Exam : 350-030

Title : CCIE Voice Written

Ver : 02-28-07

**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.h

---

**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: C

---

**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?

A. Voice: IP Precedence 5: Signaling 3
B. Voice: DSCP AF ; Signaling : DSCP EF31
C. Voice: IP Precedence 5; Signaling 3: 0
D. Voice: DSCP EF; Signaling : DSCP EF31
E. Voice: DSCP EF; Signaling : 0

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
CoS Layer 2
http://www.cisco.com/univercd/cc/td/doc/product/software/ios120/120newft/120limit/120s/120s28/12swred.ht

m
ToS Layer 3
http://www.cisco.com/en/US/products/hw/switches/ps672/products_qanda_item09186a00800a8922.shtml

---

**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 RADIUS 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

---

## 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 on 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

---

**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: D

Explanation:
You can't use ISDN MAP on a VOIP dial-peer.
What we can do is making a translation rule, which can modify the field for the ISDN fields.
Reference: Implementing Cisco Voice Gateways and Gatekeepers Volumes 1 & 2

**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 the US 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 with only analog capability.

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? Select two.

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

Answer: A, C

Explanation:
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

---

**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

---

**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 dscp ef
class-map match-all VOICE-CONTROL
match ip dscp af31
!
policy-map WAN-EDGE
class VOICE
priority percent 33

class VOICE-CONTROL
bandwidth percent 2
class class-default
fair-queue
!
C. !
version 12.2
!
class-map match-all VOICE
match ip dscp 5
class-map match-all VOICE-CONTROL
match ip dcsp 3
!
policy-map WAN-EDGE
class VOICE
priority queue percent 33
class VOICE-CONTROL
bandwidth queue percent 2
class class-default
fair-queue
!
D. !
version 12.2
!
class-map match-all VOICE
match ip dscp 46
class-map match-all VOICE-CONTROL
match ip dscp 26
!
policy-map WAN-EDGE
class VOICE
priority 33 percent
class VOICE-CONTROL
bandwidth 2 percent
class class-default
fair-queue
!

Answer: B

---

**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
Voice Enabling the Data Network; Durkin: Cisco Press p124

---

**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: B
The answer should be 8Kbps

*Table* *Recommended Bandwidth for Call Control Traffic With and Without Signaling Encryption*

| Branch Office Size (Number of IP Phones and Gateways) | Recommended Bandwidth for Control Traffic (no encryption) | Recommended Bandwidth for Control Traffic (with encryption) |
|---|---|---|
| 1 to 10 | 8 kbps | 8 kbps |
| 20 | 8 kbps | 9 kbps |
| 30 | 8 kbps | 13 kbps |
| 40 | 11 kbps | 17 kbps |
| 50 | 14 kbps | 21 kbps |
| 60 | 16 kbps | 25 kbps |
| 70 | 19 kbps | 29 kbps |

---

**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, B

Explanation: DNIS information is sended in a setup message containing all the digits necessary to fully route a call.

---

**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: B

Explanation: Data traffic is served in a Weighted Fair Queue strategy.

---

**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: A, B

Explanation:
Not C: C is not valid because it does not exist IP ports.

---

**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. Gateway.
C. Gatekeeper.
D. All of the above.

Answer: D

**QUESTION 38:**

What is the sampling rate used by PCM as specified by Nyquist?

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

Answer: D
Cisco Voice over Frame Relay, ATM & IP; Mc Querry; Cisco Press p57

**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
G711 payload is 160 bytes and IP; UPD & RTP headers total 40 bytes

**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 from show 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

---

**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#1492881

---

**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: B

Explanation: We think that B is the correct answer because frame-relay mincir of 768000 guarantees all the voice traffic in the link due congestion (assuming Voice is correctly placed in the PQ queue)

---

**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
Deploying Cisco Voice over IP Solutions; Davidson; Cisco Press p42

---

**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. Erlanb B and Erlang C
C. Erlang-C
D. Poisson
E. Binomial
F. Engset
G. Extended Erlang-B
H. None of the above.

Answer: A

Explanation: A is the correct answer because the model with infinite sources and with no call retries (lost calls cleared) is Erlang B. Extended Erlang B (answer G theoreticaly correct) is for systems with a percentage of call retries that should be taken into account.

---

**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

---

**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

---

**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

---

**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_chapter09186a00

---

**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.
E. It restricts calls to numbers such as 1-900 and International long distance calls

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_chapter09186a00

---

## 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 types 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? Select two.

A. ISDN
B. PSTN
C. WAN
D. CMs
E. VLANs
F. LAN

Answer: B, F

---

## 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#998

---

**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_chapter09186a0

---

**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

---

**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: A
There is actually not enough information provided to give an answer but of the answers available the best choice would be that the PBX would continue to process normal telephone calls that it has control over (normal telephony would work). Calls between the two systems would not work nor would signaling.

---

**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_chapter09186a0

---

**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
Cisco Voice over Frame Relay, ATM & IP; Mc Querry; Cisco Press p43

---

**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
D. Locations bandwidth Shared media resources.
E. Device configuration replication.
F. 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

Answer: C

**QUESTION 98:**

AMIS is used to:

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

Answer: B

**QUESTION 99:**

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 100:**

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 101:**

Test Technologies and Bill, 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_chapter09186a00

---

## QUESTION 102:

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.C
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. 5551212555
B. 4441212555
C. 555911911444
D. 5551444555
E. 5554442555

Answer: A

---

## QUESTION 103:

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 104:**

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 105:**

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_example09186a0080160b

---

**QUESTION 106:**

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: A

Explanation: The correct answer is 'Line Speed' because in cisco's documentation you can see a table regarding Line speed-Frame Relay fragment relationship to achieve about 10ms of serialization delay.

---

**QUESTION 107:**

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

---

**QUESTION 108:**

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
power inlms auto
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 109:**

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: A

Explanation:
The correct answer should be 66%
G.729 20 bytes for payload and 40 bytes for IP, UDP and RTP.
60/40 = 66%

---

**QUESTION 110:**

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 111:**

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 112:**

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 113:**

Class of Service (CoS) is a:

A. Method of classifying different time periods which have the greatest call volume; assist telephone companies 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 114:**

Exhibit:

```
interface ATM4/0.60 point-to-point
 description ATM Link to BRANCH#60
 bandwidth 3000
 ip address 10.108.90.1 255.255.255.252
 pvt BRANCH#6000/60
  vbr-nrt 3000 3000
  tx-ring-limit __
  service-policy output WAN-EDGE
```

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 115:**

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 116:**

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 117:**

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 118:**

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 119:**

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 120:**

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: B

Explanation: The Nyquist theorem is all about sample rate.
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.
http://www.cisco.com/warp/public/788/signalling/waveform_coding.html
Not C: The Bellman Ford theorem has nothing to do with PCM.Bellman Ford theorem is for best path computation within a routing protocol.

