QUESTION 1
When using an Inline-Power enabled Catalyst Switch, which pins are used to supply Inline-Power to an IP Phone?

A. Pins 1, 2, 3, and 6
B. Pins 1, 2, 5, and 6
C. Pins 2, 4, 6, and 8
D. Pins 3, 4, 7, and 8
E. Pins 4, 5, 7, and 8

Answer: A
CISCO CATALYST 4000 SERIES SWITCHES, Cisco Catalyst 4000 Series Inline Power Solution
http://www.cisco.com/en/US/products/hw/switches/ps663/products_data_sheet09186a00800924d0.html

QUESTION 2
When using a Cisco Inline-Power Patch-Panel, which pins are used to supply Inline-Power to the IP Phone?

A. Pins 1, 2, 3, and 6
B. Pins 1, 2, 5, and 6
C. Pins 2, 4, 6, and 8
D. Pins 3, 4, 7, and 8
E. Pins 4, 5, 7, and 8

Answer: E
CISCO NETWORK MODULES, Catalyst Inline Power Patch Panel
http://www.cisco.com/en/US/products/hw/modules/ps2797/products_data_sheet09186a00800a9ea3.html

QUESTION 3
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what protocol IP Phone uses to learn the Voice VLAN ID it should use for Voice traffic.
What will your reply be?

A. Skinny Station Protocol
B. 802.1q
C. LLQ
D. VTP
E. CDP

Answer: E

QUESTION 4
You are the network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what MailStore options are supported in Unity version 2.4.6.
What will your reply be?

A. MS Mail
B. Domino
C. Exchange 2000
D. Exchange 5.5
E. None of the above

Answer: D
http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_pre_installation_guide09186a00800ea54b.html
#73

QUESTION 5
You are a network administrator at Certkiller . Certkiller has a Unity3.0/ and Exchange 2000 system.
Which of the following attributes will be stored in Active Directory? (Choose three.)

A. Transfer Type (Supervised, Release to switch)
B. Location ID
C. Alternate Extensions
D. Recorded Name
E. All of the above

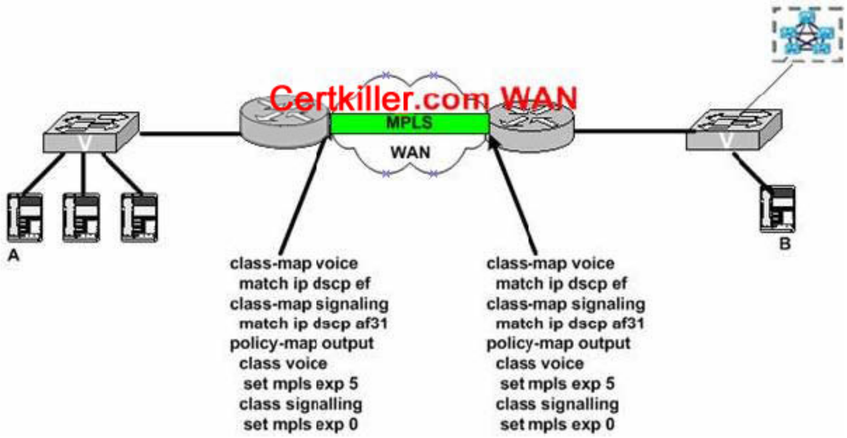
Answer: B, C, D

QUESTION 6
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what Network Management Server (NMS) application she can use to monitor Voice quality by polling the SNMP MIB for MQC.
What will your reply be?

A. Voice Health Monitor
B. Quality of Service Policy Manager
C. Internetwork Performance Monitor
D. Resource Manager Essentials
E. None of the above

Answer: B

QUESTION 7
You are the network administrator at Certkiller . The Certkiller network and the complete MPLS router QoS configuration is shown in the following exhibit:



The LAN switches (and any other equipment in the cloud) do not mark or remark the packets.
With regard to the QoS configuration in the exhibit, when IP Phone A calls IP Phone B, how will the voice and signalling packets be marked by the time they arrive at IP Phone B?

Answer: D
The DSCP field remains unchanged in the entire process

QUESTION 8
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what protocol an IP Phone uses to learn the IP Address of its TFTP Server.
What will your reply be?

A. CDP
B. OSPF
C. HSRP
D. EIGRP
E. DHCP

Answer: E

QUESTION 9
With regard to jitter, which of the following statements are true?

A. Jitter is the variation from the time that a packet is expected to be received and when it is actually received.
Voice devices have to compensate for jitter by setting up a playout buffer to accept voice in a smooth fashion and avoid discontinuity in the voice stream.
B. Jitter is the actual delay from the time that a packet is expected to be transmitted and when it actually is transmitted.
Voice devices have to compensate for jitter by setting up a playin buffer to play back voice in a smooth fashion and avoid discontinuity in the voice stream.
C. Jitter is the actual delay from the time that a packet is expected to be transmitted and when it actually is transmitted.
Voice devices have to compensate for jitter by setting up a playout buffer to play back voice in a smooth fashion and avoid discontinuity in the voice stream.
D. Jitter is the variation from the time that a packet is expected to be received and when it is actually received.

Voice devices have to compensate for jitter by setting up a playin buffer to accept voice in a smooth fashion and avoid discontinuity in the voice stream.

Answer: A

QUESTION 10
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what the differences between Type of Service (ToS) and Class of Service (CoS) are.
What will your reply be? (Choose two.)

A. CoS allows a class based access to the media, but ToS prioritizes this access according to the precedence bit.
B. CoS is a field in the IP header, but ToS is evaluated by the routing protocol.
C. CoS is a Layer 2 mechanism, but ToS is a Layer 3 mechanism.

Answer: A, C

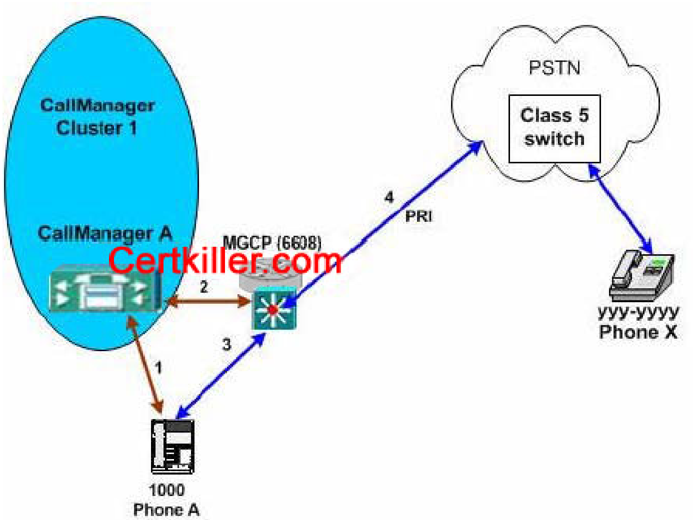
QUESTION 11
You are a network engineer at Certkiller . The Certkiller network is shown in the following exhibit:

You have just configured the Catalyst 6000 with the following commands:
set qos enable
set port qos 5/1-48 vlan-based
set port qos 5/1-48 trust-ext untrusted
set port qos 5/1-48 trust trust-cos
We assume that the IP Phone is connected to port 5/1, which of the following statements are true? (Choose three.)

A. The IP Phone will re-write the CoS of 802.1p/Q-tagged packets form the PC to CoS=0.
B. The IP phone will not modify the DSCP of packets from the PC.
C. The Catalyst 6000 will not modify the CoS of any packets received on port 5/1.
D. The Catalyst 6000 switch port 5/1 will re-write the CoS of all packets received on VLAN 110 with CoS=5.
E. The Catalyst 6000 switch port 5/1 will re-write all packets with a Cos=0.

Answer: A, B, C
set port qos trust
http://www.cisco.com/univercd/cc/td/doc/product/lan/cat6000/sw_7_2/cmd_ref/set_po_r.htm#39814

QUESTION 12
You are the network administrator at Certkiller . The Certkiller network is shown in the following exhibit:



The default gateway in the Certkiller network is a 6608 blade that is running MGCP. Call Manager runs version 3.1. Certkiller users will make calls from Phone A to Phone X. All IP streaming is G.711. Each of the logical links in the Certkiller network carries different types of traffic.
On which link can q.921 traffic be sent?

A. 1 only
B. 2 and 3
C. 2 and 4
D. 3 and 4
E. 4 only

Answer: E

QUESTION 13
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what the term "MGCP backhaul" means.
What will your reply be?

A. Transporting T1 CAS messaging to the MGCP Call Agent across IP.
B. Transporting ISDN Q.931 messaging to the MGCP Call Agent across IP.
C. Transporting ISDN Q.921 messaging to the MGCP Call Agent across IP.
D. Transporting ISDN Q.931 messaging into MGCP events to the MGCP Call Agent.
E. Encapsulating ISDN Q.931 CDR records to a RADIUD server.

