic streams properly
- C. if the access layer switch cannot trust or re-mark incoming traffic from endpoints properly
- D. if there are endpoints that cannot be trusted and connect directly to the distribution layer

Answer: A

Explanation:

QUESTION NO: 39

Refer to the exhibit.

Default CoS-to-DSCP Map								
CoS Value	0	1	2	3	4	5	6	7
DSCP Value	0	8	16	26	32	40	48	56
mls qos trust device cisco-phone								
mls qos trust cos								

How does a switch port that receives marked traffic from a Cisco IP phone use the mls qos trust cos command?

- A. The CoS setting is modified according to the CoS-to-DSCP map.
- B. CoS is used to select the ingress and egress queues.
- C. For non-IP packets, the CoS is set to 7 and DSCP-to-CoS mapping is not applied.
- D. The DSCP-to-CoS map is applied.

Answer: A

Explanation:

QUESTION NO: 40

Refer to the exhibit. Your company's QoS policy states that all traffic that is arriving at access layer switches from IP phones should be marked with a DSCP value of 46 and that all untagged traffic that is arriving from a PC that is attached to an IP phone should be marked with a CoS value of 1. Which two options will satisfy the requirements for the CoS-to-DSCP map and are the correct QoS commands? (Choose two.)

```
!  
mls qos  
!  
interface FastEthernet0/1/0  
description incoming port from IP Phone 2000  
switchport mode access  
switchport voice vlan 120  
switchport access vlan 20  
mls qos trust cos  
!  
interface FastEthernet0/1/1  
description trunk to MDF router 22  
switchport mode trunk  
mls qos trust cos  
!
```

- A. mls qos 1
- B. mls qos map cos-dscp 0 10 18 26 34 46 48 56
- C. mls qos cos 1
- D. mls qos map dscp 0 8 16 26 32 40 48 56
- E. mls qos map cos 0 8 18 26 40 48 50 56
- F. mls qos dscp 1

Answer: B,C

Explanation:

To define the ingress Class of Service (CoS)-to-differentiated services code point (DSCP) map for trusted interfaces, use the `mls qos map cos-dscp` command in global configuration mode.

`mls qos map cos-dscp dscp1...dscp8`

`dscp1...dscp8` - Defines the CoS-to-DSCP map. For `dscp1...dscp8`, enter eight DSCP values that correspond to CoS values 0 to 7. Separate consecutive DSCP values from each other with a space. The supported DSCP values are 0, 8, 10, 16, 18, 24, 26, 32, 34, 40, 46, 48, and 56.

To define the default multilayer switching (MLS) class of service (CoS) value of a port or to assign the default CoS value to all incoming packets on the port, use the `mls qos cos` command in interface configuration mode.

`mls qos cos cos-value`

`cos-value` - Assigns a default CoS value to a port. If the port is CoS trusted and packets are untagged, the default CoS value is used to select one output queue as an index into the CoS-to-DSCP map. The CoS range is 0 to 7. The default is 0.

http://www.cisco.com/en/US/docs/ios/qos/command/reference/qos_m2.html#wp1041343

QUESTION NO: 41

Which command should be included in order to trust the DSCP-marked traffic from the distribution layer?

- A. `mls qos trust cos`
- B. `mls trust dscp-cos`
- C. `mls qos trust dscp`
- D. `mls qos trust dscp-cos`

Answer: C

Explanation: To configure the multilayer switching quality of service port trust state and to classify traffic by examining differentiated services code point (DSCP) value, use the `mls qos trust dscp` command in interface configuration mode. This will enable the device to trust incoming packets that have DSCP values (the most significant 6 bits of the 8-bit service-type field).

http://www.cisco.com/en/US/products/hw/switches/ps5023/products_tech_note09186a0080883f9e.shtml

QUESTION NO: 42

Refer to the exhibit. Which class is always present even though it is not in the configuration snip?

```
!  
class-map voip  
match ip precedence 5  
!  
class-map voip-signaling  
match ip precedence 3  
!  
class-map mission_critical  
match ip precedence 4  
!  
class-map transactional  
match ip precedence 1 2  
!  
policy-map all traffic  
  class voip  
    priority percent 20  
  class voip-signaling  
    bandwidth percent 10  
    random-detect  
  class mission_critical  
    bandwidth percent 60  
    random-detect  
  class transactional  
    bandwidth 10  
    random-detect  
!
```

- A. class best-effort
- B. class class-default
- C. default class
- D. best-effort class
- E. class class-scavenger

Answer: B

Explanation: The class-default is in every policy-map by default and it cannot be removed. The class-default class is used to classify traffic that does not fall into one of the defined classes. Once

a packet is classified, all of the standard mechanisms that can be used to differentiate service among the classes apply. The class-default class was predefined when you created the policy map, but you must configure it. If no default class is configured, then by default the traffic that does not match any of the configured classes is flow classified and given best-effort treatment.

http://www.cisco.com/en/US/docs/ios/12_0t/12_0t5/feature/guide/cbwfq.html#wp25297

QUESTION NO: 43

An access layer switch is configured to extend priority to an IP phone. Cisco Discovery Protocol is enabled on all ports. What are the three possible ways that an IP phone can be instructed to treat the Layer 2 CoS priority value of the attached PC? (Choose three.)

- A. trusted IEEE 802.1Q
- B. configured DSCP level
- C. configured CoS level
- D. trusted
- E. configured IEEE 802.1Q
- F. untrusted

Answer: C,D,F

Explanation:

QUESTION NO: 44

A new Cisco 7965 IP phone is installed on a Cisco Unified Communications Manager Express system. When the phone requests the .loads file from the TFTP server, it sees that the versions are different. What does the IP phone do to resolve this issue?

- A. The IP phone requests the SEP<mac>.cfg file and reboots.
- B. The IP phone attempts to obtain the new firmware file image from the TFTP server.
- C. The IP phone boot requests the XMLDefault.cnf.xml file and boots up.
- D. The IP phone does not boot up and will require manual intervention to factory reset the phone before a new firmware image can be downloaded.

Answer: B

Explanation: Cisco IP Phone Initialization Process:

1. At initialization, the Cisco IP phone sends a request to the DHCP server to get an IP address, DNS server address, and TFTP server name or address, if appropriate. Options are set in DHCP server (Option 066, Option 150, and so on). It also gets a default gateway address if set in DHCP

server (Option 003).

2. If a DNS name of the TFTP sever is sent by DHCP, then a DNS sever IP address is required to map the name to an IP address. This step is bypassed if the DHCP server sends the IP address of the TFTP server. In this case study, the DHCP server sent the IP address of TFTP because DNS was not configured.

3. If a TFTP server name is not included in the DHCP reply, then the Cisco IP phone uses the default server name.

4. The configuration file (.cnf) file is retrieved from the TFTP server. All .cnf files have the name SEP<mac_address>.cnf, where "SEP" is an acronym for Selsius Ethernet Phone. If this is the first time the phone is registering with the Cisco CallManager, then a default file, SEPdefault.cnf, is downloaded to the Cisco IP phone.

5. All .cnf files include the IP address(es) of the primary and secondary Cisco CallManager(s). The Cisco IP phone uses the IP address to contact the primary Cisco CallManager and register.

6. Once the Cisco IP phone has connected and registered with Cisco CallManager, the Cisco CallManager tells the Cisco IP phone which executable version (called a load ID) to run. If the specified version does not match the executing version on the Cisco IP phone, the Cisco IP phone will request the new executable from the TFTP server and reset automatically.

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a0080129d92.shtml

QUESTION NO: 45

When a Cisco Unified Border Element is deployed to support RSVP-based CAC, which media flow method is required?

- A.** RSVP-based CAC can be supported with either media flow-through or media flow-around if the Cisco Unified Communications Manager is configured as an RSVP agent.
- B.** RSVP-based CAC only supports media flow-around.
- C.** The Cisco Unified Border Element does not have to participate in the RSVP message exchange and will pass RSVP messages through unchanged using media flow-around.
- D.** RSVP-based CAC requires Cisco Unified Border Element to use media flow-through.

Answer: D

Explanation:

QUESTION NO: 46

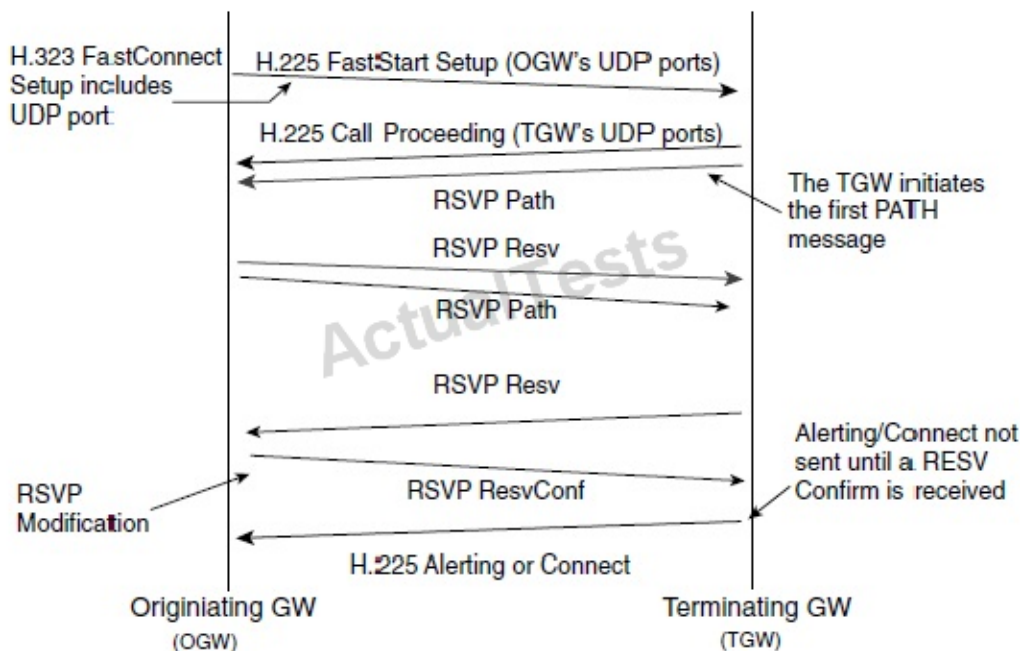
When Cisco Unified Border Element is configured to support RSVP-based CAC, at which point

during call setup are the RSVP path and reservation messages sent and received?

- A.** The path message is sent immediately after the call setup message is received and the reservation message is received after H.245 capabilities negotiation is completed.
- B.** The reservation message is sent immediately after the call setup message is received and the path message is received after H.225 call setup messages have been sent.
- C.** The path and reservation messages are sent and received after the H.245 capabilities negotiation is completed.
- D.** The path and reservation messages are sent and received immediately after the call setup message is received.

Answer: D

Explanation: The H.323 setup is suspended before the destination phone, triggered by the H.225 alerting message, starts ringing. The RSVP reservation is made in both directions because a voice call requires a two-way speech path and therefore bandwidth in both directions. The terminating gateway ultimately makes the CAC decision based on whether or not both reservations succeed. At that point the H.323 state machine continues either with an H.225 Alerting/Connect (the call is allowed and proceeds), or with an H.225 Reject/Release (call is denied). The RSVP reservation is in place by the time the destination phone starts ringing and the caller hears ringback.



QUESTION NO: 47

You have a Cisco Unified Border Element configured to provide H.323 to SIP interworking. Which

command will verify that you have a single H.323 and a single SIP call leg when the call is placed?

- A. show call active voice
- B. debug voip ipipgw
- C. show dialpeer voice
- D. debug voice dialpeer

Answer: A

Explanation: The show call active voice command allows you to display the contents of the active call table. The show call active voice command displays data from the plain old telephone service (POTS) and VoIP call legs on the voice gateway. The information presented includes call times, dial peers, connections, quality of service parameters, and gateway handling of jitter. This information can be useful when you troubleshoot a range of voice quality problems.

<http://www.cisco.com/en/US/docs/ios/voice/cube/configuration/guide/vb-gw-h323sip.html#wp1342172>

QUESTION NO: 48

Which QoS technology provides a strict priority queuing scheme that allows delay-sensitive data such as voice to be dequeued and sent before packets in other queues are dequeued, and also works with WFQ and CBWFQ.

- A. header compression
- B. IP RTP Priority and Frame Relay IP RTP Priority
- C. RSVP
- D. low latency queuing
- E. FRF.12

Answer: B

Explanation:

QUESTION NO: 49

How does Packet Loss Concealment improve voice quality?

- A. Cisco Packet Loss Concealment technology decreases the voice sampling rate to 10 ms of the voice payload to smooth gaps in the voice stream.
- B. Packet Loss Concealment intelligently analyzes missing packets and generates a reasonable replacement packet to improve the voice quality.

C. Packet Loss Concealment will buffer 20 to 50 ms of a voice stream to minimize lost or out-of-order voice packets.

D. Packet Loss Concealment will compensate for packet loss rates between 1 and 5 percent by generating a reasonable replacement packet to improve the voice quality.

Answer: B

Explanation: Packet loss concealment is a technology designed to minimize the practical effect of lost packets in VOIP. PLC mitigates against the effects of packet loss, which is the failure of one or more transmitted packets to arrive at their destination, by artificially regenerating the packet received prior to the lost one, followed by insertion of the duplicated packet into the gap. The digital value of the dropped packet is estimated by interpolation and an artificially generated packet inserted on that basis.

http://www.cisco.com/en/US/partner/tech/tk652/tk698/technologies_tech_note09186a00800f6cf8.shtml

QUESTION NO: 50

When a Cisco Unified Border Element connects two VoIP streams using flow-around media, which of the following options describes the components of the call that flow around and the components that flow through the device?

A. All security information flows through the Cisco Unified Border Element, and all call signaling and RTP flows around the device.

B. Call signaling flows through and call media flows around the device.

C. Call media flows through and call signaling flows around the device.

D. The initial call-signaling traffic flows through the device to initiate the call and then all subsequent calls flow around the device.

Answer: B

Explanation:

QUESTION NO: 51

Refer to the exhibit. What will the class map do if a packet arrives that is marked with a CoS of 6 and a DSCP value of EF?

```
!  
class-map match-all ip-voice  
match cos 5  
match dscp ef  
!  
policy-map ip-voice  
class ip-voice  
set dscp ef  
!  
interface FastEthernet 0/1  
service-policy output ip-voice  
!
```

- A. The class map will match the packet and forward it to the policy map to be marked.
- B. The class map will not map the packet and no QoS will be applied
- C. The class map will wait for the next packet in the stream to see if it has a CoS marking of 5 and then forward both packets to the policy map.
- D. For the packet to be forwarded to the policy map, it must have either a CoS of 5 or a DSCP value of EF.

Answer: B

Explanation: If there is no match for a packet, no QoS processing occurs on the packet and the switch offers best-effort service to the packet.

http://www.cisco.com/en/US/docs/switches/lan/catalyst2960/software/release/12.2_25_see/configuration/guide/swqos.html

QUESTION NO: 52

Refer to the exhibit. Consider an outgoing call that is being placed in all three scenarios that are shown in the exhibit. What is the result of the call, going down the table from top to bottom?

