Exam : 642-414

Title : Telephony Design Exam (IPTD)

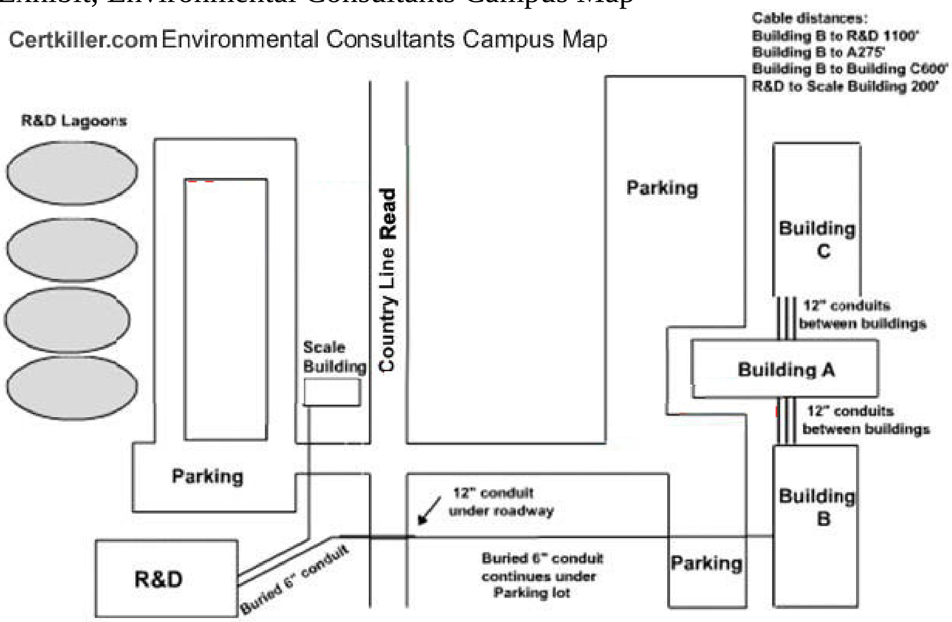
Ver : 07.02.07

QUESTION 1:

Scenario

Certkiller .com Environmental consulting (Certkiller .com), an environmental consulting company, based near Atlanta, Georgia, uses a PBX for its campus phone system. The PBX cannot support all of the new features that Certkiller .com requires, so the company has decided to change its system form traditional TDM to IP telephony. The campus currently has approximately 922 users in three four-story office buildings, a single story R&D facility, and a scale building. It will be your job to determine what information is needed for the proper design of the company's converged network.
During preliminary investigation, here is what there is to be found:
922 employees, of which 868 have phones that are DID. The rest are lobby and break room phones, departmental phones, etc. for a total of 895 phones.
Each employee with DID has a personal mailbox, as does every department with a group phone, for a total of 881 voice mail boxes.
There are 12 people in marketing, each of whom uses the phone considerably more than the average user does. Most of these calls are external calls.
There are 7 people in technical support, each of whom uses the phone considerably more than the average user does. Most of these calls are internal calls.
The current dial plan uses four-digit dialing for extension-to-extension calls, and dialing 9 for outside calls (local and long distance).
Certkiller .com doesn't have a current traffic analysis of their network. They believe they have enough bandwidth for anything they might want to run, including IP telephony, but have no concrete documentation to back that up.
Each of buildings A, B and C has a combination MDF/IDF on the ground floor, with an IDF on each upper floor. The IDFs are connected to the MDFs via multiple 25-pair cable bundles for phones and two pairs each multimode fiber optic cable for data.
Each of the 895 phones is cables using Category 3 UTP cable out from the IDFs.
Each station in the R&D building has two Category 5 UTP cable drops, plus phone.
All buildings on the campus are data-connected via an FDDI ring with the exception of the scale building, which is connected to the R&D building via Category 5 UTP cable.
The Certkiller .com facility spans two counties, so the R&D building and the scale building are services by a different PSAP than are buildings A through C.
Building C houses the main computer room that contains all the company servers.
The main computer room in Building C also contains the PBX with PSTN connectivity, and the Internet connection(s).
There is a pair of Cisco 2514 routers providing connectivity to the Internet, they are set up with HSRP.
The data network was built using token ring LANs connected via the FDDI ring.
Certkiller .com is interested in migrating its phone systems to IP telephony, vs. a massive weekend cutover.
Certkiller .com has been growing slowly over the last four years. During that time the company has become the leader in methane recover systems for dairy and hog operations.
Certkiller .com expect more growth from recent changes to federal air quality regulations.
The company is planning to expand as follows:

Marketing will double in size to 24 employees.
The Southeast Region will grow from 90 to 150 employees.
The Midwest region will grow from 95 to 130 employees.
The Southwest region will grow from 85 to 95 employees.
R&D will grow from 25 to 60 employees and will be split between the R&D building and the bottom floor in Building B.
The Project Management group will grow from 30 to 90 employees. Of the 202 new users will have a dedicated phone.
Certkiller .com has 12 department, each of which has a music-on-hold message. All the departments would like to play their messages when they place a caller on hold. There is also a standard corporate message that is played when people outside of these 12 departments place a user on hold.
On the basis of information discovered during the investigation phase of the design, it has been decided that the single-site call processing model is the proper deployment model for Certkiller .com. The access layer devices will be placed in the IDFs, the distribution layer devices will be placed in the MDFs, and the core will be deployed in the compute room. Each IDF services approximately 70 to 75 users.
Certkiller .com is using Token Ring with an FDDI backbone. The network is to be migrated to an Ethernet network. Certkiller .com has had many network outages in their current network, and is concerned with network availability, especially as the phone system will now be residing on the same network.
Topology
Exhibit, Environmental Consultants Campus Map



Exhibit, Environmental Consultants Campus Map #2

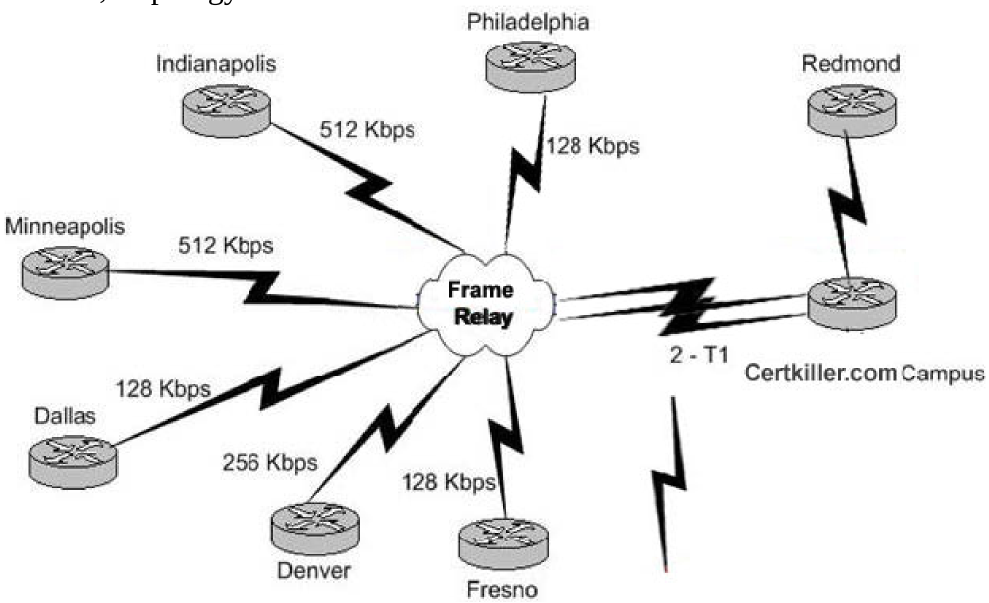
Certkiller.com Environmental Consultants Campus Map

Main Telephone Number (770) 555 0100
Internal Directory

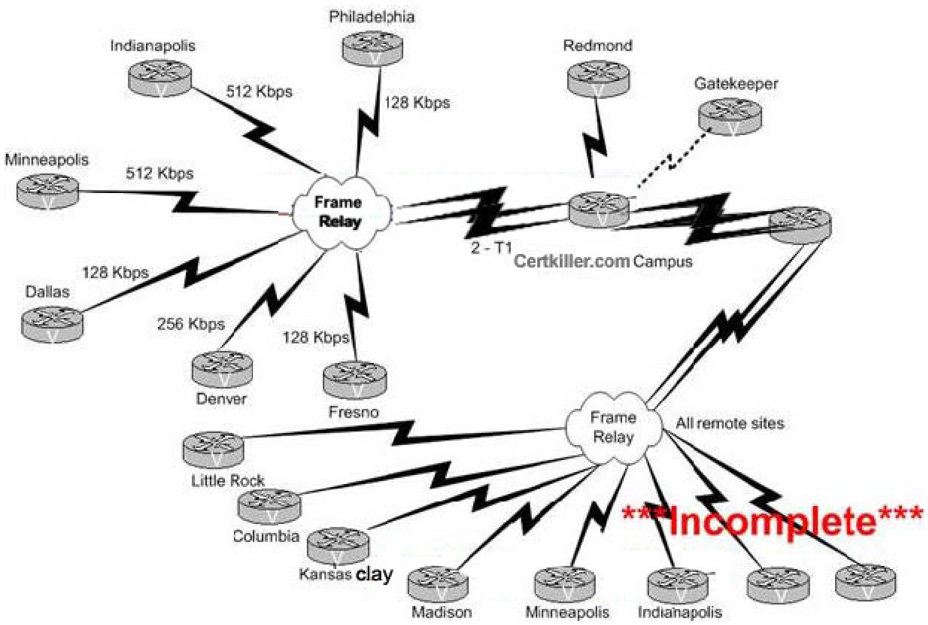
Department	Main Number
Accounting	900
Fax	901
Administration	
Operator	100
Voice Mail	150
Customer Service	1300
Fax	1301
Training	1375
Document Center	1400
Fax	1401
Environmental Services	1800
Fax	1801
Human Resources	1800
Fax	1801
Project Management	1100
Fax	1101
R&D	700
R&D Fx	701
Scale Building	799
Sales-External	
Northwest Region - Redmond (425) -555-0100	
Southwest Region - Dallas (214) -555-0100	
- Denver (303)-555-0100	
- Fresno (559)-555-0100	

Department	Main Number
Sales - External cont.	
Midwest Region - Indianapolis	(317)- 555-0100
Mineapolis	(612)- 555-0100
Northwest Region - Phildelphia	(215)- 555-0100
Southwest Region - Atlanta	(770)- 555-0100
Sales - Internal	
Northwest Region	200
Fax	201
Southwest Region	300
Fax	301
Midwest Region	400
Fax	401
Northeast Region	500
Fax	501
Southeast Region	600
Fax	601
Geotechnical Group	1700
Fax	1701
Technical Support	1000
Fax	1001
Testing Lab	1400
Fax	1401
Marketing	1600
Fax	1601
Geotechnical Group	1500

Exhibit, Topology



Exhibit, Topology #2



Certkiller .com (84 Questions)

QUESTION 2:

From the following list of customer attributes, choose the correct IP telephony call processing model:
A large campus that spans multiple PSAP area.
A single group of buildings each with its own computer room.

A. single-site call processing
B. centralized call processing
C. hybrid call processing
D. distributed call processing

Answer: A
The single-site model is used for a single campus or site with less than 30,000 lines. All traffic beyond the LAN or MAN uses the PSTN.
Single-site call processing is needed for a large campus that spans multiple PSAP area?
ref: http://www.developer.com/voice/article.php/3400571
Single-site call processing model: In the single-site model, each site or campus has its own IP PBX or media server to perform call processing functions; also, there are no voice calls communication over the WAN network. If you want to implement external calls or call remote sites, you can use PSTN.