Answer: B
CISCO IOS SOFTWARE RELEASES 12.2 SPECIAL AND EARLY DEPLOYMENTS
MGCP-Controlled Backhaul of BRI Signaling in Conjunction with Cisco CallManager
http://www.cisco.com/en/US/products/sw/iosswrel/ps5012/products_feature_guide09186a00801a8bc4.html

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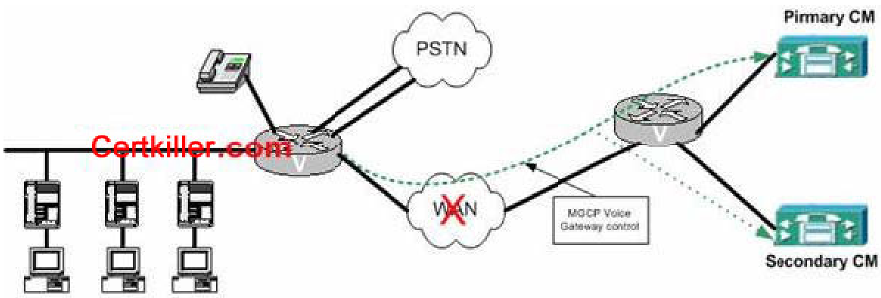
QUESTION 14
You are the network administrator at Certkiller . Certkiller has a CM network deployed with MGCP to the branch office GWs. The Certkiller network is shown in the following exhibit:

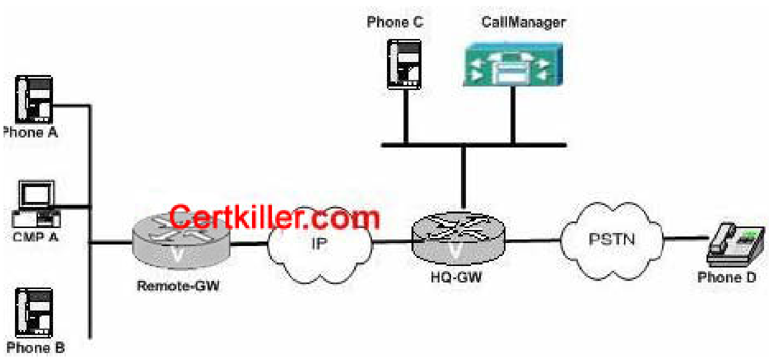


You want to protect the protect branch office telephony (IP phone to IP Phone, and IP Phone to PSTN) in the event of a WAN failure.
Which design methods could you implement? (Choose two.)

A. SRST
B. MGCP gateway fallback
C. CM clustering
D. Primary and Secondary CMs
E. CAC

Answer: A, B

QUESTION 15
You are the network administrator at Certkiller . The Certkiller network is shown in the following exhibit:



A Certkiller user at PhoneA has finished a call. He later notice that "CM Fallback Service Operating" is displayed on Phone A.
What is the possible cause of this? (Choose two.)

A. The TCP connection between CallManager and PhoneA has been disrupted.
B. The FE on HQ-GW is out of service.
C. The FXO port on Remote-GW is out of service.
D. The FE and Remote-GW is out of service.
E. Remote-GW has not received any messages from CallManager within the timeout period.

Answer: A, B
VOICE SOLUTIONS FOR BRANCH/SMALL OFFICE, Highly Available IP Communications White Paper
http://www.cisco.com/en/US/netsol/ns340/ns394/ns346/ns383/net_value_proposition09186a00801c6097.html

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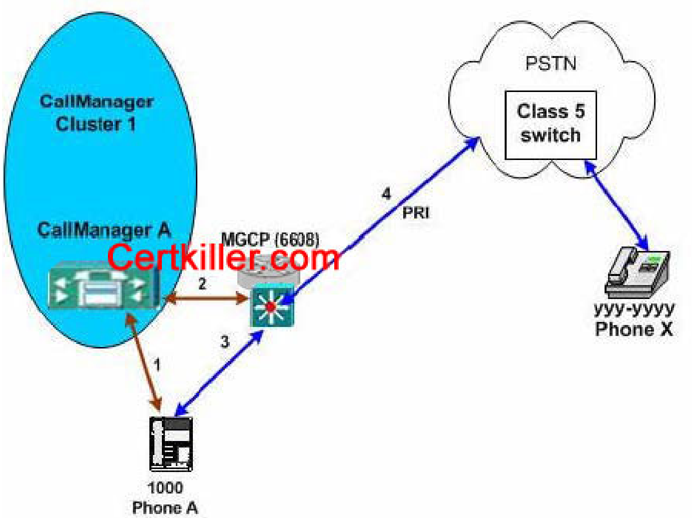
QUESTION 16
You are the network administrator at Certkiller . The Certkiller Voice network is shown in the following exhibit:



The default gateway in the Certkiller network is a 6608 blade that is running MGCP. Call Manager runs version 3.1. Certkiller users will make calls from Phone A to Phone X. All IP streaming is G.711. Each of the logical links in the Certkiller network carries different types of traffic.
On which link can Skinny (SCCP) traffic be seen?

A. 1 only
B. 1 and 2
C. 1, 3, and 4
D. 2, and 3
E. 2, 3, and 4

Answer: A
Skinny is only used between CCM and IP phone

QUESTION 17
On which of the following does the maximum device weight capacity a Cisco MCS server depend? (Choose three.)

A. The quantity and the type of phones configured on the Cisco MCS server.
B. CCM software release version.
C. The amount of memory, CPU and I/O throughput.
D. The server model and type.

Answer: B, C, D

QUESTION 18
You are the network administrator at Certkiller . The Certkiller WAN is shown in the following exhibit:

The remote Certkiller user at Analog Phone A tries to make a call to Analog Phone D. The call is rejected. You troubleshoot the problem and discover that the ISDN Plan and Type that the PSTN was receiving was not setup correctly. The PSTN was expecting the following:
Plan: Unknown
Type: Unknown
While the PSTN was receiving the following:
Plan:ISDN
Type: Unknown
Which methods will resolve this problem?

A. Use the isdn map command on the POST dialpeer on Remote-GW.
B. Use the isdn map command on the VOIP dialpeer on HQ-GW.
C. Use the isdn map command on the POTS dialpeer on HQ-GW.
D. Implement an outgoing translation rule on the VOIP dialpeer on HQ-GW.
E. Use the isdn map command on the VOIP dialpeer on Remote-GW.

Answer: E

Explanation:
The isdn map command is used to override the default ISDN type and plan generated by the router which has custom values. To undo the changes you made with the isdn map command use the following command:
isdn map{ address address |regexp |plan plan |typetype}
no isdn map{ address address |regexp |plan plan |typetype}
Syntax Description

addressaddress	Address map, which can be to the calling, called,
	or redirecting number.
regexp	Regular expression for pattern matching.
planplan	ISDN numbering plan.
typetype	ISDN number type.

QUESTION 19
Migrating form TDM voice requirements to VoIP usually does not cause migration issues for customers who expect to be:

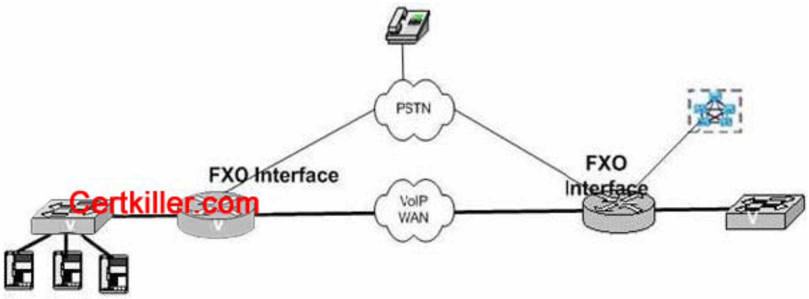
A. Deploying in a Green-Field scenario.
B. Fully IP in 1 to 3 years.
C. Fully IP within 12 months.

D. All of the above.

Answer: A

QUESTION 20

You are the network administrator at Certkiller . The Certkiller network is shown in the following exhibit:



You want to implement Caller ID for calls from PSTN to the IP Phones. Analog trunks are equipped to the PSTN from both the GWs. The PSTN CO switch is capable of delivering Caller ID on the connection to the Cisco voice GW. The Cisco voice gateway is using a VIC-2FXO-M1 card.
What would be the correct voice GW design?

A. None. Caller ID is not supported on analog FXO.
B. GW connected via SCCP to the CallManager.
C. GW must be a 2600/3600/3700 series platform.
D. GW connected via MGCP to the CallManager.
E. GW connected via H.323 to the CallManager.

Answer: E

QUESTION 21

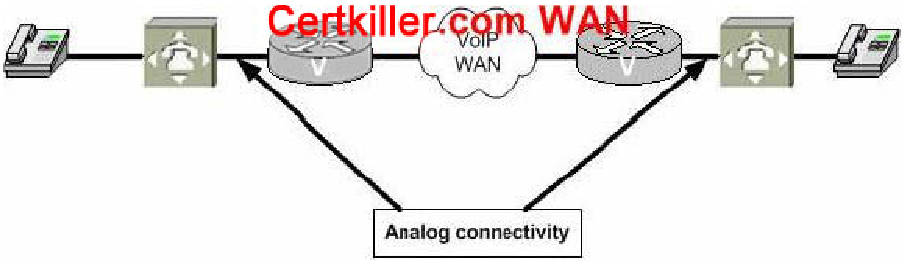
You are a network administrator at Certkiller . Certkiller has recently joined a ISP that offers debit card calling from theUS to other locations around the world where they only have FXO connectivity to the PSTN. The Certkiller network is shown in the following exhibit:

It is crucial to the ISP to ensure that their consumers are not charged for the call unless the call successfully reached the called party in the PSTN.
Which of the following design requirements will achieve the ISP's desired results?

A. Use Voice Activity Detection (VAD) to determine whether or not a person answered the call.
B. Use CDR records to determine which calls resulted in "ring no answer" and which calls were answered.
C. Configure call progress tone detection on the FXO interface to indicate disconnect supervision.
D. Avoid use of FXO for this application.
E. Configure call progress tone detection on the FXO interface to indicate answer supervision.

Answer: E

QUESTION 22
You are the network administrator at Certkiller . The Certkiller network is shown in the following exhibit:



You want Caller ID be displayed for all phones connected to the PBXs, as well as for calls going in both directions across the IP network. The PBXs only have analog (FXS, FXO and E&M) capabilities to connect to the Cisco voice GWs.
What design will satisfy this requirement?

A. All of FXS, FXO and E&M, provided that the FXO cards support Caller ID.
B. 2-wire and 4-wire E&M.
C. 4-wire E&M and FXO.
D. FXS and E&M only.
E. None of the above.

Answer: A

QUESTION 23
Your newly appointed Certkiller trainee wants to know what type of signaling can provide automatic Number Identifications (ANI) on a T1/E1 line.
What will your reply be?