COR List on Incoming Dial Peer	COR List on Incoming Dial Peer	Result
No COR	No COR	
No COR	The COR List is applied for outgoing calls	
The COR List is applied for incoming calls	No COR	

- A. success, success, success
- B. success, success, fail
- C. success, fail, success
- D. success, fail, fail
- E. fail, success, success
- F. fail, success, fail

Answer: A

Explanation: Various combinations of COR lists and the results are shown in this table:

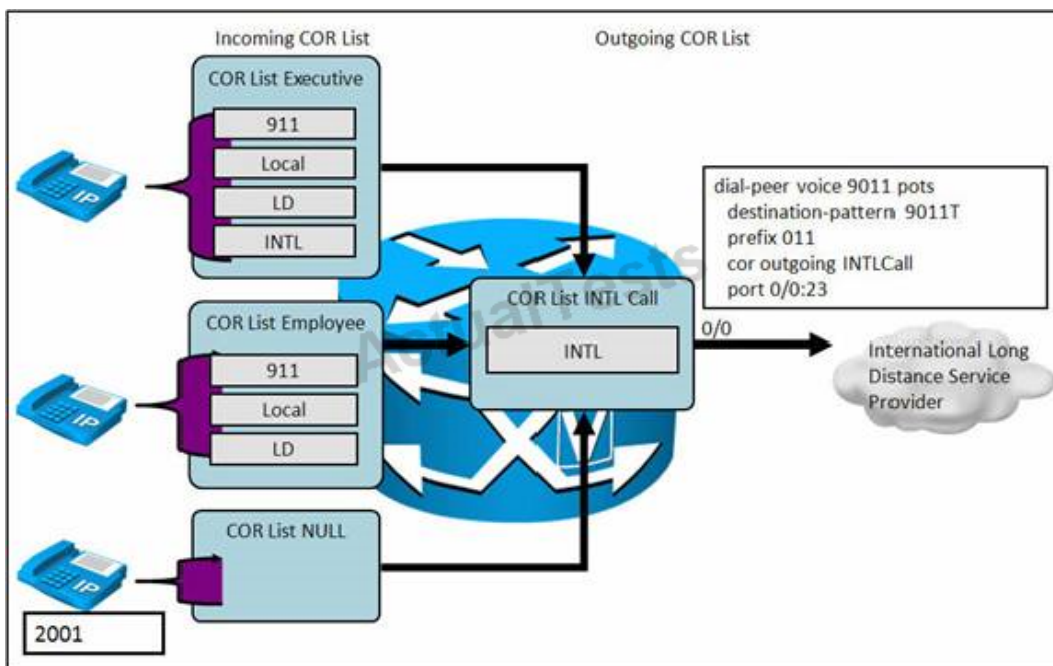
COR List on Incoming dial-peer	COR List on Outgoing dial-peer	Result	Reason
No COR	No COR	Call succeeds	COR is not in the picture
No COR	COR list applied for outgoing calls	Call succeeds	The incoming dial-peer, by default, has the highest COR priority when no COR is applied. Therefore, if you apply no COR for an incoming call leg to a dial-peer, then this dial-peer can make calls out of any other dial-peer, irrespective of the COR configuration on the outgoing dial-peer
The COR list applied for incoming calls	No COR	Call succeeds	The outgoing dial-peer, by default, has the lowest priority. Since there are some COR configurations for incoming calls on the incoming/originating dial-peer, it is a super set of the outgoing call COR configurations on the outgoing/terminating dial-peer

The COR list applied for incoming calls (super set of COR lists applied for outgoing calls on the outgoing dial-peer)	The COR list applied for outgoing calls (subset of COR lists applied for incoming calls on the incoming dial-peer)	Call succeeds	The COR list for incoming calls on the incoming dial-peer is a super set of COR lists for outgoing calls on the outgoing dial-peer
The COR list applied for incoming calls (subset of COR lists applied for outgoing calls on the outgoing dial-peer)	The COR list applied for outgoing calls (super set of COR lists applied for incoming calls on the incoming dial-peer)	Call <i>cannot</i> be completed using this outgoing dial-peer	COR lists for incoming calls on the incoming dial-peer are <i>not</i> a super set of COR lists for outgoing calls on the outgoing dial-peer

http://www.cisco.com/en/US/tech/tk652/tk90/technologies_configuration_example09186a008019d649.shtml

QUESTION NO: 53

Refer to the exhibit. When an international call to 90114989531212001 is placed from extension 2001, which of the following statements is true?



- A. The call will fail because no incoming COR list is applied.
- B. The call will succeed because the incoming COR list is a superset of the outgoing COR list.
- C. The call will fail because the incoming COR list is not a superset of the outgoing COR list
- D. The call will succeed because the incoming COR list has the highest priority, by default, when no incoming COR list is applied.

Answer: D

Explanation: By default, an incoming call leg has the highest COR priority and the outgoing COR list has the lowest COR priority. This means that if there is no COR configuration for incoming calls on a dial-peer, then you can make a call from this dial-peer (a phone attached to this dial-peer) going out of any other dial-peer, irrespective of the COR configuration on that dial-peer.

http://www.cisco.com/en/US/tech/tk652/tk90/technologies_configuration_example09186a008019d649.shtml

QUESTION NO: 54

Calculate how many IP phone calls can be sent across a 64 kbps Frame Relay link that uses the G.729 codec being sampled 50 times a second, 20 bytes a sample, and has 6 bytes of Frame Relay header overhead with no checksum and uses header compression.

- A. 3
- B. 4
- C. 5
- D. 7

Answer: C

Explanation:

QUESTION NO: 55

Which three methods are used by a Cisco Unified Border Element to provide network hiding? (Choose three.)

- A. Back-to-back user agent, replacing all H.323-embedded IP addressing
- B. IP network security boundary
- C. Media flow-through
- D. RSVP
- E. IP network privacy and topology hiding
- F. Intelligent IP address translation for RTP flows

Answer: B,E,F

Explanation: Cisco Unified Border Element can protect the network by hiding the network addresses and names for both the access (customer) side and the backbone (network core) side. A CUBE is designed to provide IP network privacy and topology hiding, IP network security boundary, Intelligent IP address translation for call media and signaling, Back-to-back user agent, replacing all SIP-embedded IP addressing, History information based topology hiding and call routing.

http://www.cisco.com/en/US/docs/routers/asr1000/configuration/guide/sbc/sbc_topology_hide.html

QUESTION NO: 56

Which of the following describes SIP Early Offer?

- A.** In SIP Early Offer mode, the SDP media capabilities are sent in the INVITE message of the calling device.
- B.** SIP Early Offer always uses session indicator 183.
- C.** In SIP Early Offer mode, the SDP media capabilities are sent in the 200 OK messages of the calling device.
- D.** In SIP Early Offer mode, the INVITE and the 200 OK messages use non-SDP message format to indicate SIP Early Offer

Answer: A

Explanation:

QUESTION NO: 57

Voice packets are arriving at a destination with a variance of between 20 and 50 milliseconds. If the jitter buffer has a capacity of 30 milliseconds, what is the impact on the audio at the receivers IP phone? Voice packets are arriving at a destination with a variance of between 20 and 50 milliseconds. If the jitter buffer has a capacity of 30 milliseconds, what is the impact on the audio at the receivers IP phone?

- A.** The jitter buffer will replay the previous voice packets to replace those packets that exceed 30 milliseconds to avoid speech gaps.
- B.** There will be no impact the audio stream because the audio packets are arriving in the jitter buffer window.
- C.** The DSP will automatically increase the jitter buffer size after sampling the range of incoming voice packets to accommodate the wider range in variation of voice packet arrival times to avoid voice gaps.
- D.** The IP phone will negotiate in mid-call a lower bandwidth codec to reduce the delay in the arrival of voice packets to avoid voice gaps.

Answer: B

Explanation:

QUESTION NO: 58

When deploying an 802.3af switch what is the default number of Watts consumed by each port if 802.3af compliant devices are attached to the switch?

- A. 4 Watts
- B. 6.3 Watts
- C. 7 Watts
- D. 15.4 Watts
- E. 22.3 Watts

Answer: D

Explanation:

QUESTION NO: 59

When configuring AutoQoS VoIP on a Cisco Catalyst switch how is the configuration performed?

- A. The auto qos voip command is applied to each interface.
- B. The auto qos voip command is applied globally in the switch.
- C. Each interface will need either the auto qos voip cisco-phone or auto qos voip trust on each interface depending on the upstream device.
- D. Each interface will need either the auto qos voip trust cisco-phone or auto qos voip trust trust on each interface depending on the upstream device.

Answer: C

Explanation: The QoS mechanisms on a Catalyst switch differ from those QoS mechanisms found on a router. For example, while a router uses LLQ as a priority queuing strategy, a Catalyst switch might use weighted round-robin (WRR) as a priority queuing strategy. Fortunately, the AutoQoS feature available on some Catalyst switch models applies voice-specific QoS features globally to a Catalyst switch and also at the port level. To configure AutoQoS on supported Catalyst switch platforms, issue the following command from interface configuration mode:

```
Switch(config-if)#auto qos voip [trust | cisco-phone]
```

If the trust option is used in the previous command, the Catalyst switch makes queuing decisions based on Layer 2 Class of Service (CoS) markings. However, if the cisco-phone option is used, the Catalyst switch makes queuing decisions based on CoS markings originating from a Cisco IP phone. The switch detects the presence of a Cisco IP phone via the CDP.

http://www.cisco.com/en/US/docs/ios/12_2t/12_2t15/feature/guide/ftautoq1.html

QUESTION NO: 60

Assuming no cRTP or header compression. How many VoIP G.729 calls can be made simultaneously over a 128-kb/s Frame Relay circuit (Layer 3) if 50 percent of the circuit is

dedicated to voice and 50 percent is dedicated to data?

- A. 1
- B. 2
- C. 3
- D. 4
- E. 5

Answer: B

Explanation:

Bandwidth Calculation Formulas

These calculations are used:

Total packet size = (L2 header: MP or FRF.12 or Ethernet) + (IP/UDP/RTP header) + (voice payload size)

Codec bit rate = codec sample size / codec sample interval

PPS = (codec bit rate) / (voice payload size)

Bandwidth = total packet size * PPS

http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a0080094ae2.shtml

QUESTION NO: 61

How are firmware images implemented and which file type describes the contents of the firmware image?

- A. Firmware images are implanted as firmware groups that are described by a file that has a .cnf suffix.
- B. Firmware images are implemented as individual files that are described by a file that has a .loads suffix.
- C. Firmware images are implemented as a file loader group and are described by a file that ends with a .sbn suffix.
- D. Firmware images are implemented as file bundles that are described by a file that ends with a .loads suffix.

Answer: D

Explanation:

QUESTION NO: 62

Which three methods are used by a Cisco Unified Border Element to provide network hiding?

(Choose three.)

- A. Back-to-back user agent, replacing all SIP-embedded IP addressing
- B. IP network security boundary
- C. media flow-through
- D. RSVP
- E. IP network privacy
- F. Intelligent IP address translation for RTP flows

Answer: B,E,F

Explanation:

QUESTION NO: 63

What is the function of class-based marking?

- A. Marking packets is based only on CoS value, IP precedence value or DSCP value allows Layer 3 frames to be identified and distinguished from other packets.
- B. Marking frames based only on CoS value or IP precedence value allows Layer 2 frames to be identified and distinguished from other frames.
- C. Marking frames or packets sets information in the Layer 2 and Layer 3 headers of a packet so that the frame or packet can be identified and distinguished from other frames or packets in the same traffic flow.
- D. Marking frames only sets information in the Layer 2 headers of a frame so that the frame can be identified and distinguished from other packets or frames.
- E. Marking allows network devices to classify a packet or frame, based on a specific traffic descriptor.

Answer: E

Explanation:

QUESTION NO: 64

A small office needs to provide outbound dialing and in-bound DID without the cost of a T1 circuit. All signaling is loop start. Which analog port configuration will support these requirements?

- A. voice-port 0/0/0
description fxs-did
signal did loop-start
!

```
voice-port 0/1/0
description fxo
signal loop-start
!
dial-peer voice 1 pots
incoming called-number .
direct-inward-dial
port 0/0/0
!
dial-peer voice 90 pots
destination-pattern 9T
port 0/1/0
B. voice-port 0/0/0
signal loop-start
!
voice-port 0/1/0
signal loop-start
!
dial-peer voice 1 pots
incoming called-number T
direct-inward-dial
!
dial-peer voice 90 pots
destination-pattern 9T
port 0/1/0
C. voice-port 0/1/0
signal did loop-start
!
dial-peer voice 1 pots
incoming called-number .
!
dial-peer voice 90 pots
destination-pattern 9T
port 0/1/0
D. voice-port 0/0/0
signal did loop-start
!
dial-peer voice 1 pots
incoming called-number .
direct-inward-dial
!
dial-peer voice 90 pots
destination-pattern 9T
port 0/0/0
```

Answer: A

Explanation:

QUESTION NO: 65

Which statement best describes dial peers in a voice gateway. (Choose two.)

- A.** Dial peers are call legs that are used to identify call source and destination endpoints and to define the characteristics that are applied to each call leg in the call connection.
- B.** Dial peers are configured with call legs that are essential to implementing dial plans and providing voice services over an IP packet network.
- C.** A dial peer is a physical addressable endpoint in a voice gateway.
- D.** Dial peers create physical connections called call legs to complete an end-to-end call.

Answer: A,C

Explanation:

QUESTION NO: 66

Which QoS mechanism for VoIP works with weighted fair queuing (WFQ) and class-based weighted fair queuing (CBWFQ)?

- A.** Header compression
- B.** FRF.12
- C.** IP RTP Priority and Frame Relay IP RTP Priority
- D.** Multilink PPP
- E.** RSVP

Answer: C

Explanation:

QUESTION NO: 67

How does LLQ ensure that voice traffic is always expedited?

- A.** LLQ adds WRED to CBWFQ. This allows delay-sensitive data such as voice to be dequeued and sent first.
- B.** LLQ uses CBWFQ to prioritize voice traffic and by dequeuing the voice packets so they can be handled first.
- C.** The strict priority queue has a higher weight than the queues in CBWFQ. This weight allows the

delay-sensitive data such as voice to be dequeued and sent first.

D. The LLQ strict priority queue allows delay-sensitive data such as voice to be dequeued and sent first (before packets in other queues are dequeued), giving delay-sensitive data preferential treatment over other traffic.

Answer: D

Explanation: Without Low Latency Queueing, CBWFQ provides weighted fair queueing based on defined classes with no strict priority queue available for real-time traffic. This scheme poses problems for voice traffic that is largely intolerant of delay, especially variation in delay. For voice traffic, variations in delay introduce irregularities of transmission manifesting as jitter in the heard conversation. The Low Latency Queueing feature provides strict priority queueing for CBWFQ, reducing jitter in voice conversations. Configured by the priority command, Low Latency Queueing enables use of a single, strict priority queue within CBWFQ at the class level, allowing you to direct traffic belonging to a class to the CBWFQ strict priority queue.

http://www.cisco.com/en/US/docs/ios/12_0t/12_0t7/feature/guide/pqcbwfq.html

QUESTION NO: 68 DRAG DROP

Drag the appropriate IOS command from the left and drop them in the spaces on the right in order to configure Cisco Unified Border Element. The ITSP does not support early offer. Not all boxes are used.

allow-connections sip to h323	
register server	
call start fast	
allow-connections sip to sip	
voice service voip	
allow-connections h323 to sip	
sip	
call start interwork	
h323	


```

dial-peer voice 10 voip
description To Cisco Unified CME
destination-pattern 1...
session target ipv4:10.1.5.2
dtmf-relay h245-alphanumeric
!
dial-peer voice 20 voip
description To ITSP
destination-pattern #2
session protocol sip
session target ipv4:10.140.1.1
dtmf-relay sip-nte digit-drop
  
```

Answer:

Drag the appropriate IOS command from the left and drop them in the spaces on the right in order to configure Cisco Unified Border Element. The ITSP does not support early offer. Not all boxes are used.

allow-connections sip to h323	voice service voip
register server	allow-connections sip to h323
call start fast	allow-connections h323 to sip
allow-connections sip to sip	h323
voice service voip	call start interwork
allow-connections h323 to sip	sip
sip	
call start interwork	
h323	

```

dial-peer voice 10 voip
description To Cisco Unified CME
destination-pattern 1...
session target ipv4:10.1.5.2
dtmf-relay h245-alphanumeric
!
dial-peer voice 20 voip
description To ITSP
destination-pattern #2
session protocol sip/2
session target ipv4:10.140.1.1
dtmf-relay rtp-nte digit-drop
  
```

Explanation:

Voice Service Voip

Allow-Connections sip to h323

Allow-Connections h323 to sip

H323

Call Start Interwork

SIP

Configuring an IP IP Gateway:

Call direction and translation section

voice service voip - Enters VoIP voice-service configuration mode

allow-connections from-type to to-type - Allows connections between specific types of endpoints in an Cisco Unified Border Element. Arguments are as follows:

- from-type - Type of connection. Valid values: h323, sip.
- to-type - Type of connection. Valid values: h323, sip.