---

**QUESTION 121:**

In VoIP, once TCP receives a request for opening a voice channel on port 1720, 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: B

Explanation:

The figure shows an H.323 basic call setup exchange between two gateways. The optional gatekeeper is not present in this example. Although gateways are shown, the same procedure is used when one or both endpoints are an H.323 terminal.

The flow procedure with a gatekeeper includes these steps:

1. The originating gateway initiates an H.225.0 session with the destination gateway on registered TCP port 1720. The gateway determines the IP address of the destination gateway internally; the gateway has the IP address of the destination endpoint in its configuration or it knows a DNS resolvable domain name for the destination.
2. Call setup procedures based on Q.931 create a call signaling channel between the endpoints.
3. The endpoints open another channel for the H.245 control function. The H.245 control function negotiates capabilities and exchanges logical channel descriptions.
4. The logical channel descriptions open Real-Time Transport Protocol (RTP) sessions.
5. The endpoints exchange multimedia over the RTP sessions.

---

**QUESTION 122:**

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 123:**

Click the Exhibit button.

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 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 124:**

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

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

Answer: B

---

**QUESTION 125:**

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 126:**

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 127:**

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

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

Answer: D

---

**QUESTION 128:**

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 129:**

Real-Time Transport Protocol (RTP) provides: (Choose three.)

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, D

---

**QUESTION 130:**

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? Select two

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: B, D

---

**QUESTION 131:**

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

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

Answer: B

---

**QUESTION 132:**

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 133:**

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 134:**

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 135:**

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 136:**

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 137:**

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 138:**

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 139:**

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 140:**

AMIS is used to:

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

Answer: B

---

**QUESTION 141:**

In a CallManager cluster, Intra Cluster Communications Signaling Includes: (Choose two.)

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

Answer:

---

**QUESTION 142:**

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 143:**

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

Explanation: C can also be considered.

---

**QUESTION 144:**

Which pins are used to supply Inline-Power to an IP Phone when using an Inline-Power enabled Catalyst Switch?

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

Answer: D

---

**QUESTION 145:**

Which pins are used to supply Inline-Power to an IP Phone when using a Cisco Inline-Power Patch-Panel?

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

Answer: C

---

**QUESTION 146:**

What protocol does an IP Phone use to learn the Voice VLAN ID it should use for Voice Traffic?

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

Answer: C

---

**QUESTION 147:**

What protocol does an IP Phone use to learn the IP Address of its TFTP Server?

A. HSRP
B. DHCP
C. Skinny Station Protocol
D. STP
E. CDP

Answer: B

---

**QUESTION 148:**

Exhibit

Consider the QoS configuration in the picture shown for a VoIP call across an MPLS network. If IP Phone A calls IP Phone B, how will voice and signaling packets be marked by the time they arrive at the IP Phone B? NOTE: Assume the LAN switches (and any other equipment in the cloud) do not mark or remark the packets, and the complete MPLS router QoS configuration is shown in the picture.

A. Voice: DSCP EF ; Signaling : DSCP AF31
B. Voice: DSCP EF ; Signaling : 0
B. Voice: IP Precedence 5; Signaling : 0
C. Voice: IP Precedence 5; Signaling : 3
D. Voice: 0 ; Signaling : 0

Answer: A, D

---

**QUESTION 149:**

When using the Low Latency Queuing feature of Cisco IOS:

A. All the RTP traffic is serviced by the PQ and the data traffic is serviced using the CBWFQ.
B. All the RTP and data traffic is send to the PQ and serviced according to the IP Precedence.
C. All the RTP traffic is serviced using the CBWFQ and data traffic is serviced by the PQ.
D. None of the above

Answer: D

---

**QUESTION 150:**

Network Topology Exhibit.

Assume Router Certkiller 1 has been correctly configured to allow the phone on the left to call the phone on the right using Voice over IP. If the phone on the left calls the phone on the right using an IP Precedence setting in both directions, what changes should be made to Router Certkiller 2?

A. Create a default dial-peer VoIP statement with the corresponding IP precedence on Router Certkiller 2 which will be used for all calls without matching destinations patterns
B. Create a dial-peer VoIP statement on Router Certkiller 2 with a matching destination pattern for Phone Certkiller 1's number and a corresponding IP precedence statement
C. Create a policy route map using the policy-route command on the inbound serial interface of Router Certkiller 2 that will set the IP precedence by matching the source and destination IP address of Router Certkiller 1 and Router Certkiller 2
D. Create a committed access rate using the rate-access command on the inbound serial interface of Router Certkiller 2 that will set the IP precedence by matching the source and destination IP address of Router Certkiller 1 and Router Certkiller 2
E. Use the set-reverse-direction keyword on the IP precedence line in the corresponding dial-peer on Router Certkiller 2

Answer: B

---

**QUESTION 151:**

Network topology exhibit

Consider the WAN network in the picture shown. The speed of the FR PVC connecting Router Certkiller 1 to the FR network is such that fragmentation is required when a voice call is present across the WAN. The speed of the ATM PVC connecting Router Certkiller 2 is high enough not to require fragmentation. Which technologies are involved in the QoS solution needed for this scenario.

A. FRF.12
B. FRF.8
C. FRF.5
D. MLPPP
E. FRF.16

Answer: A, B

Explanation: The correct answers are FRF.8 and FRF.12.
Not D: Not MLPPP, because fragmentation in frame-relay is considered FRF.12 standard.

---

**QUESTION 152:**

Which standards outline a fragmentation implementation to help control the delay variation when real-time traffic, such as voice, is carried on the same interface as data? (multiple answer)

A. MLPPP LFI
B. FRF.12
C. FRF.11
D. H.245
E. H.323

Answer: A, B

Explanation:
Not C: FRF.12 was added to do this fragmentation and that FRF.11 doesn't do this.
Reference: http://www.mfaforum.org/frame/Whitepaper/4017.shtml

---

**QUESTION 153:**

A T1 (1.536M) FR PVC must be configured for voice and data traffic. It is expected that voice will never require more than half of the bandwidth. What is the most appropriate FRTS configuration for this scenario?

A. map-class frame-relay FRTS-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 1536
frame-relay be 0
frame-relay mincir 1536000
C. map-class frame-relay FRTS-voice
frame-relay cir 1536000
frame-relay bc 15360
frame-relay be 1536
frame-relay mincir 1536000
D. map-class frame-relay FRTS-voice

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

Answer: E

---

**QUESTION 154:**

What is the ITU G.114 specification for one-way delay for Voice?