A. E&M-fgd
B. Loop Start
C. PRI
D. E&M-fgb

Answer: A
TELEPHONY SIGNALING, Understanding How Digital T1 CAS (Robbed Bit Signaling) Works in IOS Gateways
http://www.cisco.com/en/US/tech/ CK6 52/ CK6 53/technologies_tech_note09186a00800e2560.shtml

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QUESTION 24
You are the network administrator at Certkiller . The Certkiller network is shown in the following exhibit:

The PBX does not receive the initial few digits from the IP side of the 2611 network.
What should you do?

A. Configure prefix delay, in the dial-peer POTS to add the delay.

B. Configure interdigit timing 1 under the voice-port.
C. Configure prefix, in the dial-peer POTS to forward the necessary digits.
D. Configure delay-dial under the voice-port to add the delay.

Answer: D
TELEPHONY SIGNALING, Understanding and Troubleshooting Analog E&M Start Dial Supervision Signaling
http://www.cisco.com/en/US/tech/ CK6 52/ CK6 53/technologies_tech_note09186a0080093f61.shtml

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QUESTION 25
You are contracted as a network administrator for a small company, Certkiller Inc. Your newly appointed Certkiller trainee wants to know
What settings have to be in place in order to pass hook-flash on h.323 from FXS to FXO:
What will your reply be?

A. connection trunk must be configured on the voice-port (FXS) and (FXO)
B. connection plar must be configured on the voice-port (FXS)
C. connection trunk must be configured on the voice-port (FXS) and (FXO)
D. connection plar must be configured on the voice-port (FXS) and (FXO)

Answer: D
IP TELEPHONY / VOIP, Configuring Hookflash Relay on FXS/FXO Voice Ports The Following link will provide you with information on IP TELEPHONY / VOIP, Configuring hookflash Relay onFXS/FXO Voice Ports
http://www.cisco.com/en/US/tech/ CK6 52/ CK7
01/technologies_configuration_example09186a008009431b.shtml

QUESTION 26
Which of the following is the best configuration for provisioning for VoIP at the WAN Edge?

A. !
version 12.2
!
class-map match-all VOICE
match ip rtp 16384 32767
class-map match-all VOICE-CONTROL
match protocol skinny
!
policy-map WAN-EDGE
class VOICE
low-latency queuing 33 percent
class class-default
weighted-fair-queue
!
B. !
version 12.2

```
!
class-map match-all VOICE
match ip rtp 16384 32767
class-map match-all VOICE-CONTROL
match protocol skinny
!
policy-map WAN-EDGE
class VOICE
low-latency queuing 33 percent
class class-default
weighted-fair-queue
!
```

C. !

```
version 12.2
!
class-map match-all VOICE
match ip dscp 45
class-map match-all VOICE-CONTROL
match ip dcsp 26
!
policy-map WAN-EDGE
class VOICE
priority queue 33 percent
class VOICE-CONTROL
bandwidth queue 2 percent
class class-default
fair-queue
!
```

D. !

```
version 12.2
!
class-map match-all VOICE
match ip dscp 5
class-map match-all VOICE-CONTROL
match ip dscp 3
!
policy-map WAN-EDGE
class VOICE
priority percent 33
class VOICE-CONTROL
bandwidth percent 2
class class-default
fair-queue
!
```

Answer: A

QUESTION 27
You are a network administrator at Certkiller . Certkiller has a Cisco voice gateway that will be connected to a PBX/PSTN-switch via ISDN (PRI, QSIG, BRI).
Which attributes of the PBX/PSTN-switch must you know to understand which features to configure on the voice GW in order to connect successfully to the PBX/PSTN-switch? (Choose two.)

A. Whether Q.921 or Q.931 is supported by the PBX/PSTN-switch.
B. Whether H.225 or H.323 is supported by the PBX/PSTN-switch.
C. Whether symmetric mode is supported by the PBX/PSTN-switch.
D. What PRI/BRI switch-type is supported by the PBX/PSTN-switch.
E. Whether the network or users side is supported by the PBX/PSTN-switch.

Answer: D, E

QUESTION 28
Which of the following statements is an attribute of ISDN Non-Facility Associated Signaling (NFAS)?

A. Enables the D-channel to transmit "data" information unrelated to any voice call, such as inter-switch status updates.
B. Applicable to voice calls and PRI only, but not data PRI calls.
C. Is available on both T1 and E1 PRIs.
D. Single D-channel controls B-channels on the same T1 span, as well on other T1-spans.
E. Single T1 span can be split into two "trunk groups", each with its own dedicated D-channel.

Answer: D

QUESTION 29
You are a network administrator at Certkiller , Inc. Certkiller runs a centralized Automatic Message Accounting (CAMA) trunk that allows enterprise voice GW connectivity to the North American (911) services of the PSTN.
A newly appointed Certkiller trainee wants to know what the difference between CAMA trunk signalling and FXO trunk signalling are.
What will your reply be?

A. They do not differ in basic signaling, but CAMA is used exclusively for 911 calls, while FXO is used for general PSTN calls.
B. FXO allows for dialed digit delivery, while CAMA does not.
C. CAMA provides for dialed digit delivery, while FXO does not.
D. CAMA provides for ANI digit delivery, while FXO does not.
E. CAM supports only loopstart, while FXO supports ground- and loopstart.

Answer: A
CIPT Solution Reference Network Design 6-3

QUESTION 30
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know how much Layer 3 is required for Call Control on an IP Phone on average.
What will your reply be?

A. 2 kbps
B. 8 bps
C. 16 kbps
D. 120 bps
E. 150 kbps

Answer: E
CIPT Solution Reference Network Design 2-5

QUESTION 31
With regard to Analog DID connections to the PSTN, which of the following statements about are true? (Select two.)
A.DID trunks can only send calls from the CO.
B.DNIS information is send out-of-band.
C.DID trunks can only send calls towards the CO.
D.DNIS information is send in-band.

Answer: A, D

QUESTION 32
Supplementary Services in ISDN for Echo. (Select two.)
A.) G.711
B.) H.323
C.) G.165
D.) G.174
E.) G.168

Answer: C, E
The echo canceller complies with ITU-T standards G.164, G.165, and G.168.

QUESTION 33
Queuing technology in Voice Configurations at the edge gateway (design guide)
Voice (RTP-Traffic), Signaling and normal data traffic.
A.) PQ, CBWFQ, FQ
B.) LLQ, CBWFQ, WFQ
C.) LLQ, CBWFQ, FQ
D.) PQ, WFQ, FQ

Answer: C

QUESTION 34
A newly appointed Certkiller trainee wants to know what protocol that H.255 utilizes a scaled-down version of is also used to set up the connection between two H.323 endpoints.
What will your reply be?

A. SS7
B. Q.921
C. Q.931
D. H.323
E. H.245

Answer: C

QUESTION 35
What are the most important functions of H.245? (Choose two.)

A. It allows both sides of the call to perform IP address exchange and UDP port negotiations.
B. It provides Coder/Decoder (CODEC) type negotiation such as G.711, between the calling and the called parties.
C. It allows both sides of the call to perform IP port negotiation.
D. It allows both sides of the call to perform H.255 port negotiation.

Answer: B, C

QUESTION 36
TCP receives a request to open a voice channel on port 1720 in a VoIP network. There is no Fast Start. For what purpose will a new TCP port automatically be allocated for?

A. H.225 call compression.
B. H.323 call compression.
C. G.726 call compression.
D. H.225 call setup negotiation
E. H.323 call setup negotiation.
F. H.245 capability exchange negotiation

Answer: D

QUESTION 37
You are a network administrator at Certkiller and your newly appointed Certkiller trainee wants to know from you what is considered to be a node on a H.323 network?
What will your reply be?

A. Proxy.
B. None of available options.
C. Gateway.
D. Options A, C.

E. Gatekeeper.
F. All of the above.

Answer: F

QUESTION 38
What is the sampling rate used by PCM asspecified by Nyquist?

A. 800 per second
B. 1600 per second
C. 4000 per second
D. 8000 per second

Answer: D

QUESTION 39
What happens when you use the Low Latency Queuing feature of the Cisco IOS? (Select two.)

A. All the data traffic is serviced by the PQ.
B. All the RTP traffic is serviced by the PQ.
C. All the RTP traffic is serviced by the CBWFQ.
D. All the data traffic is serviced using the CBWFQ.
E. None of the above.

Answer: B, D

QUESTION 40
Your newly appointed Certkiller trainee wants to know what percent of a standard G.711 packet is taken by IP, UDP and RTP headers when CRTP is not used.
What will your reply be?

A. 15%
B. 20%
C. 25%
D. 33%
E. 45%

Answer: B

QUESTION 41
You are the technician at Certkiller . Your newly appointed Certkiller trainee wants to know what command would enable CRTP.
What will your reply be?

A. ip crtp compress
B. ip rtp compress stac

C. ip crtp compress stac
D. ip rtp header-compression
E. ip crtp header-compression

Answer: D
To enable Real-Time Transport Protocol (RTP) header compression, use the ip rtp header-compression command in interface configuration mode.
http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123tcr/123tqr/qos_i1gt.htm#1133081

QUESTION 42
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know which fields in the output fromshow active voice
command indicates that packet loss is occurring.
What will your reply be?

A. The Receive delay field.
B. The High Water playout delay field.
C. The Low Water playout delay field.
D. The Interarrival packet rate field.

Answer: A
VOICE QUALITY, Using the show call active voice Command to Troubleshoot Voice Quality Issues
http://www.cisco.com/en/US/tech/ CK6 52/ CK6 98/technologies_tech_note09186a008019ab88.shtml

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QUESTION 43
What is the default fax relay connection rate?

A. 14400 bps
B. 7200 bps
C. 9600 bps
D. 28800 bps
E. 11000 bps

Answer: B
14400 for G711, 7200 for G729. G729 is default codec

QUESTION 44
Which of the following can AAA NOT be used for? (Choose two.)

A. Administration
B. Authentication
C. Architecture
D. Admission
E. Security

Answer: A, C

QUESTION 45
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what the proper configuration for VoIP authentication via Authentication, Authorization, and Accounting (AAA) is.
What will your reply be?

