350-030

TestKing's CCIE® Voice Written

Version 3.0

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QUESTION NO: 1
Which are the three elements to MQC?

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

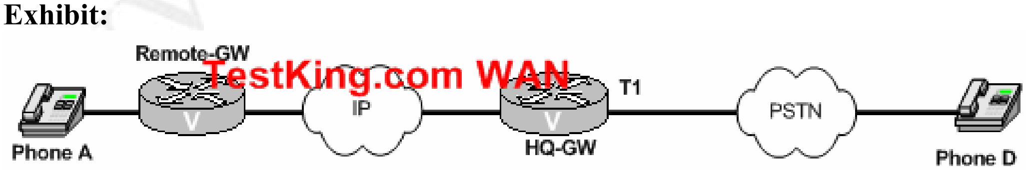
Answer: D

QUESTION NO: 2
The default fax relay connection rate is:

A. 4800 bps
B. 9600 bps
C. 14400 bps
D. 7200 bps

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

QUESTION NO: 3
Exhibit:



In the figure shown, the user as Analog phone A places a call to the user at Phone D. This call is rejected and the issue is troubleshot down to a mismatch of ISDN Plan and Type that the PSTN is receiving. The PSTN is expecting Plan: Unknown, Type: Unknown while the GW is sending Plan: ISDN Type: Unknown.
Which methods will resolve this mismatch in the networks?

A. Outgoing translation rule on the VOIP dialpeer on HQ-Gw

B. Use the "isdn map" command on the POTS dialpeer on HQ-Gw
C. Use the "isdn map" command on the VOIP dialpeer on HQ-Gw
D. Outgoing translation rule on the POTS dialer on Remote-Gw
E. Use the "isdn map" command on the VOIP dialpeer on Remote-Gw

Answer: C (????)

isdn map

To override the default ISDN type and plan generated by the router with custom values, use the **isdn map** command in interface configuration mode. To revert to the default ISDN type and plan, use the **no** form of this command.

isdn map {**address** address | ***regexp*** | **plan** plan | **type** type}
no isdn map {**address** address | ***regexp*** | **plan** plan | **type** type}

Syntax Description

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

QUESTION NO: 4
In the 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 plar** must be configured on the voice-port (FXS) and (FXO)
C. **connection trunk** must be configured on the voice-port (FXS)
D. **connection trunk** must be configured on the voice-port (FXS) and (FXO)
E. None of the above

Answer: B
IP TELEPHONY / VOIP, Configuring Hookflash Relay on FXS/FXO Voice Ports
http://www.cisco.com/en/US/tech/tk652/tk701/technologies_configuration_example09186a008009431b.shtml

QUESTION NO: 5
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 NO: 6
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: B
CIPT Solution Reference Network Design 6-3

QUESTION NO: 7
What is considered a node in a H.323 network?

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

Answer: D

QUESTION NO: 8
A Centralized Automatic Message Accounting (CAMA) trunk allows enterprise voice GW connectivity to the North American emergency (911) services of the PSTN. How does CAMA trunk signaling differ from FXO trunk signaling?

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

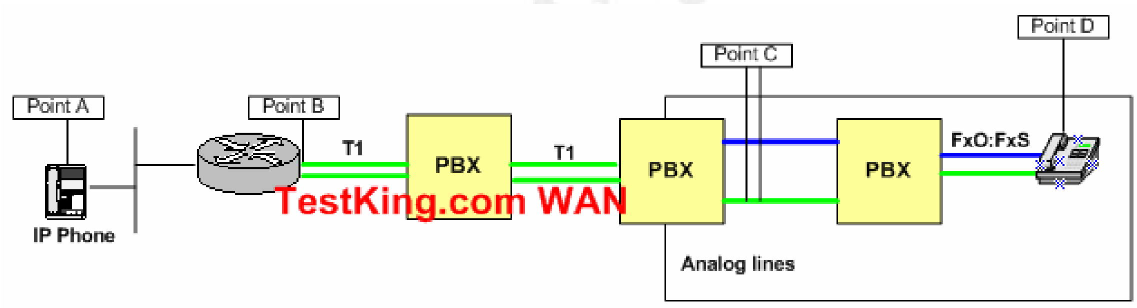
Answer: C
CIPT Solution Reference Network Design 6-3

QUESTION NO: 9
An H.323 proxy Gatekeeper Request (GRQ) Registration, Admission, and Status (RAS) message is sent by which endpoints? (Multiple answer)

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

Answer: A, C, D

QUESTION NO: 10
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: B

QUESTION NO: 11
Exhibit:

An ISP offers debit card calling from the US to countries around the world where there is only FXO connectivity to the PSTN. It is important to the ISP that customers not be charged for the call unless the call successfully reached the called party in the PSTN. What design requirement of the terminating FXO voice GW will achieve this ISP's requirements?

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

Answer: B

QUESTION NO: 12 (echo)
What is NOT a primary cause of echo in a voice network?