QUESTION 3:

From the following list, select the information that is relevant to choosing an IP telephony centralized call processing model.

A. A single 6-story building with an IDF on each floor and an MDF in the computer room on the second floor.
B. Three small regional sales offices located in the three Western time zones.
C. Centralized order processing, shipping, and billing for all customer products.
D. Connectivity to a single service provider that hosts the company web site and provides for Internet access.
E. Multiple PRIs to the PSTN.

Answer: B
IP telephony centralized call processing model works for different locations?
ref: http://www.ciol.com/content/flavour/voip/102052801.asp
IP telephony deployment, building blocks
Many enterprises have already implemented VoIP as a form of toll bypass. But by deploying a complete IP telephony solution, enterprises leverage the inherent cost savings of a converged network across the organization, while adding new features and functions.
Deployment models. There are four basic models for IP telephony deployment in the enterprise:
1. Single-site deployment -- IP telephony is deployed within a building or campus, but no voice traffic is carried over the IP WAN.
2. Independent call-processing approach -- IP telephony is deployed within various remote locations, but calls are transmitted across the public switched telephone network (PSTN).
3. Multisite deployment with distributed call processing -- Calls are transmitted using both the IP WAN (primary path) and the PSTN (secondary path) to connect enterprise locations. Call-processing and voice-messaging equipment are present at each location, but interconnect across the WAN.
4. Multisite deployment with centralized call processing -- Calls are transmitted using both the IP WAN (primary path) and the PSTN (secondary path), but call-processing and voice-messaging equipment are centrally located. This is often the most efficient solution for multisite enterprises.
The single-site and independent call-processing models are similar in the calls that continue to be transferred across the PSTN, but the enterprise can still take advantage of the benefits of IP telephony applications. The single-site and independent call-processing models often serve as the first step towards implementing an all-IP enterprise telephony network.
In a multisite with distributed call processing model, each site contains its own call processing and resources. Voice calls between sites use the IP WAN as the primary path. However, if the IP WAN is down or has insufficient resources to handle calls, the PSTN is used as a secondary path. The actual path used to connect the call, either the IP WAN or the PSTN, is transparent to both the calling and called party.
The multisite with centralized call processing model has all call-processing resources concentrated in a single site. Remote locations have only the basic infrastructure, such as switches, routers, and gateways, and endpoints such as IP or analog phones. The primary advantage of this model is the ability to centralize call processing, which reduces the equipment required at remote branches and eliminates the need for administration of

multiple private-branch exchanges (PBXs) and key systems.
In addition, this model allows for single-point, dial-plan implementation as opposed to requiring dial plans in multiple locations. This model is particularly attractive for enterprises with small branch offices and telecommuters. To facilitate deployment, Cisco recently released the Catalyst 4224, a two-rack unit (RU) access gateway switch that combines the functionality of a switch, router, and gateway.
Ref 2: http://www.informit.com/articles/article.asp?p=360064&seqNum=4&rl=1
WAN centralized call-processing-In this model, multiple sites deploy IP telephony. These sites might be connected to a central campus over a private WAN or through the use of VPNs. The headend site, or campus, contains the only call-processing manager cluster; however, remote sites can have local voice
Incorrect answers:
C, D, E These have nothing to do with the choice of IP Telephony Call Processing Model.
A This is a Single-Site Call Processing Model choice

QUESTION 4:

Which of these pieces of information most be addresses in the design of the IP telephony system for Certkiller .com? (Choose three)

A. Certkiller .com is using a public IP addressing scheme. They have four contiguous Class C addresses.
B. Certkiller .com has seen a 15% increase in traffic to their website with the announcement of a new methane monitoring solution.
C. The active Internet gateway is running at 55% of peak capacity.
D. The current PBX has a proprietary connection to the current voice-mail system.
E. The property that houses the Certkiller .com R&D facility, scale house, and test lagoons has recently been annexed by a neighboring city.

Answer: A, D, E
Answers B and C are not as relevant as the other 3 answers. Since this question asks specifically about information needed for the design of the IP Telephony System, the three correct answers are A, D, E.
A - IP scheme is always an important factor
D - Existing PBX and voice-mail must be factored into the design
E - The 911 and E911 service is a design consideration and having part of the campus reside in a different city or county can have an impact on how this service is configured
Certkiller .com use public IP addressing scheme. they have four contiguous class C address
The current PBX has proprietry connection with current voice mail system
Ref: http://phoenix.swarthmore.edu/2003-11-20/news/13505
Swarthmore's campus phone system is run through what is called a private bridge exchange, known as PBX, which is the gateway between Swarthmore's phone network and the outside phone system. The PBX is what makes it possible for Swarthmore residents to simply dial four digits and reach people on campus and makes it necessary

for users to dial 9 before placing outside calls.
Swarthmore's current PBX is a proprietary digital phone system, according to Dumic.
This system requires that every user use the phones the college provided. Under the current setup, the college has a phone and voice mail service for all students.

QUESTION 5:

You are in a meeting with the Certkiller .com telephony services manager and the data network manager.
What information do you need to obtain to assist you in the design if the IP telephony solution? (Choose four)

A. Bandwidth available to connect to the Internet.
B. The number of phones that have DID.
C. The number of servers supporting the R&A facility.
D. What type of dialing plan is deployed at Certkiller .com.
E. Which group of users contributes the most traffic to the network.
F. Which group of users spends the most time on their phones.
G. How the PBX is cabled from each MDF to each IDF and to each desk location.
H. The measures Certkiller .com has implemented to secure the network.

Answer: B, D, F, G
Correct:
B - Direct Inward Dial is an important factor in the design
D - The existing dialing plan may be sufficient or may need to be altered in the design
F - Users who are using greater then 6 BHCAs have a greater weight
G - Cabling is an important consideration, is it sufficient or will higher capacity cabling be needed?
Incorrect:
A - While this can be important, remember that you can usually add more bandwidth if required
C - Not relevant
E - This may be important on the section of the network where these users are but is not nearly as important to consider as the other four correct answers
H - While very important, existing security is not relevant since the new design will specify what security needs to be in place, regardless of what already exists
System Design
An IP telephony system must be designed taking into account the specific needs of the organization. A number of factors impact system design, including establishing a preferred mechanism for supplying power to clients, identifying desired PSTN interfaces, establishing adequate levels of QoS, and the need for advanced features, such as conferencing and transcoding
http://24.234.143.242/p17/VOIP/pimvo_wp.pdf.
1. Considerations for deploying the data network: (1) LAN/Campus environment: collect information about the topology, average/peak bandwidth, LAN QoS functionality, where servers and gateways will be located (2) WAN environment: decide on a topology (build

using a hub and spoke model or multimeshed site model), investigate impact of WAN outage, available bandwidth and scalability on the existing network, QoS requirements for current network usage.
2. Considerations for the telecom infrastructure: type and size of voice mail systems/PBXs/number of phones/fax requirements, how to route redundant/back-up paths, how to design/improve current cabling/power infrastructure.

QUESTION 6:

You are in a meeting with the Certkiller .com telephony services manager and the data network manager.
What four pieces of information will be important for you to capture from this meeting? (Choose four)

A. The manufacturer, type and number of devices in the network.
B. The IP addressing scheme.
C. The type of network design currently in place.
D. The current integration of voice and data in the Certkiller .com network.
E. The capacity of the link to the Internet.
F. The type and size of the power circuits in each MDF and IDF.

Answer: A, B, C, F
Obtain the following details when conducting your site survey
Existing LAN infrastructure
WAN infrastructure
L2 and L3 infrastructure
IP addressing scheme
Directory and messaging architecture
L3 routing and routed protocols in use
Legacy telephony and voice mail infrastructure as well as telephony features deployed
Legacy dial plan
Current utilization and performance of voice and data networks
Current network management infrastructure
Ref1:
http://www.ciol.com/content/flavour/voip/102052801.asp
System Design
An IP telephony system must be designed taking into account the specific needs of the organization. A number of factors impact system design, including establishing a preferred mechanism for supplying power to clients, identifying desired PSTN interfaces, establishing adequate levels of QoS, and the need for advanced features, such as conferencing and transcoding.
Power supply: A key concern in designing an IP telephony system is often how to supply power to IP phones. Users have become accustomed to their phones receiving all necessary power from the PBX switch.
Gateways:
Selecting an appropriate gateway, the equipment that provides the connection between

the IP and time-division multiplexing (TDM) telephony worlds, is a crucial element in designing an enterprise IP telephony system. There are four gateway protocols to choose from: Simple Gateway Control Protocol (SGCP) also known as skinny gateway, H.323, MGCP, and Session Initiation Protocol (SIP). The criteria for choosing the appropriate gateway for an enterprise include support for a full range of PSTN interfaces, sufficient port density, support for WAN interfaces, and the gateway's ability to supply a high level of QoS. Whether to choose a standalone or integrated router-gateway depends on the relative importance of cost, flexibility, functionality, and manageability to the enterprise. There are other considerations when selecting a gateway. For example, many endpoints in the voice network, such as IP phones and voice-messaging units, require out-of-band dual tone multifrequency (DTMF) transmissions to avoid in-band DTMF distortions. In these cases certain gateway protocols must be used, namely H.323 v.2, MGCP, or Skinny. To ensure call survivability, a gateway that supports MGCP is recommended.
DIAL plans & Digital signal processing (DSP) resources
Ref 2:
http://www.informit.com/articles/article.asp?p=360064&seqNum=4&rl=1
Campus Module
The Campus module contains the end-user systems and the corporate servers, such as voice-mail servers, e-mail servers, management servers, IP phones, and the Layer 2 infrastructure. VLANs are enabled on the Layer 2 switch to provide segmentation between the voice and data traffic. Host IDS (HIDS) is deployed across all critical servers. The role of HIDS is more important in this design because of the lack of a Layer 3 router within the Campus module to provide access control between the VLANs.
Design Alternatives for the Small IP Telephony Network
One alternative design is to provide two completely separate VLANs, with a Layer 3 access device providing traffic filtering between the VLANs. Another alternative is to place the voice-mail/e-mail server in the voice segment; however, this design is not recommended because the voice-mail/e-mail server is running additional services that are required in the data segment.

QUESTION 7:

You are doing a physical site survey of the Certkiller .com campus.
You noticed on the site map that the facility is divided by Country Line Road. This is actually the boundary between Cobb and Fulton counties. What issue needs to be addressed for the IP telephony design?

A. If calls from the Certkiller .com buildings in Cobb County will incur a toll charge when calling the Certkiller .com buildings in Fulton County.
B. If the tax rate for telephony information needs to be kept for both counties.
C. If a PSTN connection in bidg C can route emergency calls to the correct PSAP for the buildings in Cobb County.
D. If the phones in each county require overlapping extension numbers.

Answer: C

It is possible that the two counties are serviced by a different PSAP. It is critical that the design assure that calls coming from each building are routed to the proper PSAP.