A. aaa new-model
aaa authentication login voip radius
B. aaa new-model
aaa authentication login h225 radius
C. aaa new-model
aaa authentication h323 login radius
D. aaa new-model
aaa authentication login default radius
E. aaa new-model
aaa authentication login h323 radius

Answer: A
http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/secur_r/sec_a1g.htm#18239

QUESTION 46
You are a network administrator at Certkiller . You want to configure
an AS5300 to authenticate Certkiller users for Authentication, Authorization, and Accounting (AAA) RADIUS server by prompting the user for a PIN number, etc. You want AS5300 to use application clid_authen_collect. Certkiller users are currently dialing 9592000.
What configuration should you use?

A. dial-peer voice 1 pots
destination-pattern 9592.....
port 0:D
application clid_authen_collect
B. dial-peer voice 1 pots
destination-pattern 2.....
port 0:D
application clid_authen_collect
C. dial-peer voice 1 pots
incoming called number 9592000
destination-pattern 2.....
port 0:D
application clid_authen_collect
D. dial-peer voice 1 pots
incoming called-number 959....
destination-pattern 2.....
port 0:D
application clid authen collect

Answer: D
Cisco IOS Voice Commands, application
http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122tcr/122tvr/vrg_a1.htm#149288
1

QUESTION 47
On what is busy hour traffic for voice gateway port/trunk is based upon in a call center deployment?
(Choose two.)

A. Queue time.
B. Agent talk time.
C. Agent wrap up time.
D. Agent after call work time.
E. All of the above.

Answer: A, B

QUESTION 48
You are a network technician at Certkiller . Your newly appointed Certkiller trainee asks you what three elements make up the MQC.
What will your reply be?

A. Class-map, Policy-map and Service-policy statement
B. DSP, Codec and Sampling Rate
C. Mean Opinion Scores, representative sampling, Standard Deviation
D. Gatekeeper, H 323 Proxy and RSVP
E. Firewall, Router, Bios

Answer: A

QUESTION 49
You are a network administrator at Certkiller . You want to configure a T1 (1,536M) FR PVC for voice and data traffic. You do not expect voice to require more than half the bandwidth.
Which of the following would be the most sufficient FRTS configuration?

A. map-class frame-relay FRST-voice
frame-relay cir 1536000
frame-relay bc 15360
frame-relay be 0
frame-relay mincir 1536000
B. map-class frame-relay FRTS-voice
frame-relay cir 1536000
frame-relay bc 15360
frame-relay be 0
frame-relay mincir 768000

Answer: A

QUESTION 50
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what is NOT a primary cause of echo in a voice network.
What will your reply be?

A. Delay in the IP Network.
B. Acoustical Reflections.
C. 4 wire to 2 wire Hybrids.
D. Packet Loss in the IP Network.
E. All of the above.

Answer: D

QUESTION 51
There are 100 calls with duration of 6 minutes in busy hour time. How many Erlangs is this?
A.) 100 Erlang
B.) 10 Erlang
C.) 3600 CCS, Centum Call Seconds
D.) B and C

Answer: B

Explanation:
Points of contention:
1. To calculate this the formula for Erlangs is- 1Erlang= (BCHA x AHT) /3600 or /60
2. In the above example [100 (BHCA) x 6 minutes (aht)] /60 =10Erlangs
3. http://www.cisco.com/en/US/partner/tech/ CK6 52/ CK7
01/technologies_white_paper09186a00800d6b74.shtml
One erlang is 3600 seconds of calls on the same circuit, or enough traffic load to keep one circuit busy for 1 hour. Traffic
in erlangs is the product of the number of calls times AHT divided by 3600, as shown in the following example:
(23 calls * 172.87 AHT)/3600 = 1.104 erlangsWhich unit you use depends highly on the equipment you use and what unit of measurement they record in. Many switches use CCS because it is easier to work with increments of 100 rather than 3600. Both units are recognized standards in the field. The following is how the two relate: 1 erlang = 36 CCS.

QUESTION 52
What is the purpose of the Erlang-C Traffic Model in an IP Contact Center deployment?

A. To provision ports on an IP-IVR interfacing with Cisco CallManager.
B. To provision agents initiating/handling the outbound calls only.
C. To provision agents receiving/handling the inbound calls.
D. To provision ports on a voice gateway interfacing to the PSTN.

Answer: C
According the website http://www.erlang.com/whatis.html , the source given by Cisco as a study resource, the Erlang C is used for Call Center Agent staffing / queuing.

QUESTION 53
Which of the following can generate 100 Erlangs during the busy hour? (Choose two.)

A. 1 call per hour averaging 100 minutes.
B. 2000 calls per hour averaging 3 minutes each.
C. 3000 calls per hour averaging 2 minutes each.
D. 3000 calls per hour averaging 3 minutes each.
E. None of the above.

Answer: B, C

QUESTION 54
With regard to echo, which of the following statements is false?

A. The term "ERL" refers to a measurement of the volume of Echo heard by the user.
B. Increasing the Echo-Cancellation coverage in an Echo Canceller may also increase Echo Canceller convergence time.
C. Analog components in the voice path causes echo.
D. Echo exists in a Circuit Switched environment, but usually goes unnoticed because of the flow delay.

Answer: A

QUESTION 55
You are a network administrator at Certkiller . You are troubleshooting an IOS Voice Gateway. Which command will produce detailed information about the codec, ERL, tx/rx packets, and dial peers on the current active calls?

A. The show voice port command.
B. The show voice call command.
C. The show voice call active command.
D. The show call active voice command.

Answer: D
http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123tcr/123tvr/vrht_sh1.htm#34981

QUESTION 56
What is the traffic model typically used when dimensioning call center agents receive calls from infinite sources (PSTN callers) where call are queued during the busy hour?

A. Erland-B
B. Erland-C

C. Binomial
D. Extended Erlandg-B
E. Engset

Answer: B
Not A: This model, Erlang-B, block calls and do not queue.

QUESTION 57
You are a network engineer at Certkiller . Certkiller has a voice gateway that is receiving calls from infinite sources (PSTN callers) during the busy hours when lost calls should be cleared (blocked).
What typical traffic model is required to dimension the number of gateway ports/trunks?

A. Erlang-B
B. B and C
C. Erlang-C
D. Poisson
E. None of the above.

Answer: A

QUESTION 58
You are the technician at Certkiller . Your newly appointed Certkiller trainee wants to know what could cause a user to hear echoes of her own voice.
What will your reply be? (Choose three.)

A. Mismatch in impedance in the hybrid transformer.
B. Gain in local loop.
C. ERL is low at the trail circuit.
D. A-3 db loss is taking place in local loop.

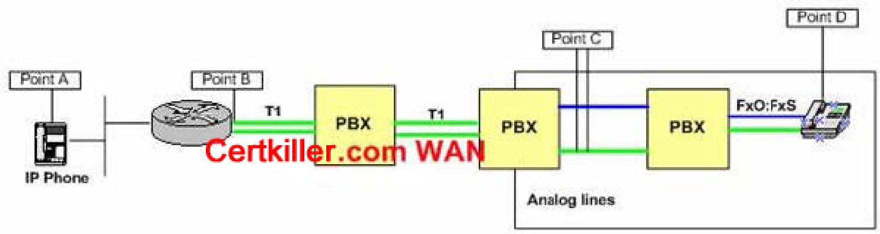
Answer: A, B, C

QUESTION 59
What weighting are packets destined for the PQ given when PQ-WFQ is configured on an interface?

A. 1024
B. 56
C. 12
D. 0

Answer: D
http://www.cisco.com/en/US/tech/ CK6 52/ CK6 98/topic1

QUESTION 60
You are a network administrator at Certkiller . The Certkiller network is shown in the following exhibit:



With reference to the exhibit, what section of the voice path represents the Tail Circuit?

A. Between Point A and Point B.
B. Between Point B and Point C.
C. Between Point B and Point D.
D. Between Point C and Point D.

Answer: C
Reference:
1. Cisco IP troubleshooting P416
2. http://www.cisco.com/en/US/tech/ CK6 52/ CK6 98/technologies_tech_note09186a0080149a1f.shtml

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QUESTION 61
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee
Wants to know what, according to Cisco's design guide, is the typical bandwidth needed for a single VoIP call (including layer 2) when using a g.729 codec, and no header compression.
What will your reply be?

A. 64 Kbps
B. 32 Kbps
C. 16 Kbps
D. 8 Kbps
E. 4 Kbps

Answer: B
Actually for Ethernet it is 29.6kbps

QUESTION 62
What is the range of UDP port numbers used in Cisco's VoIP implementation?

A. 1699 to 3210
B. 1100 to 2100
C. 1225 to 2245
D. 16384 to 32767
E. 32769 to 64535

Answer: D

QUESTION 63
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know

how a H.323 proxy Gatekeeper Request (GRQ) Registration, Admission, and Status (RAS) message is sent and by which endpoints.
What will you tell your trainee? (Choose three.)

A. Gateway.
B. H.323 Terminal.
C. Proxy.
D. Gateway.
E. Firewall.

Answer: A, B, C

QUESTION 64
Certkiller has recently appointed a new trainee. The trainee wants to know what the target overall loss plan across a telephone network is.
What will your reply be?

A. 12dBm - 24dBm.
B. 8dBm - 16dBm.
C. 4dBm - 12dBm.
D. 0dBm - 8dBm.

Answer: D

QUESTION 65
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what a gatekeeper is.
What will your reply be? (Choose two.)

A. A compulsory component in a H.323 system which provides call control services to the H.323 endpoints.
B. An optional component in a H.323 system which provides call control services to the H.323 endpoints.
C. A compulsory component in a SIP system which provides call control services to the H.323 and SIP endpoints.
D. Logically separate from the endpoints, but its physical implementation may coexist with a terminal, multipoint conference unit (MCU), gateway, multipoint controller (MP), or other non-H.323 LAN device.
E. An optional component in a SIP system which provides call control services to the H.323 and SIP endpoints.