A. 4 wire to 2 wire Hybrids
B. Packet Loss In the IP Network
C. Delay in the IP Network
D. Acoustical Reflections

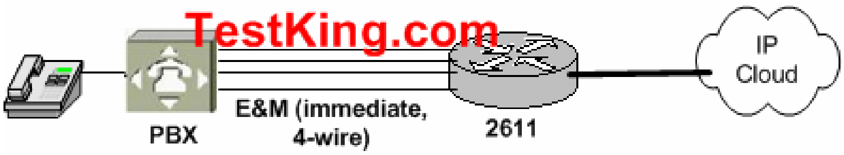
Answer: B

QUESTION NO: 13 (SMDI)
Select the valid SMDI packet:

A. ND0010001A0002222 00012324
B. RD0010001B0002222 00012324
C. MC0010001D0002222 00012324
D. MVI010001N0002222 000112324
E. MD0010001A0002222 000112324

Answer: E
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 NO: 14
Exhibit:



What should an administrator do if the PBX does not receive the initial few digits from the IP side of the 2611?

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

Answer: B
TELEPHONY SIGNALING, Understanding and Troubleshooting Analog E&M Start Dial Supervision Signaling
http://www.cisco.com/en/US/tech/tk652/tk653/technologies_tech_note09186a0080093f61.shtml

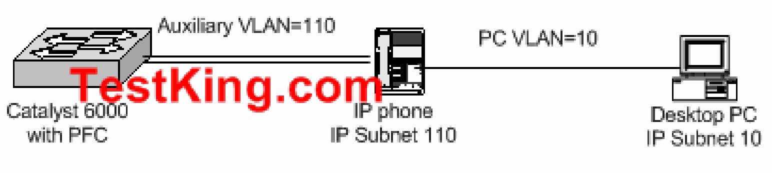
QUESTION NO: 15
When troubleshooting a FailSafe problem in Unity, the first place you should look for detailed error messages is the:

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

Answer: C

QUESTION NO: 16

Exhibit:



The Catalyst 6000 in the shown diagram has been configured 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
```

Assuming that the IP Phone is connected to port 5/1, which statements are true?

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

Answer: A, D, E
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 NO: 17
From the perspective of the CallManager, the Unity TSP looks and behaves most like a:

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

Answer: D

QUESTION NO: 18
What statement is an attribute of ISDN Non-Facility Associated Signaling (NFAS)?

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

Answer: A
Voice Enabling the Data Network; Durkin: Cisco Press p124

QUESTION NO: 19
What type of signaling provides Dialed Number Information Service (DNIS) on a T1/E1?

A. E&M
B. Loop Start
C. Ground 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/tk652/tk653/technologies_tech_note09186a00800e2560.shtml

QUESTION NO: 20
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 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 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 45
class-map match-all VOICE-CONTROL
 match ip dcsp 26
!
policy-map WAN-EDGE
 class VOICE
  priority queue 33 percent
 class VOICE-CONTROL
  bandwidth queue 2 percent
 class class-default
  fair-queue
!
```

Answer: B

QUESTION NO: 21
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 NO: 22
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
CIPT Solution Reference Network Design 2-5

QUESTION NO: 23
Migrating form TDM voice requirements to VoIP does not typically cause migration issues for customers who expect to be:

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

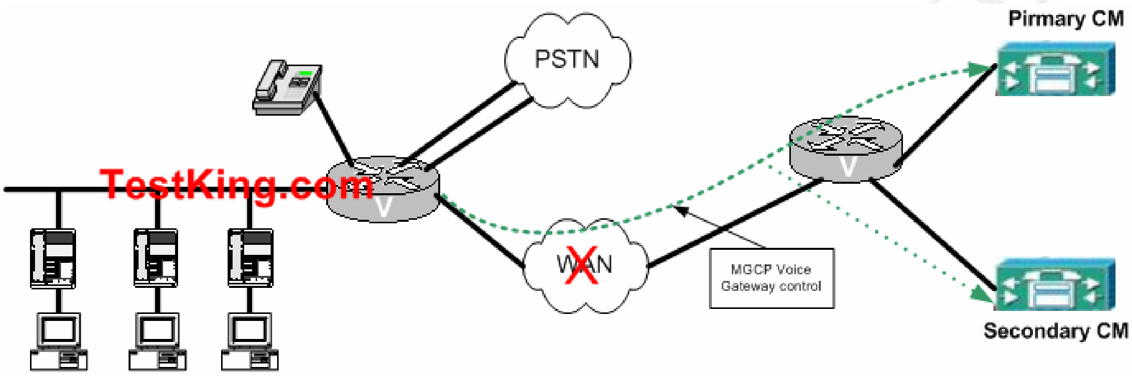
Answer: C

QUESTION NO: 24
Which are possible reasons when a user hears echoes of her own voice? (Multiple answer)

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

Answer: A, B, D

QUESTION NO: 25
Exhibit:



In a CM network deployed with MGCP to the branch office GWs, which two design methods should be used to protect branch office telephony (IP phone to IP Phone, and IP Phone to PSTN) when a WAN failure occurs?

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

Answer: B, D

QUESTION NO: 26
A voice gateway is receiving calls form infinite sources (PSTN callers) during the busy hours where lost calls are cleared (blocked).
The traffic model typically used to dimension the number of gateway ports/trunks required is:

A. Erlang-C
B. Poisson
C. Erlang-B
D. B and C

E. None of the above.

Answer: C
Answer is Extended Erlang B - Deploying Cisco Voice over IP Solutions; Davidson; Cisco Press p15

QUESTION NO: 27
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. Restrict calls to numbers such as 1-900 and International long distance calls.
D. Enable the use of E911 services in a Centralized Call Processing model.
E. All of the above.