QUESTION 8:

You are doing a physical site survey of the Certkiller .com campus.
What four issues are related to the physical placement of network hardware?
(Choose four)

A. physical security
B. adequate rack space
C. a dust-free environment
D. access to building distribution cabling
E. sufficient HVAC
F. adequate lightning

Answer: A, B, D, E
While all answers seem appropriate, these four are the most relevant.
physical security , HVAC and adequate rack place is related to physical placement of network hardware?
Ref:
www.sbc.com/Large-Files/RIMS/ Missouri/Local_Access/mo-la-02.pdf
RATE ELEMENTS (Continued)
20.3 Caged Collocation (Continued)
D. Safety and Security
This charge represents reasonable costs incurred by SWBT to secure its equipment contained
within Eligible Structure. This charge is expressed as a recurring rate on a per square foot basis and was developed based on implementation of varying combinations of the following
security measures and devices. This rate may include only the costs associated with the most
cost-effective method of security systems, which may consist of a sub set of the following:
- Interior Security Partition separating SWBT equipment
- Provisioning of door locks and keying of existing doors
- Door access controller and network controller necessary for a card reader system
- Security camera systems
- Locking cabinets for network equipment
- Combination door locks
- Cable locks for computer terminals and test equipment
- Secure ID/password protection for computer systems
- Emergency exit door alarms
In the event SWBT elects to erect an interior security partition in a given Eligible Structure to

separate its equipment, the lesser of the costs of the partition or a security camera system for
such eligible structure shall be applicable. In no event shall a CLEC be required to pay for
both an interior security partition to separate SWBT's equipment in an Eligible Structure and a
security camera system for such Eligible Structure. Construction of interior security partition
shall not impair access to CLEC's equipment that is collocated under the cageless option.
E. Cage Preparation
Consists of the following elements and represents charges unique to the Collocator making
the request. Rates and charges are as found in paragraph 21.2 following.
- Grounded wire partition
- Door key Set
- Lights
- Outlets
- Cable rack and support structure inside the cage
- Cage sign
F. RSM Option
The additional Dedicated Heating Ventilating and Air Conditioning (HVAC) Charge consists
of the necessary dedicated ductwork extensions from the branch duct to the caged common
collocation area including downturns and diffusers required to handle the additional heat load
created by the RSM option. The Dedicated Power Plant Space Charge is a floor space rental
charge based on the square footage required for a power plant layout with batteries.
&
http://www.calstate.edu/CPDC/AE/TIP_Guidelines/TIP-Sec_2.doc
It is important to give the design and location of these spaces high priority within a
building plan. At a minimum, these spaces must meet the following design constraints:
In a multi-story building, these rooms must be stacked and should be centrally located,
reducing the distance from the room to all user locations.
These spaces must be dedicated to the telecommunications function and must not be
shared with electrical, janitorial, fire alarms, security systems, or storage functions.
The telecommunications rooms must be located near the center of the building but no
farther than 290 feet (cable pathway distance) from the most distant user outlet. The
average distance should be 150 feet or less.
The room must be designed and situated to eliminate overhead obstructions (including
false ceilings) and minimize any potential damage from items such as water or drain
pipes, electrical interference, dust or other airborne contaminants, and physical hazards.
The environment of these rooms must be equal to or better than a normal office (positive
air flow/cooling, office-level lighting, sealed or tiled floor - no carpet). These rooms are
intended to house terminal resources (network electronics) and must be equipped with

increased electrical service and additional cooling equipment to provide 24 hour a day, seven days a week support.
The minimum room size is ten feet by eight feet. Additional square footage should be provided if the space would need to accommodate optical fiber cable to individual station outlets and/or to house significant network routing or computing server equipment.
Each major building should be equipped with a conference room configured to be utilized as a teleconferencing or video-conferencing room. This will require additional acoustic material on the floor, ceiling, walls, and windows; storage space for equipment; enhanced power, lighting, and HVAC controls; blackout shades; and appropriate furniture.

QUESTION 9:

To provide the fastest response to an outage in the connection between the access layer, distribution layer and core of the Certkiller .com network, what protocols should be deployed?

A. L2 access layer with per-VLAN spanning tree (PVST) with an L2 distribution layer, with common spanning tree (CST) running with an L3 core, with OSPF in the core.
B. L3 at the access and distribution layers running OSPF across a loop-free L2 core with no spanning tree.
C. L3 at the access, distribution, and core layers, with OSPF as the routing protocol running on all devices.
D. L3 at the access layer running OSPF with the L3 core over the L2 distribution layer.
E. L2 at the access layer with per-VLAN spanning tree (PVST) with an L3 distribution layer, which runs OSPF with an L3 core

Answer: E

QUESTION 10:

During migration from Token Ring to Ethernet, routers will be deployed to allow access between the existing Token Ring and the new Ethernet networks. Where in the network should these routers be deployed?

A. At the core of the network so that each IDF can continue to use the FDDI backbone.
B. In the individual IDFs so that individual users can be migrated from the Token Ring to the Ethernet network to minimize each individuals down time.
C. At each MDF so that each IDF can be migrated separate and avoid possible wide-spread network outages.
D. In the computer room so that the Token Ring and Ethernet networks are only connected at one location to minimize risks.

Answer: C
This is the best place as it allows a better control of migration and reduces potential downtime.
Ref:

http://www.cdpa.nsysu.edu.tw/~zmx/dormnet/switchbook.pdf.
Fiber-Optic backbone from the MDF to each IDF.
In effect our plan is make each pod of classrooms function as a LAN. Each pod of classrooms is fed by an ATM switch, which via star topology is connected by fiber to the MDF. There are also Gigabit Ethernet switches at strategically positioned IDFs to feed each pod as well, thus providing parallel network paths.
Scalability
One of the criteria used for the selection of data equipment is its ability to scale to the growing network infrastructure needs of these educational institutions. In the classroom, selection of the Catalyst 3500 switches allows stacking of multiple units to provide more ports as necessary working as a "single" device. At the IDF, use of the Catalyst 5509 allows for the use of a combination 10/100 BaseT and Gigabit ports making it a very versatile closet switch. In some cases, we are using the Catalyst 3508 Gigabit aggregation switch instead of the Catalyst 5509 since that level of versatility is not need. At the MDF use of the Catalyst 5500 allows for a one box solution for collapsing both the Gigabit and ATM network. It offers a combination of 10/100 Base-T, Gigabit and ATM OC3 interface options that can be used to optimally use this as a Core Switch.

QUESTION 11:

Each IDF currently supports a single ring per floor to matter how many different departments are on that floor. Which VLAN deployment scheme would provide each department with load balancing, high availability, and security on a per-floor basis?

A. Each department would have a separate data and voice VLAN. Both departmental VLANs would be trunked over a single path from the IDF to the MDF.
B. Since most departments are small, each department would use a single VLAN for data and voice. Each departmental VLAN would be trunked from the IDF to the MDF over two paths, a primary and a backup.
C. Each department would use two VLANs, one for data and one for voice. These VLANs would be trunked over redundant uplinks from the IDF to the MDF.
D. Each floor would use two VLANs no matter how many departments are located there. The two VLANs would have a primary and backup path on each up link from the IDF to the MDF.

Answer: C
This fulfills all the requirements of the question, separate VLANS for each department provides security between departments, separate VLANS for data and voice also help with security. Using per vlan trunking will allow load balancing. Using redundant uplinks provides higher availability.
The primary function of the Campus module is to switch data, voice, and management traffic while enforcing the network and voice VLAN separation. The VLAN separation is augmented by the use of filtering on the Layer 3 switch and also a stateful firewall. HIDS are used to protect both key voice services and the PC-based IP phone hosts. The stateful firewall and the Layer 3 switch control the traffic flows between the data and voice

VLANs. The proxy server provides data services to IP phones; it also is located on the same VLAN as the call-processing manager. Private VLANs are used to mitigate local trust-exploitation attacks between the proxy server and the call-processing manager. For secure management, Layer 3 and Layer 4 filtering limits administration of key systems to authorized administration hosts. In addition, application-level security provides user authenti-cation and confidentiality.
Performance is not a limitation in this design because all devices are situated on a Fast Ethernet network. The only limitation to this design is the number of IP telephony devices that the call-processing manager can support. If the number of IP telephony devices exceeds the capacity of the call-processing manager, additional call-processing managers are required.
http://www.informit.com/articles/article.asp?p=360064&seqNum=4&rl=1

QUESTION 12:

Certkiller .com's traffic distribution is as follows:
Oracle -27%
Internal HTTP 11%

A. Queue 1 - Microsoft Office, external HTTP Queue 2 - UNIX RPC, Internet HTTP Queue 3 - voice signaling traffic, Oracle, ERP, overhead Queue 4 - voice bearer traffic
B. Queue 1 - voice bearer traffic Queue 2 - voice signaling traffic, Oracle, ERP, overhead Queue 3 - UNIX RPC, internal HTTP Queue 4 - Microsoft Office, external HTTP
C. Queue 1 - Microsoft Office, external Http Queue 2 - UNIX RPC, internal HTTP Queue 3 - voice bearer traffic Queue 4 - voice signaling traffic, Oracle, ERP, overhead
D. Queue 1 - Oracle, EPR, overhead Queue 2 - Microsoft Office, external HTTP Queue 3 - UNIX RPC, internal HTTP Queue 4 - voice signaling traffic, voice bearer traffic

Answer: A
Queue 4 is the high-priority queue, voice bearer traffic should be placed there and is processed before the lower priority queues. Voice signaling is placed in the next highest queue. Different data traffic types make up the queues 1 to 3 depending on their importance.

QUESTION 13:

Where in the existing network would the optimum location be to trust network devices to apply QoS correctly?

A. core layer switch
B. distribution layer switch
C. access layer switch
D. PC

Answer: C

A trust boundary is the point at which the device allowing traffic into the network either applies classification of traffic or recognizes the trust classification has been applied by the end station.
The primary issue when you are deploying QoS is where packets should be classified, which device is to be trusted to properly mark pakets. The three trust boundaries are distribution switches, access switches and IP phones.
The ideal trust boundary is at the IP phone, this is the most scalable solution and involves only a few configuration tasks.
Since IP Phone is not an offered choice, the Access Layer Switch becomes the most appropriate choice.

QUESTION 14:

IT wants to implement the simplest MoH transport mechanism, and is unconcerned about bandwidth consumption. What type of MoH transport mechanism will support this requirement, and how many simultaneous MoH sessions should be planned for? (Choose two)

A. multicast
B. unicast
C. 12
D. 13
E. 18
F. 36
G. 45

Answer: B, E
Unicast is easier to set up since multicast requires switches and routers that are multicast-enabled. If bandwidth was an issue, this would be a reason to select multicast.
The number of simultaneous MoH sessions is usually 2% of the total number of users.
Since we are talking about ~900 users, 18 is the correct answer.

QUESTION 15:

What is the maximum number of conferences that a s ingle DSP resource will support?

A. 6 if the callers are using medium complexity codecs, 3 for high complexity codecs.
B. 6 callers, there is no codec complexity issue.
C. 12 if the callers are using medium complexity codecs, 6 for high complexity codecs.
D. 12 callers, there is no codec complexity issue.

Answer: B
One conference DSP can support up to six participants, conferences cannot span multiple DSPs.
There is no concept of complexity in conferencing and transcoding.

For advanced activities such as conference calling and transcoding, the IP telephony system must have adequate DSP resources. With conference calls, the system must be able to simultaneously blend voice transmissions from a number of users. Likewise, most calls are transmitted across the WAN using a low bandwidth codec such as G.729a. At the opposite end of the WAN, transmissions will need to be transcoded to G.711 codec if the destination device does not "speak" G.729a. These functions require DSP resources. DSP resources can be added to the network by placing a DSP blade in the Catalyst 6000 or Catalyst 4000.

QUESTION 16:

What are the conferencing guidelines for a single-site deployment? (Choose two)

A. Use a single type of codec.
B. Use hardware conferencing only for small deployments.
C. Make certain the Meet-Me and Ad-hoc conference resources each account for a minimum of 5% of the user base.
D. Group any conferencing resources into MRGLs based on their location, to manage admission control.