A. 50 ms
B. 100 ms
C. 150 ms
D. 200 ms
E. 250 ms

Answer: C

---

**QUESTION 155:**

What is MQC?

A. Modular Quality of Service Command Line Interface
B. Mean Quality Coefficient
C. Maximum Quality Carrier
D. Minimum Quality Call
E. Maximum Quality Call

Answer: A

---

**QUESTION 156:**

On average, how much Layer 3 is required for Call Control for an IP Phone?

A. 150 bps
B. 600 bps
C. 2 kbps

D. 4 kbps
E. 8 kbps

Answer: A

---

**QUESTION 157:**

The difference between Type of Service (ToS) and Class of Service (CoS) is:

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

Answer: D
CoS Layer 2
http://www.cisco.com/univercd/cc/td/doc/product/software/ios120/120newft/120limit/120s/120s28/12swred.htm
ToS Layer 3
http://www.cisco.com/en/US/products/hw/switches/ps672/products_qanda_item09186a00800a8922.shtml

---

**QUESTION 158:**

Which queueing algorithms are recommended for Voice, Call Control and generic data traffic (respectively)?

A. Priority Queuing, Custom-Queueing, Weighted-Fair Queueing
B. Low-Latency Queueing, Class-Based Weighted-Fair Queuing and Weighted-Fair Queuing
C. Low-Latency Queueing, Class-Based Weighted-Fair Queuing and Default Queuing
D. Priority Queueing, Bandwidth Queuing and Fair Queuing
E. Low-Latency Queueing, Class-Based Weighted-Fair Queuing and Fair Queuing

Answer: B

---

**QUESTION 159:**

What 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 queueing 33 percent
class VOICE-CONTROL
class-based queueing 2 percent
class class-default
weighted-fair-queue
!
B. !
version 12.2
!
class-map match-all VOICE
match ip dscp ef
class-map match-all VOICE-CONTROL
match ip dscp af31
!
policy-map WAN-EDGE
class VOICE
priority percent 33
class VOICE-CONTROL
bandwidth percent 2
class class-default
fair-queue
!
C. !
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
!
D. !

```
version 12.2
!
class-map match-all VOICE
match ip dscp 46
class-map match-all VOICE-CONTROL
match ip dscp 26
!
policy-map WAN-EDGE
class VOICE
priority queue 33 percent
class VOICE-CONTROL
bandwidth queue 2 percent
class class-default
fair-queue
!
```

Answer: B

---

**QUESTION 160:**

What is the optimal recommended interval for traffic-shaping converged Frame-Relay circuits on non-distributed platforms and how is this set?

A. 8 ms Interval set by configuring Bc to equal CIR/125
B. 8 ms Interval set by configuring Be to equal CIR/125
C. 10 ms Interval set by configuring Bc to equal CIR/100
D. 10 ms Interval set by configuring Be to equal CIR/100
E. 12.5 ms Interval set by configuring Bc to equal CIR/80
F. 12.5 ms Interval set by configuring Be to equal CIR/80

Answer: C

---

**QUESTION 161:**

How many bytes are saved per VoIP packet by enabling cRTP on a standard G.711 call?

A. 2-5 bytes per VoIP packet
B. 12-15 bytes per VoIP packet
C. 22-35 bytes per VoIP packet
D. 35-38 bytes per VoIP packet
E. 42-45 bytes per VoIP packet

Answer: D

**QUESTION 162:**

What Network Management Server (NMS) application leverages the Service Assurance Agent within IOS to gather statistics on VoIP latency, jitter and loss?

A. Resource Manager Essentials
B. Device Fault Monitor
C. Voice Health Monitor
D. Internetwork Performance Monitor
E. Quality of Service Policy Manager

Answer: D

**QUESTION 163:**

Pulse Code Modulation (PCM) sampling rate was specified by Nyquist to accurately recreate the voice signal on the opposite end. What is the sample rate used in PCM?

A. 4000 per second
B. 8000 per second
C. 16000 per second
D. 64000 per second

Answer: B

**QUESTION 164:**

What statement regarding G.729 is correct?

A. G.729 has an algorithmic delay of about 10 milliseconds. In the Cisco IOS Voice over IP (VoIP) product, the DSP generates a frame every 10 milliseconds. Each speech frame is then placed within one packet; the packets delay is, therefore, 10 milliseconds.
B. G.729 has an algorithmic delay of about 20 milliseconds. In the Cisco IOS Voice over IP product, the DSP generates a frame every 10 milliseconds. Two of these speech Frames are then placed with in one packet; the packet delay is therefore, 20 milliseconds.
C. G.729 has an algorithmic delay of about 30 milliseconds. In the Cisco IOS voice over IP product, the DSP generates a frame every 10 milliseconds. Three of these speech Frames are then placed with in one packet; the packet delay is therefore, 30 milliseconds.
D. G.729 has an algorithmic delay of about 20 milliseconds. In the Cisco IOS voice over IP product, the DSP generates a frame every 20 milliseconds. Each speech frame is then placed with in one packet; the packet delay is therefore, 20 milliseconds.

Answer: B

---

**QUESTION 165:**

Based upon Cisco's design guide, using a G.729 codec, and no header compression, what is the typical bandwidth needed for a single VoIP call (including layer 2)?

A. 8 KBps
B. 10 KBps
C. 16 KBps
D. 24 KBps
E. 32 KBps

Answer: D

---

**QUESTION 166:**

If Direct Inward Dialing (DID or DDI) is required by the customer, what signaling type can be considered for implementation to meet this requirement?

A. T1 CAS
B. Analog E&M
C. T1/E1 PRI
D. E1 R2
E. All of the above

Answer: E

---

**QUESTION 167:**

What statement is correct regarding the "Fax Relay?"

A. Fax Relay is a real-time fax over the packet network. A call starts as a voice call, and upon detecting the v.21 flags from the answering fax machine, the call is switched to fax relay. There is a dial peer configuration that is applicable to fax relay known as fax-rate.
B. Fax Relay is similar to Store & Forward Fax. A call starts as a voice call, and upon detecting the v.21 flags from the answering fax machine, the call is switched to fax relay.
C. Fax Relay is a real-time fax over the packet network. A call starts as a voice call, and upon detecting the v.90 flags from the answering fax machine, the call is switched to fax relay.
D. Fax Relay is a real-time fax over the packet network. A call starts as a voice call, and upon detecting the T.30 flags from the answering fax machine, the call is switched to fax

relay.
E. Fax Relay is an asynchronous-time fax over the packet network. A call starts as a voice call, and upon detecting the T.120 flags from the answering fax machine, the call is switched to fax relay.