Answer: E
CIPT Solution Reference Network Design Chapter 7

QUESTION NO: 28
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 clid_authen_collect.
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 clid_authen_collect
B. dial-peer voice 1 pots
   incoming called-number 5551000
   destination-pattern 1…..
   application clid_authen_collect
C. dial-peer voice 1 pots
   destination-pattern 1…..
   port 0:D
   application clid_authen_collect
D. dial-peer voice 1 pots
   destination-pattern 5551…
   port 0:D
   application clid_authen_collect

Answer: A

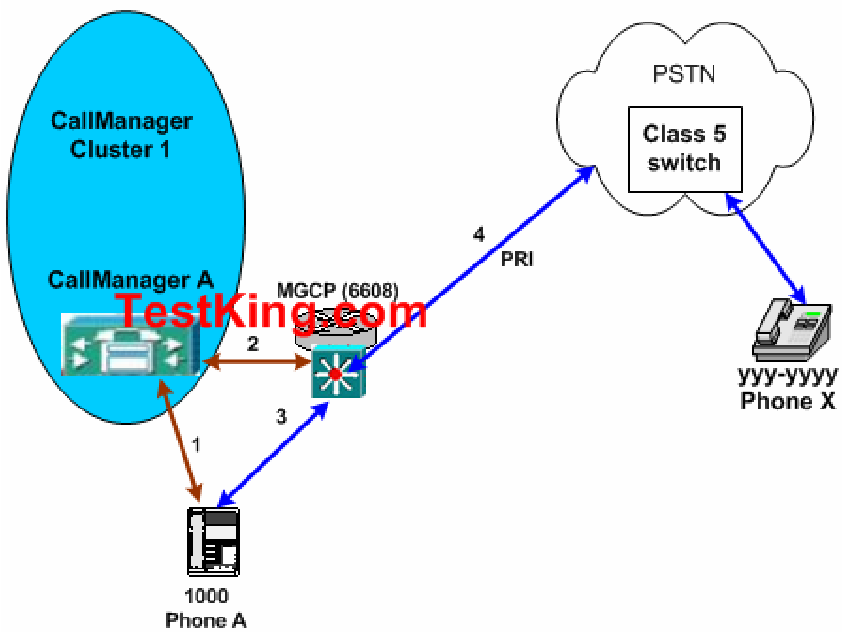
Cisco IOS Voice Commands, application
http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122tcr/122tvr/vrg_a1.htm#1492881

QUESTION NO: 29
What percentage of standard G.711 packet is taken by IP, UDP and RTP headers? (Note: cRTP not used)

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

Answer: E
G711 payload is 160 bytes and IP, UDP & RTP headers total 40 bytes

QUESTION NO: 30
Exhibit:



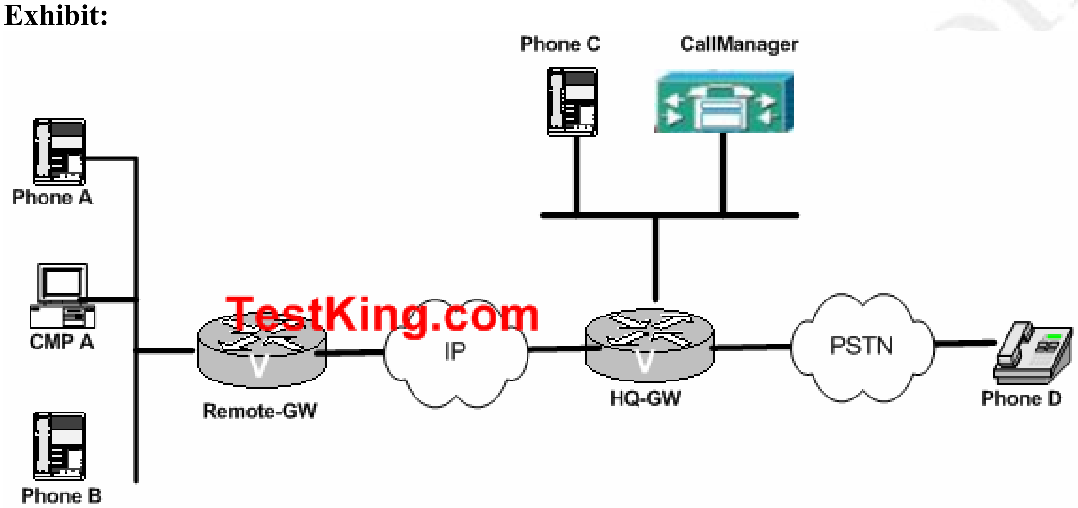
Assume that the gateway is a 6608 blade configured as a gateway and running MGCP; Call Manager runs version 3.1, and that a call is made from phone A to phone X. All IP streaming is G711. Each of the logical links represented carries certain types of traffic. On which links can Skinny (SCCP) traffic be seen?

A. 1, 2, 3, and 4

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

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

QUESTION NO: 31
Exhibit:



In the shown diagram the user at Phone A is hearing persistent echo on calls to the PSTN. The ERL has been determined to be 15db. Note the following configuration on the HQ-GW voice T1:

```
voice-port 1/0:15
 echo-cancel coverage 8
end
```

What step should the user initiate to attempt to resolve the echo?

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

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

QUESTION NO: 32
What does thus SMDI packet represent?