Answer: A, C
For a single-site deployment it is recommended to use only G.711 codec for all endpoints.
General conferencing guidelines are to provide Ad-Hoc and Meet-Me conferencing resources for at least 5 percent of the user base.
Digital signal processing (DSP) resources: For advanced activities such as conference calling and transcoding, the IP telephony system must have adequate DSP resources. With conference calls, the system must be able to simultaneously blend voice transmissions from a number of users. Likewise, most calls are transmitted across the WAN using a low bandwidth codec such as G.729a. At the opposite end of the WAN, transmissions will need to be transcoded to G.711 codec if the destination device does not "speak" G.729a. These functions require DSP resources.
Ref:
http://www.ciol.com/content/flavour/voip/102052801.asp

QUESTION 17:

Certkiller .com has been looking at a number of new applications it would like to deploy on the company IP telephone system. They include a company directory on the IP phone, Personal Assistant, unified messaging, and a small call center. Which of the following applications would influence the choice of signaling and type for the PSTN connection?

A. small call center
B. company directory
C. Personal Assistant

D. Unified messaging

Answer: A
Call centers usually influence the choice of signaling and type for the PSTN connection. (example: number of voice channels needed, ANI/DNIS manipulation, etc)
Size of call center influence the type of signaling and type of pstn connection
IP PBX and Media Server
The IP PBX and media server is the core component of the IP voice solution. IP PBX and media server perform call processing capabilities and PBX features over the IP network infrastructure as well as extend and manage enterprise telephony features and capabilities to IP telephony network devices, media terminals, and applications such as media processing devices, messaging devices, IP phones, VoIP gateway, IVR, CTI applications, and so forth. The IP PBX and media server could be worked on single-site models and multi-site WAN models:
Single-site call processing model: In the single-site model, each site or campus has its own IP PBX or media server to perform call processing functions; also, there are no voice calls communication over the WAN network. If you want to implement external calls or call remote sites, you can use PSTN.
Multi-site WAN model with centralized call processing: In the multi-site WAN model, the IP PBX can either resides at a central campus or each site, and communication with remote branch offices or between sites normally takes place over the IP WAN or PSTN.
http://www.ciol.com/content/flavour/voip/102052801.asp

QUESTION 18:

In their business day, Certkiller .com has 35,422 total minutes of external traffic. Given that the busiest hour is 17% of the daily total, calculate the number of erlangs from Certkiller .coms daily call minutes (round to the nearest 10th)

A. 1.7
B. 60.2
C. 73.8
D. 100.4
E. 167.2

Answer: D
Formula is : Total minutes in the day multiplied by 17% and then divided by 60 gives you the Erlangs
So, 35422 X .17 = 6021.74 6021.74 / 60 = 100.36 (which is 100.4 when rounded up)

QUESTION 19:

In their busiest day, Certkiller .com has 35,422 total minutes of external traffic. Using the answer you determined in question 1, use the following erlan char snippets to determine the number of T1 circuits needed to connect Certkiller .com to the PSTN, assuming one call blocked on 100 attempts is acceptable.

A. They only need five trunk lines. Purchasing a T1 would be a waste of money.
B. They only need six trunk lines. Purchasing a T1 would be a waste of money.
C. 3
D. 4
E. 5
F. 6
G. 7
H. 8

Answer: E
Using the ErlangB calculator gives us a requirement of 118 lines.
One T1 had 24 DSOs (when using CAS) so 5 T1s will give us 120 lines.

QUESTION 20:

Certkiller .com has contacted its LEC to obtain an additional range of DIDs. The company's current DID range is 555-0000 through 555-0999. The LEC can provide Certkiller .com with an additional range of numbers, 556-0000 through 556-3999. The LEC is currently sending Certkiller .com four digits inbound. The two DID ranges overlap. What can be used to resolve this solution? (Choose two)

A. Contact an alternative ILEC to see if it can provide a DID range that does not overlap with the current range.
B. Ask the current LEC to send 5 digits.
C. Pick a number that would be used to prefix all existing three digit extensions.
D. Move to a 6-digit dial plan to provide more dialing granularity for all extension numbers.

Answer: B, C
Difficult to choose an answer since additional information may be required
A - This is the ideal solution but is unlikely to be possible in many cases
B - ?
C - The question says they use 4 digit dialing and this answer mentions using a number to prefix the existing three digit dialing. Manipulating the dialed number by making 4XXX route to the new 0000 - 0999 range would work. The routing would look like this:
0000 - 0999 à 555-0000 - 555-0999
1000 - 3999 à 556-1000 - 556-3999
4000 - 4999 à 556-0000 - 556-0999
D - This solution would set an access number and/or a site code followed by the 4 digits representing the number dialed. OR, it would just be the last 6 digits of the phone number to be dialed.
Possible solutions (when non-overlapping ranges are not available):
Move to a dial plan which uses more digits for on-net calls and set all sites to use this, regardless of if calls are for the local site or a remote site (5 or 6 digit dial plan)
Systems with overlapping site extension ranges can benefit from the use of a

variable-length dial plan with the following characteristics:
Within a site, the system retains the use of abbreviated dialing for calls to on-net extensions
Between sites, users dial an access code followed by a site code and the destination's on-net extension
Off-net calls require an access code followed by a PSTN number

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QUESTION 21:

Currently, Certkiller .com has not decided if it will implement an IP contact center for the technical support group. As an interim step, calls to the technical support group need to be distributed in a round-robin fashion. Can this be accomplished?

A. No, IP Contact Center is needed.
B. Yes, by using call pick-up groups.
C. Yes, by implementing hunt groups.
D. Yes, with route groups.
E. Yes, by routing calls via an IVR unit.

Answer: C
It is important to note that different releases of CCM used different terms to describe this capability and not all features were available in older CCM releases.
Release 3.3 and earlier - Hunt Groups
Release 4.0 - Route/Hunt List
Release 4.1 - Hunt Pilot
A route/hunt list consists of a pool of line groups and/or route groups. This route/hunt list is assigned to a hunt pilot number. When the hunt pilot receives a call, Cisco CallManager distributes the call sequentially through a list of line groups and/or route groups. The line group and/or route group then distributes the call to its members by following the associated hunt distribution algorithm and hunt options. Cisco CallManager disconnects the call if it is not routed to any of the hunt list members.
The hunt distribution algorithm specifies the way in which the call can be distributed through the members in a single line or route group. The hunt options specify the action to be taken if the call, once distributed to a line group member, is not answered or if the member is busy or unavailable (unregistered). Route groups do not have hunt options; they have only a distribution algorithm configuration option
You can assign an order to the devices within a route group, and Cisco CallManager will send calls to the devices in that specified order. If you want to use round-robin ordering for outgoing calls, set the orders of all devices in a route group to 1, and set the service parameter ReorderRouteList to True. With these settings, all route group members with the same order priority will then take turns being selected to route calls.
Ref:
http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_implementation_design_guide_chapter09186a00802c37f9.h

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QUESTION 22:

Which four of the following parameters must be considered of the Certkiller .com CallManager hardware selection? (Choose four)

A. The BHCA multiplier to device weights for busier devices.
B. The MoH transport mechanism.
C. The number of devices the cluster will support.
D. The dial plan weights for the cluster.
E. The type of Call Admission Control (CAC) the cluster will support.
F. Whether load balancing is required between servers in the cluster.

Answer: A, C, D, F
This question requires you to select a platform, not to determine how many platforms are required.
When selecting the platform, the number of device (C) and dial plan weights (D) to be supported are considered. The BHCA multiplier for busier devices (A) is important because it will increase the total device weight. Certain platforms do not support load balancing (F) so this possible requirement becomes a factor to consider.
The MoH transport mechanism (B) selection will affect the number of required platforms (standalone or co-resident) so this is not a correct choice for this question.
The type of CAC supported (E) plays no roll in the selection of the platform

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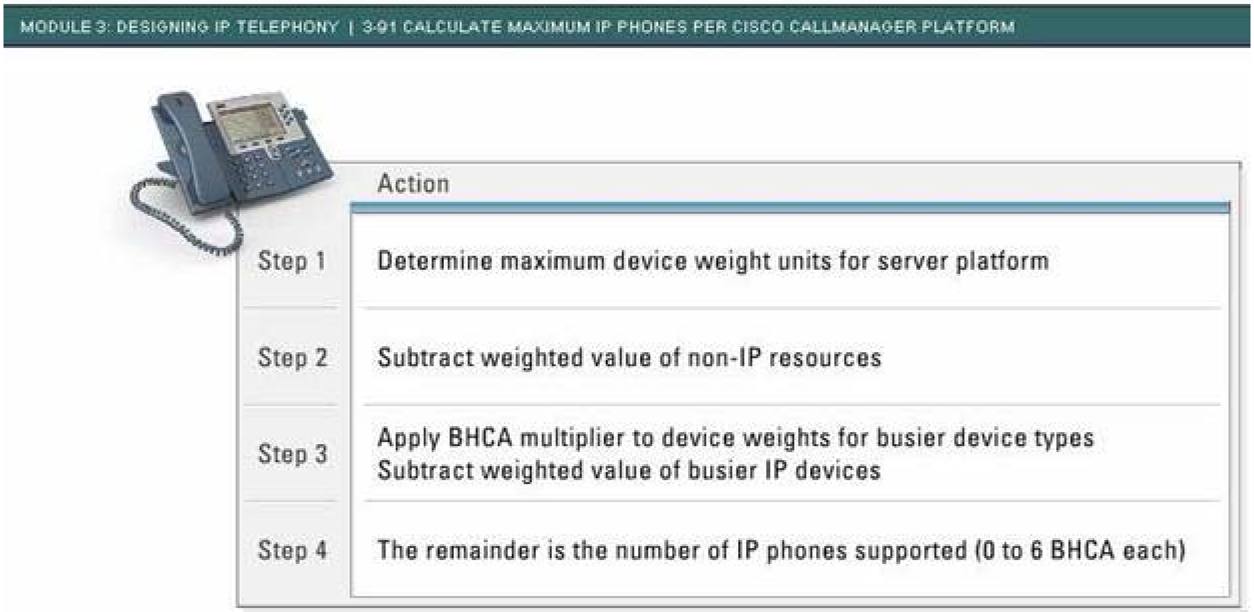
QUESTION 23:

One possible cluster arrangement for Certkiller .com is to use two MCS-7830 servers in a 1:1 redundancy scheme with 50-50 load balancing. The cluster will support 3000 device units and 1500 phones. You have calculated that one of the servers will be supporting 312 non-IP phone weight values. There will be 8 phones with a BHCA of 15 registered to this server along with 12 phones with a BHCA of 8. What is the maximum numbers of phones with a BHCA of 6 or fewer that this one server can support in this arrangement?

A. 390
B. 414
C. 438
D. 702
E. 1140
F. 1188
G. 1452

Answer: E

Explanation:
Formula is: 1500 - 312 - (8 x 3) - (12 x 2) = 1140



QUESTION 24:

When deciding upon a cluster server redundancy scheme, what must be considered? (Choose three)

A. The failover time for each scheme.
B. The impact on the users during system backups.
C. The likelihood of multiple primary server failures.
D. The impact on the users during software upgrades.
E. The number of IP phones registered on the subscribers.
F. Whether the publisher is dedicated or has devices registered to do it.