Answer: A

---

**QUESTION 168:**

The most important functions of H.245 include: (multiple answer)

A. Coder/ Decoder (CODEC) type negotiation such as G.711, between the calling and called parties
B. Both sides of the call perform IP address exchange and UDP port negotiation.
C. Both sides of the call perform H.225 port negotiation.
D. Both sides of the call perform IP port negotiation.

Answer: A, B

---

**QUESTION 169:**

In VoIP, once TCP receives a request for opening a voice channel on port 1720, 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: B

---

**QUESTION 170:**

The range of UDP port numbers used in Cisco's VoIP implementation is:

A. 225 to 245
B. 16384 to 32767
C. 1718 to1720
D. 11000 to12000
E. 32768 to 64535

Answer: B

**QUESTION 171:**

Compressed Real-Time Transport Protocol (CRTP) is used on a link-by-link basis to compress the:

A. IP/UDP/RTP header from 44 bytes to 2-4 bytes most of the time
B. IP/RTP header from 44 bytes to 6-8 bytes most of the time
C. IP/UDP/RTP header from 40 bytes to 2-4 bytes most of the time
D. IP/UDP header from 66 bytes to 2-4 bytes most of the time
E. All of the above as it depends on the application type

Answer: C

**QUESTION 172:**

H .225 utilizes a scaled-down version of what protocol that is used to set up the connection between two H.323 endpoints?

A. Q.931
B. Q.Sig
C. SS7
D. Frame Relay SVC signaling
E. ATM UNI signaling

Answer: A

**QUESTION 173:**

What standard defines the supplementary service for ISDN?

A. Q.931
B. Q.822
C. Q.932
D. Q.930
E. Q.742

Answer: C

**QUESTION 174:**

What issue is most commonly encountered with an analog Foreign Exchange Office

(FXO) loopstart port connection to PBX?

A. Disconnect supervision
B. Battery reversal
C. Busy tone
D. On-hook and off-hook issues

Answer: A

---

**QUESTION 175:**

What interface does SMDI traditionally use?

A. Serial
B. Parallel
C. Ethernet
D. Firewire
E. USB

Answer: A

---

**QUESTION 176:**

AAA Can NOT be used for: (multiple answer)

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

Answer: E, F

---

**QUESTION 177:**

A voice gateway processing 100 calls in the busy hour averaging six minutes each would be equivalent to:

A. 100 Erlangs
B. 360 CCS (call centum seconds)
C. 10 Erlangs

D. 60 Erlangs
E. B and C

Answer: E

---

**QUESTION 178:**

In an IP Contact Center deployment, the Erlang-B Traffic Model is used to provision the following resources:

A. Agents receiving/handling inbound calls
B. Ports on a voice gateway interfacing to the PSTN
C. Ports on an IP-IVR interfacing with Cisco CallManager
D. B and C
E. None of the above

Answer: D

---

**QUESTION 179:**

In a call center deployment, busy hour traffic for voice gateway port/trunk is based upon:

A. Agent talk time (the time agent spends talking to a caller)
B. Agent after call work time (AKA "agent wrap up time")
C. Queue time (the time caller spends waiting in queue waiting for an agent to become available)
D. A and C
E. All of the above

Answer: D

---

**QUESTION 180:**

A Cisco H.323 gatekeeper can resolve an address using:

A. An H.323 ID
B. An E.164 address
C. An Email-ID
D. A URL
E. Any of the above

Answer: B

Explanation: The correct answer is B because the resolution of a H.323 Gateway is e.164-to-IP address.

---

**QUESTION 181:**

A Registration Request (RRQ) Registration, Admission, and Status (RAS) message is NOT sent by what endpoint?

A. H.323 Gateway
B. Gatekeeper
C. H.323 Terminal
D. Proxy

Answer: B

---

**QUESTION 182:**

In a remote office in a CM network, which types of call processing functions do SRST preserve?

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

Answer: A, D, E

Explanation:
D: SRST preserve IP-to-IP calls (local calls), IP-to-PSTN calls and global Voicemail calls.

---

**QUESTION 183:**

A user at Phone Certkiller 1 finishes a call. Later, he notes that "CM Fallback Service Operating" is displayed on Phone Certkiller 1. Which are possible explanations for this?

A. The TCP connection between phone Certkiller 1 and call manager has been disrupted.
B. Remote-GW has not received any messages from CallManager within the appropriate timeout period.
C. The FE on Remote-GW is out of service.
D. The FXO port on Remote-GW is out of service.
E. The FE on HQ-GW is out of service.

Answer: A, B, C E

---

**QUESTION 184:**

Exhibit:

In the diagram shown, what section of the voice path represents the Tail Circuit?

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

Answer: D

---

**QUESTION 185:**

When troubleshooting a FailSafe problem in Unity, the first place you should look for detailed error messages is the:

A. tempu.log
B. System Log
C. Application Log
D. SDL Trace
E. Status Monitor

Answer: C

---

**QUESTION 186:**

PRI is the preferred method for inter-connecting CallManager 3.2 and below to

PBX's because:

A. It is the cheapest solution available
B. It offers the highest level of inter-operability currently available between CallManager and PBX's
C. It allows a customer to share their existing Voicemail system with CallManager subscribers whilst delivering full functionality
D. Caller ID is available

Answer: A, B

Explanation: A would be more correct than D.

---

**QUESTION 187:**

In an AVVID environment, where admission control is handled through the configuration of locations, a g711 call reserves the following amount of bandwidth:

A. 100
B. 64
C. 24
D. 80

Answer: D

---

**QUESTION 188:**

In a distributed AVVID call-processing model, an IOS gatekeeper is used for call Admission Control. What function does the IOS gatekeeper perform?

A. The gatekeeper will send an ARQ if there is enough available bandwidth.
B. The gatekeeper will send an ACF if there is enough available bandwidth.
C. The gatekeeper will send an LRQ message to another gatekeeper if there is not enough bandwidth.
D. The gatekeeper will fall back to the PSTN is there is not enough bandwidth.

Answer: B

---

**QUESTION 189:**

In AVVID architecture, what happens if the TFTP server goes down? Select two.

A. All the phones will un-register from the CallManager because they cannot download

their configuration.
B. If a phone is reset, it will fail to register again.
C. New phones plugged to the network will fail to register.
D. All the existing phones in the network will stay operational.

Answer: C, D

---

**QUESTION 190:**

How often, by default, is a keepalive sent between an IP Phone and its active Call Manager?:

A. 10 seconds and it can be adjusted in the call manager configuration
B. 10 seconds and it can not be adjusted
C. 60 seconds and it can be adjusted in the call manager configuration
D. 30 seconds and it can be adjusted in the call manager configuration
E. 60 seconds and it can not be adjusted

Answer: D

---

**QUESTION 191:**

Is it possible to disable silence suppression on a per IP phone basis?