Answer: B, D

QUESTION 66
You are a network administrator at Certkiller . Certkiller has a converged 512 Kbps MLP circuit. What set of commands will guarantee a maximum serialization delay of 15 ms on this circuit?

A. ppp multilink fragment 960
B. ppp multilink fragment 320
C. ppp multilink fragment 640

ppp multilink interleave
D. ppp multilink fragment-delay 15
ppp multilink interleave
E. ppp multilink fragment-delay 15

Answer: D
To disable packet fragmentation, use the ppp multilink fragment disable command in interface configuration mode.
http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/dial_r/dia_n1g.htm#1134762

QUESTION 67
You are a network administrator at Certkiller . Certkiller has a H.323 network. Which of the following functions is NOT performed by the Gatekeeper on the Certkiller network?

A. Call routing
B. The Number to IP address translations
C. Call authorization
D. Call admission control
E. Codec negotiation

Answer: E

QUESTION 68
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what type of signalling provides Dialled Number Information Service (DNIS) on a T1/E1.
What will your reply be?

A. E&M
B. Ground start
C. Loop Start
D. All of the above

Answer: D
TELEPHONY SIGNALING, Understanding How Digital T1 CAS (Robbed Bit Signaling) Works in IOS Gateways
http://www.cisco.com/en/US/tech/ CK6 52/ CK6 53/technologies_tech_note09186a00800e2560.shtml

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QUESTION 69
What is a DPA (Digital PBX Adapter) used for?

A. To allow a customer to network VoiceMail systems and PBX together.
B. To enable Calling-Name between CallManager and Meridian Mail systems.
C. To connect an Octel 200/300/250/350 to CallManager.
D. To connect CallManagerr to PBX.
E. None of the above.

Answer: C
CISCO DPA 7600 SERIES GATEWAYS
http://www.cisco.com/en/US/products/hw/gatecont/ps821/index.html

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QUESTION 70
Where can the Gatekeeper be when Gateways are registering with a Gatekeeper?

A. In a different subnet.
B. On a remote LAN.
C. On the same subnet.
D. On the same LAN.
E. Any of the options above.

Answer: E

QUESTION 71
You are a network administrator at Certkiller . You want to implement a standards-based protocol that will allow CallManager to seamlessly integrate with another vendor's traditional PBX system. What protocol should you implement? (Choose three.)

A. MGCP
B. QSIG
C. PRI NI-2
D. SRST
E. OSPF

Answer: A, B, C
The standards based protocols to connect to other pbx's are mgcp, qsig and pri.
CM utilizes a VoiceGateWay with one of three protocols:
MGCP
SIP
H.323
http://www.cisco.com/en/US/partner/products/sw/voicesw/ps556/products_usage_guidelines09186a00803b2b0a.html
Not D:SRST = Survivable Remote Site Telephony and is not a protocol as it is more of a philosophy. The question does not have anything to do with SRST.
SRST
CiscoSRST provides CiscoCallManager with fallback support for Cisco IP phones if a WAN failure is detected. Cisco SRST runs under the Cisco IOS stack and uses the Skinny protocol (SCCP) to interact with Cisco IP phones.
http://www.cisco.com/en/US/partner/products/sw/voicesw/ps556/prod_troubleshooting_guide_chapter09186a00801ef30

QUESTION 72
You are a telephony trainee at Certkiller . The network administrator asks you what the advantage of Survivable Remote Site Telephony (SRST) design method is.

What will your reply be?

A. It enhances the availability of CM distributed call processing.
B. It enhances the availability of CM single site campus design.
C. It enhances the availability of CM tool bypass.
D. It enhances the availability of CM centralized call processing.

Answer: D

QUESTION 73
What types of call processing functions do SRST preserve in a remote office in a CM network? (Choose two.)

A. IP Phone to VoiceMail transcoding services.
B. IP Phone to conference DSP resources.
C. IP Phone to GW calls.
D. IP Phone to IP Phone calls.
E. CTI applications such as IP SoftPhones.

Answer: A, C
CISCO CATALYST 4200 SERIES SWITCHES, Configuring Survivable Remote Site Telephony
http://www.cisco.com/en/US/products/hw/switches/ps669/products_configuration_guide_chapter09186a008007f1e

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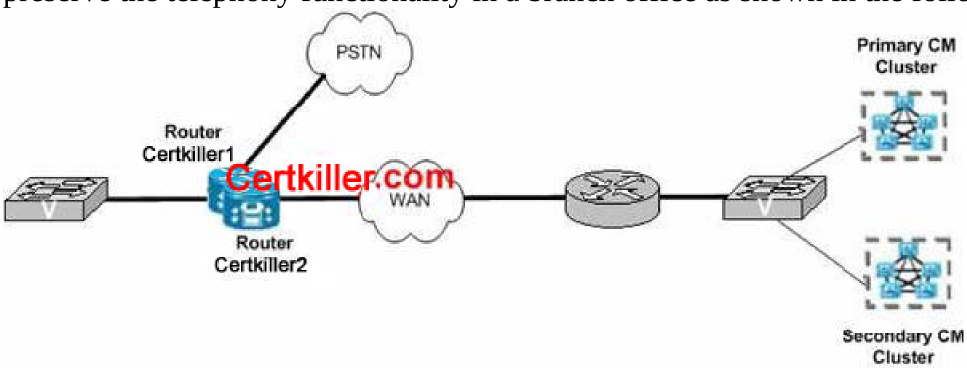
QUESTION 74
You are the network administrator at Certkiller . Certkiller uses HSRP in conjunction with SRST to preserve the telephony functionality in a branch office as shown in the following exhibit:



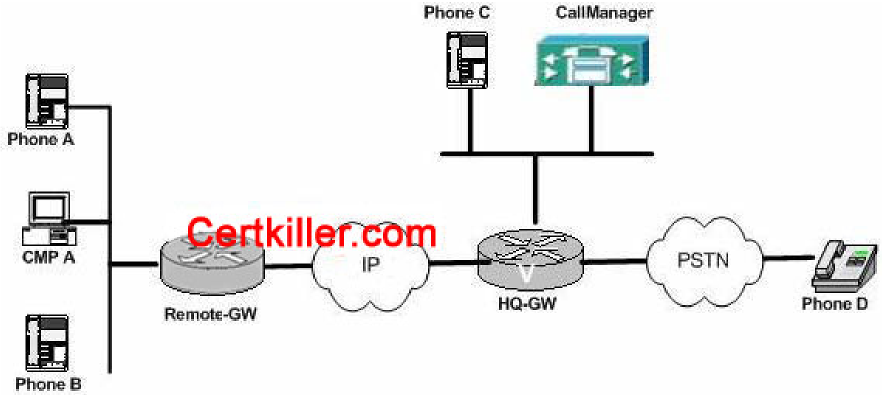
Should a WAN failure occur while the primary router, Router Certkiller 1 is in use, Router Certkiller 1 should switch to SRST mode to preserve telephony functions. However, if Router Certkiller 1 also fails, the HSRP backup router, Router Certkiller 2, must become the active router for the branch office and must take over SRST and routing functions for the branch office.
What type of physical connectivity must you duplicate on Routers Certkiller 1 and Certkiller 2 to ensure that Router Certkiller 2 will be effective in running SRST for the branch office?

A. ISDN
B. PSTN
C. WAN

D. CMs
E. VLANs

Answer: B

QUESTION 75
You are the network administrator at Certkiller . The Certkiller Voice network is shown in the following exhibit:



A Certkiller user at Phone A complains about a persistent echo on calls to the PSTN. The ERL has been determined to be 15db. The configuration on the HQ-GW voice T1 is as follows:
voice-port 1/0:15
echo-cancel coverage 8
end
What should be done resolve this problem?

A. Increase the echo tail coverage
B. Decrease the NLP threshold
C. Decrease the output gain
D. Increase the input gain
E. Increase the output gain

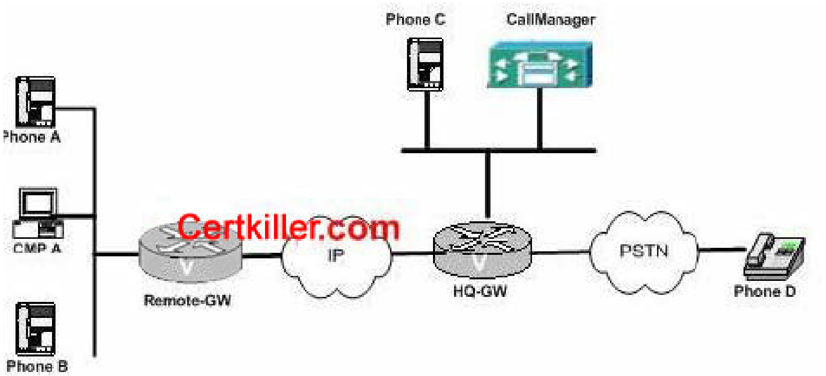
Answer: A
Cisco IOS Voice, Video, and Fax Commands, echo-cancel coverage
http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122tcr/122tvr/vrg_e1.htm#998290

QUESTION 76
You are contracted as a network administrator for a small company, Certkiller Inc. Your newly appointed Certkiller trainee wants to know
What combination will give him the best video quality using a 128kbs video conference call?
What will your reply be?

A. H.263 video and G.728 audio
B. H.261 video and G.728 audio
C. H.263 video and G.711 audio
D. H.261 video and G.711 audio
E. H.263 video and G.722 audio

Answer: A

QUESTION 77
You are the network administrator at Certkiller . The Certkiller Voice network is shown in the following exhibit:



Everything in the Certkiller Voice network is under your control except for the PSTN.
The Certkiller user at IP Phone A complains of a persistent echo during all calls from IP Phone A to Analog Phone D.
What would be the best way to resolve this echo issue?

A. Adjust the Echo Cancellation parameters on Phone D.
B. Adjust the Echo Cancellation parameters on HQ-GW.
C. Adjust the Echo Cancellation parameters on Phone A.
D. Adjust the Echo Cancellation parameters on Remote-GW.
E. Adjust the Echo Cancellation parameters on CallManager.