Answer: C, D, E
The 1:1 redundancy scheme allows upgrades with only the failover periods impacting the cluster. The failover time is the same (about 100 registrations per second) regardless of the scheme being used.
A 1:1 redundancy must be used when there are more then 10,000 IP phones since no single backup subscriber can have more than 10,000 backup registrations.
With a 2:1 redundancy, the cluster can function with one primary subscriber failure but not if both primary subscribers fail (in a 3 server cluster). Therefore the likelihood of this happening is a factor.
Number of IP phones registered on the subscribers is considered while deciding upon cluster server redundancy scheme.
Ref for E : swpat.ffii.org/pikta/txt/ep/1177/666/
system includes a loosely confederated network of server clusters along with any number of client terminals (i e., clients) that connect to the clusters. Terminals/clients can be software entities running under some operating system or any other device running on

some communication network that can have access to the cluster. Users are registered within some specific cluster and given a unique user ID. This user ID along with the ID of the cluster (CID) constitutes a globally unique user IDref for C //
With regard to being robust, given an error free run of the hardware, each server's uptime is preferably above 99.9%, and the uptime of the network preferably above 99.99% in certain embodiments. When exceptional errors occur, such as hardware errors, a maJcimum 3-5 min lag ls accepted. Essentially these state that when a server 3 is taken down, or breaks down, another server must automatically take over its role. Further, any single point of failure, such as databases or even hardware parts (such as networks) are preferably redundant and automatically taken over by other parts if they fail.

QUESTION 25:

In designing the CallManager cluster for high availability, what must you include in your design? (Choose two)

A. Redundant gateway connections to the PSTN.
B. Using a 1:1 redundancy scenario in the CallManager cluster.
C. The deployment of a dedicated publisher and dedicated TFTP server.
D. Dedicated servers for MoH and conferencing.
E. All members of the cluster needing to be in the same LAN or MAN.

Answer: B, C
1:1 redundancy is obvious
A dedicated publisher and TFTP server are also the best choices for high availability.
The other answers are not as crucial as the two correct answers
A cluster can contain as many as 8 servers, of which 6 are capable of call processing .the other 2 servers can be configured as dedicated database publisher and a dedicated TFTP server respectively

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QUESTION 26:

In the new Redmond office what must be verified from the perspective of the LAN before IP telephony can be deployed? (Choose three)

A. The number of public IP addresses available.
B. The type of wiring in the office.
C. The amount of rack space in the equipment rack.
D. The number of PSTN connections needed.
E. The amount of power that is available to support new LAN switches.

Answer: B, C, E
The type of wiring is very important, older category cabling may not be sufficient for IP Telephony. The equipment rack must have the space needed for new switches. Power considerations are important in two aspects: power to the IP phone and power for the switches in the rack

Ref:System Design
An IP telephony system must be designed taking into account the specific needs of the organization. A number of factors impact system design, including establishing a preferred mechanism for supplying power to clients, identifying desired PSTN interfaces, establishing adequate levels of QoS, and the need for advanced features, such as conferencing and transcoding.
Power supply: A key concern in designing an IP telephony system is often how to supply power to IP phones. Users have become accustomed to their phones receiving all necessary power from the PBX switch

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QUESTION 27:

In the traditional three-tier network design, at what layer will the Redmond office connect to the existing Certkiller .com network?

A. The Redmond office will become a sublayer to the access layer of the Certkiller .com campus.
B. The Redmond office will become another access layer attachment to the distribution layer of the Certkiller .com campus.
C. The Redmond office will become the equivalent to a server block and will attach to the core of the Certkiller .com network.
D. The Redmond office will become a WAN block attachment to the distribution layer of the Certkiller .com campus network.

Answer: D
The Redmond office connects over a WAN to the main site. The distribution layer connects to the two offices via the Enterprise Edge - WAN module.
http://www.cisco.com/warp/public/788/AVVID/ACU_casestudy.pdf
The connectivity method to the RNO differs slightly from state to state, as shown in the following table.
Victoria is based on Classical IP over ATM (RFC 1577) while the other RNOs have a straight PVC setup with
RFC 1483 encapsulation. The routing protocol used between ACU and RNO is Open Shortest Path First
(OSPF).
The IP phones are on one subnet, and the PCs are on another. The IP phone subnet is trusted by the AARNet core, and traffic to and from this subnet is subject to LLQ.
Ref 2
http://www.ciol.com/content/flavour/voip/102052801.asp
Client layer -- Desktop equipment such as IP phones, Cisco IP SoftPhones, and videoconferencing equipment
Infrastructure layer -- Gateways, routers, and switches
Call-processing layer -- Redundant call control and directories
Application layer -- Voice mail or unified messaging, personal productivity applications, and business productivity applications

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QUESTION 28:

When you analyze the distribution of the packet sizes from the Redmond facility, you see quickly that there is a potential for delay where voice packets could be trapped behind large data packets. Certkiller .com is looking to initially determine if the WAN circuit can be reduced in size to 384 Kbps. What is the recommended packet fragmentation size for a 384 Kbps WAN circuit using a G.729 codec for voice operating at 50 pps?

A. 48 bytes
B. 100 bytes
C. 288 bytes
D. 320 bytes
E. 480 bytes

Answer: E
Fragment size = (maximum allowed jitter x link speed in kbps) / 8
Max jitter is usually set to 10ms
Therefore, (10 x 384) / 8 = 480 bytes
At the WAN edge, the router or gateway prioritizes voice traffic, queuing it using priority queues and, in the case of low-speed WAN links, shaping the traffic using Link Fragmentation Interleaving (LFI) and compressed RTP (cRTP). To ensure QoS, the WAN backbone must not only have adequate capacity, but it must also be able to handle provisioning based on coder/decoder (codec) type. Typically a G.711 codec call consumes about 80 to 90 kbps of bandwidth depending on the media. For G.729a the corresponding consumption rate is 24 kbps, but with cRTP this can be brought down to about 12 kbps per voice call.
Ref: http://www.ciol.com/content/flavour/voip/102052801.asp

QUESTION 29:

There is a concern about voice quality over the link. In reviewing the estimated WAN traffic patterns, you see that voice will be approximately 45% of the traffic and there are no other real-time applications on the link. Which three of the following tools will help to insure that the bandwidth is used as efficiently as possible and the highest voice quality is attainted? (Choose three)

A. CBWFQ
B. cRTP
C. LFI
D. LLQ
E. Traffic policing
F. Traffic shaping

Answer: B, C, D
At the WAN edge, the router or gateway prioritizes voice traffic, queuing it using priority

queues and, in the case of low-speed WAN links, shaping the traffic using Link Fragmentation Interleaving (LFI) and compressed RTP (cRTP). To ensure QoS, the WAN backbone must not only have adequate capacity, but it must also be able to handle provisioning based on coder/decoder (codec) type. Typically a G.711 codec call consumes about 80 to 90 kbps of bandwidth depending on the media. For G.729a the corresponding consumption rate is 24 kbps, but with cRTP this can be brought down to about 12 kbps per voice call.
There are three separate areas of the network where QoS is established: at the campus level, at the WAN edge, and across the WAN backbone. At the campus level, Cisco routers identify traffic as voice and then prioritize calls to ensure priority over less time-sensitive traffic, such as typical data streams.
Ref: http://www.ciol.com/content/flavour/voip/102052801.asp

QUESTION 30:

With respect to the function of LLQ, how does the operation of the priority queue (PQ) differ from the class-based weighted fair queues (CBWFQ)?

A. Both queuing methods work in a similar manner. The difference is in the classification methods.
B. The CBWFQs are set to a minimum bandwidth that is policed. The PQ is set to a maximum bandwidth that is also policed.
C. The PQ is set to a maximum bandwidth that is polices. Each queue within the CBWFQ is set for e minimum bandwidth that can be metered on a per queue basis to control bandwidth consumption.
D. The PQ uses a minimum bandwidth that uses an internal metering method that drops packets as the bandwidth gets closer the maximum threshold. The CBWFQs are set to a minimum bandwidth per queue and controlled through metering as each queue reaches a maximum level.

Answer: C
priority Reserves bandwidth and provides Low Latency Queuing (LLQ) with CBWFQ
ref: www.informit.com/articles/ article.asp?p=358548&seqNum=4

QUESTION 31:

According to best practices, how should outgoing WAN traffic be queued so that all IP traffic will be handled properly?

A. Place voice bearer and signaling traffic in the high priority queue, UNIX RPC and Oracle into the medium priority queue, HTTP and Exchange/Outlook into the normal queue, all remaining traffic placed into the low priority queue.
B. Place a custom signaling traffic in the priority queue with the voice bearer traffic. Place the remainder if the traffic in a Custom Queue as follows. UNIX PC and Oracle will go into the normal queue, HTTP and Exchange/Outlook will go into the default queue, and the overhead traffic will go into the best-effort queue.

C. Place the voice bearer traffic in the AF41 queue, the voice signaling traffic in the AF 31 queue, UNIX RPC and Oracle traffic in the AF43 queue, and HTTP and Exchange/Outlook traffic in best-effort queue.
D. Place the voice bearer traffic in the priority queue, the UNIX RPC and Oracle traffic in the AF41 queue, the voice signaling traffic in the AF31 queue, the Exchange/Outlook traffic in the AF21 queue, and the HTTP traffic in the best effort queue.

Answer: D
QoS: Voice applications require a high level of QoS to avoid loss, delay, and delay variation or jitter. The term QoS refers to a collective set of tools used to prioritize competing IP traffic as well as ensure the quality of voice transmissions.
There are three separate areas of the network where QoS is established: at the campus level, at the WAN edge, and across the WAN backbone. At the campus level, Cisco routers identify traffic as voice and then prioritize calls to ensure priority over less time-sensitive traffic, such as typical data streams.
At the WAN edge, the router or gateway prioritizes voice traffic, queuing it using priority queues and, in the case of low-speed WAN links, shaping the traffic using Link Fragmentation Interleaving (LFI) and compressed RTP (cRTP). To ensure QoS, the WAN backbone must not only have adequate capacity, but it must also be able to handle provisioning based on coder/decoder (codec) type.

QUESTION 32:

It has been calculated that data will consume 60% of the available L3 bandwidth on the T1 circuit. What is the maximum number of voice calls that could be placed between Redmond and the Certkiller .com campus without using header compression?

A. 9
B. 17
C. 19
D. 23
E. 26
F. 32
G. 35

Answer: D

Explanation:
1500 - (1500 x .6) = 600
600 / 25.6 (Frame Relay) = 23.4

Codec	Ethernet	PPP	ATM	Frame Relay	VPN	V3PN
Header	18 bytes	6 bytes	5 bytes	6 bytes		
G.711 at 50 pps	85.6 kbps	82.4 kbps	106 kbps	81.6 kbps		
G.711 at 33 pps	77.6 kbps	75.5 kbps	84 kbps	75 kbps		
G.729A at 50 pps	29.6 kbps	26.4 kbps	42.4 kbps	25.6 kbps		
G.729A at 33 pps	22.2 kbps	20 kbps	28 kbps	19.5 kbps		

Bandwidth consumption per conversation for various transport types (includes Layer 2 overhead)

QUESTION 33:

What problem occurs if a user from the Redmond facility is temporarily assigned to the Atlanta campus, and the user takes their IP phone and uses it in Atlanta?

A. There is no problem. The phone will register with the same CallManager subscriber as before and operate correctly.
B. Every internal call placed by the user will cross the WAN to the Redmond gateway to be routed correctly.
C. Every call received by the user will first traverse the WAN link and be rerouted by the Redmond gateway.
D. Every on-net call placed by the user will reduce the available WAN voice bandwidth even if it does not cross the WAN link.

Answer: D
IF a user takes IP phone from one place and use it at other place, every on-net call placed by the user will reduce the available voice bandwidth even if it doesn't cross WAN link
On-Net: A term used in VoIp telephony services refers to calls over a 'virtual private network'.

QUESTION 34:

Even though the Atlanta campus is highly available, a distribution router failure could result in a telephone outage in Redmond. What can be done to minimize the impact of such a failure?

A. Configure CoR on the Atlanta WAN router.
B. Enable AAR on the Redmond WAN router.
C. Implement SRST on the Redmond gateway.
D. Start CallManager Express on the Redmond gateway when communications with Atlanta are lost.