MD0010013D 0002914

A. A “Forward All Calls”, extension 10013 calling 2914
B. A “Call Forward No Answer” extension 1914 from extension 10013
C. Extension 2914 calling into voicemail on port 13
D. MWI OFF command for extension 2914
E. MWI ON command for extension 2914

Answer: C
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 NO: 33
What SMDI message from CallManager CMI or VG248 allows a Voicemail system to provide a “Hearth-beat” function on the RS-232 serial link?

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

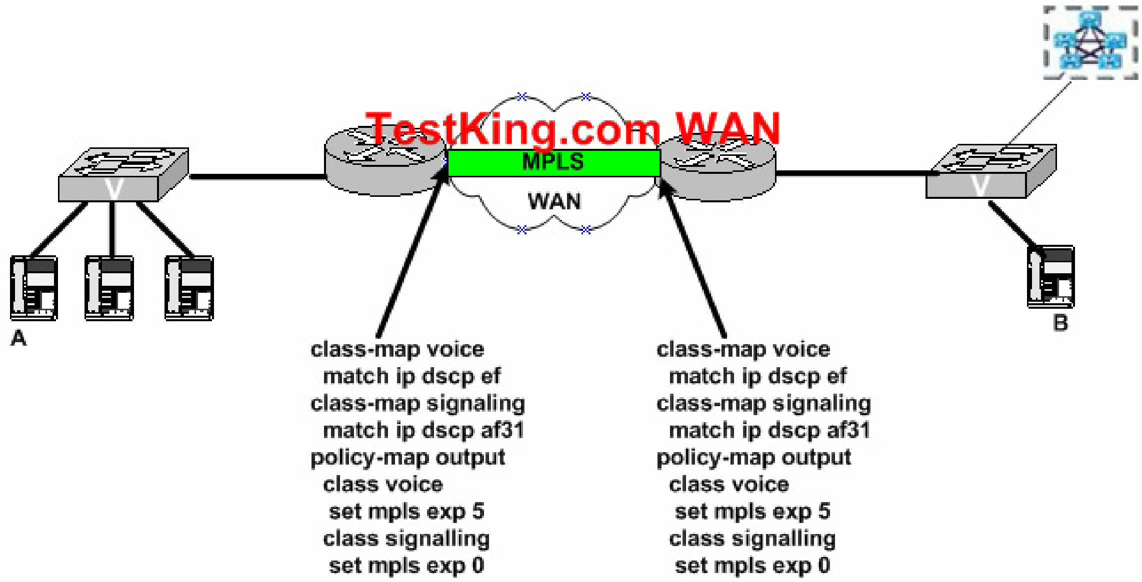
Answer: ~~A~~ (should be D)
Configuring CallManager 3.x for Integration to Voice Mail Systems via SMDI
http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_configuration_example09186a0080160b6f.shtml

QUESTION NO: 34
The maximum device weight capacity a Cisco MCS server can have does NOT depend upon:

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

Answer: D

QUESTION NO: 35
Exhibit:



Consider the QoS configuration in the picture shown for 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
C. Voice: IP Precedence 5; Signaling: 0
D. Voice: IP Precedence 5; Signaling: 3
E. Voice: 0; Signaling: 0

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

QUESTION NO: 36
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
http://www.cisco.com/en/US/tech/tk652/tk698/topic1

QUESTION NO: 37
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

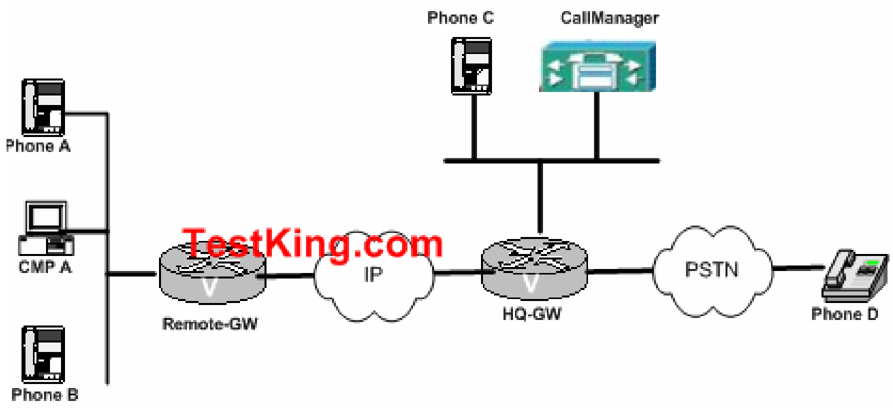
Answer: E
Actually for Ethernet it is 29.6kbps

QUESTION NO: 38
What command enabled 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
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 NO: 39
Exhibit:



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

A. The TCP connection between phone A and call manager has been disrupted.
B. Remote-GW has not received any messages from CallManager within the appropriate timeout period.
C. The FE and 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, E
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 NO: 40
Consider:

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 chosen by Call Manager?

