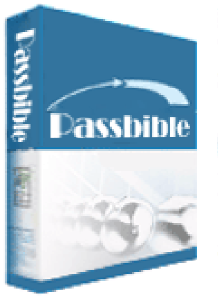
Cisco 642-432 Exam

642-432: Cisco Voice Over IP
Practice Exam: 642-432 Exams
Exam Number/Code: 642-432
Exam Name: Cisco Voice Over IP
Questions and Answers: 97 Q&As
(IP Communications)



Exam : 642-432

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Title : Cisco(r) Voice Over IP

1. You are working with your customer in their lab to test the effect of jitter on voice quality. You have set the maximum playout delay to 40 ms on the voice enabled routers. What will be the impact on voice quality if after severe congestion the playout buffer empties and the source sends packets to the buffer faster than they are leaving?
A. There will be no noticeable drop in quality.
B. The jitter buffer will adapt to the faster-arriving packets by expanding the buffer size.
C. The jitter buffer will speed up delivery of packets to the DSP so that packets are not dropped.
D. After the jitter buffer fills up, subsequent packets are discarded.
Answer: D

2. In a VoIP environment when speech samples are framed every 20 ms, a payload of 20 bytes is generated. What is the size of the combined IP, UDP, and RTP headers?

A. 40 bytes

B. 12 bytes

C. 20 bytes

D. 8 bytes

E. 4 bytes

F. 2 bytes

Answer: A

3. Examine the output.

Your customer has sent you their MGCP gateway configuration. They are unable to get the gateway to communicate with the call agent. What command needs to be inserted to resolve the problem?

A. ccm-manager mgcp 172.16.1.1

B. mgcp call-agent 172.16.1.1

C. application MGCPAPP 172.16.1.1

D. mgcp 5036 172.16.1.1

Answer: B

4. A customer would like to be able to retire their existing permanent PSTN trunk connection between their PBXs and migrate to permanent connections over the IP WAN. There is one configuration statement missing from each router in the figure. What are the two missing statements? (Choose two.)

A. connection tie-line e&m

B. connection trunk 4045551200

C. connection tie-line 2015551000

D. connection trunk 2015551000

E. connection tie-line 4045551200

F. connection tie-line

Answer: BD

5. You have been forwarded some questions by a prospective VoIP customer who would like to know the Cisco default sample size for the G.729 codec. What is it?

A. 40 ms

B. 30 ms

C. 20 ms

D. 10 ms

Answer: C

6. Examine the example output.

Choose the command that will restore communication with gatekeeper functionality to this device.

A. h323-gateway voip h323-id GK1

B. gateway

C. h323-gateway voip bind srcaddr 172.16.2.2

D. h323-gateway voip GW1-zone2.abc.com abc.com ipaddr 172.16.2.1

Answer: B

7. In T1 CAS, where are the signaling states and control features carried for Super Frame robbed-bit signaling?

A. 6th and 12th frame

B. 6th, 12th, 18th, and 24th frame

C. the first and seventeenth time slot

D. the first and sixteenth time slot

Answer: A

8. Your customer has forwarded this diagram and configuration. The customer wishes to have a connection between its PBXs, a connection that is created and dropped as required. There is one configuration statement missing from each router. What are the two missing statements? (Choose two.)

A. connection trunk 2015551000404555….

B. connection trunk 4045551200

C. connection tie-line 4045551200

D. connection tie-line 404555….

E. connection tie-line 2015551000

F. connection trunk

Answer: CE

9. You are working with a potential customer that would like to integrate its existing PBX telephone system into its IP network. The accompanying figure shows that the customer has two offices that need to be connected to the IP network so that the customer can exchange telephone calls without using the PSTN. Both PBXs use a proprietary signaling type.

Which signaling type will allow you to support your customer?

A. E&M

B. CCS

C. CAS

D. T-CCS

E. FXO

F. FXS

Answer: D

10. You are meeting with a customer that has deployed IP telephony at their headquarters location. They would like to roll out IP telephony to their regional office as well. They are now using the G.711 codec at headquarters. They want to be able to maximize the number of calls carried without impacting voice quality or forcing a WAN upgrade. Which codec would be appropriate for their WAN?

A. G.726

B. G.723.1

C. G.711

D. G.729B

Answer: D

11. Examine the following PBX system parameters:

The calling side seizes the line by going off-hook on its E-lead and sends information as DTMF digits.

The voice path is 4-wires, and the voice enabled router is in another building from the PBX.

Select the correct set of commands to allow communication between a voice enabled router and a PBX.

A. voice port 1/0/0

signal immediate-start

operation 4-wire

type 2

B. voice-port 1/0/0

signal delay-dial

operation 4-wire

type 1

C. voice port 1/0/0

signal wink-start

operation 4-wire

type 3

D. voice port 1/0/0

signal immediate-start

operation 4-wire
type 4
Answer: A

12. You have a customer that is interested in determining the number of VoIP calls their Frame Relay WAN links can support. Each of their Frame Relay WAN links has 84 kbps of bandwidth available outside all other applications and overhead. How many G.729 calls using the 8 kbps codec and 20 byte sample size can be supported?
A. 1
B. 2
C. 3
D. 4
Answer: C

13. You are working with a potential customer that would like to integrate its existing PBX telephone system into its IP network. The accompanying figure shows that the customer has two offices that need to be connected to the IP network so that the customer can exchange telephone calls without using the PSTN. Both PBXs use an in-band signaling type.
Which signaling type will allow you to support your customer?
A. QSIG
B. CCS
C. CAS
D. T-CCS
E. E&M
F. FXO
Answer: C

14. You are working with a potential customer that would like to integrate its existing PBX telephone system into its IP network. The accompanying figure shows that the customer has two offices that need to be connected to the IP network so that the customer can exchange telephone calls without using the PSTN. Both PBXs are currently connected to T1 ISDN circuits.
Which signaling type will allow you to support your customer?
A. QSIG
B. CCS
C. CAS
D. T-CCS
E. E&M
F. FXO
Answer: A

15. In a VoIP environment when speech samples are framed every 20 ms, a payload of 20 bytes is generated. Assuming a total packet length of 60 bytes, what is the length of the packet header if cRTP is deployed without redundancy checks?
A. 1 byte
B. 2 bytes
C. 3 bytes
D. 4 bytes
E. 20 bytes
F. 40 bytes
Answer: B

16. Which command parameter specifies that the router should not attempt to initiate a trunk connection but should wait for an incoming call before establishing the trunk?
A. codec clear-channel

B. connection-trunk 404555.... answer-mode

C. voice-port 1/0:1

D. ds0-group timeslots 1-23 type ext-sig

Answer: B