Answer: C
Cisco Survivable Remote Site Telephony: When a central Cisco Call Manager cluster also handles call processing for users at distributed sites, Cisco SRST, a Cisco IOS software image for Cisco routers, can ensure continuous phone service. If a WAN link fails, Cisco SRST in the router provides basic Cisco Call Manager functionality until the link is restored.

QUESTION 35:

The Dallas, Denver, and Fresno regional sales offices have the same seven-digit telephone numbers. How can the dial plan be developed so that extension 1023 in Dallas doesn't overlap with extension 1023 in Denver and Fresno?

A. Use a two-digit site code and four-digit dialing.
B. Use eleven-digit dialing.
C. Dial the last seven DID digits.
D. Dial the last six DID digits.
E. Use translation patterns with four-digit dialing to route calls properly.

Answer: A

QUESTION 36:

There may be times when the IP WAN is out of capacity. What CallManager technology should be deployed so that calls can be completed when the WAN is at full capacity?

A. AAR
B. ARC
C. CAC
D. NEHO
E. TEHO

Answer: A
Automated Alternate Routing
The automated alternate routing (AAR) feature enables Cisco CallManager to establish an alternate path for the voice media when the preferred path between two intra-cluster endpoints runs out of available bandwidth, as determined by the locations mechanism for call admission control.
The AAR feature applies primarily to centralized call processing deployments. For

instance, if a phone in branch A calls a phone in branch B and the available bandwidth for the WAN link between the branches is insufficient (as computed by the locations mechanism), AAR can reroute the call through the PSTN. The audio path of the call would be IP-based from the calling phone to its local (branch A) PSTN gateway, TDM-based from that gateway through the PSTN to the branch B gateway, and IP-based from the branch B gateway to the destination IP phone.
AAR can be transparent to the users. You can configure AAR so that users dial only the on-net (for example, four-digit) directory number of the called phone and no additional user input is required to reach the destination through the alternate network (such as the PSTN).

QUESTION 37:

There may be times when the IP WAN is out of capacity. What CallManager technology should be deployed so that calls will not cross the WAN when the WAN is at full capacity?

A. AAR
B. ARC
C. CAC
D. NEHO
E. TEHO

Answer: C
CAC should be deployed so that call will not cross WAN when WAN is at full capacity.
CAC provides the ability to support resource-based call admission control processes. These resources include system resources such as CPU, memory, and call volume, and interface resources such as call volume.
If system resources are not available to admit the call, two kinds of actions are provided: system denial (which busyouts all of T1 or E1) or per-call denial (which disconnects, hairpins, or plays a message or tone). If the interface-based resource is not available to admit the call, the call is dropped from the session protocol (such as H.323).

QUESTION 38:

DRAG DROP

Drag and drop the terms to their definitions. Some terms by be used more than once.

Calling Search Space | Calling Restrictions | Partition

This defines which partitions are accessible to a particular device.	Place here	The dial plan entries that me be placed in this include IP phone directory numbers, translation patterns, route patterns, CTI route points and voice mail ports.	Place here
This refers to partitions and calling search spaces.	Place here	An attempt to dial a DN in a partition not listed here will fail or result in a busy tone.	Place here

Answer:

This defines which partitions are accessible to a particular device.	Calling Search Space	The dial plan entries that me be placed in this include IP phone directory numbers, translation patterns, route patterns, CTI route points and voice mail ports.	Partition
This refers to partitions and calling search spaces.	Calling Restrictions	An attempt to dial a DN in a partition not listed here will fail or result in a busy tone.	Calling Search Space

Explanation:
Understanding Partitions and Calling Search Spaces
A partition comprises a logical grouping of directory numbers (DNs) and route patterns with similar reachability characteristics. Devices that are typically placed in partitions include DNs and route patterns. These entities associate with DNs that users dial. For simplicity, partition names usually reflect their characteristics, such as "NYLongDistancePT," "NY911PT," and so on.
A calling search space comprises an ordered list of partitions that users can look at before users are allowed to place a call. Calling search spaces determine the partitions that calling devices, including IP phones, soft phones, and gateways, can search when attempting to complete a call.
When a calling search space is assigned to a device, the list of partitions in the calling search space comprises only the partitions that the device is allowed to reach. All other DNs that are in partitions not in the device calling search space receive a busy signal.

QUESTION 39:

From the following list select the devices that can be accessed by route groups. (Choose four)

A. H.248 intracluster trunks
B. MGCP gateways
C. SIP gateway
D. H.323 gateways

E. H.225 trunk
F. Intercluster trunk not gatekeeper controlled.

Answer: B, D, E, F
B, D:
MGCP, H.323 gateways are accessed by route groups.

Route Group Configuration Settings

Field	Description
Route Group Information	
Available Devices (select device, then select port below)	Choose a device in the Available Devices list box and add it to the Selected Devices list box by clicking Add to Route Group. If the route group contains a gateway that uses the QSIG protocol, only gateways that use the QSIG protocol display in the list. If the route group contains a gateway that uses the non-QSIG protocol, gateways that use the QSIG protocol do not display in the list. If you included the route group in a route list that contains QSIG gateways, the H.323 gateways do not display in the list.

If this device supports individually configurable ports, choose the port. (Devices that allow you to choose individual ports include Cisco Access Analog and Cisco MGCP Analog gateways and T1 CAS.) Otherwise, choose the default value (All or None Available, depending upon the device that is chosen). For a device that has no ports available (None Available), the device may be already added to the Route Group, or cannot be added to the route group.

QUESTION 40:

In a centralized call processing model, what is a disadvantage of using the tradition approach for building classes of service (as compared to the line/device approach)?

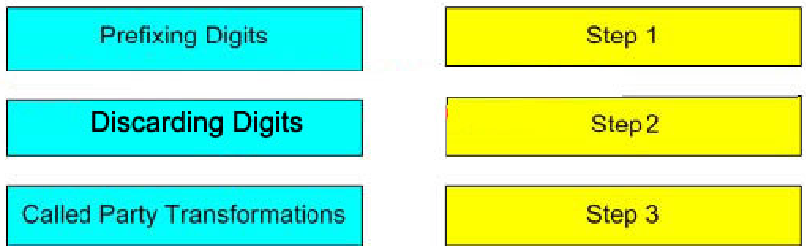
A. The minimum number of partitions = (number of classes of service) + (number of sites) + (1 partition for all IP phone DNs)
B. The maximum number of calling search spaces = (number of classes of service) + (number of sites)
C. The minimum number of partitions = (number of classes of service) * (number of sites) + (1 partition for all IP phone DNs)
D. The maximum number of calling search spaces = (number of classes of service) * (number of sites)

Answer: C

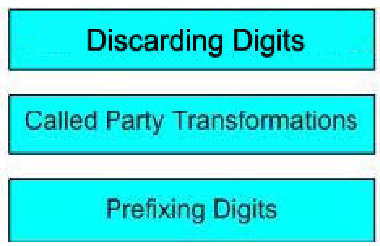
QUESTION 41:

DRAG DROP

Place in order the major types of digit manipulations that Cisco CallManager does.



Answer:



QUESTION 42:

A route pattern exists in Partition "Site1_Local" and an identical route pattern exists in Partition "HQ_Local". A calling search space containing "Site1_Local" is assigned to a device and a calling search space containing "HQ_Local" is assigned to a line on that device. How does CallManager determine which route pattern to use?

A. CallManager uses only the CSS associated with the device.
B. CallManager uses only the CSS associated with the line.
C. CallManager concatenates the two CSSs and gives priority to partitions in the CSS assigned to the line.
D. CallManager concatenates the two CSSs and gives priority to partitions in the CSS assigned to the device.
E. CallManager does a logical OR of the partitions in the two CSSs and searches for the longest match.

Answer: C

QUESTION 43:

Which gateway signaling protocol will provide Certkiller .com with the highest call survivability at the remote sites in the event of a WAN failure?

A. H.323
B. SIP
C. MGCP
D. SCCP

Answer: C
If the call involves only survivable endpoints and one or more Cisco Call Managers fail, the streaming connections between the endpoints is maintained. However, the endpoints do not have call-processing services available to them after the failure (transfer, conference, hold, park, pickup and resume). In general, MGCP gateways provide the highest degree of call survivability.

QUESTION 44:

In the event of a WAN failure, Certkiller .com would like the regional offices to be able to continue to function without having to resort to cell phones. Which technology will be in use during a WAN failure, and which gateway protocol will be used? (Choose two)

A. CME
B. SIP
C. H.323
D. SRST
E. MGCP

Answer: D, E
Cisco Survivable Remote Site Telephony: When a central Cisco Call Manager cluster also handles call processing for users at distributed sites, Cisco SRST, a Cisco IOS software image for Cisco routers, can ensure continuous phone service. If a WAN link fails, Cisco SRST in the router provides basic Cisco Call Manager functionality until the link is restored.
If the call involves only survivable endpoints and one or more Cisco Call Managers fail, the streaming connections between the endpoints is maintained. However, the endpoints do not have call-processing services available to them after the failure (transfer, conference, hold, park, pickup and resume). In general, MGCP gateways provide the highest degree of call survivability.

QUESTION 45:

Certkiller .com would like to conserve as much bandwidth as possible when faxes are sent across the WAN. Which two methods of fax transmission will satisfy that requirement? (Choose two)

A. Cisco fax relay
B. Cisco fax pass-through
C. T.4
D. T.30
E. T.38

Answer: A, E
The T.37/T.38 Fax Gateway functionality provides store-and-forward fax and fax relay

support on the voice ports of the AS5300 voice gateway. These capabilities are available on both the C542 and C549 voice ports. This functionality allows dynamic switching from one application to another in the same call (IVR, fax relay, and fax store and forward).
T.38 preserves the semantics of faxing over the PSTN by establishing a real-time fax connection across the IP infrastructure. T.37 converts fax data to a TIFF attachment of an e-mail message and forwards via SMTP.

QUESTION 46:

At the Indianapolis facility you plan to have a 512 kbps Frame Relay connection. The codec type used on the WAN is G.729 with cRTP. What is the maximum number of calls that this link can support when a fax pass-through call is in progress, and 192Kbps is reserved for data? Assume best practices for overhead are being used.

A. 13
B. 14
C. 15
D. 16
E. 17
F. 18

Answer: B

QUESTION 47:

In what locations will DSP resources need to be configured to minimize WAN traffic and support conferencing and transcoding?

A. The only location that will need DSP resources will be the Certkiller .com campus in Atlanta.
B. DSP resources will only be required at the regional sales office to convert G.711 to G.729 for WAN access.
C. DSP resources will be required at the central site to support transcoding for the voice mail system, and at all sites to support conferencing.
D. DSP resources will be required at the central site to support conferencing, and at the remote sites to support transcoding for the voice mail system.

Answer: C
DSP resources will be required at central site to support Transco ding for voicemail system
The number of DSPs needed for each voice interface depends on the following two factors:
Codec complexity
Type of codec selected

DSP firmware is included with each IOS release for the Cisco ICS 7750. Five DSP firmware images are available for use on ASI and MRP cards. Two of the DSP firmware images are intended for MRPs (which contain analog VICs) and for ASI cards (which contain FXS ports); two images are intended for digital trunks (such as T1 CAS and T1/E1 PRI); and one image is intended for transcoding.
Each DSP firmware image supports a particular set of codecs. High-complexity DSP firmware supports more codecs than medium-complexity firmware supports. However, in order to support more codecs, the number of voice channels supported by the firmware has to be reduced.
Codec InteroperabilityCodec interoperability is the ability of one codec to decode another codec. If a DSP is configured with a certain codec, it should be able to decode voice coded using any codec with which it is interoperable.
The following G.729 codec combinations interoperate:
* G.729 and G.729a
* G.729 and G.729
* G.729a and G.729a
* G.729 Annex-B and G.729a Annex-B
* G.729 Annex-B and G.729 Annex-B
* G.729a Annex-B and G.729a Annex-B
The following G.723.1 codec combinations interoperate:
* G.723.1 (5.3 kbps) and G.723.1 (6.3 kbps)
* G.723.1 (5.3 kbps) and G.723.1 (5.3 kbps)
* G.723.1 (6.3 kbps) and G.723.1 (6.3 kbps)
* G.723.1 Annex-A (5.3 kbps) and G.723.1 Annex-A (6.3 kbps)
* G.723.1 Annex-A (5.3 kbps) and G.723.1 Annex-A (5.3 kbps)
* G.723.1 Annex-A (6.3 kbps) and G.723.1 Annex-A (6.3 kbps)

QUESTION 48:

According to best practices, what should the two CallManager clusters use to communicate with each other?