A. 2XXX in partition Part_1.
B. 20XX in partition Part_2.
C. 200X in partition Part_3.

D. None of the above (user gets re-order tone).

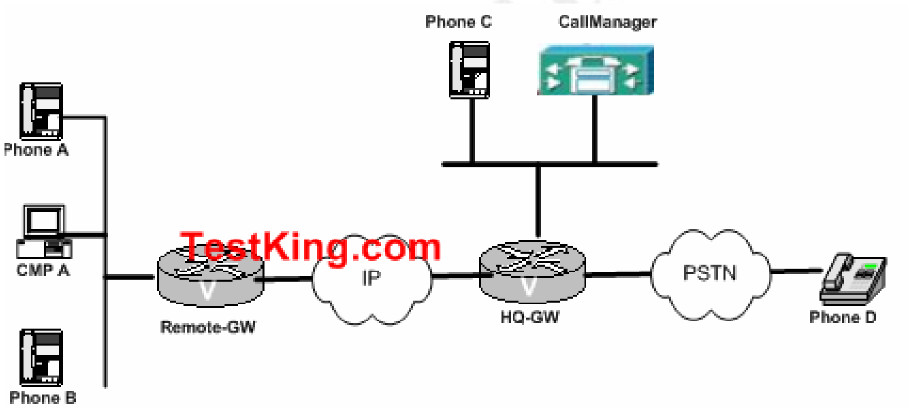
Answer: A
CIPT Course v3.3 p3-76

QUESTION NO: 41
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 Erlandg-B
B. Engset
C. Erland-C
D. Binomial
E. None of the above.

Answer: C
Deploying Cisco Voice over IP Solutions; Davidson; Cisco Press p16

QUESTION NO: 42
Exhibit:



During all calls from IP Phone A to Analog Phone D the user at IP Phone A hears persistent echo. During all calls from IP Phone A to IP Phone C no echo is heard. What is the best way to resolve the echo issue in the equipment under your control? (Note: Everything is under your control except for the PSTN)

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

Answer: E

QUESTION NO: 43
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
CISCO NETWORK MODULES, Catalyst Inline Power Patch Panel
http://www.cisco.com/en/US/products/hw/modules/ps2797/products_data_sheet09186a00800a9ea3.html

QUESTION NO: 44
For what purpose is a DPA (Digital PBX Adapter) used?

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

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

QUESTION NO: 45
What is the proper configuration for VoIP authentication via Authentication, Authorization, and Accounting (AAA)?

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

E. aaa new-model
   aaa authentication login voip radius

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

QUESTION NO: 46
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
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 NO: 47
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 NO: 48
The concept of Location is used by CallManager in order to:

A. Define what CODEC to use between devices which may be separated by a WAN link.
B. Define the bandwidth that can be used between devices.
C. Define groups of devices based on physical location, for the purpose of assigning Primary and Backup CallManager servers.

D. Group devices based upon physical location, in order to delegate Administrative Control.

Answer: B
CIPT Course v3.3 p3-98

QUESTION NO: 49
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

QUESTION NO: 50
What SIP header is a SIP Proxy allowed to change?

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

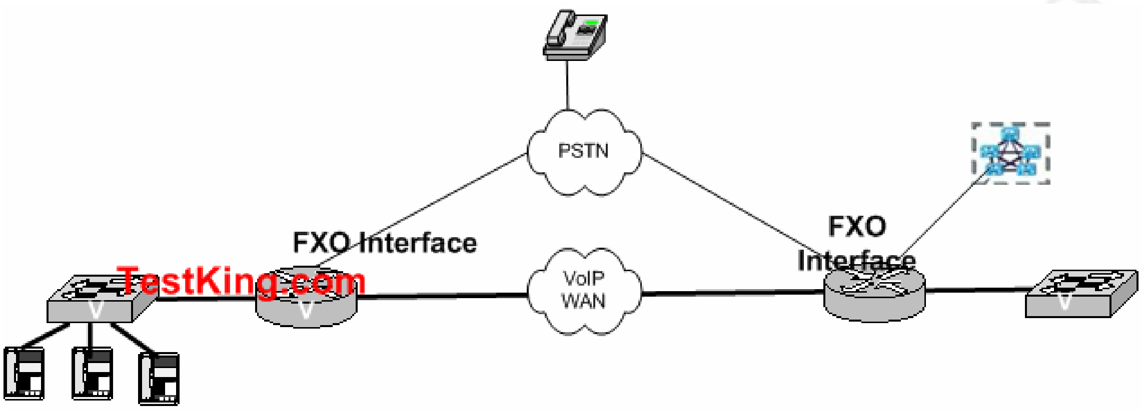
Answer: D
CISCO SIP PROXY SERVER, F Call-Flow Scenarios
http://www.cisco.com/en/US/products/sw/voicesw/ps2157/products_administration_guide_chapter09186a00801a1d94.html

QUESTION NO: 51
Two Unity Servers can be placed in the same Dialing Domain if:

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

Answer: B
CISCO UNITY, Network Settings
http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_administration_guide_chapter09186a00801ba7bb.html

QUESTION NO: 52
Exhibit:



In the figure shown, the customer requires that Caller ID for calls from the PSTN to the IP Phones must be supported. Analog trunks are equipped to the PSTN from both GWs. What is the correct voice GW design for this customer? (Note: Assume the PSTN CO switch is capable of delivering Caller ID on the connection to the Cisco voice GW, and the cisco voice gateway is using a VIC-2FXO-M1 card)

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

Answer: C

QUESTION NO: 53
Which fields in the output from show active voice indicate that packet loss is occuring?