Main protocol section

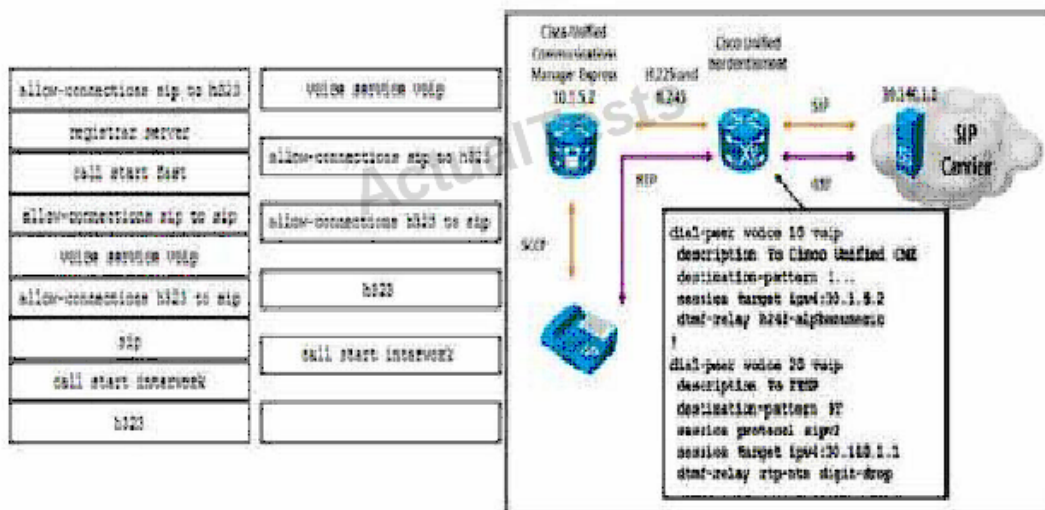
h323

call start interwork - Enables slow-start to fast-start interworking

sip

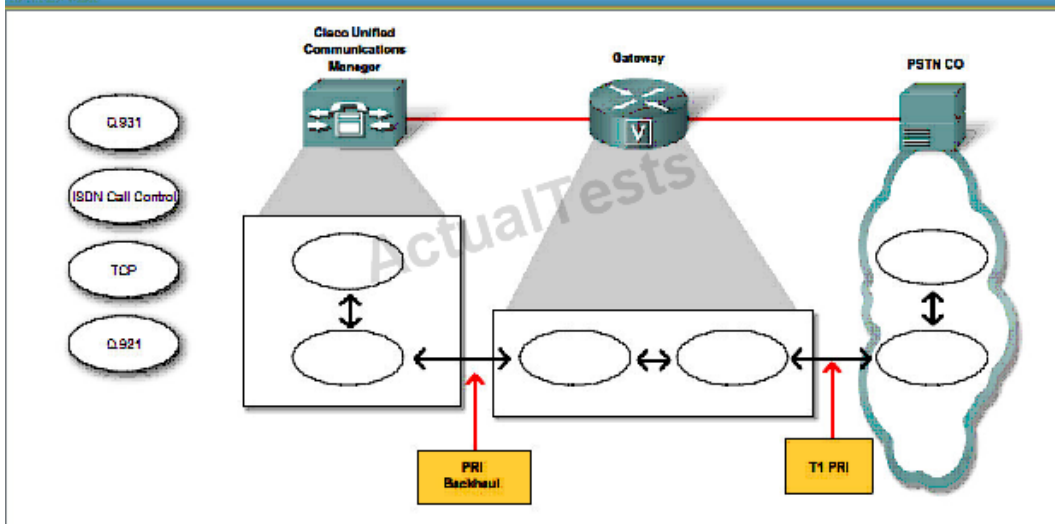
<http://www.cisco.com/en/US/docs/ios/voice/cube/configuration/guide/vb-gw-config.html>

Drag the appropriate IOS command from the left and drop them in the spaces on the right in order to configure Cisco Unified Border Element. The IOS does not support each option. Not all boxes are used.



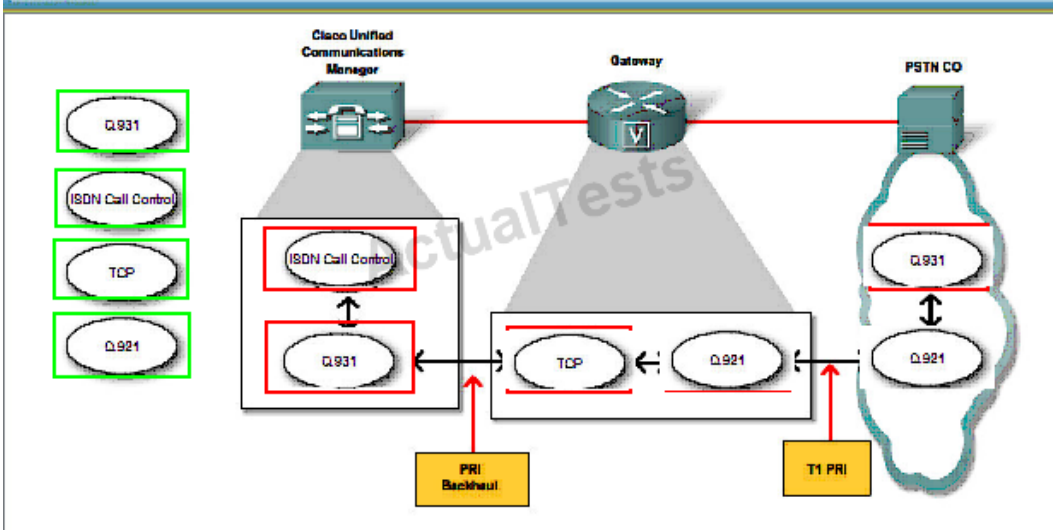
QUESTION NO: 69 DRAG DROP

Drag the signaling methods from the left and drop them in the correct position in the graphic on the right. Some methods are used more than once, and some methods may not be used at all.

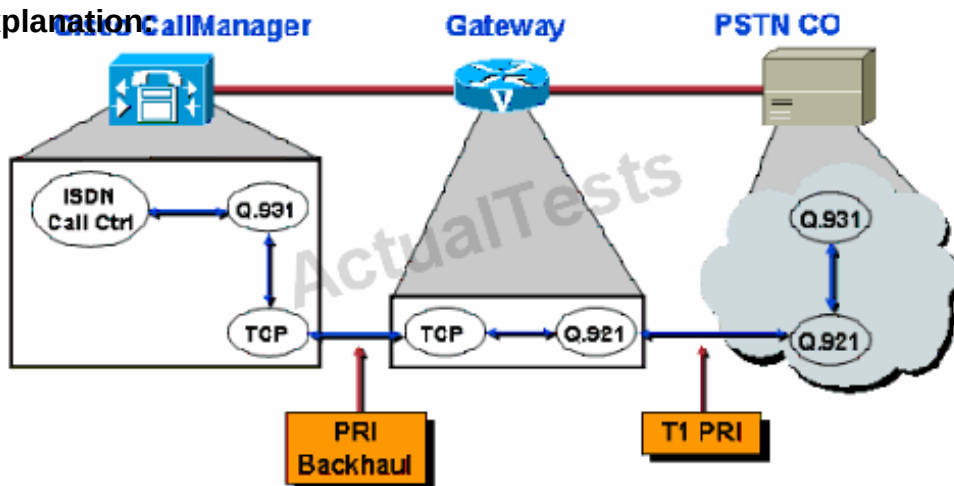


Answer:

Drag the signaling methods from the left and drop them in the correct position in the graphic on the right. Some methods are used more than once, and some methods may not be used at all.



Explanation



The one thing that distinguishes a PRI from other interfaces is the fact that the data that is received from the PSTN on the D-channel and needs to be carried in its raw form back to the Cisco CallManager to be processed. The gateway does not process or change this signalling data, it simply passes it onto the Cisco CallManager through TCP port 2428. The gateway is still responsible for the termination of the Layer 2 data. That means that all the Q.921 data-link layer connection protocols are terminated on the gateway, but everything above that (Q.931 network layer data and beyond) is passed onto the Cisco CallManager. This also means that the gateway does not bring up the D-channel unless it can communicate with Cisco CallManager to backhaul the Q.931 messages contained in the D-channel.

http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00801da84e.shtml

QUESTION NO: 70 DRAG DROP

The voice gateway selects an inbound VoIP dial peer by matching the information elements in the setup message with the dial-peer attributes. From the list on the left, drag the elements to the right and drop them in the order in which a voice gateway matches inbound calls. Not all options are used.

default dial peer	
answer-address	
destination-pattern	
incoming called-number	
port	

Answer:

The voice gateway selects an inbound VoIP dial peer by matching the information elements in the setup message with the dial-peer attributes. From the list on the left, drag the elements to the right and drop them in the order in which a voice gateway matches inbound calls. Not all options are used.

default dial peer	incoming called-number
answer-address	answer-address
destination-pattern	destination-pattern
incoming called-number	port
port	

Explanation:

default dial peer	incoming called-number
answer-address	answer-address
destination-pattern	destination-pattern
incoming called-number	port
port	

Inbound Dial Peers Matching Process

When the Cisco IOS router or gateway receives a call setup request, a dial peer match is made for the incoming call in order to facilitate routing the call to different session applications. This is not a digit-by-digit match; rather the full digit string received in the setup request is used to match against configured dial peers. The router or gateway matches the information elements in the setup message with the dial peer attributes to select an inbound dial peer. The router or gateway matches these items in this order:

Called number (DNIS) with the incoming called-number command:

First, the router or gateway attempts to match the called number of the call setup request with the configured incoming called-number of each dial peer. Because call setups always include DNIS information, it is recommended to use the incoming called-number command for inbound dial peer matching. This attribute has matching priority over the answer-address and destination-pattern commands.

Calling Number (ANI) with the answer-address command:

If no match is found in step 1, the router or gateway attempts to match the calling number of the call setup request with the answer-address of each dial peer. This attribute can be useful in situations where you want to match calls based on the calling number (originating).

Calling Number (ANI) with the destination-pattern command:

If no match is found in step 2, the router or gateway attempts to match the calling number of the call setup request to the destination-pattern of each dial peer. For more information about this, see the first bullet in the Dial Peer Additional Information section of this document.

Voice-port (associated with the incoming call setup request) with configured dial peer port (applicable for inbound POTS call legs):

If no match is found in the step 3, the router or gateway attempts to match the configured dial peer port to the voice-port associated with the incoming call. If multiple dial peers have the same port configured, the dial peer first added in the configuration is matched.

If no match is found in the first four steps, then the default dial peer 0 command is used.

http://www.cisco.com/en/US/tech/tk652/tk90/technologies_tech_note09186a008010fed1.shtml#con

QUESTION NO: 71 DRAG DROP

Drag the delay type on the left and drop it on the correct description on the right.

Processing Delay	Time required for sampling, encoding, packetization, and the reverse of the process.
Serialization Delay	The time it takes for a router to take the packet from an input interface, examine it, and put it into an output interface.
Dejitter Buffer	The time a packet resides in the output queue of a router.
DSP Delay	The time it takes to transmit "bits on the wire."
Queuing Delay	Buffer to compensate for variation in the arrival times of packets.
Propagation Delay	The time it takes for a packet to traverse a line from one end to the other end.

Answer:

Drag the delay type on the left and drop it on the correct description on the right.

Processing Delay	DSP Delay
Serialization Delay	Processing Delay
Dejitter Buffer	Queuing Delay
DSP Delay	Serialization Delay
Queuing Delay	Dejitter Buffer
Propagation Delay	Propagation Delay

Explanation:



Processing Delay: Coder delay is the time taken by the digital signal processor (DSP) to compress a block of PCM samples. This is also called processing delay (n). This delay varies with the voice coder used and processor speed.

Serialization Delay: Serialization delay (n) is the fixed delay required to clock a voice or data frame onto the network interface. It is directly related to the clock rate on the trunk.

Dejitter Buffer: Because speech is a constant bit-rate service, the jitter from all the variable delays must be removed before the signal leaves the network. In Cisco router/gateways this is accomplished with a de-jitter (n) buffer at the far-end (receiving) router/gateway. The de-jitter buffer transforms the variable delay into a fixed delay. It holds the first sample received for a period of time before it plays it out. This holding period is known as the initial play out delay.

DSP Delay: The time the packet spends inside the DSP is known as DSP Delay. Sampling, Encoding, Decoding etc. takes place inside the DSP.

Queuing Delay: After the compressed voice payload is built, a header is added and the frame is queued for transmission on the network connection. Voice needs to have absolute priority in the router/gateway. Therefore, a voice frame must only wait for either a data frame that already plays out, or for other voice frames ahead of it. Essentially the voice frame waits for the serialization delay of any preceding frames in the output queue. Queuing delay (βn) is a variable delay and is dependent on the trunk speed and the state of the queue. There are random elements associated with the queuing delay.

Propagation Delay: Caused by the length a signal must travel via light in fiber or electrical impulse in copper-based networks

http://www.cisco.com/en/US/tech/tk652/tk698/technologies_white_paper09186a00800a8993.shtml

QUESTION NO: 72 DRAG DROP

Drag the components that make up Cisco fax relay and T.38 from the left and drop them under the appropriate category on the right.

T.30 fax signaling spools a virtual fax machine to the locally attached fax machine.	Cisco Fax Relay
ITU-T standard that allows two fax machines to communicate as if there was a direct phone line. between	
RTP-based transmission method that uses proprietary signaling and encoding mechanisms.	T.38

Answer:

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ITU-T standard that allows two fax machines to communicate as if there was a direct phone line. between	
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Explanation:

T.30 fax signaling spools a virtual fax machine to the locally attached fax machine.	Cisco Fax Relay
ITU-T standard that allows two fax machines to communicate as if there was a direct phone line. between	
RTP-based transmission method that uses proprietary signaling and encoding mechanisms.	T.38

Cisco fax relay is the oldest method of supporting fax on Cisco IOS gateways and has been supported since Cisco IOS Release 11.3. Cisco fax relay uses Real-Time Transport Protocol (RTP) as the method of transport. In Cisco fax relay mode, gateways terminate T.30 fax signaling by spoofing a virtual fax machine to the locally attached fax machine. The gateways use a Cisco-proprietary fax-relay RTP-based protocol to communicate between them.