Answer: B

QUESTION 78
On which criteria does a Cisco SIP Proxy Server use to make routing decisions?

A. From: header
B. To: header
C. SDP parameters
D. User-Portion on the Request -URI

Answer: D

QUESTION 79
Which of the following tasks can the Cisco SIP Proxy not perform?

A. Registrar Server.
B. Redirect Server.
C. Proxy Server.
D. User Agent.

Answer: D

QUESTION 80
Regardless of next-hop SIP device, what is the order in which CSPS will determine how to route the packet when you process a SIP message?

A. Static Routes, GKTMP, Registry, LRQ to H.323 Gatekeeper
B. Domain Routes, GKTMP, Registry, LRQ to H.323 Gatekeeper
C. Static Routes, GKTMP, LRQ to H.323 Gatekeeper, Registry
D. Registry, GKTMP, Static Route, LRQ to H.323 Gatekeeper

Answer: D
Cisco SIP Proxy Server Administration Guide p1-8

QUESTION 81
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what SIP header a SIP Proxy is allowed to change.
What will your reply be?

A. The To header.
B. The Request-URI.
C. The Contact header.
D. The From header.

Answer: B
CISCO SIP PROXY SERVER, F Call-Flow Scenarios
http://www.cisco.com/en/US/products/sw/voicesw/ps2157/products_administration_guide_chapter09186a00801a1

QUESTION 82
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what the following SMDI packet represents:
MD0010013D 0002215
What will your reply be?

A. MWI OFF commands for extension 2215
B. A "Call Forward No Answer" extension 2215 from extension 10013
C. MWI ON command for extension 2215
D. A "Forward All Calls", extension 10013 calling 2215
E. Extension 2215 calling into voicemail on port 13

Answer: E
Troubleshooting Legacy Voice Mail Integration with Cisco CallManager 3.0 and 3.1
http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a00800a8956.shtml

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QUESTION 83
You are a network administrator at Certkiller . An IP Phone named Phone A is configured with the extension 1500. Another IP Phone named Phone B is configured with the extension 1501. What is the minimum configuration that will allow the two IP Phones to contact each other?

A. Both phones must be on the same trunk.
B. Both phones must be assigned the same Calling Search Space.
C. Both phones extensions must be in the same partition.
D. None of the above.

Answer: C

QUESTION 84
When the CallManager goes down, what will the PBX do?

A. Normal telephony will work
B. There will nothing work
C. There are only some problems with signaling, but calls will be forwarded
D. ?

Answer: Pending

QUESTION 85
When is it possible for two Unity Servers to be placed in the same Dialing Domain? (Select two.)

A. When they are attached to the same PBX.
B. When they are both assigned the same location ID.
C. When they do not have to dial trunk access codes to reach each other's subscribers.
D. When their subscribers do not have overlapping extensions.
E. When they are in the same Exchange Site/Routing Group.

Answer: A, D
CISCO UNITY, Network Settings
http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_administration_guide_chapter09186a0080 1ba

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QUESTION 86
You are a network administrator at Certkiller . Certkiller has a router that is connected to a PBX via a 4 wire E&M circuit. All the calls to the trunk are failing. You suspect that the PBX is not receiving the incoming calls on the trunk.
To determine if this is the case, you want PBX to generate a dial tone when it receives an incoming call. Which of the following will cause the PBX to generate dial tone?

A. Short the E pin to the Tip pin.
B. Short the E pin to the ground.
C. Short the M pin to the ground.

D. Short the M pin to the Ring pin.
E. Short the Tip pin to the Ring pin.

Answer: B

QUESTION 87
You are the network administrator at Certkiller . Certkiller has a Cisco IPCC deployment environment. CallManager is the routing client.
In the Certkiller network, perform what function will the ICM upon the receiving a call? (Choose two.)

A. ICM plays pre-recorded announcements for callers waiting in queue if no agents are available.
B. ICM plays music for callers waiting in queue if no agents are available.
C. ICM Identifies and selects an available agent and determines the label to be returned to the routing client.
D. ICM maps the Dialed Number to a call type and then maps the call type to a routing script.
E. None of the above.

Answer: C, D.

QUESTION 88
What happens when configuration changes are made to the CCMAdmin page of a Subscriber CallManager?

A. The configuration changes are made locally in the SQL Database, and replicated in the publisher SQL Database.
B. The configuration changes are made in the publisher SQL Database, and replication to subscribers.
C. The configuration changes are made locally in the SQL Database, and replicated up to the publisher immediately.
D. The configuration changes are made locally in the SQL Database, and replicated up to the publisher at the next scheduled replication.

Answer: B

QUESTION 89
You are a network administrator at Certkiller . The new Certkiller trainee wants to know what Inter-Cluster communications signalling includes in a CallManager cluster.
What will your reply be? (Select two.)

A. Locations bandwidth and Shared media resources.
B. Call detail records (CDR) database replication.
C. Registration of devices Locations bandwidth Shared media resources.
D. Device configuration replication.
E. None of the above.

Answer: A, C
CIPT Course v3.3 p1-37

QUESTION 90
What does CallManager use the concept of Location for?

A. To define groups of devices based on physical location, in order to assign Primary and Backup CallManager servers.
B. To group devices based on physical location, in order to delegate Administrative Control.
C. To define the CODEDC to be used between two devices separated by a WAN link.
D. To define the bandwidth that can be used between two devices.

Answer: D
CIPT Course v3.3 p3-98

QUESTION 91
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know what the difference is between a route-group and a route-list.
What will your reply be?

A. Route-lists contain gateways for route-groups.
B. Route-group contains a list of route-patterns.
C. Route-lists contain a list of gateways.
D. Route-groups contain route-lists which points to the gateways.
E. Route-lists contain route-groups which point to the gateways.

Answer: E
CIPT Course v3.3 p3-22

QUESTION 92
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know why PRI is the preferred method for inter-connecting CallManager3.2 and below to PBX's.
What will your reply be? (Select two.)

A. It allows a customer to share their exiting Meridian Mail system with CallManager subscribers whilst delivering full functionality.
B. It provides Caller ID functionality.
C. It is the most cost effective solution available.
D. It offers the highest level of inter-operability between CallManager and PBX's.
E. It allows a customer to share their exiting VoiceMail system with CallManager.

Answer: C, D

QUESTION 93
You are a network administrator at Certkiller . The Certkiller Voice network is configured as follows:
* Phone A's device calling search space is CSS_Dev_A.
* Phone A's Line 1 is assigned calling search space CSS_Line_A
* Route Pattern 2XXX is placed in Partition Part_1.
* Route Pattern 20XX is placed in Partition Part_2.

* Route Pattern 200X is placed in Partition Part_3.
* CSS_Dev_A contains partition(s) Part_1.
* CSS_Line_A contains partition(s) Part_2.
If a call is made to 2001 from Phone A, using Line1, what route pattern will be used by Call Manager?

A. 200X in partition Part_3.
B. 20XX in partition Part_2.
C. 2XXX in partition Part_1.
D. None of the above.

Answer: B

QUESTION 94
Your newly appointed Certkiller trainee wants to know by you what CallManager uses a Calling Search Space for.
What will your reply be?

A. To restrict calls to a particular range of numbers.
B. To enable the use of E911 services.
C. To enable the use of an overlapping dial plan.
D. To provide access-list-like security.
E. All of the above.

Answer: E
CIPT Solution Reference Network Design Chapter 7

QUESTION 95
You are a network administrator at Certkiller . Your newly appointed Certkiller trainee wants to know how many different types of devices can register with a Cisco CallManager.
What will your reply be?

A. The number of calls a device handles in the busy hour.
B. All of the above.
C. The total number of each device type.
D. Memory and CPU resources each device type requires from the server.
E. Pagefile and CPU usage.

Answer: D
CIPT Solution Reference Network Design 6-3

QUESTION 96
Exhibit:
phone -> gw -> fr -> atm -> gw -> phone
ATM to FR and Back over WAN for Voice-Transfer (FR <-> ATM). (Select two.)
A.) FRF.8
B.) FRF.11

C.) FRF.12
D.) FRF.15
E.) FRF.16

Answer: A, B

QUESTION 97
Transfer types from AMIS to Email-server.
A.) Email-Message
B.) Email-Message and Voice-Mail-Message
C.) Voice-Mail-Message
D.) ?

Answer: C

QUESTION 98
You are a network engineer for Certkiller , Inc. Your newly appointed Certkiller trainee asked you about the perspective of the CallManager. In particular, she wants to know which device the Unity TSP looks and behaves mostly like.
What will your reply be?

A. A TAPI Device
B. A MGCP Gateway
C. A Cisco IP Phone
D. A H.323 Gateway
E. A CTI Port

Answer: A

QUESTION 99
You are a network administrator at Certkiller . You want to troubleshoot a FailSafe problem in Unity.
Where should you look for detailed error messages?

A. In the Application.Log.
B. On the Status Monitor.
C. In the Tempu.log.
D. In the SDL Trace.
E. In the System.log.

Answer: A

QUESTION 100
Test Technologies and King, Inc recently merged to form Certkiller , Inc. You are appointed as the network administrator for Certkiller , Inc. An IP Phone from Test Technologies, named Phone A has been assigned to Calling Search Space
A. Calling Search Space A contains the following partitions in the order

shown, listed with their respective Routing Patterns:
Partition_A1, containing Route Pattern 1XXX
Partition_A2, containing Route Pattern 10XX
Phone A dials "1001". Which of the following statements is true?

A. None of the route pattern is an exact match. Therefore, none will match and the caller will hear the re-order tone.
B. Both patterns are equivalent matches. Therefore Call Manager will choose them the round robin fashion.
C. Route Pattern 1XXX and 10XX both match, but since 1XXX is listed at the top it will be chosen first.
D. Route Pattern 1XXX and 10XX both match, but since the 10XX is a better match it will be chosen.