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

Answer: A, B, D

VOICE QUALITY, Using the show call active voice Command to Troubleshoot Voice Quality Issues
http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a008019ab88.shtml

QUESTION NO: 54
What type of signaling can provide Automatic Number Identification (ANI) on a T1/E1?

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

Answer: C
TELEPHONY SIGNALING, Understanding How Digital T1 CAS (Robbed Bit Signaling) Works in IOS Gateways
http://www.cisco.com/en/US/tech/tk652/tk653/technologies_tech_note09186a00800e2560.shtml

QUESTION NO: 55
When troubleshooting an IOS Voice Gateway, what command will produce detailed information (codec, ERL, tx/rx packets, dial peers, etc) on currently active calls?

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

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

QUESTION NO: 56
In a CallManager cluster, Intra Cluster Communications Signaling includes:

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

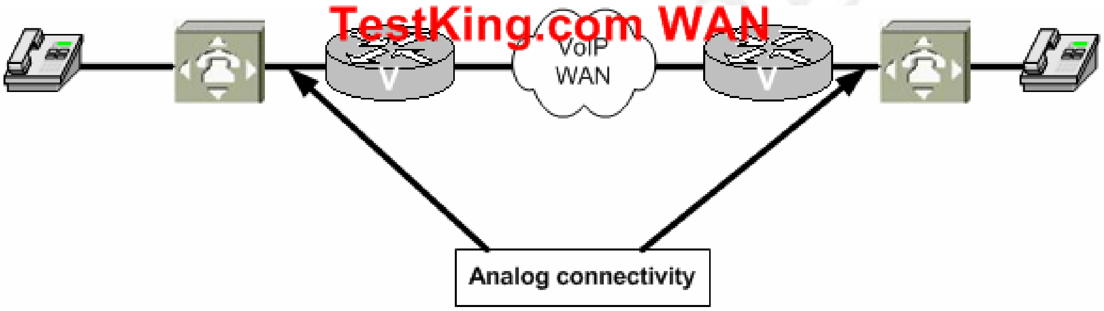
Answer: A
CIPT Course v3.3 p1-37

QUESTION NO: 57
When processing a SIP message, regardless of next-hop SIP device, what is the order in which CSPS will determine how to route the packet?

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

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

QUESTION NO: 58
Exhibit:



In the figure shown, the customer requires that Caller ID be displayed for all phones connected to the PBXs, and for calls in both directions across the IP network. The PBXs have only analog (FXS, FXO and E&M) capabilities to connect to the Cisco voice GWs. What design will achieve the customer's requirements?

A. 4-wire E&M
B. FXS and E&M only
C. 2-wire and 4-wire E&M
D. None with only analog capability.
E. All of FXS, FXO and E&M, provided the FXO cards are a vintage that support Caller ID.

Answer: E

QUESTION NO: 59
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, E
CISCO CATALYST 4200 SERIES SWITCHES, Configuring Survivable Remote Site Telephony
http://www.cisco.com/en/US/products/hw/switches/ps669/products_configuration_guide_chapter09186a008007f1e1.html

QUESTION NO: 60
What does the term "MGCP backhaul" mean?

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

Answer: D
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 NO: 61
On a T1/E1 Frame Relay circuit, what factor determines the fragmentation size?

A. Minimum CIR
B. Average CIR
C. Burst Count
D. Line speed
E. None of the above.

Answer: A
VOFR, Frame Relay Fragmentation for Voice
http://www.cisco.com/en/US/tech/tk652/tk692/technologies_tech_note09186a00801142de.shtml

QUESTION NO: 62

What is the target overall loss plan across a telephone network?

A. 0dBm – 8dbm
B. 8dBm – 12dBm
C. 12dBm – 16dBm
D. 16dBm – 20dBm

Answer: A

QUESTION NO: 63
A gatekeeper is: (multiple answer)

A. An optional component in an H.323 system which provides call control services to the H.323 endpoints.
B. A compulsory component in an H.323 system which provides call control services to the H.323 endpoints.
C. 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.
D. A compulsory component in an SIP system which provides call control services to the H.323 and SIP endpoints.

Answer: A, C

QUESTION NO: 64
Survivable Remote Site Telephony (SRST) is a design method to enhance the availability of what type of telephony network design?

A. CM tool bypass
B. CM centralized call processing
C. CM distributed call processing
D. CM single site campus design

Answer: B

QUESTION NO: 65
In a H.323 network, what function is NOT performed by the gatekeeper?

A. Call admission control
B. Number to IP address translation
C. Codec negotiation

D. Call routing
E. Call authorization

Answer: C

QUESTION NO: 66
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: B, D

QUESTION NO: 67
Which Cisco Products can produce SMDI packets?