T.38 Fax Relay provides an ITU-T standards-based method and protocols for fax relay. Data is packetized and encapsulated according to the T.38 standard. The encoding of the packet headers and the mechanism to switch from VoIP mode to fax relay mode are clearly defined in the specification.

http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/voice_troubleshooting/old/vts_fax.html

QUESTION NO: 73 DRAG DROP

Drag the functions that are associated with H.245 from the list on the left to the boxes on the right.

	H.245
Carries capabilities exchange information.	
Used to establish connections between two H.323 endpoints.	
Controls opening and closing of logical channels that are used to carry media streams.	
Opens a signaling channel between an endpoint and a gatekeeper before any other channels.	
Opens a call-signaling channel between two H.323 endpoints or between an endpoint and a gatekeeper.	
Determines which endpoint is a master and which is a responder. It is used to resolve conflicts during the call.	

Answer:

Drag the functions that are associated with H.245 from the list on the left to the boxes on the right.

- Carries capabilities exchange information.
- Used to establish connections between two H.323 endpoints.
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- H.245
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- H.245
- Carries capabilities exchange information.
 - Controls opening and closing of logical channels that are used to carry media streams.
 - Determines which endpoint is a master and which is a responder. It is used to resolve conflicts during the call.

H.225 is responsible only for setting up the call and routing it to the proper destination. H.225 does not have any mechanism for exchanging capabilities or setting up and tearing down media streams. The called H.323 device is responsible for sending the IP address and port number that are used to establish the TCP connections for H.245 signaling. This information can be sent by the called device in either the Alerting or Connect message. When the originating H.323 device receives the IP address and port number for H.245 negotiations, it initiates a second TCP connection to carry out the necessary capabilities exchange and logical channel negotiations. This TCP session is primarily used to do four things:

Master/slave determination-This is used to resolve conflicts that might exist when two endpoints in a call request the same thing, but only one of the two can gain access to the resource at a time.

Terminal capabilities exchange-This is one of the most important functions of the H.245 protocol. The two most important capabilities are the supported audio codecs and the basic audio calls.

Logical channel signaling-This indicates a one-way audio stream. With H.323 version 2, it is possible to open and close logical channels in the middle of a call. Because H.245 messages are independent of the H.225 signaling, a call can still be connected in H.225 even if no logical

channels are open. This is typical with such features as hold, transfer, and conference.

DTMF relay-Because voice networks typically do not carry DTMF tones inband because of compression issues, these tones are carried on the signaling channel. Ensure that the type of DTMF relay configured on your gateway is compatible with your gatekeeper.

http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/voice_troubleshooting/old/vts_h323.html#wp1068085

QUESTION NO: 74 DRAG DROP

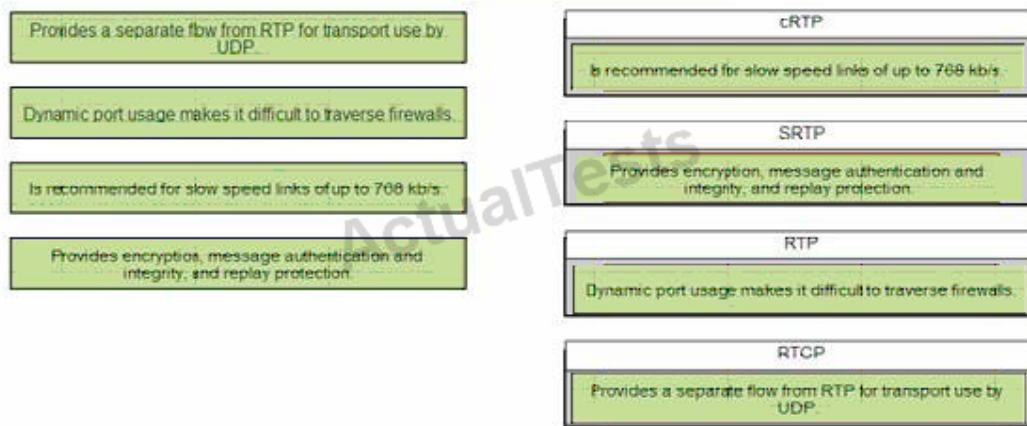
Drag the statement from the left to the protocol name that is associated with it on the right.

Provides a separate flow from RTP for transport use by UDP.	cRTP
Dynamic port usage makes it difficult to traverse firewalls.	SRTP
Is recommended for slow speed links of up to 768 kb/s.	RTP
Provides encryption, message authentication and integrity, and replay protection.	RTCP

Answer:

Drag the statement from the left to the protocol name that is associated with it on the right.

Provides a separate flow from RTP for transport use by UDP.	cRTP
Dynamic port usage makes it difficult to traverse firewalls.	SRTP
Is recommended for slow speed links of up to 768 kb/s.	RTP
Provides encryption, message authentication and integrity, and replay protection.	RTCP

Explanation:

The Real-Time Transport Protocol (RTP) is an Internet protocol standard that specifies a way for programs to manage the real-time transmission of multimedia data over either unicast or multicast network services. RTP is commonly used in Internet telephony applications. RTP does not in itself guarantee real-time delivery of multimedia data; it does, however, provide the wherewithal to manage the data as it arrives to best effect. RTP combines its data transport with a control protocol (RTCP), which makes it possible to monitor data delivery for large multicast networks. When protocols are used in conjunction, RTP is originated and received on even port numbers and the associated RTCP communication uses the next higher odd port number. Monitoring allows the receiver to detect if there is any packet loss and to compensate.

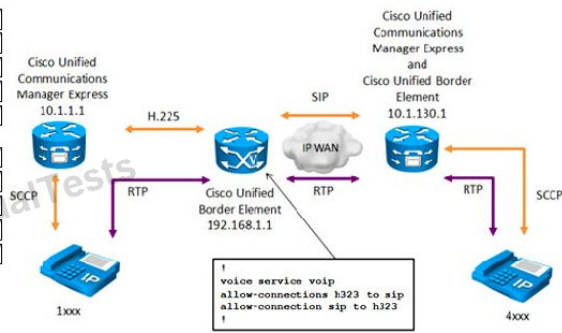
The Secure Real-time Transport Protocol (or SRTP) defines a profile of RTP (Real-time Transport Protocol), intended to provide encryption, message authentication and integrity, and replay protection to the RTP data in both unicast and multicast applications. Since RTP is closely related to RTCP (Real Time Control Protocol) which can be used to control the RTP session, SRTP also has a sister protocol, called Secure RTCP (or SRTCP); SRTCP provides the same security-related features to RTCP, as the ones provided by SRTP to RTP. Utilization of SRTP or SRTCP is optional to the utilization of RTP or RTCP; but even if SRTP/SRTCP are used, all provided features (such as encryption and authentication) are optional and can be separately enabled or disabled. The only exception is the message authentication feature which is indispensably required when using SRTCP.

On slow links, it may be advantageous to compress the IP/UDP/RTP headers using Compressed RTP (cRTP). If you use cRTP then the 40 bytes of overhead incurred by the IP/UDP/RTP headers can typically be compressed down to 2 to 4 bytes (2 bytes when no UDP checksums are sent, and 4 bytes when checksums are sent). Enabling compression on both ends of a low-bandwidth serial link can greatly reduce the network overhead if it carries a lot of RTP traffic. cRTP is supported on serial lines using Frame Relay, HDLC, or PPP encapsulation. It is also supported over ISDN interfaces. CRTP should not be used on links greater than 2 Mbps.

QUESTION NO: 75 DRAG DROP

Drag the appropriate IOS commands from the left and drop them in the spaces on the right in order to configure the dial peers for the Cisco Unified Border Element. The IP WAN connection between the two Cisco Unified Border Elements uses SIP. Not all commands and spaces are used.

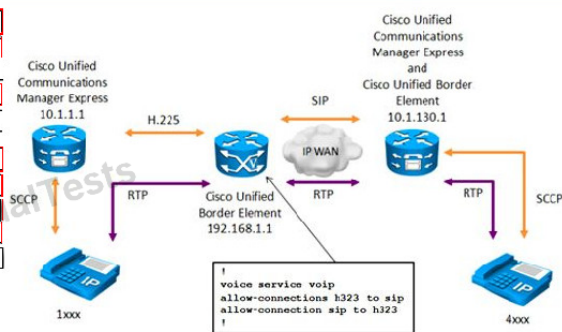
dial-peer voice 4000 pots	
destination-pattern 1...	
codec transparent	
session target ipv4:10.1.130.1	
dial-peer voice 1000 voip	
destination-pattern 4...	
media flow-around	
session target ipv4:10.1.1.1	
session protocol sipv2	
codec transparent	
dial-peer voice 4000 voip	
media flow-through	



Answer:

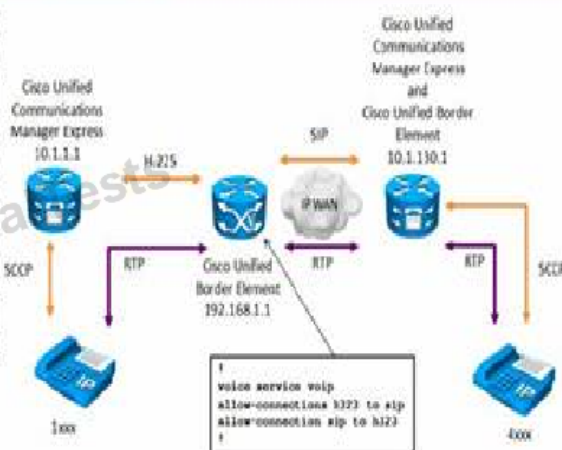
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dial-peer voice 4000 pots	dial-peer voice 1000 voip
destination-pattern 1...	destination-pattern 1...
codec transparent	session target ipv4:10.1.1.1
session target ipv4:10.1.130.1	codec transparent
dial-peer voice 1000 voip	media flow-through
destination-pattern 4...	dial-peer voice 4000 voip
media flow-around	destination-pattern 4...
session target ipv4:10.1.1.1	session target ipv4:10.1.130.1
session protocol sipv2	codec transparent
codec transparent	session protocol sipv2
dial-peer voice 4000 voip	
media flow-through	



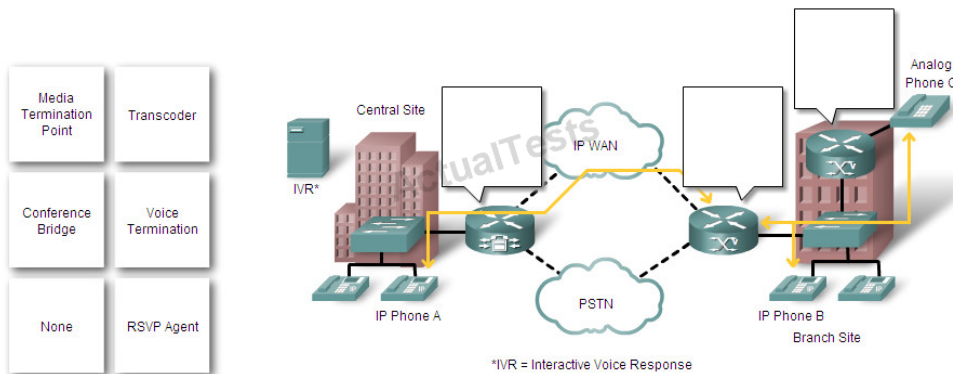
Explanation:

dial-peer voice 4000 pots	dial-peer voice 1000 voip
destination-pattern 1...	destination-pattern 1...
codec transparent	session target ipv4:10.1.1.1
session target ipv4:10.1.130.1	codec transparent
dial-peer voice 1000 voip	media flow-through
destination-pattern 4...	dial-peer voice 4000 voip
media flow-around	destination-pattern 4...
session target ipv4:10.1.1.1	session target ipv4:10.1.130.1
session protocol sipv2	codec transparent
codec transparent	session protocol sipv2
dial-peer voice 4000 voip	
media flow-through	



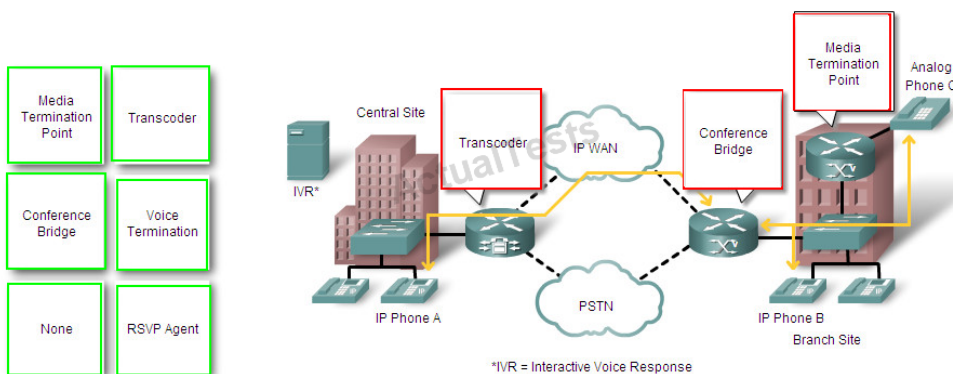
QUESTION NO: 76 DRAG DROP

All calls over the IP WAN use G.729. IP phones A and B use Cisco Unified Communications Manager Express. IP Phone A is on a call with IP Phone B. IP Phone A conferences in Analog Phone C with IP Phone B. Software conference resources are not being used. Drag the appropriate DSP resource for each gateway from the list to the correct locations in the graphic so the call can be completed.



Answer:

All calls over the IP WAN use G.729. IP phones A and B use Cisco Unified Communications Manager Express. IP Phone A is on a call with IP Phone B. IP Phone A conferences in Analog Phone C with IP Phone B. Software conference resources are not being used. Drag the appropriate DSP resource for each gateway from the list to the correct locations in the graphic so the call can be completed.