Answer: D
http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_administration_guide_chapter09186a00800c2e

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QUESTION 101
You are the network administrator at Certkiller . The Certkiller network is shown in the following exhibit:

A Certkiller user located at phone A dials 5551212555. The GWs the Certkiller network are configured as follows:

REMOTE-GW

```
voice translation-profile  CertKiller_1
translate called 1
voice translation-rule 1
rule 1 /'\(555\)+\(.+\)//444\2/ type any national any isdn
dial-pee voice 1 voip
translation-profile outgoing  CertKiller_1
session target ipv4.x.x.x.x
Port 0/0
```

HQ-GW

```
Interface FastEthernet0
Ip Address x.x.x.x y.y.y.0
voice translation-profile  CertKiller_2
translate called 1
voice translation-rule 1
rule 1 /'\(12\)+\(.+\)//911\2/type national unknown plan unknown isdn
dial-peer voice 1 pots
translation-profile outgoing  CertKiller_2
Port 1/0:23
```

The call routing is working properly through the IP Network.
What digit string will be sent to the PSTN for termination?

A. 555121255
B. 444121255
C. 555911911444
D. 5551444555
E. 5554442555

Answer: A

QUESTION 102
You are a telephony trainee at Certkiller Inc. Your instructor asks you what industry-standard protocol the CallManager uses for integration to VoiceMail systems.
What will your reply be?

A. T1-CAS
B. H.323
C. SMDI
D. Q.931
E. H.225.

Answer: C

QUESTION 103
Which of the following Cisco Products can produce SMDI packets? (Choose three.)

A. A Cisco Unity.
B. A Cisco CallManager.
C. A Cisco VG200 Voice Gateway.
D. A Cisco VG248 Analog Phone Gateway.
E. All of the above.

Answer: A, B, D

QUESTION 104
You are a network engineer at Certkiller , Inc. Your newly appointed Certkiller trainee wants to know what SMDI message from CallManager CMI or VG248 allows a Voicemail system to provide a "Hearth-beat" function on a RS-232 serial link.
What will your reply be?

A. MWI INV
B. MWI BLK
C. OP:MWI
D. None of the above

Answer: C
Configuring CallManager 3.x for Integration to Voice Mail Systems via SMDI
http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_configuration_example09186a0080160b6f.shtm

QUESTION 105
What factor determines the fragmentation size on a T1/E1 Frame Relay circuit?

A. Line speed.
B. Burst Count.
C. The Average CIR.
D. The minimum CIR.
E. The maximum CIR.

Answer: D
VOFR, Frame Relay Fragmentation for Voice
http://www.cisco.com/en/US/tech/ CK6 52/ CK6 92/technologies_tech_note09186a00801142de.shtml

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QUESTION 106
You are a network administrator at Certkiller . You have been instructed to select the correct SMDI packet from the options below:
Select the correct SMDI packet:

A. MVI010001N0002222 000112324
B. MD0010001A0002222 000112324
C. MC0010001D0002222 00012324
D. ND0010001A0002222 00012324
E. RD0010001B000222 00012324

Answer: B
CISCO CALLMANAGER, Troubleshooting Legacy Voice Mail Integration with Cisco CallManager 3.0 and 3.1
http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a00800a8956.shtml

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QUESTION 107

If a 7960 IP phone sends voice media frames towards the access switch, how will these frames be observed at point A
NOTE: Assume that a frame sniffer capturing data between the phone and the access switch

Assume that the access switch, a Cisco Catalyst 3524XLi is configured as:
interface FastEthernet0/1
pow[illegible]
speed auto
switchport trunk encapsulation dot1q
switchport trunk native vlan 12
switchport mode trunk
switchport voice vlan 112
switchport priority extend cos 0
spanning-tree portfast

NOTE: Assume the phone is connected into port FastEthernet0/1.

A. The frame will b tagged with 802.1Q VLAN ID of 12, and will have an 802.1p cos value of 3.
B. The frame will be tagged with 802.1Q VLAN ID of 112, and will have an 802.1p cos value of 5.
C. The frame will be tagged with 802.1Q VLAN ID of 0, and will have an 802.1p cos value of 3.
D. The frame will be un-tagged.

Answer: B
Tagged 802.1Q VLAN 112, and have 802.1p cos value 5.

QUESTION 108
What percentage of a standard G.729a packet is taken by IP, UDP and RTP headers? (NOTE: cRTP not used)

A. 66%
B. 50%
C. 40%
D. 33%
E. 20%

Answer: E

QUESTION 109
What IOS feature can synthesize VoIP packets and measure latency, jitter and loss statistics?

A. RTP Probe
B. Extended Ping - VoIP Feature
C. Real-Time Voice Responder
D. Class-Based QoS MIB
E. Service Assurance Agent

Answer: E

QUESTION 110
Click the Exhibit button.
Consider the Low Latency Queuing (LLQ) configuration segment shown. How will the traffic in the two priority classes be handled by the LLQ algorithm?

```
policy-map WAN-EDGE
class VOICE1
  priority 100
class VOICE2
  priority 50
class VOICE-CONTROL
  bandwidth 20
class class-default
  fair-queue
```


A. There are two priority queues and traffic from each class will be funneled to its own queue.
B. There is a single priority queue of 100K as that is the first statement encountered.

C. This is an invalid LLQ configuration segment - you can only define one priority class.
D. There is a single priority queue of 150K and traffic from both classes are treated FIFO within that queue.
E. There is a single priority queue of 150K and traffic from both classes are treated WFQ within that queue.

Answer: D

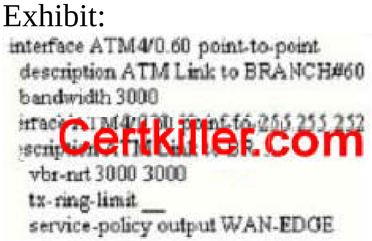
QUESTION 111
What command will guarantee a maximum serialization delay on 10 ms on a converged256 kbps Frame-Relay circuit?

A. frame-relay fragment-delay 10
B. frame-relay fragment 320
C. frame-relay serialization-delay 10
D. frame-relay fragment 640
E. frame-relay fragment 160

Answer: B
Frame relay fragment command for 256k link syntax is frame-relay fragment so answer can be B, D or E.
Fragment is expressed in byte. 256k/8 > 32000 bytes for 1s for 10ms 320 bytes.
Answer is B.

QUESTION 112
Class of Service (CoS) is a:
A.
with designing their network to a certain capacity
B. Portion of the IP header that relates to the service level of the packet
C. General term that describes a level of service necessary for a specific application
D. Method of classifying different traffic flows into a category and applying a particular quality of service (QoS) for that flow

Answer: D

QUESTION 113
Exhibit:


The low-speed ATM PVC shown carries both voice and data traffic. What is the most appropriate value for the tx-ring?

A. 0
B. 3
C. 10
D. 15

E. 60

Answer: B

QUESTION 114
One of the most important aspects of design criteria is minimizing total one-way end-to-end delay. This total delay has been found to be acceptable as long as it remains within:

A. 0 to 1 second
B. 0 to 150 milliseconds
C. 0 to 500 milliseconds
D. 0 to 300 milliseconds

Answer: B

QUESTION 115
When PQ-WFQ is configured on an interface, the packets destined for the PQ are given a weighting of:

A. 0
B. 128
C. 4096
D. 32767

Answer: A

QUESTION 116
Which are the three elements to MQC?

A. CallManager, IP Phones and SRST
B. Gatekeeper, H.323 Proxy and RSVP
C. Mean Opinion Score, representative sampling, Standard Deviation
D. Class-map, Policy-map and Service-policy statement
E. DSP, Codec and Sampling Rate

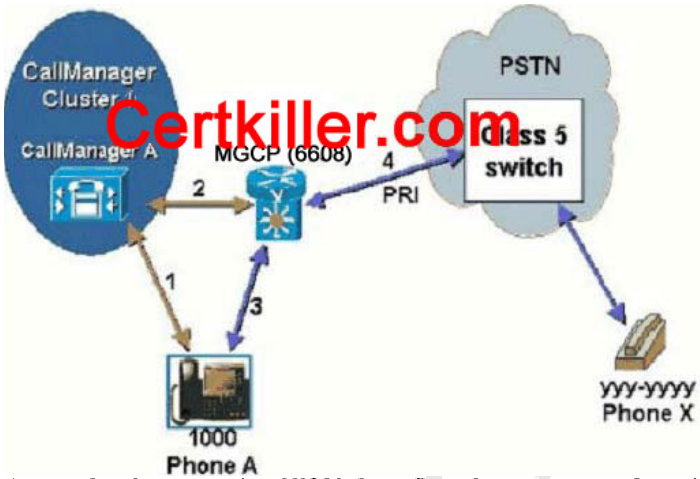
Answer: D

QUESTION 117
The jitter buffer is used for:

A. Concealing variable interframe gaps
B. Re-ordering out of sequence voice packets
C. Encoding analog voice into digital voice
D. Multiplexing multiple voice calls

Answer: A

QUESTION 118
Exhibit:



Assume that the gateway is a 6608 blade configured as a gateway and running MGCP; Call Manager runs version 3.1, and that a call is made from phone A to phone X. All IP streaming is G.711. Each of the logical links represented carries certain types of traffic. On which links can RTP traffic be seen?

A. 2, 3, and 4
B. 3 only
C. 2 and 3
D. 1, 2, 3, and 4
E. 1 and 4

Answer: B

QUESTION 119
What paragraph best describes Pulse Code Modulation (PCM)?