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 NO: 68
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 NO: 69
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

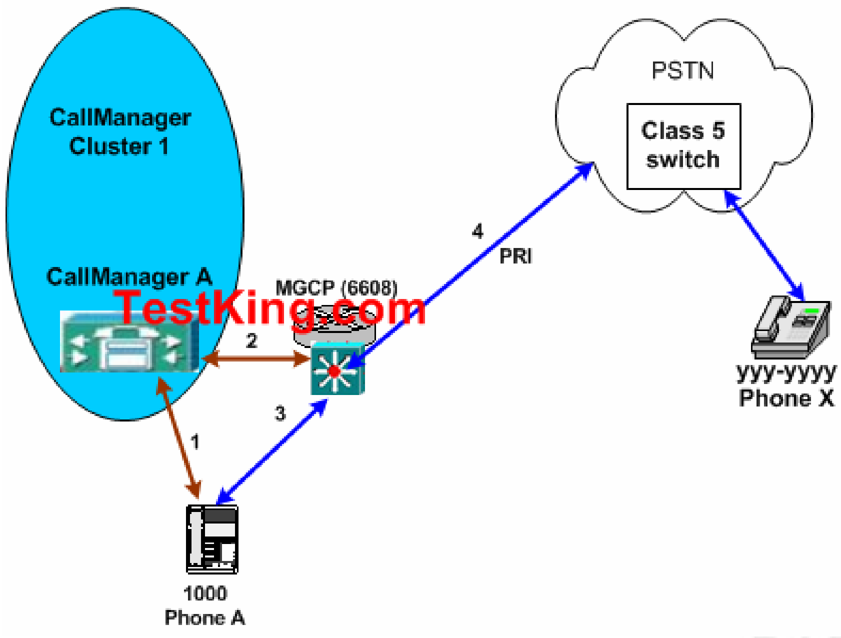
Answer: A

QUESTION NO: 70
What standards-based protocol will allow CallManager to seamlessly Integrate with other vendor's traditional PBX systems?

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

Answer: D

QUESTION NO: 71
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 types of traffic.
On which links can q.921 traffic be sent?

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

Answer: A

QUESTION NO: 72
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: A

QUESTION NO: 73
Consider phone A, assigned to Calling Search Space A. Calling Search Space A contains the following partitions (on the order shown), listed with their respective Route Patterns:

Partition_A1, containing Route Pattern 1XXX
Partition_A2, containing Route Pattern 10XX

If Phone A dials "1001", what statement is true?

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

Answer: B
http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_administration_guide_chapter09186a00800c2ef1.html#10870

QUESTION NO: 74
When connecting a Cisco voice gateway to a PBX or the PSTN via ISDN (PRI, QSIG, BRI), which are the attributes of the PBX/PSTN-switch that must be known to understand which features to configure on the voice GW in order to connect successfully to it?

A. What PRI/BRI switch-type is supported by the PBX/PSTN-switch.
B. Whether symmetric mode is supported by the PBX/PSTN-switch.
C. Whether the network or user side is supported by the PBX/PSTN-switch.
D. Whether Q.921 or Q931 is supported by the PBX/PSTN-switch.
E. Whether wink, delay dial or immediate dial is supported by the PBX/PSTN-switch.

Answer: A, C

QUESTION NO: 75
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, D

QUESTION NO: 76
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 NO: 77
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
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 NO: 78
When using the CCMAdmin page of a Subscriber CallManager, changes made to the configuration are:

A. Made locally in the SQL Database, and replicated up to the publisher immediately.

B. Made locally in the SQL Database, and replicated up to the publisher at the next scheduled replication.
C. Made locally in the SQL Database, and in the Publisher SQL Database.
D. Made in the Publisher SQL Database, and replicated to subscribers.

Answer: D

QUESTION NO: 79
What is the difference between a route-group and a route-list?

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

Answer: B
CIPT Course v3.3 p3-22

QUESTION NO: 80
With Unity 3.0/ and Exchange 2000, which attributes are stored in Active Directory?

A. Recorded Name
B. Recorded Greeting(s)
C. Alternate Extensions
D. Transfer Type (Supervised, Release to Switch)
E. Location ID

Answer: A, B, C, E

QUESTION NO: 81
When using the Low Latency Queuing feature of Cisco IOS:

A. All the RTP traffic 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: A

QUESTION NO: 82
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 FRST-voice
frame-relay cir 1536000
frame-relay bc 15360
frame-relay be 0
frame-relay mincir 1536000
B. map-class frame-relay FRST-voice
frame-relay cir 1536000
frame-relay bc 1536
frame-relay be 0
frame-relay mincir 1536000
C. map-class frame-relay FRST-voice
frame-relay cir 1536000
frame-relay bc 15360
frame-relay be 1536
frame-relay mincir 1536000
D. map-class frame-relay FRST-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: A (or D – they are the same)

QUESTION NO: 83
What MailStore option does Unity version 2,4,6 support?

A. Exchange 5.5
B. Exchange 2000
C. Domino
D. MS Mail

Answer: A

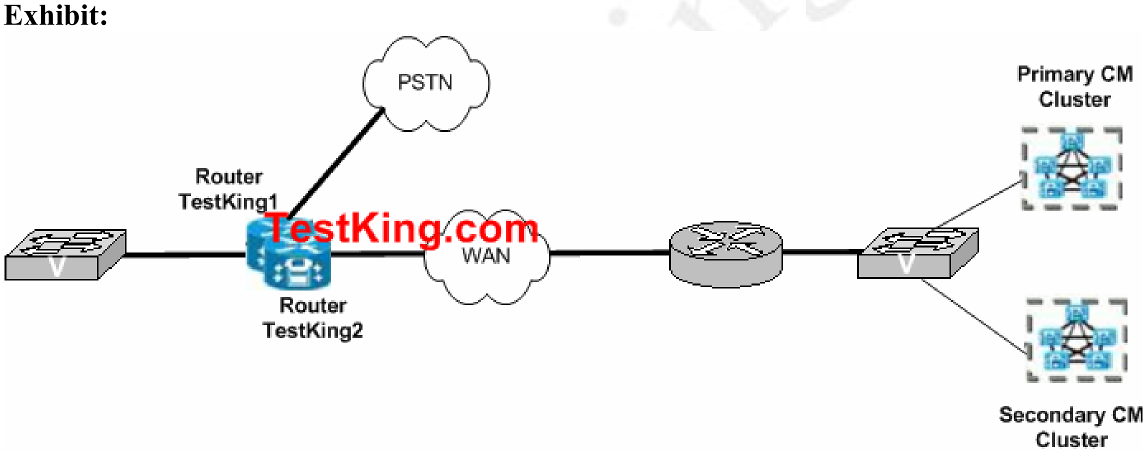
http://www.cisco.com/en/US/products/sw/voicesw/ps2237/prod_pre_installation_guide09186a00800ea54b.html#73967