Explanation:

Order is:

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_0_1/ccmsys/a05dsp.html

QUESTION NO: 77 DRAG DROP

Drag the appropriate IOS commands from the left and drop them in the spaces on the right to create a dial peer that will match all inbound calls and prevent two-stage dialing on a T1 PRI circuit. Not all boxes are used and not all options are used.

voice-port 0/0/0:15	
dial-peer voice 1 voip	
direct-inward-dial	
answer-address T	
no dial-peer outbound status-check pots	
incoming called-number .	
destination-pattern T	
dial-peer voice 1 pots	
port 0/0/0:15	

Answer:

Drag the appropriate IOS commands from the left and drop them in the spaces on the right to create a dial peer that will match all inbound calls and prevent two-stage dialing on a T1 PRI circuit. Not all boxes are used and not all options are used.

voice-port 0/0/0:15	dial-peer voice 1 pots
dial-peer voice 1 voip	incoming called-number .
direct-inward-dial	direct-inward-dial
answer-address T	
no dial-peer outbound status-check pots	
incoming called-number .	
destination-pattern T	
dial-peer voice 1 pots	
port 0/0/0:15	

Explanation:

voice-port 0/0/0:15	dial-peer voice 1 pots
dial-peer voice 1 voip	incoming called-number .
direct-inward-dial	direct-inward-dial
answer-address T	
no dial-peer outbound status-check pots	
incoming called-number .	
destination-pattern T	
dial-peer voice 1 pots	
port 0/0/0:15	

In the case of Digital Interfaces, when the PBX or central office (CO) switch sends a setup message that contains all the digits necessary to fully route the call, those digits can be mapped to an outbound Voice over IP (VoIP) dial-peer (or hairpin to plain old telephone service (POTS) dial-peer directly). The router/gateway does not present a secondary dial tone to the caller and does not collect digits. It forwards the call directly to the configured destination. In the case of analog interfaces, the user only hears the dial tone once (either local or remote), and then dials the digits and gets through to the destination phone. This is called one stage dialing. When one receives an inbound call from a POTS interface, the *Direct Inward Dial* (DID) feature in dial-peers enables the router/gateway to use the called number (dialed number identification service (DNIS)) to directly match an outbound dial-peer. When DID is configured on the inbound POTS dial-peer, the called number is automatically used to match the destination pattern for the outbound call leg. The *incoming called number* command will match the dial-peer that has the DID configured. http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a00800e00d0.shtml

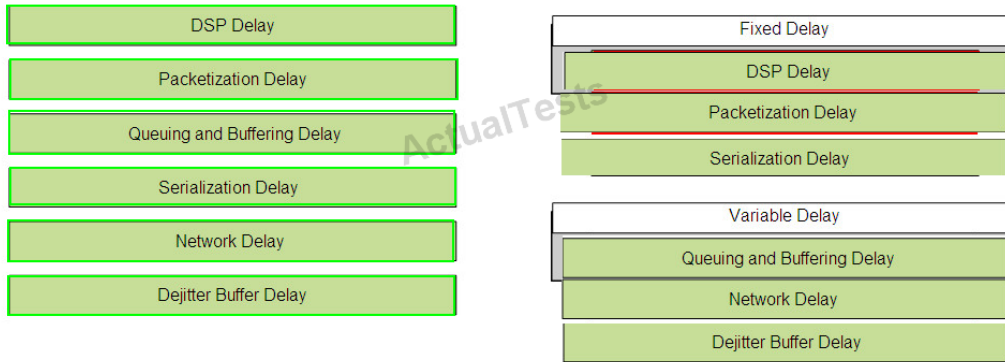
QUESTION NO: 78 DRAG DROP

Drag the components from the left and drop them under the appropriate categories on the right.

DSP Delay	Fixed Delay
Packetization Delay	
Queuing and Buffering Delay	
Serialization Delay	
Network Delay	Variable Delay
Dejitter Buffer Delay	

Answer:

Drag the components from the left and drop them under the appropriate categories on the right.



Explanation: DSP delay, Packetization delay, Serialization delay & De jitter Buffer delay are Fixed delay types. Queuing and Buffering delay & Network delay are Variable Delay types.

http://www.cisco.com/en/US/tech/tk652/tk698/technologies_white_paper09186a00800a8993.shtml

Fixed Delay

DSP Delay

Serialization Delay

De jitter Buffer Delay

Processing Delay

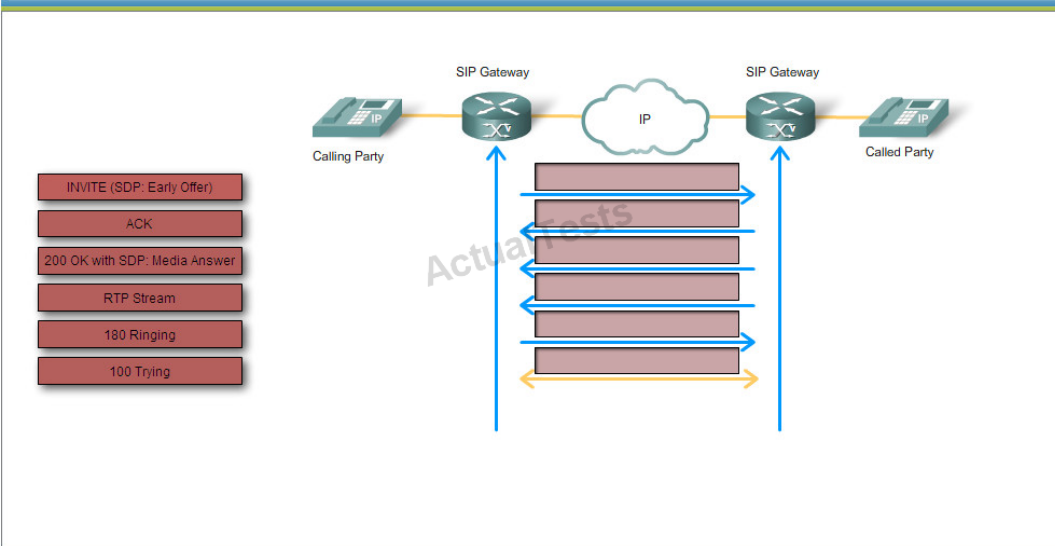
Variable Delay

Network Delay

Queuing Delay

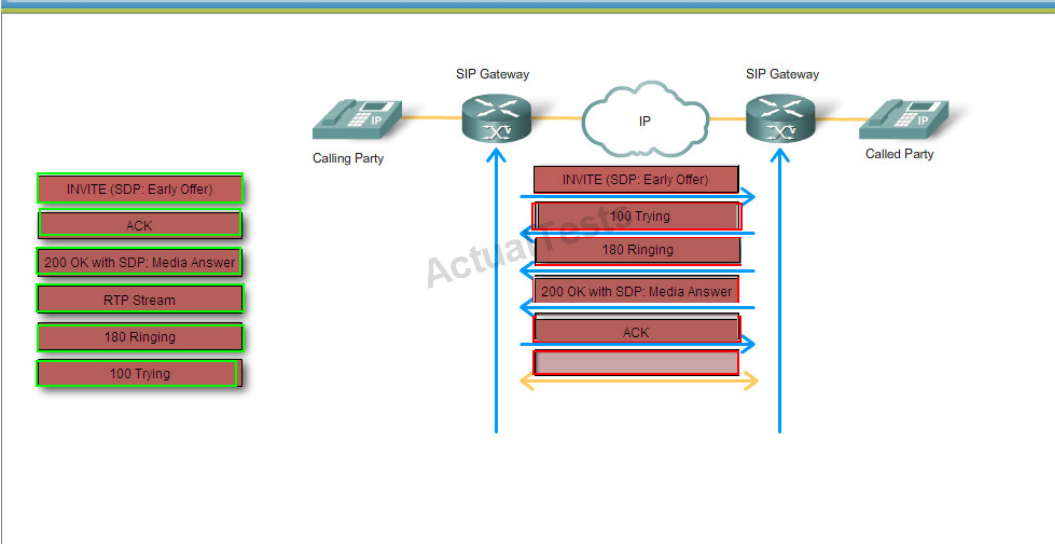
QUESTION NO: 79 DRAG DROP

Drag the signaling streams to support SIP Early Offer from the left and drop them in the correct box in the graphic on the right.



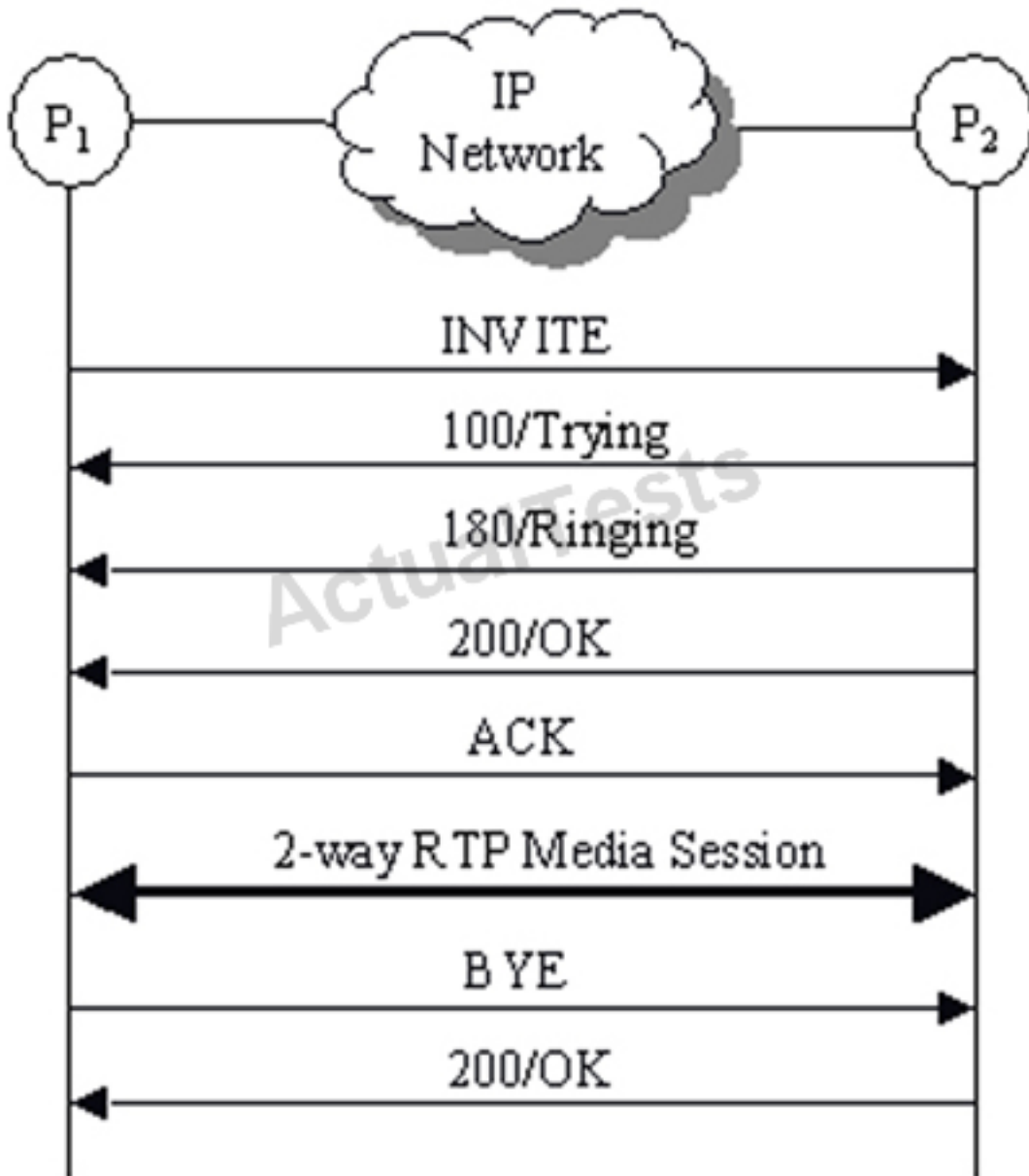
Answer:

Drag the signaling streams to support SIP Early Offer from the left and drop them in the correct box in the graphic on the right.



Explanation:

Call Flow of a Typical sip Session



QUESTION NO: 80 DRAG DROP

Drag the attributes of a scalable numbering plan from the left and place them in the boxes on the right.

Provides summarization.	
Flat numbering plan.	
Distribution of dial plan logic.	
Conforms to international and private standards.	
Simplicity in provisioning.	
Reduces post-dial delay.	
Availability and fault tolerance.	
Complex call routing.	
Requires management from multiple points in the network.	

Answer:

Drag the attributes of a scalable numbering plan from the left and place them in the boxes on the right.

Provides summarization.	Distribution of dial plan logic.
Flat numbering plan.	Provides summarization.
Distribution of dial plan logic.	Simplicity in provisioning.
Conforms to international and private standards.	Reduces post-dial delay.
Simplicity in provisioning.	Availability and fault tolerance.
Reduces post-dial delay.	
Availability and fault tolerance.	
Complex call routing.	
Requires management from multiple points in the network.	

Explanation:

When designing a large-scale dial plan, Cisco recommends you adhere to the following attributes:

- Logic distribution: Good dial plan architecture relies on the effective distribution of the dial plan logic among the various components. Devices that are isolated to a specific portion of the dial plan reduce the complexity of the configuration. Each component focuses on a specific task accomplishment. Generally, the local switch or gateway handles details that are specific to the local point of presence (POP). Higher-level routing decisions are passed along to the gatekeepers and PBXs. A well-designed network places the majority of the dial plan logic at the gatekeeper devices.
- Hierarchical design (scalability): You should attempt to keep the majority of the dial plan logic (routing decisions and failover) at the highest-component level. Maintaining a hierarchical design makes the addition and deletion of number groups more manageable. Scaling the overall network is much easier when configuration changes are made to a single component.
- Simplicity in provisioning. Keep the dial plan simple and symmetrical when designing a network. Try to keep consistent dial plans on the network by using translation rules to manipulate the local digit dialing patterns. These number patterns are normalized into a standard format or pattern before the digits enter the VoIP core. Putting digits into a standard format simplifies provisioning

and dial-peer management.

- Reduction in postdial delay: Consider the effects of postdial delay in the network when you design a large-scale dial plan. Postdial delay is the time between the last digit dialed and the moment the phone rings at the receiving location. In the PSTN, people expect a short postdial delay and to hear ringback within seconds. The more translations and lookups that take place, the longer the postdial delay becomes. Overall network design, translation rules, and alternate pathing affect postdial delay. Therefore, you should efficiently use these tools to reduce postdial delay.
- Availability and fault tolerance. Consider overall network availability and call success rates when you design a dial plan. Fault tolerance and redundancy within VoIP networks are most important at the gatekeeper level. By using an alternate path you help provide redundancy and fault tolerance in the network.
- Conformance to public standards: Different geographical locations might impose restrictions to your dial plan. Therefore, familiarize yourself with any such limitations prior to designing your dial plan.

QUESTION NO: 81 DRAG DROP

Assume a SIP voice network. Drag each characteristic to the type of SIP call setup the characteristics best describes.