A. H.225 trunk
B. Intercluster trunk, non-gatekeeper controlled
C. Intercluster trunk, gatekeeper controlled
D. SIP trunk

Answer: C
"Use gatekeeper-controlled intercluster trunks to route calls between Cisco CallManager clusters. This practice enables you to add or modify clusters easily in your network without reconfiguring all other clusters.

QUESTION 49:

What additional best practice should be adhered to for the distributed call-processing environment?

A. Use only one type of codec on the LAN.
B. Provide a highly available gatekeeper.
C. Have a gateway resolve the E.164 addresses to IP addresses.
D. Build full-mesh connectivity between Atlanta, Redmond, and the LT main office in Tampa.

Answer: B
Considerations for Distributed Call Processing Deployments
Observe the following best practices when designing a dial plan for a distributed call processing deployment (that is, any multi-site deployment involving multiple CiscoCallManager clusters):
Avoid splitting DID ranges across multiple Cisco CallManager clusters. This practice would make intercluster routing very difficult because summarization would not be possible. Each DID range should belong to a single Cisco CallManager cluster.
Avoid splitting devices within a remote site between multiple Cisco CallManager clusters. Locations-based call admission control is significant only within a cluster, and having devices belong to different clusters at the same remote site would lead to poor utilization of the IP WAN bandwidth because you would have to "partition" the available bandwidth between the clusters. Each remote site should belong to a single Cisco CallManager cluster.
Use gatekeeper-controlled intercluster trunks to route calls between CiscoCallManager clusters. This practice enables you to add or modify clusters easily in your network without reconfiguring all other clusters.
Implement redundancy in the connection between Cisco CallManager and the gatekeeper by using a gatekeeper cluster and by assigning the intercluster trunk to a device pool that uses a CiscoCallManager group with multiple servers configured.
When sending calls to the gatekeeper, expand the called number to the full E.164 address. This practice simplifies PSTN failover when the IP WAN is not available because no additional digit manipulation is required to reroute the call via a PSTN gateway. Also, this practice eliminates the need to configure the local (calling) Cisco CallManager with dial length information for each remote site.
Within the gatekeeper, configure one zone per Cisco CallManager cluster. For each cluster/zone, add zone prefix statements to match all DN ranges owned by that cluster.

QUESTION 50:

How should the dial plan be configured to support failover if the gatekeeper rejects a call because of insufficient bandwidth?

A. A route pattern should point to a route group to the IP WAN as the first choice and a second route group to the PSTN as a second choice.
B. A route group should point to the gateway connected to the H.323 inter-cluster trunk and also point to a second gateway connected to the PSTN as a second choice.
C. A route pattern should be configured for the IP WAN and a second route pattern should be configured for the PSTN as a second choice.

D. A rote list should be configured to point at a route group to the IP WAN as the first choice and a route group to the PSTN as the second choice.

Answer: D
A route list is configured to point at the route group to IP WAN at the first choice while configuring dial plan supports failover due to insufficient bandwidth and route group to PSTN as second choice
You can implement Tail-End Hop-Off (TEHO) across multiple Cisco CallManager clusters by following these guidelines:
Add specific route patterns for the relevant E.164 ranges to the source (originating) CiscoCallManager cluster, and point them to a route list that has the IP WAN route group as the first choice and a local PSTN route group as the second choice
Within the CiscoIOS gatekeeper configuration, add zone prefix statements for all the relevant E.164 ranges and point them to the appropriate Cisco CallManager cluster
Ensure that the intercluster trunk calling search space in the destination Cisco CallManager cluster includes partitions featuring route patterns that match the local PSTN numbers, and that digit manipulation is applied as needed (for example, stripping the area code before sending the call to the PSTN).

QUESTION 51:

Both the Certkiller .com and LT organizations have fully functional IP telephony solutions. They both use an access code with a two-digit site code followed by a four-digit extension to dial internally. How should the dial plan be designed for calls dialed from Certkiller .com to a DN at Logoon Technologies?

A. Use a different access code than used for on-net dialing and create a translation pattern to reach the DN at LT.
B. Use a fully qualified E.164 address and send it to the gateway for resolution.
C. Send a fully qualified E.164 address to the gatekeeper for resolution.
D. Deploy a new route group that use E.164 addresses and the IP WAN as the primary path and the PSTN if the IP WAN is busy.

Answer: C
Company with Fully IP telephony solution should Send fully E.164 address to gatekeeper for resolution to design dial plan with another company
Design of dial plans requires knowledge of the network topology, current telephone number dialing patterns, proposed router locations, and traffic routing requirements. No standard protocol is defined for the dynamic routing of E.164 telephony addresses. Until a standards-based dynamic routing protocol for E.164 telephony addresses is developed, H.323 VoIP dial plans are statically configured and managed on gateway and gatekeeper platforms.
Overview of the Dial Plan
The dial plan is the method by which individual blocks of telephone numbers (technically, E.164 addresses) are assigned to physical facilities, or circuits. For large-scale service provider networks, dial plans consist of the following elements:

* A grouping of E.164 prefixes with respect to zones and zone GKs
* An assignment of E.164 address blocks to POPs and POP GWs
* The normalization (number translation, prefixing, and digit stripping) of telephone numbers at the POP GWs
* The establishment of POTS and VoIP dial peers at the GWs

A dial plan is a numbering plan for the voice-enabled network. It is the way that you assign individual or blocks of telephone numbers (E.164 addresses) to physical lines or circuits. The North American telephone network is based on a 10-digit dial plan consisting of 3-digit area codes and 7-digit telephone numbers. For telephone numbers located within an area code, a 7-digit dial plan is used for the Public Switched Telephone Network (PSTN). Features within a telephone switch (such as Centrex) support a custom 5-digit dial plan for specific customers that subscribe to that service. PBXs also support variable-length dial plans, containing from 3 to 11 digits

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QUESTION 52:

The Certkiller .com domain coordinator is interested in determining how the CallManager servers can be secured. Which three unused Windows services need to be disabled on subscriber servers? (Choose three)

A. Network Time Protocol
B. IIS
C. DHCP
D. TFTP
E. NTFS
F. The registry

Answer: B, C, D
During CCM installation, most services and security are configured correctly. However, there are a few additional services that can be disabled to improve the security of CCM. Many computer attacks are against IIS and since it is recommended that all administration be done on the publisher server, IIS should be disabled on all subscribers in the CCM cluster. IIS is needed during software upgrades so the administrator must remember to re-enable it when upgrading. In addition, DHCP and TFTP can be disabled on all of the servers except where they are specifically needed.
Like any other application server, administrators should disable unused services. For example, CallManager uses a small set of services including DHCP, TFTP and the World Wide Web service which must remain enabled, but other unused services should be stopped. In addition, host-based intrusion prevention software can be used to provide further protection.
Within IIS, administrators should explicitly grant and deny access using virtual directory ACLs. Also, password policies on the Windows server should dictate a minimum length of seven characters and carry mandatory

requirements for numbers and special characters as
well. Administrators can also configure settings to stop
known PBX exploits such as voice-mail transfers to, forward
to, and transfer to long-distance or pay-per-use
phone numbers
Incorrect Answers:
A: Network Time Protocol is not a security risk.
E: NTFS is a file system, not a service.
F: The registry is not a service.

QUESTION 53:

The Certkiller .com telephony coordinator is concerned that the current E911 services are not properly configured, and wants to correct that situation. What needs to be considered at the Certkiller .com campus? (Choose two)

A. The square footage of each floor in each building.
B. The number of employees per floor.
C. The distance from the PSAP.
D. The number of digits that are sent as the CLID to the ERL.
E. The number of ESZs per building.

Answer: A, B
Most CMRS providers do not have a formal contingency plan that defines where employees are to report for work if their primary work place is inaccessible, for whatever reason. Some CMRS providers have indicated they are investigating this issue and that they intend to develop plans in the future.
Some alternate technology providers have supplied their critical employees with portable computers, equipped with high speed modems, for them to remotely access switches. Critical employees are expected to work from a location that allows them access to the switches during times of emergencies/disasters.

QUESTION 54:

What are four general E911 responsibilities of an enterprise telephony system? (Choose four)

A. Provide a detailed map of all ERLs.
B. Enable PSAP call-back to 911 call originator.
C. Allow conferencing with internal security personnel.
D. Deliver appropriate CLID digits to LEC.
E. Route calls to the appropriate point (on-net and off-net)

Answer: A, B, D, E
PSAP communications links, whether single, redundant, or diverse. These links include those from the PSAP to the Automatic Location Identification (ALI) database and, if

applicable, to police, fire, emergency medical services, poison control, trauma centers, the local media, and a dedicated ring down line to the LEC to report problems in network communications during emergencies.

QUESTION 55:

Which two types of interfaces can be used to support dynamic ANI for E911? (Choose two)

A. FXO
B. E&M
C. PRI
D. CAMA

Answer: C, D
PMI and CAMA interfaces can be used to support dynamic ANI for E911.
The dynamic aspect of ANI refers to the fact that a system has many phones sharing access to the 911 network across the same interface, and the ANI transmitted to the network might need to be different for each call.
There are two types of dynamic ANI interfaces:
1) Integrated Services Digital Network Primary Rate Interface (ISDN-PRI, or simply PRI)
2) Centralized Automatic Message Accounting (CAMA).

QUESTION 56:

The Certkiller .com campus crosses county boundaries. Suppose the two counties are in different area codes, but are serviced by the same LEC. If Certkiller .com uses two PRI interfaces to make E911 calls to the different PSAPs, what must be correctly planned for?

A. The ELINs must be properly mapped in each ANI database.
B. The ALI numbers provided yield the appropriate routing and CPN lookup.
C. Each county PSAP must maintain ALI databases information for both sides if the Certkiller .com campus.
D. It must be ensured that the CPN will be used for ANI and falls within a range of numbers acceptable on the link.

Answer: D
If two PRI are used to call 911 by same LEC.
CPIN will be used for ANI and falls within the range if numbers acceptable on the link

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QUESTION 57:

Which four attributes must be considered when designing a secure IP telephony solution? (Choose four)

A. Stateful inspection of all packets on the corporate intranet.
B. Creation and assignment of VLANs and broadcast domains.
C. Implementation of packet filters and the establishment of firewalls.
D. Protection of voice at Layer 2.
E. Placement of application layer gateways.
F. Reserve NAT and stateful inspection to segments that connect to the Internet.