QUESTION NO: 84
When Gateways are registering with a Gatekeeper, the Gatekeeper can be:

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

Answer: E

QUESTION NO: 85
Exhibit:



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 TestKing1 (the primary router) is used. Router TestKing1 switches to SRST mode to preserve telephony functions. At this point Router TestKing1 fails, and HSRP backup Router TestKing2 becomes the active router for the branch office, taking over SRST and routing functions for the office. For Router TestKing2 to be effective in running SRST for the branch, which types of physical connectivity must be duplicated on Routers TestKing1 and TestKing2?

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

Answer: D

QUESTION NO: 86
What statement about echo is false?

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

Answer: C
Deploying Cisco Voice over IP Solutions; Davidson; Cisco Press p42

QUESTION NO: 87
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: B

QUESTION NO: 88
What Network Management Server (NMS) application can monitor Voice quality by polling the SNMP MIB for MQC?

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

Answer: E

QUESTION NO: 89
During the busy hour, 100 Erlangs may be generated by:

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

Answer: D

QUESTION NO: 90
In a Cisco IPCC deployment, where CallManager is the routing client, upon receipt of a call, the ICM performs what function?

A. ICM maps the Dialed Number to a call type and then maps the call type to a routing script.
B. ICM Identifies and selects an available agent and determines the label to be returned to the routing client.
C. ICM plays Music or targeted announcements for callers waiting in queue if no agents are available.
D. A and B.

Answer: D

QUESTION NO: 91
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
Cisco Voice over Frame Relay, ATM & IP; McQuerry; Cisco Press p57

QUESTION NO: 92
A router is connected to a PBX via a 4 wire E&M circuit. All calls to the trunk are failing and it is suspected that the PBX is not seeing the incoming calls on the trunk. To

determine if this theory is true the PBX is configured to generate dialtone when it sees an incoming call.
Which of the following should cause the PBX to generate dialtone?

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

Answer: D
Cisco Voice over Frame Relay, ATM & IP; McQuerry; Cisco Press p43

QUESTION NO: 93
Exhibit 1:

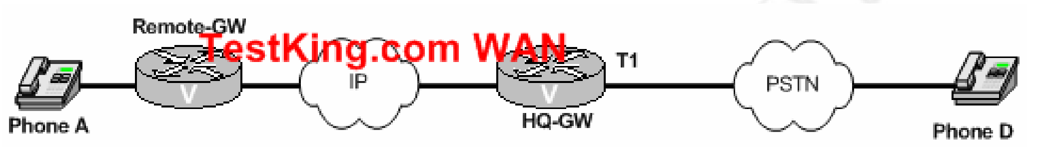


Exhibit 2:

REMOTE-GW

```
voice translation-profile TESTKING_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 TESTKING_1
session target ipv4.x.x.x.x
Port 0/0
```

HQ-GW

```
Interface FastEthernet0
Ip Address x.x.x.x y.y.y.0
voice translation-profile TESTKING_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 TESTKING_2
Port 1/0:23
```

The user at phone A dials 555121255.

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.)

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

Answer: A

QUESTION NO: 94
Cisco SIP Proxy can NOT perform what task?

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

Answer: A

QUESTION NO: 95
Which statements are true about Analog DID connections to the PSTN?

A. DID trunks can only send calls towards the CO.
B. DNIS information is send in-band.
C. DID trunks can only send calls from the CO.
D. DNIS information is send out-of-band.

Answer: B, C

QUESTION NO: 96
What statement regarding jitter is correct?

A. 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.
B. 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.

C. Jitter 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.
D. 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.

Answer: C

QUESTION NO: 97
A Cisco SIP Proxy Server can make routing decisions based upon which criteria?

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

Answer: A

QUESTION NO: 98
In an IP Contact Center deployment, the Erlang-C Traffic Model is used to provision:

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. Agents initiating/handling outbound calls only.

Answer: B

QUESTION NO: 99
What industry-standard protocol does CallManager support for Integration to Voicemail systems?

A. SMDI
B. H.323
C. T1-CAS
D. None of the above.

Answer: A

QUESTION NO: 100
The range of UDP port numbers used in Cisco's VoIP implementation is:

A. 225 to 245
B. 16384 to 32767
C. 1718 to 1720
D. 11000 to 12000
E. 32769 to 64535

Answer: B

Note: Answers to the unanswered questions will be provided shortly. First customer, if any, faster than the TestKing team in proving answers will receive TestKing credit for each answer provided.

Send answers to feedback@testking.com.