Most dynamic address resolution capability	Server reports back to a UA with destination coordinates
UA incapable of establishing its own sessions	All setup messages to through server
UA must keep data on large number of destinations	Nonscalable
Relies on cached information to resolve addresses	

Direct Call Setup	Proxy Server Call Setup
Place here	Place here
Place here	Place here
Place here	Place here

Redirect Server Call Setup
Place here

Answer:

Most dynamic address resolution capability	Server reports back to a UA with destination coordinates
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Direct Call Setup

Proxy Server Call Setup

Nonscalable	Most dynamic address resolution capability
UA must keep data on large number of destinations	All setup messages to through server
Relies on cached information to resolve addresses	UA incapable of establishing its own sessions

Redirect Server Call Setup

Server reports back to a UA with destination coordinates
--

Explanation:

Most dynamic address resolution capability	Server reports back to a UA with destination coordinates
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Direct Call Setup

Proxy Server Call Setup

Nonscalable	Most dynamic address resolution capability
UA must keep data on large number of destinations	All setup messages to through server
Relies on cached information to resolve addresses	UA incapable of establishing its own sessions

Redirect Server Call Setup

Server reports back to a UA with destination coordinates
--

Direct call setup:+ Nonscalable+ UA must keep data on large number of destinations+ Relies on cached information to resolve addresses

Redirect Server Call Setup:+ Server reports back to a UA with destination coordinates

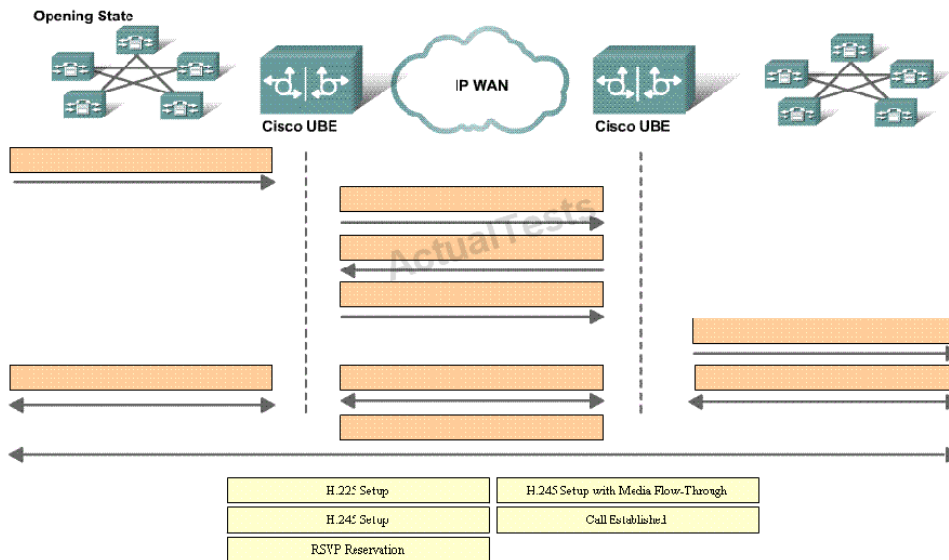
Proxy Server Call Setup:+ Most dynamic address resolution capability+ All setup messages to through server+ UA incapable of establishing its own sessions

http://www.cisco.com/en/US/tech/tk652/tk701/technologies_configuration_guide_chapter09186a0080163444.html

QUESTION NO: 82 DRAG DROP

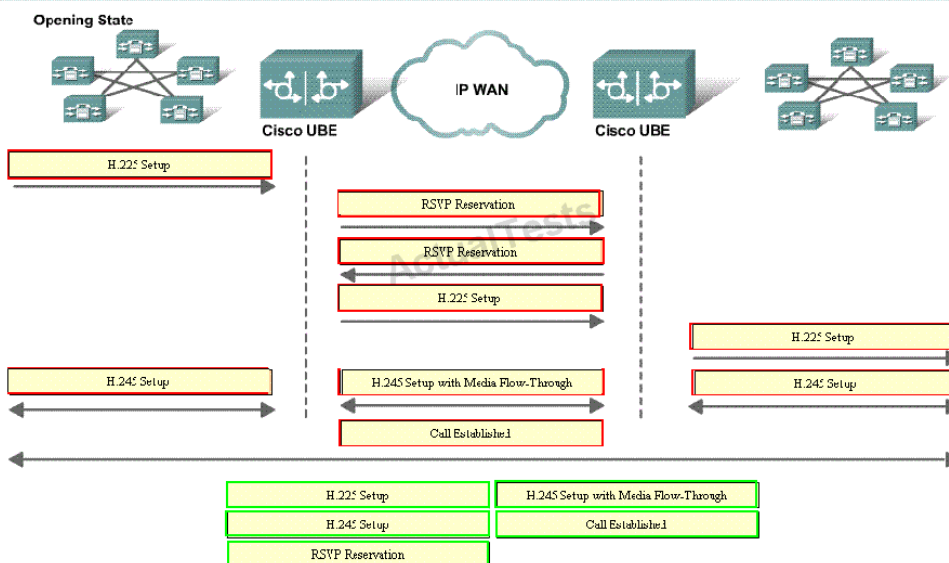
Drop

Drag and drop the appropriate call-signaling term to the correct box in the diagram to establish RSVP-based Call Admission Control between the two Cisco Unified Border Elements (Cisco UBEs). Some options may be used more than once.



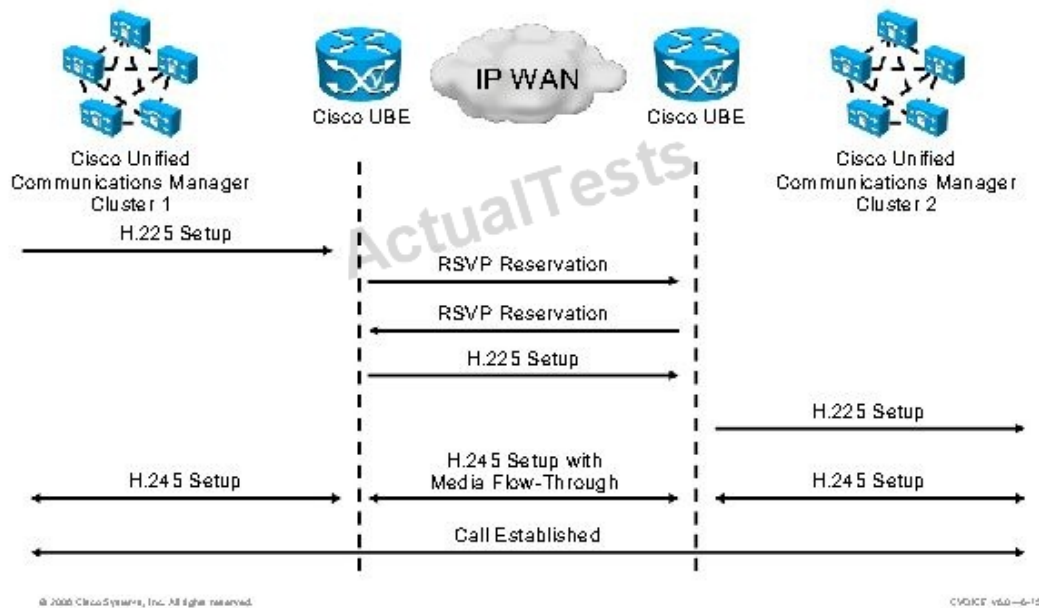
Answer:

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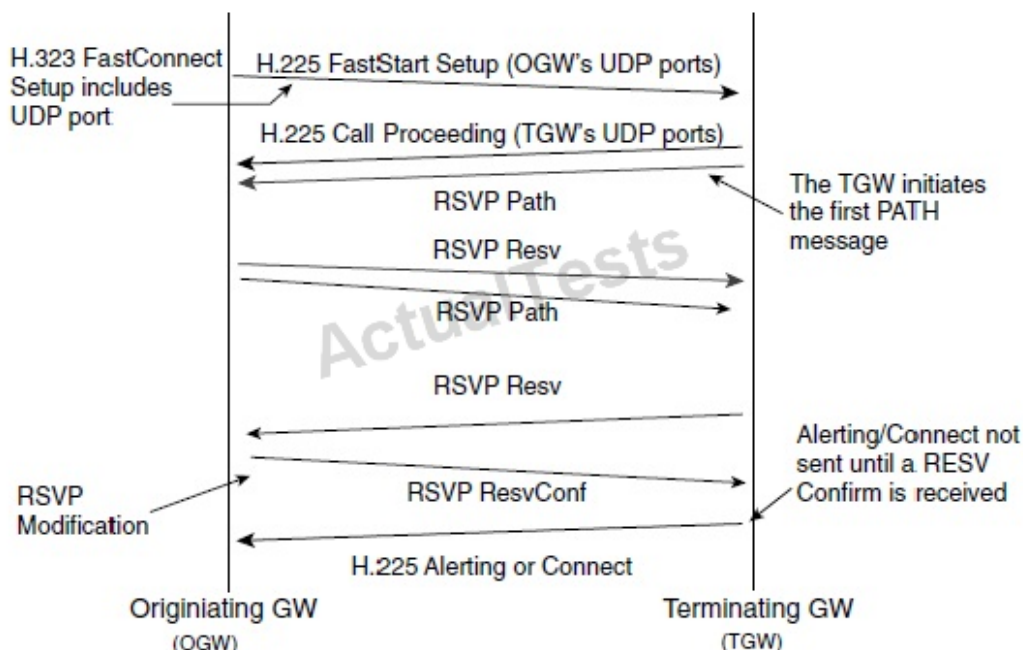


Explanation:

RSVP-Based CAC on Cisco UBE (Cont.)



The H.323 setup is suspended before the destination phone, triggered by the H.225 alerting message, starts ringing. The RSVP reservation is made in both directions because a voice call requires a two-way speech path and therefore bandwidth in both directions. The terminating gateway ultimately makes the CAC decision based on whether or not both reservations succeed. At that point the H.323 state machine continues either with an H.225 Alerting/Connect (the call is allowed and proceeds), or with an H.225 Reject/Release (call is denied). The RSVP reservation is in place by the time the destination phone starts ringing and the caller hears ringback.



http://www.cisco.com/en/US/docs/ios/12_1t/12_1t5/feature/guide/dt4trsvp.html

QUESTION NO: 83 DRAG DROP

Click and drag the feature on the left to the category it belongs to on the right.

supports analog faxes and modems on a VoIP network	Gateway
performs call setup and teardown between VoIP networks and the PSTN	
interconnects segments of the same or different VoIP networks using different media types	CUBE
interconnects segments of the same or different VoIP networks using different signaling types	

Answer:

Click and drag the feature on the left to the category it belongs to on the right.

supports analog faxes and modems on a VoIP network	Gateway
performs call setup and teardown between VoIP networks and the PSTN	
interconnects segments of the same or different VoIP networks using different media types	CUBE
interconnects segments of the same or different VoIP networks using different signaling types	

Explanation: Gateway: Supports Analog Faxes and Modems on a Voip Network

Performs Call Setup and teardown between Voip Networks & the PSTN

CUBE. Interconnects segments of the same or different VoIP networks using different media types

Interconnects segments of the same or different VoIP networks using different media types

Gateway Functionality : Gateways are responsible Media stream handling and speech path integrity, DTMF relay, Fax relay and pass-through, Digit translation and call processing, Dial peers and codec filtering, Carrier ID handling, Termination and re-origination of signaling and media. The Cisco Unified Border Element is a session border controller designed to provide easy, secure, and cost-effective connectivity between independent unified communications networks or network

domains for different enterprises. It provides interconnection between incompatible applications within the enterprise network, between different enterprises for business-to-business applications, and between enterprise networks and service provider Session Initiation Protocol (SIP) trunks. The Cisco Unified Border Element provides key session management capabilities, H.323 and SIP interworking functions, and network-to-network interface security and demarcation capabilities. It performs most of the same functions of a public switched telephone network (PSTN)-to-IP gateway but joins two VoIP call legs. Media packets can either flow through (thus hiding the networks from each other) or around the Cisco Unified Border Element platform

http://www.cisco.com/en/US/docs/ios/voice/cube/configuration/guide/vb-gw-overview_ps10591_TSD_Products_Configuration_Guide_Chapter.html

QUESTION NO: 84 DRAG DROP

Drop

Click and drag the type of call on the left to the type of voice port it applies to on the right.

local	FXO
PLAR	FXS
on-net	FXS or switch
off-net	E & M, FXO, FXS
PBX to PBX	T1 or E1 with CAS or PRI

Answer:

Click and drag the type of call on the left to the type of voice port it applies to on the right.

local	off-net
PLAR	local
on-net	on-net
off-net	PLAR
PBX to PBX	PBX to PBX

Explanation: 1) T1 or E1 with CAS or PRI: PBX to PBX2) FXO: off-net3) FXS: local4) FXS or

switchH. on-net5) E&M, FXO, FXS: PLAR

Explanation

PBX to PBX connections can use T1 or E1 with CAS or PRI: PBX can connect to a network through T1 or E1 lines with channel associated signaling (CAS) or Primary Rate Interface (PRI) signaling.

For off-net calls, the typical connection between the router and the PSTN is through FXO port.

A local call just needs FXS ports so it is the only choice for this type of call.

We can make on-net calls through FXS port (phone directly connected to the router) or FXO port (phone connected to a PBX). The “switch” here means that we can connect an IP phone through a switch and place on-net calls through Cisco Unified Communications Manager.

A PLAR call can work with any type of signaling, including E&M, FXO, FXS interfaces.

Topic 2, Volume B**QUESTION NO: 85**

In the destination patterns, which wildcard symbol indicates a single-digit placeholder?

- A. ()
- B. +
- C. . (period)
- D. %

Answer: C

Explanation:

QUESTION NO: 86

Which voice feature operates the same as a firewall on a data network?

- A. digit manipulation
- B. call coverage
- C. calling privileges
- D. call routing and path selection

Answer: C

Explanation:

QUESTION NO: 87

Which three call permissions are assigned with the Employee calling privileges? (Choose three.)

- A. long distance
- B. international
- C. 911 (emergency)
- D. local

Answer: A,C,D

Explanation:

QUESTION NO: 88

How many bits are added to a secure Real-Time Transport Protocol packet from the 160-bit SHA-1 hash?

- A. 160
- B. 32
- C. 64
- D. 128

Answer: B

Explanation:

QUESTION NO: 89

Which traditional telephony protocol was used as a basis for the H.323 suite of protocols?

- A. Q.921
- B. Q.931
- C. SS7
- D. SCCP

Answer: B

Explanation:

QUESTION NO: 90

Which component in the Media Gateway Control Protocol environment is responsible for controlling the operation of the gateways?

- A. gatekeeper
- B. gate master
- C. call agent
- D. calling authority

Answer: C

Explanation:

QUESTION NO: 91

Which proprietary voice client-server protocol sends traffic back to Cisco Unified Communications Manager with every digit pressed on the endpoint?

- A. H.323 Protocol
- B. Media Gateway Control Protocol
- C. Session Initiation Protocol
- D. Skinny Client Control Protocol

Answer: D

Explanation:

QUESTION NO: 92

If a centralized solution has to be implemented on multiple-equipment vendors devices, which signaling protocol should be used?

- A. Session Initiation Protocol
- B. Media Gateway Control Protocol
- C. Skinny Client Control Protocol
- D. H.323 protocol

Answer: D

Explanation:

QUESTION NO: 93

Which codec is the best option when a voice bandwidth of 8kbps or below is required with the highest voice quality?

- A. G.726
- B. G.728
- C. G.711
- D. G.729

Answer: D

Explanation:

QUESTION NO: 94

When Cisco Unified Communications Manager Express is used, which type of files are used to enable phone displays and operations?

- A. phone GUI files
- B. phone firmware files
- C. Unified Communications Manager Express basic files
- D. Unified Communications Manager Express TSP archive files

Answer: B

Explanation:

QUESTION NO: 95

Which command should you use to associate a Session Initiation Protocol phone using a tag of 1 with a directory number with a tag of 20?

- A. button 1:20
- B. button 20:1
- C. number 20 dn 1
- D. number 1 dn 20

Answer: D

Explanation:

QUESTION NO: 96

Which Cisco Unified Communications Manager component provides direct digital-to-digital conversion from one codec to another?

- A. media termination point
- B. media converter
- C. digital signal processor
- D. coder

Answer: C

Explanation:

QUESTION NO: 97

Which command should you use to configure a T1 CAS trunk to use the most reliable line coding technique?

- A. linecoding ami
- B. linecode b8zs
- C. linecode ami
- D. linecoding b8zs

Answer: B

Explanation:

QUESTION NO: 98

When you configure a VoIP dial peer, which command should be used to configure the remote gateway with the destination IPv4 address 172.16.1.118?

- A. session target ipv4:172.16.1.118
- B. remote target ipv4:172.16.1.118
- C. destination address 172.16.1.118
- D. destination ipv4:172.16.1.118

Answer: A

Explanation:

QUESTION NO: 99

Which digit manipulation command should be used to globally expand local 4-digit extension numbers beginning with a 4 to a full telephone number starting with 1919555 when calling outbound?

- A. prefix 1919555
- B. num-exp 4... 19195554...
- C. num-exp 4... 1919555...
- D. prefix 19195554

Answer: B

Explanation:

QUESTION NO: 100

Which command can be used to display the outgoing dial peer that is reached when the telephone number 919195551234 is dialed?

