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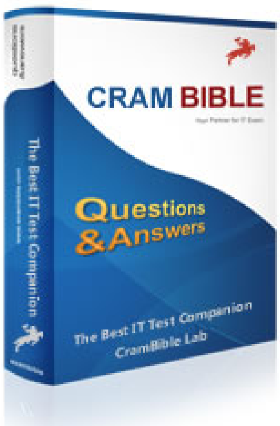

350-030

CCIE Voice Written

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THE TOTAL NUMBER OF QUESTIONS IS 267

QUESTION NO: 1 On which gateway or gatekeeper is the IOS command **call-rsvp-sync resv-timer 10** used to set the timer?

A. originating VoIP gateway for completing RSVP reservation setups in 10 seconds

B. originating and terminating VoIP gateway for completing RSVP reservation setups in 10 seconds

C. terminating VoIP gateway for completing RSVP reservation setups in 10 seconds

D. VoIP gatekeeper for completing RSVP reservation setups in 10 seconds

Answer: C

QUESTION NO: 2

Calls to an ICD queue should reserve an available agent and connect the call after a database lookup is performed. How should the script be configured to accomplish this?

A. Set the Resource Step Connect option to No and perform a Connect after the database lookup is completed.

B. Issue a Call Hold after the Resource Step selects an agent and release the hold after the database lookup is completed.

C. Issue a Queue Step followed by the database lookup and a Resource Step.

D. Issue a Queue Step followed by the database lookup and a Dequeue Step.

Answer: A

QUESTION NO: 3

What occurs if the system clocks are not synchronized between the sender and receiver of an RTP stream?

A. Packets can be placed in sequence but jitter cannot be compensated for.

B. Packets cannot be reordered, because sequence and jitter cannot be compensated for.

C. Jitter can be compensated for, but packets cannot be reordered if they arrive out of sequence.

D. Packets may be reordered and jitter may be compensated for, because the timestamp is not related to the system time.

E. When the RTP stream is opened, the sender and receiver synchronize their clocks before the stream commences so that packet sequencing and dejitter will function correctly.

Answer: D

QUESTION NO: 4

If all n MTP transcoding sessions are utilized, and an n + 1 connection is attempted, how will the next call be treated?

A. it will not use an MTP and will use the transcoding resources associated with the codec to complete the ca

B. it will be redirected to the PSTN due to a lack of MTP resources

C. it will use the alternate codec type and attempt to complete the call

D. it will complete the call without using the MTP transcoding resource

Answer: D

QUESTION NO: 5

A centralized call processing topology comprises a headquarters and a branch office. Calls within both the headquarters and the branch office utilize the G.711 codec. Calls between the headquarters and the branch office utilize the G.729 codec. Multicast MoH must always be transmitted using the G.711 codec.

Which of the following configurations would meet this requirement?

A. A total of two regions are required for all sites with G.729 codec specified between the two regions; G.711 codec is used within regions. The MoH server should belong to the headquarters region.

B. A total of two regions are required for all sites with G.729 codec between the two regions; G.711 codec is used within regions. The Cisco IP Voice Media Streaming App needs to be configured for G.711 codec only.

C. A total of two regions are required for all sites with G.729 codec between the two regions; G.711 codec is used within regions. The MoH server is placed in a separate location with the G.711 codec utilized between itself and each office.

D. The MoH server cannot be configured to transmit G.711 because the phones are negotiating G.729 codec during call setup.

E. Three separate regions are required: one for the headquarters, one for the branch office, and one for the

MoH server. Codecs between and within each region are specified accordingly.

Answer: E

QUESTION NO: 6

On a WAN PPP link, what is the required bandwidth for three G.729 VoIP calls when cRTP is turned off, and what is it when cRTP is turned on? (Note The payload size is 20 bytes.)

A. cRTP off: 72.6 kb/s; cRTP on: 24.6 kb/s

B. cRTP off: 90 kb/s; cRTP on: 36 kb/s

C. cRTP off: 79.2 kb/s; cRTP on: 33.6 kb/s

D. cRTP off: 26.4 kb/s; cRTP on: 11.2 kb/s

E. cRTP off: 48 kb/s; cRTP on: 24 kb/s

Answer: C

QUESTION NO: 7

Acme Widgets Inc. wants to compress the voice data traveling over its WAN connection to its parent company. It is currently using G.729, loading two voice frames per packet. When Acme Widgets Inc. implements cRTP using the ip rtp header-compression command, what will be the Layer 3 bandwidth consumption per call on the WAN link?

A. 8.0 kb/s

B. 8.8 kb/s

C. 9.6 kb/s

D. 12.0 kb/s

E. 16.0 kb/s

Answer: A

QUESTION NO: 8

Approximately what percentage of overall bandwidth is saved (at Layer 3) by cRTP for a G.711 VoIP call packetized at 50 p/s?

A. 60 percent

B. 50 percent

C. 40 percent

D. 30 percent

E. 20 percent

Answer: E

QUESTION NO: 9

```
IP Phone A     1000 calls 2000           IP Phone B   1000 is forwarded      Voicemail Server
Ext 1000       ----------------------->  Ext 2000     -------------------->  Ext 3000

                              Configuration

call-manager-fallback
 voicemail 3000
 call-forward busy 3000
 call-forward noan 3000 timeout 3

vm-integration
 pattern direct 2 CGN *
 pattern ext-to-ext no-answer 5 FDN * CGN *
 pattern ext-to-ext busy 7 FDN * CGN *
 pattern trunk-to-ext no-answer 4 FDN * CGN *
 pattern trunk-to-ext busy 6 FDN * CGN *
```

Refer to the exhibit. Based on the configuration shown, what digit pattern will the voice-mail server see if there is no answer when IP phone B is called from phone A? (Note: Assume that the Cisco Unified Communications Manager servers have become unreachable, and therefore the IP phones are in SRST mode and have registered to the gateway with the configuration in the exhibit.)

A. 52000*1000*

B. 72000*1000*

C. 42000*1000*

D. 21000*

E. 62000*1000*

Answer: A