A. PCM converts analog sound into digital form by sampling the analog sound 16000 times per second and converting each sample into a numeric code. The Nyquist theorem states that when sampling an analog signal at twice the rate of the highest frequency of interest, one can accurately reconstruct that signal back into its analog form. Since most speech content is below 4000 Hz (4 kHz), the sampling rate needed is 16000 times per second (225 microseconds between samples): The transmission rate is obtained by multiplying 8000 samples per second times 8 bits per sample, giving 64,000 bits per second.
B. PCM converts analog sound into digital form by sampling the analog sound 8000 times per second and converting each sample into a numeric code. The Nyquist theorem states that when sampling an analog signal at twice the rate of the highest frequency of interest, one can accurately reconstruct that signal back into its analog form. Since most speech content is below 4000 Hz (4 kHz), the sampling rate needed is 8000 times per second (125 microseconds between samples): The transmission rate is obtained by multiplying 8000 samples per second times 8 bits per sample, giving 64,000 bits per second.
C. PCM converts analog sound into digital form by sampling the analog sound 8000 times per second and converting each sample into a numeric code. The Bellman Ford theorem states that when sampling an analog signal at twice the rate of the highest frequency of interest, one can accurately reconstruct that signal back into its analog form. Since most speech content is below 4000 Hz (4 kHz), the sampling rate needed is 8000 times

per second (125 microseconds between samples): The transmission rate is obtained by multiplying 8000 samples per second times 8 bits per sample, giving 64,000 bits per second.
D. All of the above are correct and it depends what type of CODEC is used.

Answer: C
A scientist by the name of Harry Nyquist discovered that the original analog signal can be reconstructed if enough samples are taken. He determined that if the sampling frequency is at least twice the highest frequency of the original input analog voice signal, this signal can be reconstructed by a low-pass filter at the destination.
Reference: http://www.cisco.com/en/US/partner/tech/ CK6 52/ CK7 01/technologies_tech_note09186a00801149b3.shtml

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QUESTION 120
In VoIP, once TCP receives a request for opening a voice channel on port1720, a new TCP port is allocated for (Note: assume no Fast Start):

A. H.225 call setup negotiation
B. H.245 capability exchange negotiation
C. H.323 call setup negotiation
D. UDP port negotiation
E. G.726 call compression

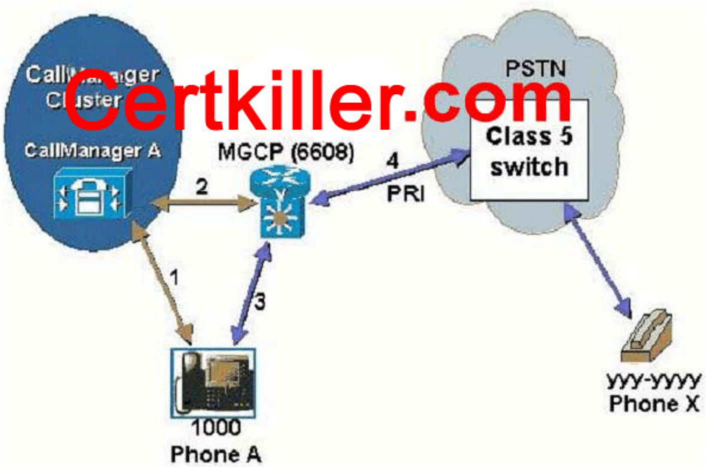
Answer: A

QUESTION 121
A method for touch-tone phones in which each digit corresponds to one of 16 combinations of pairs of sine waves chosen from eight different frequencies is called:

A. In-band signaling
B. Ear and Mouth or REceive and TransMit
C. Multifrequency
D. Out-of-Band signaling
E. Dual tone multifrequency tone detection

Answer: E

QUESTION 122
Click the Exhibit button.



runs version 3.1, and that a call is made from phone A to phone X. All IP streaming is G.711. Each of the logical links represented carries certain types of traffic. On which links can q.931 traffic be seen?

A. 2 and 3
B. 2, 3, and 4
C. 1 and 4
D. 1, 2, 3, and 4
E. 2 and 4

Answer: E

QUESTION 123
H.323 RAS (Registration, Authorization and Status) messages are sent using:

A. TCP/IP
B. UDP/IP
C. ICMP
D. RTMP

Answer: B

QUESTION 124
In general, fax relay is:

A. More tolerant than voice to packet loss
B. Less tolerant than voice to packet loss
C. As tolerant as voice to packet loss
D. Not subject to packet loss

Answer: B

QUESTION 125
A commonly used instance of ADPCM, which encodes using 4-bit samples, giving a transmission rate of 32 kbps is called:

A. ITU-T G.711
B. ITU-T G.723.1
C. ITU-T G.726
D. ITU-T G.728
E. ITU-T G.729

Answer: C

QUESTION 126
What is considered a node in a H.323 network?

A. Gateway
B. Gatekeeper
C. Proxy
D. All of the above

Answer: D

QUESTION 127
To provide for a standard approach for offering voice and fax over Frame Relay, the Frame Relay Forum released a standard, which describes frame formats, conformance requirements, and compression algorithms to support voice and fax over Frame Relay. The standard is:

A. FRF.12
B. FRF.11
C. FRF.11 & FRF.12
D. H.245

Answer: B

QUESTION 128
Real-Time Transport Protocol (RTP) provides: (Choose two.)

A. Payload header and content identification
B. Sequence numbering
C. Feedback to calling and called party about the quality of connection
D. Simple time-stamp & reconstruction
E. Lost packet statistics and round trip times

Answer: A, B

QUESTION 129
The IOS GWs support ECMA QSIG. CM, in MGCP call Agent mode, supports ISO QSIG. What implications does this have on an IP telephony network?

A. None, the ISO standard is a superset of the ECMA standard.
B. None, as long as the IOS GW remains in contact with the CM at all times.
C. During MGCP GW Fallback, no calls to the attached QSIG PBX will ork.
D. During MGCP GW Fallback, basic calls to the attached QSIG PBX will work.
E. During MGCP GW Fallback, all call functionality to the attached QSIG PBX will work.

Answer: D

QUESTION 130
What ITU-T logarithmic pulse code modulation (PCM) standard (G.711) used in the conversion between analog and digital signals is used mainly inEurope?

A. MU-law
B. A-law
C. MU-law & A-law
D. None of the above

Answer: B

QUESTION 131
The terms "Wink start", "Delay start" and "Immediate start" are applicable to:

A. Analog E&M Signaling
B. T1 CAS E&M signaling
C. E1 CAS E&M Signaling
D. Analog DID Signaling
E. All of the above

Answer: A

QUESTION 132
What does SMDI stand for?

A. Serial Message De-muxing Interface
B. Simple Message Desk Interface
C. Skinny Message De-muxing Interface
D. Simple Modular Disk Interface
E. None of the above

Answer: B

QUESTION 133
Which Cisco Products can provide SMDI packets? (Choose three.)

A. Cisco VG200 Voice Gateway
B. Cisco VG248 Analog Phone Gateway

C. Cisco Call Manager
D. Cisco Unity
E. Cisco IAD-2400

Answer: B, C, D

QUESTION 134
AAA Can be used for: (Choose three.)

A. Unified messaging
B. Admission
C. Authentication
D. Security
E. Architecture
F. Administration
G. Billing

Answer: B, C, G

QUESTION 135
When provisioning Cisco Call Manager device weights, what is taken into account?

A. The total number of devices by type and BHCA per device
B. The deployment model (Centralized or Distributed Call Processing)
C. The server model and type
D. A and C

Answer: D

QUESTION 136
Private Line Auto Ringdown (PLAR) is a way to bypass dial tone from the:

A. Originating Gateway
B. Gatekeeper
C. Terminating Gateway
D. Destination Switch

Answer: A

QUESTION 137
When Direct Inward Dialing is used on a POTS dial peer and an incoming POTS call center enters the

router: (Choose two.)

A. The number that is dialing (ANI) automatically becomes the destination-pattern number for the IP destination.
B. The number that is dialed (DNIS) automatically becomes the destination-pattern number for the IP destination.
C. The number that is dialing (ANI) automatically becomes the destination-pattern number for the telephony destination.
D. The number that is dialed (DNIS) automatically becomes the destination-pattern number for the telephony destination.
E. The number that is dialed (DNIS) & the number that is dialing (ANI) automatically becomes the destination-pattern number for the IP & telephony destination.

Answer: B, D

QUESTION 138
In a 128 kbs videoconference call, what combination will give you the best video quality?

A. H.261 video and G.711 audio
B. H.261 video and G.728 audio
C. H.263 video and G.711 audio
D. H.263 video and G.728 audio
E. H.263 video and G.722 audio

Answer: D

QUESTION 139
AMIS is used to:

A. Send Email messages
B. Send VoiceMail messages
C. Send VoiceMail & Email messages
D. Send Recorded names

Answer: B

QUESTION 140
In a CallManager cluster, Intra Cluster Communications Signaling Includes: (Choose four.)

A. Registration of devices
B. Device configuration replication
C. Locations bandwidth Shared media resources
D. Call Detail Records (CDR) database replication

Answer: A, B, C, D

QUESTION 141
What is the correct sequence of call setup between a IP phone and a H.323 Client?

A. Phone A initiates a call to an H.323 Client C via H.323 signaling with Call Manager Z. Call Manager Z performs H.323 signaling (H.225, H.245) with H.323 Client C. Phone A and H.323 Client C stream audio directly between each other.
B. Phone A initiates a call to an H.323 Client C via H.323 signaling (H.225, H.245) with Call Manager Z. Call Manager Z performs stimulus signaling with H.323 Client C. Phone A and H.323 Client C stream audio directly between each other.
C. Phone A initiates a call to an H.323 Client C via stimulus signaling with Call Manager Z. Call Manager Z performs H.323 signaling (H.225, H.245) with H.323 Client C. Phone A and H.323 Client C stream audio directly between each other.
D. Phone A initiates a call to an H.323 Client C via stimulus signaling with Call Manager Z. Call Manager Z performs stimulus signaling with H.323 Client C. Phone A and H.323 Client C stream audio directly between each other.

Answer: C

QUESTION 142
Unity 4.0 introduces what standards-based protocol to send/receive messages to/from other Vendors Voicemail systems?

A. OctelNet
B. E&M
C. VPIM
D. AMIS-A

Answer: D