- A. show dialplan 919195551234
- B. show number 919195551234
- C. show dial-peer number 919195551234
- D. show dialplan number 919195551234

Answer: D

Explanation:

QUESTION NO: 101

Which command should you use to configure a dial peer to support T.38 fax relay and to use Cisco fax relay if T.38 negotiation is unsuccessful?

- A. fax protocol t38 fallback cisco
- B. fax t38 fallback cisco
- C. fax relay t38 cisco
- D. fax relay t38 backup cisco relay

Answer: A

Explanation:

QUESTION NO: 102

The command address-hiding is entered to enable the SIP-to-SIP address hiding feature. In which configuration mode is this command entered?

- A. VoIP dial-peer configuration mode
- B. SIP configuration mode
- C. voice service VoIP configuration mode
- D. Cisco Unified Border Element configuration mode

Answer: C

Explanation:

QUESTION NO: 103

Which two commands should you use on a common gateway between the two IP telephony networks to enable SIP to H.323 interworking? (Choose two.)

- A. allow-connections h323 to sip
- B. voice service h323 to sip
- C. voice service sip to h323
- D. allow-connections sip to h323

Answer: A,D

Explanation:

QUESTION NO: 104

Which debug command can be used to show events specific to the Cisco Unified Border Element gateway?

- A. debug ip voip cube
- B. debug voip ipipgw
- C. debug ip cube
- D. debug voip cube events

Answer: B

Explanation:

QUESTION NO: 105

Which type of delay describes the amount of time it takes to place a frame onto a physical medium?

- A. propagation delay
- B. processing delay
- C. serialization delay
- D. queuing delay

Answer: C

Explanation:

QUESTION NO: 106

Which command will correctly map Class of Service mappings 0 through 7 to Differentiated Services Code Point 0, 10, 18, 26, 34, 46, 48, and 56, accordingly?

- A. mls qos map 0 10 18 26 34 46 48 56
- B. mls quos cos map 0 10 18 26 34 46 48 56
- C. mls qos cos-dscp map 0 10 18 26 34 46 48 56
- D. mls qos map cos-dscp 0 10 18 26 34 46 48 56

Answer: D

Explanation:

QUESTION NO: 107

Which queuing method is the basis for the low latency queuing?

- A. weighted random early detection
- B. custom queuing
- C. weighted fair queuing
- D. class-based weighted fair queuing

Answer: D

Explanation:

QUESTION NO: 108

To avoid unnecessary delay for high-priority traffic, on which speed link should you enable the link fragmentation and interleaving feature?

- A. less than 768 kb/s
- B. less than 1.544 Mb/s
- C. less than 2.048 Mb/s
- D. less than 512 kb/s

Answer: A

Explanation:

QUESTION NO: 109

Which command is used to enable the AutoQoS feature on an incoming interface while also trusting existing quality of service markings?

- A. router(config-if)#auto qos voip
- B. router(config-if)#auto qos voip trust
- C. router(config)#auto qos voip trust interface interface
- D. router(config)#auto qos voip interface interface

Answer: B

Explanation:

QUESTION NO: 110

Which command is used to assign 20% of the bandwidth of an interface to a traffic class with priority?

- A. router(config-cmap)#priority percent 20
- B. router(config-pmap-c)#bandwidth percent 20
- C. router(config-pmap-c)#priority percent 20
- D. router(config-cmap)#bandwidth percent 20

Answer: C

Explanation:

QUESTION NO: 111

Which type of North American Numbering Plan number code is used to designate a number for special purposes?

- A. easily recognizable codes
- B. carrier identification codes
- C. service codes
- D. Automatic Number Identification II digits

Answer: A

Explanation:

QUESTION NO: 112

What is the maximum number of digits that can be assigned to a European Subscriber Number using the European Telephony Numbering Space?

- A. 7
- B. 15
- C. 10
- D. 17

Answer: B

Explanation:

QUESTION NO: 113

Which digit manipulation feature allows a partial telephone number to be prepended with a specific set of digits and is applied to all calls?

- A. digit prefixes
- B. forward digits
- C. digit extension
- D. number expansion

Answer: D

Explanation:

QUESTION NO: 114

Which path-selection strategy can be used to avoid expensive public switched telephone network calls when an existing network (IP) link exists between sites?

- A. toll-bypass
- B. explicit preference
- C. site-code dialing
- D. Tail-End Hop-Off

Answer: A

Explanation:

QUESTION NO: 115

When you implement the tail-end hop-off path-selection strategy, which task should you complete first?

- A. Define the VoIP inbound digit manipulation.
- B. Define the VoIP outbound digit manipulation.
- C. Define the outbound VoIP dial peer.
- D. Define the inbound VoIP dial peer.

Answer: B

Explanation:

QUESTION NO: 116

When a Session Initiation Protocol user agent client initiates a call to another user agent server, what is the first message type that is sent to the Cisco SIP Proxy Server?

- A. invite
- B. trying
- C. initiate
- D. setup

Answer: A

Explanation:

QUESTION NO: 117

Which Session Initiation Protocol server role is given to the component that implements a mechanism to resolve addresses?

- A. Location server
- B. Naming server
- C. Session Initiation Protocol proxy
- D. Session Initiation Protocol router

Answer: A

Explanation:

QUESTION NO: 118

Which address type would Session Initiation Protocol address user@cisco.com classify as?

- A. E.164
- B. mixed format
- C. username at a fully qualified domain name
- D. email

Answer: C

Explanation:

QUESTION NO: 119

Which networking feature typically is used on an IP phone that is also connected to a local computer to maintain separation between the voice and data traffic?

- A. virtual LAN
- B. class of service
- C. quality of service
- D. port security

Answer: A

Explanation:

QUESTION NO: 120

Which version of Cisco Unified Communications Manager Express is recommended that is supported on endpoints running Cisco IOS 15.0(1)M?

- A. 8
- B. 7
- C. 7.1
- D. 8.5

Answer: C

Explanation:

QUESTION NO: 121

A TFTP server is configured with the IP address 10.10.10.150. Which command should you enter in the DHCP pool configuration mode to configure a client to use the defined TFTP server?

- A. option 150 10.10.10.150
- B. tftp-server ip 10.10.10.150
- C. tftp-server 10.10.10.150
- D. option 150 ip 10.10.10.150

Answer: D

Explanation:

QUESTION NO: 122

The file "apps31.9-1-1TH1-16.sbn" is located in the flash memory of the device. Which command should you enter on a Cisco IOS device to serve this file correctly using Trivial File Transfer Protocol?

- A. tftp-server flash:apps31.9-1-1TH1-16.sbn
- B. tftp-server apps31.9-1-1TH1-16.sbn
- C. copy tftp flash:apps31.9-1-1TH1-16.sbn

D. tftp server:apps31.9-1-1TH1-16.sbn

Answer: A

Explanation:

QUESTION NO: 123

Which type of ephone-dn can be used to support one virtual voice port with support for two channels?

- A. single-line
- B. dual-line
- C. dual-channel
- D. dual-voice-channel

Answer: B

Explanation:

QUESTION NO: 124

Which type of Session Initiation Protocol directory number supports the assignment of up to 10 telephone numbers and up to two active calls?

- A. multiple-number directory number (using single-line ephone-dns)
- B. dec-line
- C. multiple-number directory number (using dual-line ephone-dns)
- D. sip-ten-line

Answer: C

Explanation:

QUESTION NO: 125

Which command should you use to configure a Skinny Call Control Protocol directory number with an extension of 1003 and secondary number 9195551003?

- A. number 1003 secondary 9195551003
- B. number 1003 9195551003

- C. extension 1003 9195551003
- D. extension 1003 secondary 9195551003

Answer: A

Explanation:

QUESTION NO: 126

Which Cisco Unified Communications Manager component acts as a voice switch between multiple telephony circuits and can provide signaling and media conversion?

- A. gatekeeper
- B. gateway
- C. media exchanger
- D. PBX

Answer: B

Explanation:

QUESTION NO: 127

Which type of analog voice port would be used on a telephony device to connect to a telephone or fax machine?

- A. T1
- B. ear and mouth
- C. Foreign Exchange Office
- D. Foreign Exchange Station

Answer: D

Explanation:

QUESTION NO: 128

A call is received on a voice gateway from a Session Initiation Protocol-based Internet source. The call is destined for a telephone that is connected directly to the gateway. Which type of dial-peer is considered outgoing?

- A. plain old telephone service
- B. Foreign Exchange Station
- C. Foreign Exchange Office
- D. VoIP

Answer: A

Explanation:

QUESTION NO: 129

Which codec is considered to have high complexity that limits the number of active voice channels on a gateway?

- A. G.729AB
- B. G.729
- C. G.726
- D. G.722

Answer: B

Explanation:

QUESTION NO: 130

Which command should you use to configure an analog ear and mouth voice port to use the most popular type outside of North America?

- A. Type II
- B. Type I
- C. Type V
- D. Type IV

Answer: C

Explanation:

QUESTION NO: 131

Which calling privileges command can be used to assign a specific configuring class of restriction list named "restrict" incoming on a dial peer?

- A. COR list restrict inbound
- B. corlist restrict
- C. corlist incoming restrict
- D. cor assign restrict

Answer: C

Explanation:

QUESTION NO: 132

Which Cisco voice feature can you use to connect together two Session Initiation Protocol networks?

- A. Cisco Transcoding
- B. Cisco Unified Border Element
- C. SIP gatekeeper
- D. class of restriction

Answer: B

Explanation:

QUESTION NO: 133

Two networks have Resource Reservation Protocol and Cisco Unified Border Element gateway configuration. To RSVP reserve bandwidth and guarantee a minimum bit rate between these two networks, which command should you use on the outgoing gateway dial peer of the Cisco Unified Border Element?

- A. req-qos guaranteed-delay
- B. acc-qos guaranteed-delay
- C. ip rsvp bandwidth bandwidth
- D. h323-gateway voip rsvp-reserve

Answer: A

Explanation:

QUESTION NO: 134

Which type of delay is caused by the distance that signal must travel in fiber- or copper-based

networks?

- A. handling delay
- B. processing delay
- C. serialization delay
- D. propagation delay

Answer: D

Explanation:

QUESTION NO: 135

Which feature can be used to avoid traffic congestion by dropping low-priority packets rather than dropping high-priority packets and is not recommended for voice networks?

- A. weighted random early detection
- B. tail-drop
- C. class-based weighted fair queuing
- D. low latency queuing

Answer: A

Explanation:

QUESTION NO: 136

Which queuing mechanism guarantees low-latency propagation and high-bandwidth priority?

- A. Class-based weighted fair queuing
- B. Low latency queuing
- C. Priority queuing
- D. Custom queuing

Answer: B

Explanation:

QUESTION NO: 137

What is the maximum amount of one-way delay that is considered acceptable per G.114 for a

voice call with little or no concern for quality issues?

- A. 200 ms
- B. 1 sec
- C. 150 ms
- D. 500 ms

Answer: C

Explanation:

QUESTION NO: 138

What is the maximum percentage of packet loss that can be accepted on VoIP traffic?

- A. less than 5%
- B. 0
- C. less than 2%
- D. less than 1%

Answer: D

Explanation:

QUESTION NO: 139

Which of the available quality of service models offers the greatest amount of scalability while maintaining service quality?

- A. Diffserv
- B. IntServ
- C. Best Effort
- D. PriServ

Answer: A

Explanation:

QUESTION NO: 140

Which of the Diffserv differentiated services code point per-hop behaviors is used for low-delay

services including VoIP and video over IP?

- A. Assured Forwarding
- B. Expedited Forwarding
- C. default
- D. class selector

Answer: B

Explanation:

QUESTION NO: 141

What is the total number of differentiated services code point bits in the DiffServ field?

- A. 2
- B. 4
- C. 6
- D. 8

Answer: C

Explanation:

QUESTION NO: 142

At what point in a VoIP network are the existing class of service or differentiated services code point values considered valid and are used throughout the rest of the network?

- A. trust boundary
- B. marking boundary
- C. access layer
- D. distribution layer

Answer: A

Explanation:

QUESTION NO: 143

Which command correctly enables a trust of the existing differentiated services code point value

coming into a switch?

- A. switch(config-mls)#mls qos trust dscp
- B. switch(config-if)#mls qos trust dscp
- C. switch(config-if)#mls dscp trust
- D. switch(config-mls)#dscp trust

Answer: B

Explanation:

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QUESTION NO: 144

Which three IOS commands are used by Cisco Unified Communications Manager Express for digit manipulations? (Choose three.)

- A. prefix digits
- B. session target
- C. forward-digit numbers
- D. num-exp match
- E. no dtmf-relay
- F. target session

Answer: A,C,D

Explanation:

QUESTION NO: 145

What is the purpose of a COR list?

- A. to identify the incoming call leg with the highest priority
- B. to create a set of dial peers for IP phones
- C. to restrict the ability of phones to make certain types of calls
- D. to list a detailed configuration for an IP phone and voice port

Answer: C

Explanation:

QUESTION NO: 146

Which command is used to view the COR list?

- A. show voip ccapi inout
- B. show dial-peer cor
- C. show dial-peer summary
- D. show ip interface brief

Answer: B

Explanation:

QUESTION NO: 147

Which two protocols are peer-to-peer protocols? (Choose two.)

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

Answer: C,D

Explanation:

QUESTION NO: 148

Which protocol has a centralized configuration?

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

Answer: A

Explanation:

QUESTION NO: 149

Which signaling protocol is proprietary to Cisco?

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

Answer: D

Explanation:

QUESTION NO: 150

Which four are benefits of VLANs in an organization? (Choose four.)

- A. increased performance
- B. improved manageability
- C. better VLAN database
- D. quick response to ping times
- E. physical topology independence
- F. better LAN infrastructure
- G. switch is able to reboot faster
- H. increased security

Answer: A,B,E,H

Explanation:

QUESTION NO: 151

Which codec supports VAD and CNG?

- A. G.723.1
- B. G.726
- C. G.729B
- D. G.711

Answer: C

Explanation:

QUESTION NO: 152

Which command should you use when checking for phone registration status or verifying button assignments?

- A. show IP phone
- B. show ephone-dn 9
- C. show ephone
- D. show telephony-service

Answer: C

Explanation:

QUESTION NO: 153

Which option inside DHCP pool configuration gives a TFTP server to a Cisco IP phone?

- A. option 82
- B. option 110
- C. option 150
- D. option 176

Answer: C

Explanation:

QUESTION NO: 154

Which two signaling methods are supported on Cisco FXO and FXS ports? (Choose two.)

- A. loop-start
- B. ground-start
- C. AMI
- D. E&M
- E. kewlstart

Answer: A,B

Explanation:

QUESTION NO: 155

Which media resource requires hardware digital signal processors?

- A. conference bridge
- B. music on hold
- C. annunciator
- D. transcoding

Answer: D

Explanation:

QUESTION NO: 156

Which four commands are used to configure VoIP dial peers? (Choose four.)

- A. forward-digits
- B. destination-pattern
- C. description
- D. dial-peer voice
- E. codec
- F. port number
- G. prefix
- H. session target

Answer: B,D,E,H

Explanation:

QUESTION NO: 157

Which four are critical functions that are performed by the Cisco Unified Border Element? (Choose four.)

- A. handling of DoS attacks
- B. migration control
- C. QoS accounting and billing
- D. choice of deployment support
- E. flexibility on a single platform
- F. critical log captures
- G. monitoring and troubleshooting deployments
- H. investment protection

Answer: B,D,E,H

Explanation:

QUESTION NO: 158

Which command allows verification of RSVP configuration?