A. Yes, there is a checkbox under the phone configuration.
B. No, the only configuration is a global one.
C. Yes, in the device pool configuration for the phones.
D. No, silence suppression is not necessary for phones.

Answer: B

---

**QUESTION 192:**

The IP phone uses:

A. TCP/IP and UPD/IP to and from the call Manager in order to Tx/Rx stimulus; RTP/UDP/IP to and from the IP phones or H.323 terminals for audio.
B. UPD/IP to and from the call Manager in order to Tx/Rx stimulus; RTP/IP to and from the IP phones for audio.
C. TCP/IP to and from the call Manager in order to Tx/Rx stimulus; RTP/UPD/IP to and from the IP phones for audio, video and data.
D. TCP/IP to and from the call Manager in order to Tx/Rx stimulus; RTP/UPD/IP to and from the IP phones for audio.

Answer: D

---

**QUESTION 193:**

Gatekeeper call admission control is a policy-based scheme. What statements concerning it are true?

A. It requires static configuration of available resources.
B. Is not aware of the network topology.
C. It is not necessary to restrict gatekeeper call admission control schemes to hub-and-spoke topologies.
D. It is aware of the network topology.

Answer: A, B

---

**QUESTION 194:**

What interface does the Cisco CRS Engine use to communicate with Cisco CallManager, regardless of release version?

A. TAPI (Telephony Application Programming Interface)
B. CAPI (Common ISDN Application Programming Interface)
C. JTAPI (Java Telephony Application Programming Interface)
D. HSSI (High Speed Serial Interface)

Answer: C

---

**QUESTION 195:**

When provisioning Cisco CallManager and IPCC Express (CRS), what IPCC Express agent provisioning configurations are possible?

A. One pool of agents shared among multiple scripts
B. One pool of CTI ports shared among multiple scripts
C. NxN mesh of agents and ports shared among N scripts
D. All of the above

Answer: D

---

**QUESTION 196:**

In a Cisco IPCC deployment, an ICM routing client is anything that can generate a route request to the ICM Central Controller. Routing clients include:

A. Cisco CallManager
B. Cisco IP-IVR using the CRS platform
C. PSTN
D. All of the above

Answer: D

---

**QUESTION 197:**

In a Cisco IPCC deployment, the CallManager communicates route requests to the ICM Central Controller via:

A. A Peripheral Gateway
B. A voice Gateway
C. A router
D. PSTN
E. None of the above

Answer: A

---

**QUESTION 198:**

When deploying multiple Unity-Bridge's what is true?

A. Each Unity-Bridge requires a dedicated Unity server.
B. Multiple Unity-Bridges can be connected to one Unity server acting as a "bridge-head."
C. All Unity-Bridge's must be connected directly to the customers MS Exchange network.
D. None of the above

Answer: A

---

**QUESTION 199:**

Consider phones A and B. Both phones are registered in the same cluster. Phone A is configured with extension 1000. Phone B is configured with extension 2000. Indicate what choice below is necessary and sufficient to allow phone A to be able to call phone B AND phone B to be able to call phone A

A. Both phone extensions are in the same partition
B. Both phones are assigned the same Calling Search Space
C. Both (A) and (B)
D. None of the above

Answer:

---

**QUESTION 200:**

What command enables cRTP?

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

Answer: A

Explanation: A is the correct because it compresses IP/UDP/RTP headers
Not D: Option D compresses IP/TCP/RTP and RTP uses UDP transport protocol

---

**QUESTION 201:**

When dimensioning call center agents receive calls from infinite sources (PSTN callers) where calls are queued during the busy hour, the traffic model typically used is:

A. Extended Erlang-B
B. Engset
C. Erlang-C
D. Binomial
E. None of the above

Answer:A

---

**QUESTION 202:**

Many types of devices can register with a Cisco CallManager Examples are IP phones, voice mail ports, CTI (TAPI/JTAPI) devices, gateways, and DSP resources such as transcoding and conferencing A weight is assigned for each of these devices when provisioning CallManager based upon

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

Answer:

---

**QUESTION 203:**

From the perspective of the callManager, the Unity TSP looks and behaves most like a:

A. H.232 Gateway
B. CTI Port
C. Cissco IP Phone
D. TAPI Device
E. MGCP Gateway

Answer: D

---

**QUESTION 204:**

In Cisco IP Contact Center solution(IPCC), what is in charge of agent state management, selection and reservation?

A. Voice Gateway
B. CallManager
C. ICM Central Controller
D. None of the above

Answer: C

---

**QUESTION 205:**

What command will guarantee a maximum serialization delay of 10 ms on a converged 512 Kbps MLP circuit?

A. ppp multilink fragment 960
B. ppp multilink fragment 320
C. ppp multilink fragment 640
ppp multilink interleave
D. ppp multilink fragment-delay 10
ppp multilink interleave

E. ppp multilink fragment-delay 10

Answer: D

---

**QUESTION 206:**

What is the proper configuration for VoIP authentication via Authentication, Authorization, and Accounting (AAA)?

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

Answer: C

---

**QUESTION 207:**

A Calling Search Space can be used by CallManager to:

A. Enable the use of an overlapping dial plan
B. All of the above
C. Provide access-list-like security
D. Restrict calls to numbers such as 1-900 and International long distance calls
E. Enable the use of E911 services in a Centralized Call Processing model

Answer: B

---

**QUESTION 208:**

In order to pass hook-flash on h.323 from FXS to FXO:

A. connection plar must be configured on the voice-port (FXS)
B. connection trunk must be configured on the voice-port (FXS) and (FXO)
C. None of the above
D. connection plar must be configured on the voice-port (FXS) and (FXO)
E. connection trunk must be configured on the voice-port (FXS)

Answer: C

---

**QUESTION 209:**

A user at Phone A finishes a call. Later, he notes that "CM Fallback Service Operating" is displayed on Phone
A. Which are possible explanations for this?

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

Answer: B,D

---

**QUESTION 210:**

What standards-based protocol will allow CallManager to seamlessly Integrate with other vendors' traditional PBX systems?

A. QSIG
B. All of the above
C. MGCP
D. PRI NI-2

Answer: A

---

**QUESTION 211:**

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 Skinny (SCCP) traffic be seen?

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

Answer: B

---

**QUESTION 212:**

A Cisco SIP Proxy Server can make routing decisions based upon which criteria?

