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Vendor: Cisco

Exam Code:642-437

Exam Name: Implementing Cisco Unified Communications Voice over IP and QoS v8.0 (CVOICE v8.0)

Version: Demo

QUESTION 1

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.

Correct Answer: D

QUESTION 2

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.

Correct Answer: B

QUESTION 3

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

Correct Answer: ABEG

QUESTION 4

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

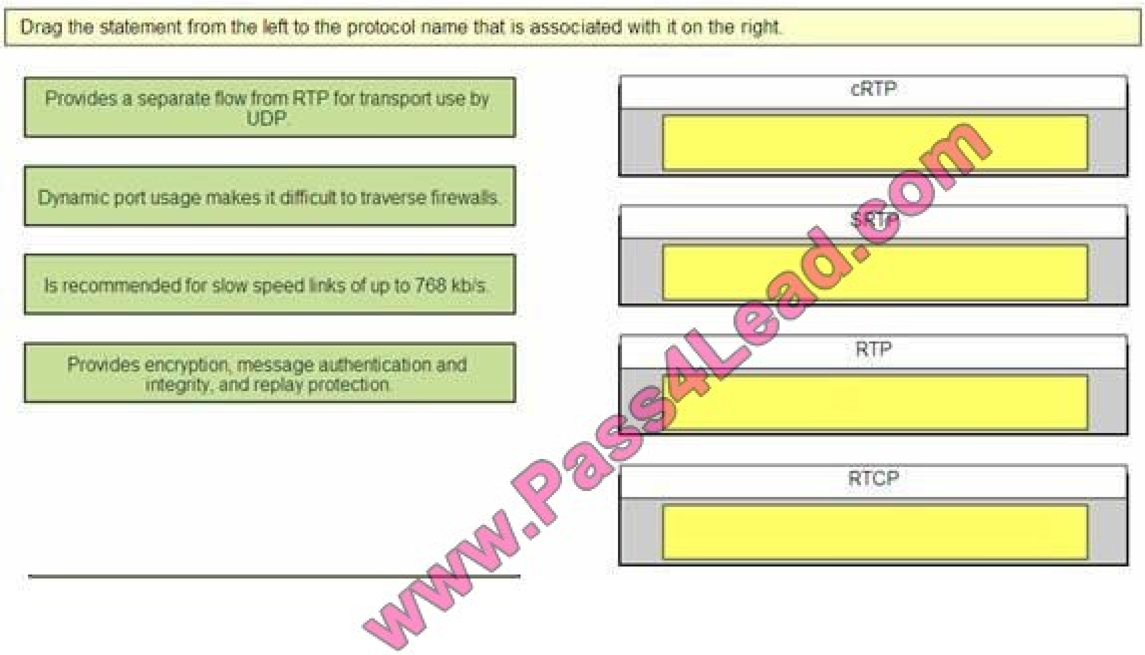
D. VoIP

Correct Answer: A

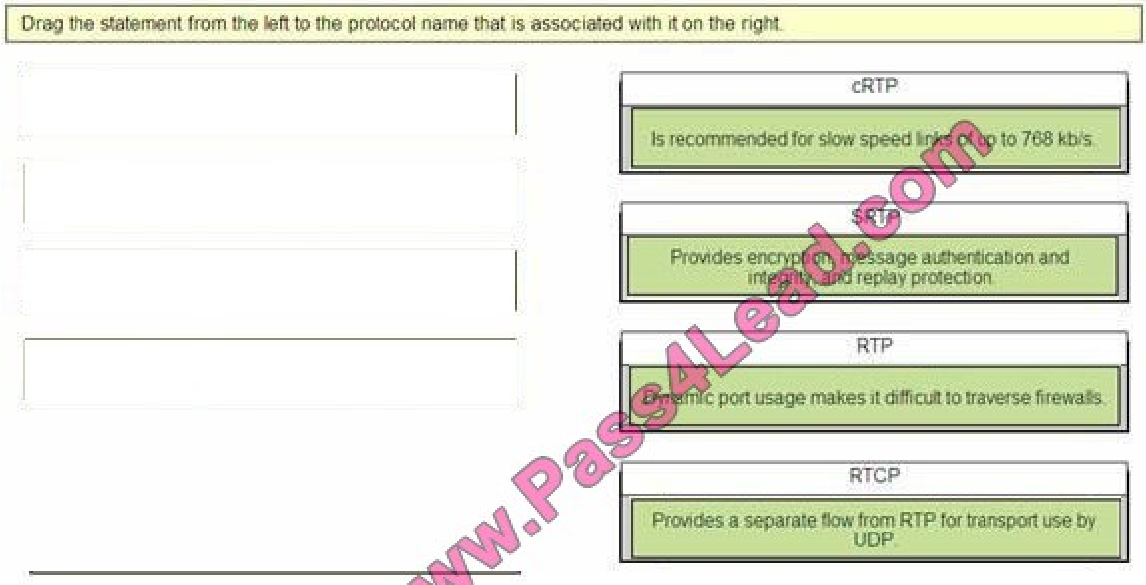
QUESTION 5

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

Select and Place:



Correct Answer:



Explanation/Reference:

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 lowbandwidth 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 6

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

Correct Answer: A

QUESTION 7

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

Correct Answer: A

QUESTION 8

Which signaling protocol is proprietary to Cisco?

A. H.323

B. SIP

C. MGCP

D. SCCP

Correct Answer: D

QUESTION 9

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.

Correct Answer: ACF

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.

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

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QUESTION 10

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.

Correct Answer: AC

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/administratio n/guide/sccp/sccpaaph.pdf