- A. show ip rsvp interface
- B. show ip rsvp installed detailed
- C. show ip interface detail
- D. show ip rsvp policy

Answer: A

Explanation:

QUESTION NO: 159

What is the function of the command `crypto pki trustpoint name` when you are configuring Cisco Unified Border Element for SRTP-RTP Internetworking?

- A. It declares the trustpoint that the router uses and enters CA-trustpoint configuration mode.
- B. It specifies the key pair to associate with the certificate.
- C. It specifies the enrollment parameters of a CA.
- D. It obtains the certificate for the router from the CA.

Answer: A

Explanation:

QUESTION NO: 160

Which four voice and video issues are caused by the absence of QoS? (Choose four.)

- A. jitter
- B. delays
- C. circuit issues
- D. call setup failure
- E. echo
- F. defective CAT cable

G. dropped calls

Answer: A,B,E,G

Explanation:

QUESTION NO: 161

What is the Cisco-recommended network requirement for voice and video?

- A.** over 3.0 megabytes circuit
- B.** Cisco AutoQoS
- C.** packet loss of 1 percent or less
- D.** -6 on gateway attenuation

Answer: C

Explanation:

QUESTION NO: 162

Which statement about the DiffServ model is true?

- A.** DiffServ provides traffic management using QoS classifications.
- B.** DiffServ specifies a mechanism for classifying and managing network traffic and providing QoS on modern IP networks.
- C.** DiffServ is based on building a virtual circuit in the Internet using the bandwidth reservation technique.
- D.** DiffServ is a way to provide the assurance of sufficiently low delay and packet loss for certain types of applications or traffic.

Answer: B

Explanation:

QUESTION NO: 163

The IP precedence marking uses which bits of the ToS IP header field?

- A.** the two left-most bits
- B.** the three left-most bits

- C. the four left-most bits
- D. the seven left-most bits

Answer: B

Explanation:

QUESTION NO: 164

Using the Cisco QoS baseline model, what are the PHB, DSCP, and CoS values for voice traffic, respectively?

- A. EF, 42, and 3
- B. AF41, 42, and 7
- C. EF, 46, and 5
- D. AF41, 46, and 1

Answer: C

Explanation:

QUESTION NO: 165

What does policy-based routing use to determine the next hop?

- A. routing tables
- B. packet options that were developed by the network admin
- C. destination address
- D. MAC address lookup table

Answer: D

Explanation:

QUESTION NO: 166

Which WAN connection speeds are recommended for using link fragmentation and interleaving?

- A. speeds at 768 and lower
- B. speeds at T1 and below
- C. speeds between 512 and 768

D. speeds between 64 and 512

Answer: A

Explanation:

QUESTION NO: 167

Which two statements describe ways that AutoQoS VoIP configuration is applied on a Cisco catalyst switch? (Choose two.)

- A. The auto qos voip command is applied to each interface.
- B. The auto qos voip command is applied globally in the switch.
- C. The auto qos voip cisco-phone command is applied to each interface.
- D. The auto qos voip trust command is applied to each interface.

Answer: C,D

Explanation:

QUESTION NO: 168

Which of these is "trusted" when the auto qos voip command is configured with the trust parameter on routers?

- A. source address
- B. MAC address of sender
- C. DSCP
- D. DES keyword

Answer: C

Explanation:

QUESTION NO: 169

How many rules does a voice translation profile allow?

- A. up to 15
- B. up to 25
- C. up to 8

D. up to 12

Answer: A

Explanation:

QUESTION NO: 170

Which signaling protocol allows voice devices to operate without compromising call processing capabilities?

A. SCCP

B. SIP

C. H.323

D. MGCP

Answer: B

Explanation:

QUESTION NO: 171

Which three features are embedded in an RTP header? (Choose three.)

A. payload type

B. bandwidth type

C. DSP

D. packet size

E. sequence number

F. time stamp

Answer: A,E,F

Explanation:

QUESTION NO: 172

What is the difference between medium and high complexity codecs?

A. The difference is the number of voice channels that can be supported by a single DSP.

B. With higher codec complexity, more calls can be processed.

- C. The DSP processes codecs that only support high complexity transparently.
- D. Medium complexity codecs were used prior to Cisco 2800 and 3800 Series routers, whereas high complexity codecs are used in Cisco 2800 and 3800 Series and newer routers.

Answer: A

Explanation:

QUESTION NO: 173

Which three commands are used to configure a T1 CAS interface? (Choose three.)

- A. framing
- B. show run int s0/0
- C. linecode
- D. dial-peer voice
- E. ds0-group 1 timeslots
- F. show frame pvc

Answer: A,C,E

Explanation:

QUESTION NO: 174

Which statement is true for an ephone-dn that has no COR applied?

- A. Calls can be made from this ephone-dn to all possible numbers from the local router.
- B. Calls cannot be made from this ephone-dn.
- C. Calls can be made from this ephone-dn only to another ephone-dn on the local router.
- D. Calls can be made from this ephone-dn only to external numbers.

Answer: A

Explanation:

QUESTION NO: 175

Which dial plan component is responsible for choosing the appropriate path for a call?

- A. endpoint addressing

- B. call route and path selection
- C. call coverage and path selection
- D. calling privileges

Answer: A

Explanation:

QUESTION NO: 176

Which scenario requires QoS to be deployed on a network?

- A. consistent bandwidth
- B. an end-to-end audit
- C. slow performance
- D. an echo

Answer: B

Explanation:

QUESTION NO: 177

Which three are primary applications for classifying data on the network using QoS? (Choose three.)

- A. transactional applications
- B. any applications
- C. best-effort applications
- D. mission-critical applications
- E. World Wide Web applications
- F. database applications

Answer: A,C,D

Explanation:

QUESTION NO: 178

Which three connections can utilize the priority command? (Choose three.)

- A. ATM PVCs
- B. serial links
- C. Fast Ethernet links
- D. Frame Relay links
- E. EtherChannel
- F. Fibre Channel

Answer: A,B,D

Explanation:

QUESTION NO: 179

Switch port fa0/2 has been configured to connect an IP phone with an attached PC. Given the set of commands in the example, where does the trust boundary lie?

```
interface fa0/2  
  
mls qos trust cos  
  
mls qos trust device cisco-phone  
  
switchport voice vlan 112
```

- A. between the IP phone and the switch
- B. between the IP phone and the PC
- C. between the access layer switch and the distribution layer switch
- D. between the PC port and the LAN port on the IP phone

Answer: A

Explanation:

QUESTION NO: 180

What is the function of TEHO in a path selection strategy?

- A. Allows users to place an intersite call by dialing a site code.
- B. Uses the WAN link for call routing to avoid PSTN charges.
- C. Uses the WAN link for call routing to avoid PSTN charges; in addition, the PSTN breakout should be as close as possible to the final PSTN destination.
- D. Defines each site with a unique site code.

Answer: C

Explanation:

QUESTION NO: 181

Which two gateway protocols can be configured if you have Cisco 2600XM or 3700 Series Multiservice Routers installed? (Choose two.)

- A. MGCP**
- B. SIP**
- C. H.323**
- D. SCCP**

Answer: A,C

Explanation:

QUESTION NO: 182

What is the Cisco recommended best practice for using data VLANs and voice VLANs?

- A. Use the same VLAN ID for both data and voice.**
- B. Use one VLAN ID for data and use one VLAN ID for voice.**
- C. Use only data VLANs.**
- D. Use only voice VLANs.**

Answer: B

Explanation:

QUESTION NO: 183

What should you use to configure endpoint addressing if you are using Cisco Unified Communications Manager Express with SRST?

- A. POTS**
- B. VoIP**
- C. ephone-dn**
- D. dial-peer**

Answer: C

Explanation:

QUESTION NO: 184

What do you use an FXS for?

- A. an analog telephone
- B. a PBX
- C. an IP phone
- D. an E1

Answer: A

Explanation:

QUESTION NO: 185

What is matched first in an inbound dial peer?

- A. incoming called-number
- B. answer-address
- C. destination-pattern
- D. a dial peer with a voice port

Answer: A

Explanation:

QUESTION NO: 186

What is the total bandwidth per call in a link, if you use a G.729 (high-complexity codec) with a 40-byte sample and Frame Relay with cRTP?

- A. 19,200
- B. 11,200
- C. 9,600
- D. 7,420

Answer: C

Explanation:**QUESTION NO: 187**

What is the command sequence to configure an FXS port as a signal loop-start?

- A.** router(config)# voice-port slot/port
router(config-voiceport)# signal loopstart
router(config-voiceport)# shutdown
router(config-voiceport)# no shutdown
- B.** router(config)# voice-port slot/port
router(config-voiceport)# signal loopstart
- C.** router(config)# voice-port slot/port
router(config-voiceport)# signal groundstart
router(config-voiceport)# shutdown
router(config-voiceport)# no shutdown
- D.** router(config)# voice-port slot/port
router(config-voiceport)# signal groundstart

Answer: A

Explanation:**QUESTION NO: 188**

Which dial peer is matched when you call 2550?

- A.** dial-peer voice 10 pots
destination-pattern 2...
forward-digits all
port 1/1/1
- B.** dial-peer voice 20 pots
destination-pattern 2T
forward-digits all
port 1/1/4
- C.** dial-peer voice 30 pots
destination-pattern 255
forward-digits all
port 1/1/2
- D.** dial-peer voice 40 pots
destination-pattern 255.
forward-digits all

port 1/1/3

Answer: D

Explanation:

QUESTION NO: 189

What is a Cisco Voice Gateway feature that enables class of service or calling privileges to be assigned?

- A. COR
- B. dial peer
- C. partition
- D. calling search space

Answer: A

Explanation:

QUESTION NO: 190

What are two characteristics of a Cisco Unified Border Element? (Choose two.)

- A. It facilitates easy and cost-effective connectivity between independent VoIP service provider networks.
- B. It is referred to in the industry as a "session border controller."
- C. It can be an FXS port.
- D. It can be a IP phone in the network.

Answer: A,B

Explanation:

QUESTION NO: 191

What equipment should you use if you want to route calls between a branch in a private network and a branch in a public network, hiding the networks between them?

- A. Cisco Unified Border Element
- B. MGCP gateway

- C. H.323 gateway
- D. Cisco Unified Communications Manager

Answer: A

Explanation:

QUESTION NO: 192

Which two methods are used by a Cisco gateway to transmit fax traffic across the IP network? (Choose two.)

- A. fax pass-through
- B. T.30 signaling
- C. modem pass-through
- D. modem relay
- E. Cisco fax relay

Answer: A,E

Explanation:

QUESTION NO: 193

When troubleshooting poor VoIP call quality, which two tools should you use to measure the overall metrics? (Choose two.)

- A. mean opinion score
- B. quality of service
- C. quality of experience
- D. R-Factor
- E. signal-to-noise ratio

Answer: A,D

Explanation:

QUESTION NO: 194

Which two are Cisco IP phones that are Bluetooth-enabled devices? (Choose two.)

- A. Cisco Unified IP Phone 9971
- B. Cisco IP Communicator
- C. Cisco Unified IP Phone 7925G
- D. Cisco Unified IP Phone 8961 Series
- E. Cisco Unified IP Phone 6900 Series

Answer: A,C

Explanation:

QUESTION NO: 195

Which command should you use to activate the analog voice port in voice-port configuration mode?

- A. voice-port slot/ port
- B. no shutdown
- C. cptone locale
- D. dial-type {dtmf | pulse}

Answer: B

Explanation:

QUESTION NO: 196

Which statement about a dial peer is true?

- A. A dial peer is similar to static routes; it defines where calls originate and terminate and which path the calls take through the network.
- B. A dial peer describes the number and pattern of digits that a user dials to reach a particular phone number.
- C. A dial peer manipulates digits in a dial string before it passes the dial string to the telephone device.
- D. A dial peer is used to control the number of digits that are stripped before the dialed string is passed to the telephony interface.

Answer: A

Explanation:

QUESTION NO: 197

What are the two major types of VoIP gateways?

- A. E and M
- B. FXO and FXS
- C. CAS and CCS
- D. analog and digital

Answer: D

Explanation:

QUESTION NO: 198

Which two digit manipulation commands should you use in a POTS dial peer? (Choose two.)

- A. num-exp
- B. [no] digit-strip
- C. translation-profile
- D. voice translation-profile
- E. forward-digits

Answer: B,E

Explanation:

QUESTION NO: 199

Using Cisco best practices, on which application should you implement calling privileges?

- A. quality of service
- B. type of service
- C. route pattern
- D. class of restriction

Answer: D

Explanation:

QUESTION NO: 200

During the boot process, which server does a Cisco IP phone receive its IP address information from?

- A. TFTP server
- B. NTP server
- C. Cisco Unified Communications Manager Express router
- D. DHCP server

Answer: D

Explanation:

QUESTION NO: 201

Dial peers and gateways are important in the call routing process. The call routing process is done based on the inbound dial-peer and outbound dial-peer matching process. Which two statements are true? (Choose two.)

- A. Dial peer 0 for inbound POTS peers has a "no ivr application" configuration.
- B. When a Cisco router or gateway receives a call setup request, a digit-by-digit match is used to match against configured dial peers.
- C. Cisco platforms such as the Cisco AS53xx and AS5800 Series require that a configured inbound dial peer is matched for incoming POTS calls to be accepted as voice calls.
- D. In order to modify the default interdigit timeout set for 10 seconds, issue this command.
interdigit timeouts # seconds voice-port.

Answer: A,C

Explanation:

QUESTION NO: 202

For multisite deployments with distributed call processing, which three dial plan solutions does Cisco recommend? (Choose three.)

- A. using access and site codes
- B. implementing PSTN access
- C. defining remote site redundancy options
- D. defining Call Admission Control for Cisco Extension Mobility
- E. implementing PSTN backup
- F. implementing Service Advertisement Framework

Answer: A,B,E

Explanation:

QUESTION NO: 203

You are configuring a Cisco Unified Border Element. What do you need to do to achieve address hiding?

- A.** Under the voice service configuration mode, issue the media flow-around command.
- B.** Under the dial-peer configuration mode, issue the codec flow-through command.
- C.** Activate the voice service voip command at the dial-peer configuration mode.
- D.** Under the dial-peer configuration mode, issue the media flow-through command.

Answer: D

Explanation:

QUESTION NO: 204

Which two options are Cisco recommendations for enabling Cisco Unified Border Element to perform a CAC? (Choose two.)

- A.** a voice service VoIP that is configured on the unique Cisco Unified Border Element to enable the CAC service
- B.** two Cisco Unified Border Element gateways to enable RSVP-based CAC, with the media flow-through command configured
- C.** Cisco Unified Border Element gateways activating RSVP-based CAC, using the media flow-around command
- D.** two Cisco Unified Communications Manager clusters interconnected using a Cisco Unified Border Element
- E.** an interconnection of a Cisco Unified Border Element and a Cisco Unified Communications Manager to support both voice and video calls

Answer: B,D

Explanation:

QUESTION NO: 205

The two main commands that are required for low latency are priority and bandwidth. What is the

function of the bandwidth command?

- A. Identifies a class as strict priority bandwidth.
- B. Applies to a priority queue.
- C. Allocates bandwidth to nonpriority classes.
- D. Identifies a class as a strict priority and allocates a bandwidth.

Answer: C

Explanation:

QUESTION NO: 206

Which option is used within a dial plan to define the destination that a user is allowed to call?

- A. dial peers
- B. calling peers
- C. calling privileges
- D. destination patterns

Answer: C

Explanation:

QUESTION NO: 207

Which is a function of RTP?

- A. call multiplexing
- B. encryption
- C. payload identification
- D. replay protection

Answer: C