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

Answer: C

---

**QUESTION 213:**

Which are possible reasons when a user hears echoes of her own voice? (multiple answer)

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

Answer: A,B

**QUESTION 214:**

A Calling Search Space can be used by CallManager to:

A. Enable the use of an overlapping dial plan
B. Provide access-list-like security
C. Enable the use of E911 services in a Centralized Call Processing model
D. Restrict calls to numbers such as 1-900 and International long distance calls
E. All of the above

Answer: E

**QUESTION 215:**

Click the Exhibit button.
In the figure shown, HSRP is used in conjunction with SRST to preserve telephony functionality in a branch office. Consider a situation where a WAN failure occurs while router A (the primary router) is used. Router A switches to SRST mode to preserve telephony functions. At this point Router A fails, and HSRP backup Router B becomes the active router for the branch office, taking over SRST and routing functions for the office. For Router B to be effective in running SRST for the branch, which types of physical connectivity must be duplicated on Routers A and B?

A. WAN
B. VLANs
C. CMs
D. PSTN
E. LAN

Answer: D,E

**QUESTION 216:**

A voice gateway is receiving calls from infinite sources (PSTN callers) during the busy hour where lost calls are cleared (blocked). The traffic model typically used to dimension the number of gateway ports/trunks required is:

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

Answer: A

---

**QUESTION 217:**

Which standards are related to echo in a network.

A. 323
B. 165
C. 174
D. 168
E. 711

Answer: B, D

---

**QUESTION 218:**

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 q.921 traffic be seen?

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

Answer: A

---

**QUESTION 219:**

The maximum device weight capacity a Cisco MCS server can have does NOT depend upon:

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

Answer: C

---

**QUESTION 220:**

What type of signaling provides Dialed Number Information Service (DNIS) on a T1/E1?

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

Answer: B

---

**QUESTION 221:**

The user at phone A dials 5551212555. What Digit string is sent to the PSTN for termination assuming call routing is working properly through the IP Network? NOTE: There are 2 exhibits for this question.

```
Remote-GW
~~~~
voice translation-profile CCIE_1
translate called 1
voice translation-rule 1
rule 1 /^\(555\)+\(.*\)/ /444\2/ type any national plan any isdn
dial-peer voice  type
translation-profile outgoing CCIE_
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 CCIE_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 CCIE_2
Port 1/0:23
```

A. 5551444555
B. 5551212555
C. 4441212555
D. 555911911444
E. 5554442555

Answer: C

---

**QUESTION 222:**

Consider phones A and B. Both phones are registered in the same cluster. Phone A is configured with extension 1000. Phone B is configured with extension 2000. Indicate what choice below is necessary and sufficient to allow phone A to be able to call phone B AND phone B to be able to call phone A.

A. Both (A) and (B)
B. Both phones are assigned the same Calling Search Space
C. None of the above
D. Both phone extensions are in the same partition

Answer: C

---

**QUESTION 223:**

During the busy hour, 100 Erlangs may be generated by:

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

Answer: E

---

**QUESTION 224:**

AAA Can be used for: (multiple answer)

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

Answer: A,E,G

---

**QUESTION 225:**

In a remote office in a CM network, which types of call processing functions do SRST preserve?

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

Answer: B,D

---

**QUESTION 226:**

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

A. E1 CAS E&M Signaling

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

Answer: E

---

**QUESTION 227:**

AAA Can be used for: (multiple answer)

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

Answer: B,E,F

---

**QUESTION 228:**

PRI is the preferred method for inter-connecting CallManager 3.2 and below to PBX's because:

A. It is the cheapest solution available
B. It offers the highest level of inter-operability currently available between CallManager and PBX's
C. It allows a customer to share their existing Voicemail system with CallManager subscribers whilst delivering full functionality
D. Caller ID is available

Answer: B,D

---

**QUESTION 229:**

An H.323 proxy Gatekeeper Request (GRQ) Registration, Admission, and Status (RAS) message is sent by which endpoints? (multiple answer)

A. Gateway
B. Proxy
C. H.323 Terminal

D. Gatekeeper

Answer: A,C

---

**QUESTION 230:**

An AS5300 is configured to authenticate a user for Authentication, Authorization, and Accounting (AAA) RADIUS server by prompting the user for a PIN number, etc., by using application clidauthencollect. Users are dialing 5551000. What is the correct configuration?

A. dial-peer voice 1 pots
incoming called-number 555
destination-pattern 1 .
port 0:D
application clidauthencollect
B. dial-peer voice 1 pots
destination-pattern 5551
port 0:D
application clidauthencollect
C. dial-peer voice 1 pots
destination-pattern 1.
port 0:D
application clidauthencollect
D. dial-peer voice 1 pots
incoming called-number 5551000
destination-pattern 1 .
application clidauthencollect

Answer: A

---

**QUESTION 231:**

What does this SMDI packet represent?
MD0010013D 0002914

A. MWI OFF command for extension 2914
B. A "Call Forward No Answer" extension 2914 from extension 10013
C. Extension 2914 calling into voicemail on port 13
D. A "Forward All Calls", extension 10013 calling 2914
E. MWI ON command for extension 2914

Answer: C

---

**QUESTION 232:**

In CME, which command/debug will show MAC Address, IP Address, DN and Phone Model of phones that are registered/while registering? (Choose 2)

A. debug ephone detail
B. show ephone-dn registered
C. show ephone
D. debug ephone register
E. show ephone-dn

Answer: C,D

---

**QUESTION 233:**

There are 2 remote sites and one main site. Each site has a CME router with many IP phones in an IPT deployment. The Network Administrator wants to provide all of the phones voicemail access using CUE. Which way can CUE be deployed?

A. Cisco Unity Express and the CME gateway at each site may NOT be collocated in the same router chassis providing a voicemail access to local IP phones registered to local CME.
B. One Cisco Unity Express can be used at the main site with CME router providing voicemail access to all the 3 sites.
C. Cisco Unity Express and the CME gateway at each site must be collocated in the same router chassis providing voicemail access to local IP phones registered to local CME.
D. One Cisco Unity Express with CME can be used at the main site to provide voicemail access to the IP phones at the main site. Another Cisco Unity Express with CME can be used at one of the remote sites to provide voicemail access for all of the IP phones at the two remote sites.

Answer: C

---

**QUESTION 234:**

Refer to the exhibit. While debugging a problem on SIP network, one of the messages displayed when the debug ccsip messages command is entered is shown in the exhibit. What information will the server return to the caller?

```
May  9 02:12:06 370:
Send
 SIP/2.0 415 Unsupported media type
Via: SIP/2.0/UDP  10.0.1.3:54475
From: sip:7300001@10.0.1.3
To: <sip:7900001@10.0.2.3;user=phone>;tag=9B9374-18BE
Date: Tue, 09 May 1993 02:13:06 UTC
Call-ID: 7B8D93F2-6EE5001E-0-9BCB54@10.0.1.3
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Content-Type: application/sdp
CSeq: 100 INVITE
Content-Length: 442
```

A. A list of acceptable formats
B. An acceptable language code
C. The acceptable media type
D. A correct directory number
E. A list of acceptable media types

Answer: A

---

**QUESTION 235:**

Calculate the percentage of overall bandwidth saved (at Layer 3) by cRTP for a G.711 VoIP call packetized at 50 pps.

A. Approximately 50%
B. Approximately 60%
C. Approximately 30%
D. Approximately 40%
E. Approximately 20%

Answer: E

---

**QUESTION 236:**

Suppose your business has a need to send and receive real-time fax over a VoIP network. Which fax protocol(s) could you implement? (Choose 2)

A. T.37 Fax Over IP
B. MGCP
C. T.38 Fax-relay
D. Fax Passthrough

Answer: C,D

---

**QUESTION 237:**

Which ports must be opened on an IOS firewall to allow successful H.225 RAS (Registration, Admission, and Status) message exchanges between an IOS gatekeeper and IOS H.323 gateways?

A. UDP 1719 and UDP 1720
B. TCP 1718 and TCP 1720
C. UDP 1718 and TCP 1719
D. UDP 1718 and UDP 1719
E. TCP 1718 and TCP 1719

Answer: D

---

**QUESTION 238:**

Design a two-rate, three-color marker compliant with RFC 2597, RFC 2698 and Cisco best-practices to police Bulk traffic (marked to AF11) to a conforming rate of 5 Mbps and a violating rate of 10 Mbps. Which if the following policy maps meet this criteria?

A. policy-map BULK-2RATE-3COLOP-MARKER
class BULK
police conform-rate 5000000 violate-rate 10000000
conform-action transmit
exceed-action set-dscp-transmit af12
violate-action set-dscp-transmit af13
B. policy-map BULK-2RATE-3COLOR-MARKER
class BULK
police cir 5000000 burst 5000000
conform-action transmit
exceed-action set-dscp-transmit af12
violate-action drop
C. policy-map BULK-2RATE-3 COLOR-MARKER
class BULK
police cir 5000000 pir 10000000
conform-action transmit
exceed-action set-dscp-transmit 12
violate-action set-dscp-transmit 14
D. policy-map BULK-2RATE-3COLOR-MARKER
class BULK
police cir 5000000 pir 10000000
conform-action set-dscp-transmit 11
exceed-action set-dscp-transmit 12
violate-action set-dscp-transmit 13
E. policy-map BULK-2RATE-3COLOR-MARKER
class BULK

police cir 5000000 be 5000
conform-action set-dsc-transmit af11
exceed-action set-dscp-transmit af12
violate-action set-dscp-transmit af13

Answer: C

---

**QUESTION 239:**

An IP Phone configured in voice VLAN 128 is registered to a CM cluster. It has speech connection established with another IP Phone which is connected to another CM cluster. Configured codec on both phones is G.711, 20 ms Sample. A PC is connected to back of an IP Phone which is running Sniffer Program and collecting packets coming out of the IP Phone. What would be data frame size coming out of the phone as seen in the Sniffer?

A. 200 bytes
B. 218 bytes
C. 160 bytes
D. 64 bytes
E. 128 bytes
F. 204 bytes

Answer: B

---

**QUESTION 240:**

Which method could be used to determine if there is a JTAPI memory leak in a CallManager server?

A. Check for changes to IP phone settings, like ring settings reverting to default values
B. Check for an increasing number of fast busys when dialing to the PSTN
C. Review all CCM User logs
D. Look at the physical memory available of the server.
E. Determine if dialing the voice mail pilot number fails to connect to voice mail

Answer: D

---

**QUESTION 241:**

Refer to the shown output. Calls to the PSTN are being rejected. Which of the following will correct this issue?

```
Mar  8 09:43:15.383: ISDN Se2/0:23 OSBT: TX >SETUP pd= 8  callref = 0x1B5C
      Bearer Capability i = 0x8090A2
            Standard = CCITT
            TransferCapability = Speech
            Transfer Mode = Circuit
            Transfer Rate = 64 kbit/s
      Channel ID i = 0xA98397
            Exclusive, Channel 23
      Calling Party Number i = 0x2181, '4085552048'
            Plan:ISDN, Type:CallManager
      Called Party Number i = 0x89 '01144123<569999'
            Plan:Private, Type:CallManager
*Mar  8 09:43:15.451: ISDN Se2/0:23 Q931: RX <- RELEASE_COMP pd = 8  callref = 0x9B5C
      Cause i = 0x80BF - Service/option not available, unspecified
```

A. Change the ISDN Numbering Plan type to International
B. Add the command 'isdn contiguous-bchan' to the serial interface
C. Change the channel selection order to ascending
D. Add the command 'isdn negotiate-bchan' to the serial interface

Answer: A

---

**QUESTION 242:**

On authentication, if the login profile matches the login device (that is, the user has a user device profile that is configured for a Cisco IP Phone Model 7960 and logs into a Cisco IP Phone Model 7960), how does Cisco CallManager Extension Mobility behave? (Choose 2)

A. The system uses the device profile default for that phone model for phone template and softkey template configuration and, if the phone can support addon modules, for the addon module.
B. If the phone model supports Cisco IP Phone Services and they are configured, the services get copied from the user device profile.
C. The phone automatically reconfigures with the individual user device profile information
D. The user can access all of the services that are configured on their device profile.
E. The system copies all device-independent configurations (that is, user hold audio hold audio source, user locale, userid, speed dials, and directory number configuration except for the setting "line setting for this device") from the user device profile to the login device.

Answer: C,D

---

**QUESTION 243:**

Choose which one of these, that is NOT a valid CallManager service parameter:

A. Cisco Database layer Monitor
B. Cisco Webdialer
C. Cisco RTMT Data Collector
D. Cisco Serviceability Reporter
E. Cisco Telephony Call Dispatcher
F. Cisco IP Manager Assistant

Answer: C

---

**QUESTION 244:**

When using RSVP to dynamically set up end-to-end QoS across a heterogeneous network there are several QoS Levels that can be configured on the Originating and Terminating Gateway. Which of the following statements correctly describe the difference between these three QoS Levels: Controlled-Load, Guaranteed-delay and Best-effort? (Choose 3)

A. With Guaranteed-delay QoS Level; if synchronized RSVP is attempted and fails, if acceptable QoS on either gateway is anything other than best effort, the call is released.
B. With Controlled-load QoS Level; if synchronized RSVP is attempted and fails, the call is released.
C. With Guaranteed-delay QoS Level; if synchronized RSVP is attempted and fails, if acceptable QoS on the terminating gateway is Controlled-Load or Guaranteed-delay, the call receives best-effort service.
D. With Controlled-load QoS Level; if synchronized RSVP is attempted and fails, the call receives best-effort service.
E. With Best-effort QoS Level; no RSVP synchronization is attempted and the call receives best-effort service.

Answer: A,B,E

---

**QUESTION 245:**

Refer to the exhibit. Two ports on a 3600 gateway platform are stuck in the EM_PARK state. What workaround can be configured on the router to help alleviate this situation?

```
CertKiller3 # show voice call summary
  PORT       CODEC  VAD VTSP STATE VPM STATE
  =====      ====== ============== ==========
  1/0:0.1      -      -      -      EM_ONHOOK
  1/0:0.2      -      -      -      EM_PARK
  1/0:0.3      -      -      -      EM_PARK
  1/0 0.4      -      -      -      EM_ONHOOK
  1/0:0.5      -      -      -      EM ONHOOK
```

A. A call park watchdog timer
B. Fake Answer
C. Configure the interface for wink-start signaling
D. A GetDigit Timeout value

Answer: B

---

**QUESTION 246:**

An IP Phone appears to be registered to the Cisco CME system in "show ephone registered". However, it can not make any outgoing calls, nor can it receive any incoming calls. When you pick up the handset or press the speaker button of the IP phone, you do not get a dial tone. Which would be LEAST helpful in solving the problem?

A. Ensure that the IP Phone is registered to the right CME system by pressing Configuration -> Network Configuration -> TFTP server on the phone
B. Enure that the IP Phone shows at least an extension assigned to any buttons
C. Ensure that a button and ephone-dn association is shown in "show ephone register"
D. Ensure that the IP Phone is set with the right time/date information
E. Ensure that the IP Phone is set to use the right CME system as the TFTP server
F. Ensure that a 'button' command exists to associate a button with a configured ephone-dn in ephone configuration section. Add the 'button' command if needed

Answer: D

---

**QUESTION 247:**

To hide its identity when initiating calls, Phone B requests that Server B place its calls for it. What kind of device is Server B?

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

Answer: E

---

**QUESTION 248:**

Refer to the exhibit. While debugging a problem on SIP network, one of the messages displayed when the debug ccsip messages command is entered is shown in the exhibit. What information will the server return to the caller?

```
May  9 02:12:06 370:
Send
 SIP/2.0 415 Unsupported media type
Via: SIP/2.0/UDP  10.0.1.3:54475
From: sip:7300001@10.0.1.3
To: <sip:7900001@10.0.2.3;user=phone>;tag=9B9374-18BE
Date: Tue, 09 May 1993 02:13:06 UTC
Call-ID: 7B8D93F2-6EE5001E-0-9BCB54@10.0.1.3
Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Content-Type: application/sdp
CSeq: 100 INVITE
Content-Length: 442
```

A. A list of acceptable media types
B. A correct directory number
C. A list of acceptable formats
D. An acceptable language code
E. The acceptable media type

Answer: C

---

**QUESTION 249:**

A DPA 7630 is being used to integrate a CallManager cluster with an Octel voicemail system. What is configured in the CallManager cluster to establish call paths to the DPA 7630?

A. A CTI route point is added with a DN for each call path.
B. A 7902 IP phone is configured for each call path.
C. A VIP 30 IP phone is configured for each call path.
D. A voicemail port is added for each call path.
E. A CTI port is configured for each call path.

Answer: C

---

**QUESTION 250:**

In an Intra Cluster Call Flow Trace the following Skinny Call States could be observed. Which of the following is NOT a valid Skinny Call state for an IP Phone?

A. Off Hook
B. Call Park
C. Call Transfer
D. Call Routing
E. Connected
F. Call Waiting

Answer: D

---

**QUESTION 251:**

When implementing IP Communicator in CallManager, which two steps need to be accomplished in order for the application to operate correctly? (Choose 2)

A. Verify CallManager version
B. Create and associate a user ID with the communicator device
C. Configure MAC address from Ethernet interface on the PC where IP communicator will be installed in Phone Configuration screen in CallManager
D. Configure a new CTI port to connect IP Communicator to CallManager
E. Associate directory number from desk phone with IP Communicator software

Answer: A,C

---

**QUESTION 252:**

Which function does CTI Manager provide for a CallManager cluster?

A. When a Cisco TAPI application fails, the CTI Manager closes the provider and calls at JTAPI ports and route points that have not yet been terminated get redirected to the Call

Forward On Failure (CFF) number that has been configured for them.
B. When a Cisco CallManager node fails, the CTI Manager recovers the affected JTAPI ports and route points.
C. The CTI Manager and Cisco TSP provide TAPI applications access to CallManager resources and functionality without being aware of a specific CallManager.
D. The CTI Manager and Cisco TSP provide both TAPI and JTAPI application access to specific CallManager servers in a cluster.

Answer: C

---

**QUESTION 253:**

A POTS (PSTN/PBX) user places a call (through Cisco router/gateways) and does not hear a ringback tone before the call is answered. Which of the following actions would probably correct the situation?

A. Ensure that the terminating gateway has not dropped the Progress Indicator signal
B. Configure the originating gateway to respond PI=1 to force ringback
C. Ensure that all intermediate devices carry the correct Progress Indicator without modification
D. Configure the terminating gateway to send a PI=8 in the Alert message
E. Configure all intermediate devices to repeat the Progress Indicator signal

Answer: D

---

**QUESTION 254:**

Calculate the percentage of overall bandwidth saved (at Layer 3) by cRTP for a G.729 VoIP call packetized at 50 pps.

A. Approximately 50%
B. Approximately 40%
C. Approximately 30%
D. Approximately 60%
E. Approximately 20%

Answer: D

---

**QUESTION 255:**

In a centralized call processing environment there are three sites, HQ, Site A and Site B that connect over an IP WAN. Each site will use G.711 internally and G.729 between sites. Which two tasks need to be completed to support locations for this solution? (Choose 2)

A. Two regions will need to be configured for each site, one for the internal codec type

and one for the IP WAN codec.
B. Each site will require a device pool that specifies the site name in the region.
C. One region will be required for each site.
D. All sites will become a member of a single region with specific device pools for each site.
E. Each site will require a region that assigns all devices to a device pool.

Answer: B,C

---

**QUESTION 256:**

Which Service Parameter would you select to change the TCP ports used for collecting real-time information within a CallManager cluster?

A. Cisco Extended Functions service
B. Cisco Database Layer Monitor service
C. Cisco CallManager service
D. Cisco RIS Data Collector service
E. Cisco Serviceability Reporter service

Answer: D

---

**QUESTION 257:**

What is the relationship between a device pool and a region in CallManager configuration?

A. The device pool sets the default codec for the devices associated with it in a specific region and region sets only the codec that is used between other regions.
B. Devices use the settings in the region as the default pool for that region.
C. The device pool sets the intra-region and inter-region codecs and the region makes the association between it and other regions.
D. Devices acquire a region setting from the device pool to which they are assigned.

Answer: D