Answer: B, C, D, E
B: * Mixture of voice-only and data-capable CPE on routers/switches in corporate network
* Specific assumption that voice-only CPE (IP phone, access gateway) are in different VLANs from data CPE (e.g., PCs)
- Separate sub-nets allow different virtual LANs (VLANS) to be defined.
- Each VLAN is engineered differently, based on voice vs. data needs.
* On each router/switch, there are two VLANs:
- One VLAN for voice-only CPE (IP phones and access gateways).
- Second VLAN for data-capable CPE (PCs, PCs with soft IP phone, or IP phone with PC attached).
* A IP phone which supports a PC plug-in would need to be supported on Data VLAN because it has data traffic. No voice quality guarantees in this case
C: Firewalls if present in any proposed voice path over the data network must be configured to handle H.323 version 2 protocols.
D: Building an IP Telephony system requires the use of an IP infrastructure based on Layer 2 and Layer 3 switches and routers, with switch connections to the desktop. In this environment the use of hubs is not recommended since the IP phones should not be in the same broadcast area. Voice traffic destined for one phone must not be broadcast to all phones in the same broadcast area. Servers and gateways must be on individual switched ports to minimize the collision domain of the media and prevent performance degrading delay
E: With Cisco PIX Firewalls, users can take advantage of larger address classes than those they may have been assigned by the Internet's Network Information Center (NIC). Cisco PIX Firewalls provide this access through its NAT facility, described by RFC 1631. Cisco PIX Firewall versions 6.0 and 6.1 offer SIP support that solves NAT traversal issues by acting as an application-layer gateway (ALG).

QUESTION 58:

From a perspective of securing an IP telephony solution, what is a benefit of deploying separate subnets for IP phones and their attached PCs?

A. Prevents an attacker or attacking application from snooping on a common wire.
B. Preserves stateful inspection of application packets.
C. Reduces granularity of QoS deployment.
D. Conserves IP addresses.

Answer: A
IP Addressing

Within IP Telephony systems each IP phone requires an IP address, along with associated information such as subnet mask, default gateway and so on. Essentially, this means that the organization's need for IP addresses increases as IP phones are assigned to users. This information can be configured statically on the IP phone or provided by the Dynamic Host Configuration Protocol (DHCP).
1. Creating a Separate IP Subnet for IP Phones
The IP phones can also be put on a separate IP subnet. The new subnet could be in a registered address space or in a private address space, such as network 10.X.X.X. Using this scheme, data devices would be on a subnet reserved for only data and the IP phones would be on a subnet reserved for voice. Configuration on the IP phone can be minimized by having the phone learn as much information dynamically as possible. The IP phone when powered up can get its voice subnet automatically, the send a DHCP request on that subnet for an IP address.

QUESTION 59:

Which of the following technologies will allow you to design a solution for protection voice at Layer 2? (Choose three)

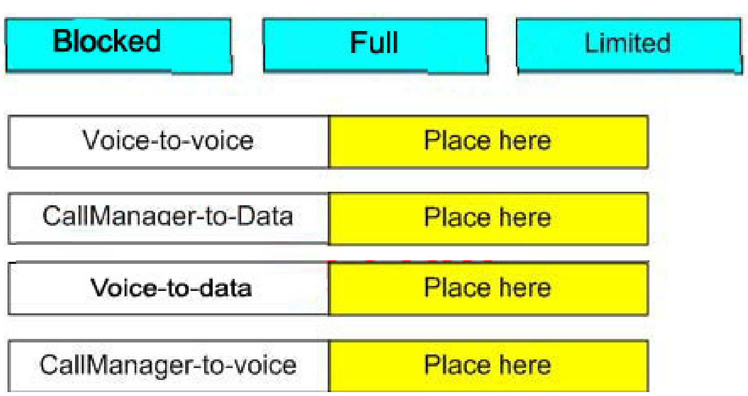
A. IP source guard
B. Internal VPNs
C. DHCP snooping
D. RFC 1918 addressing
E. Locally administered MAC addresses
F. Secure ARP detection

Answer: A, C, F
C: The IP phones can also be put on a separate IP subnet. The new subnet could be in a registered address space or in a private address space, such as network 10.X.X.X. Using this scheme, data devices would be on a subnet reserved for only data and the IP phones would be on a subnet reserved for voice. Configuration on the IP phone can be minimized by having the phone learn as much information dynamically as possible. The IP phone when powered up can get its voice subnet automatically, the send a DHCP request on that subnet for an IP address.
A, F: The trust boundary has shifted to the distribution layer. If is more than likely that there is a high-end switch in the distribution layer with features to support this function. If possible this function should not be performed in the core of the enterprise network. The trust boundary should be kept as close to the source as possible.
The following is a summary of the generic guidelines discussed above. These guidelines are independent of any specific enterprise network configuration and should apply to any network type.
1. Switched Layer 2 infrastructure. Use of Hubs in the Enterprise LAN should be avoided.
2. 100 mB Ethernet ideal, 10mB Ethernet may be used if it is determined that it provides the required traffic capacity.
3. VoIP using a Token ring or slotted ring network should be avoided

4. Category-5 Cabling is recommended (Category-3 or Coaxial is not recommended)
5. IP address may be public or private. A NAT or PAT devices, if provided, must be H.323v2 aware.
6. The enterprise must supply a Router to connect to the WAN Network; this router must support the associated WAN interface (channelized DS3, ATM, IP, SONET, etc.)
7. The enterprise network should allow the Service Provider to verify connectivity through the network to a CPE.
8. It is desirable that Routers and Layer 3 Switches in the Enterprise LAN should have the capability to support either IP Precedence or DiffServ QoS. The type of QoS supported will depend upon the Service Provider WAN Network.
9. If the enterprise has a firewall that faces the WAN network, either this firewall must support H.323v2, or else the enterprise must open up a range of UDP ports and selected TCP ports.
10. It is desirable that all Layer 2 Switches in the enterprise network support 802.1p hence ARP is configured
11. It is desirable the all Layer 2 & 3 Devices in the enterprise network support 802.1q VLANs
12. Prior to installation and during operation the corporate network should be analyzed to insure that the transport interface provides an appropriate quality of service for that traffic. Analysis should be made to determine the characteristics of the corporate network for:
13. 1. Latency
2. Jitter
3. Packet Loss

QUESTION 60:

DRAG DROP
An internetwork has three VLANs to support voice and data traffic, one for voice, one for data and one for the CallManager cluster. How should inter-subnet connectivity be deployed? Drag and drop the type of connectivity to each type of connection.



Answer:

Voice-to-voice	Full
CallManager-to-Data	Limited
Voice-to-data	Blocked
CallManager-to-voice	Full

QUESTION 61:

You are using an H.225 gatekeeper controlled trunk between a CallManager cluster and a gateway. When designing the security component for the CallManager clusters, what can result if you use a firewall between a CallManager cluster and the voice and data VLANs? (Choose two)

A. clipping of the video stream
B. echo
C. call setup failure
D. one-way audio
E. choppy audio

Answer: C, D
Firewalls and NAT: How do they affect QoS? Firewalls are not part of the H.323 standard.
Networks deploy firewalls in order to prevent unauthorized use of their network and to defend
against hackers and others who might disrupt their network operations. Because their primary
purpose is to limit access, this presents a problem for those who want to talk. Some workarounds
are:
Open ports in the firewall to allow for telephony traffic. Not Recommended!
Choose firewalls that support dynamic negotiation of ports to allow for telephony traffic while minimizing security risks in statically opening ports. (Example: Cisco's PIX)
Cisco H.323 proxy can be used to provide a secure way of tunneling through or around firewalls.
Those with firewalls should check with their vendor for the most current H.323 update.
Use intelligent firewalls that understand voice.
Stateful inspection firewall functionality is also needed to manage today's network protocols effectively.
Traditional packet filters cannot handle complex data protocols such as IP streaming and videoconferencing, which use multiple UDP and TCP ports and complicated initiation schemes. To be effective, the stateful inspection firewall must have specific knowledge of the protocols it manages.

Stateful inspection can be used for voice signaling protocols including SIP, SCCP, H.323 and MGCP on firewalls and within routers and switches. This results in increased security, as the firewall can inspect the initial signaling packet, discover the UDP port used by the RTP stream, open this pinhole for the communication, and then watch for the end-of-call signal to close the connection. This ensures that connections are not left open for hackers to exploit transmissions are secured without any additional configuration at each remote office

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QUESTION 62:

Which four methods can be used to protect IP phones? (Choose four)

A. Dedicate a TFTP server for each voice network segment.
B. Use a firewall with the CallManager cluster to prevent attacks from inside and outside the organization.
C. Disable the PC port.
D. Use signed loads.
E. Prevent/Restrict/DisableGARP.
F. Disable the Settings button.

Answer: C, D, E, F
E: Other kinds of attacks come through the network rather than through an attached PC. By default, most IP phones accept Gratuitous ARP (GARP) packets. GARP, which is used by some legitimate network devices to announce their presence, can also be used by attackers to spoof a valid network device. For example, an attacker could send out a GARP that claims to be the default router. Most administrators choose to disable Gratuitous ARP
F: Administrators can also configure settings to stop known PBX exploits such as voice-mail transfers to, forward to, and transfer to long-distance or pay-per-use phone numbers

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QUESTION 63:

Which one of the following represents a best practice for the distributed call-processing environment?

A. For the gatekeeper to be aware of the topology, use a physical hub and spoke topology.
B. Provide high availability for the gateways.
C. Use a gatekeeper to resolve the E.164 addresses to IP addresses.
D. Use only one type of codec on the WAN because the MGCP specification does not allow for header overhead in the bandwidth requests.

Answer: D

Explanation: Use only one type of codec on the WAN because the H.323 specification does not allow for L2 IP, UPD, or RTP header overhead in the bandwidth request. Using one type of codec on the WAN simplifies capacity planning by eliminating the need to overprovision the IP WAN to allow for the worst-case scenario.

QUESTION 64:

How should the two CallManager clusters be configured so that IP phone calls can be placed between them?

A. Both should be configured as H.323v1 devices and should be configured as intra-cluster H.225 devices.
B. Communications between sites with CallManager clusters requires H.232 and each cluster must be configured as an intra-cluster H.323 device.
C. Both CallManager clusters require gatekeeper control and the specific gateway to be queried.
D. Both CallManager clusters require H.323v2 and should be configured as inter-cluster H.323 devices with gatekeeper control.

Answer: D

Explanation: Communication between sites that have CCM or CCM clusters require the use of H.323v2. This means that a remote CCM or CCM Cluster must be configured as an intercluster H.323 device. Within the CCM configuration for an H.323 device, configure the device to be gatekeeper-controlled and specify the gatekeeper to be queried. This configuration means that before the CCM sets up a call with a remote CCM, it must first send an ARQ with the requested bandwidth to the gatekeeper.

QUESTION 65:

Certkiller 's telephony coordinator believes that most conferences will be local to each site. Which deployment model for resources to support conferencing and transcoding will fit the requirements for Certkiller ?

A. Local
B. Centralized
C. Decentralized
D. Hybrid, the larger sites will use decentralized DSP resources and the smaller sites will use centralized DSP resources.

Answer: D

QUESTION 66:

At the Dallas regional office you plan to have a 128 kbps Frame Relay connection with cRTP enabled. Voice calls crossing the WAN will use the G.727 codec. Considering that L3 bandwidth consumption should not exceed 75% of the link bandwidth, what is the maximum number of calls that this link can support when a fax pass-through call is in progress?

A. 0
B. 1
C. 2
D. 3
E. 4
F. 5
G. 6

Answer: A

Explanation:
For question 65 "At the Dallas Regional Office .." the question is regarding the fax passthrough mode. How many calls could go across the link with a fax pass-through call in progress?
Based on the following parameters, I chose answer A or 0.
1. 128K Frame Relay Connection
2. cRTP Enabled
3. L3 Total Bandwidth should not exceed 75%. Data + voice. 96kbps.
4. Fax Pass-Through Call In Progess
Key is the fact that a fax pass-through call is the equivalent bandwidth-wise of a super-sized G.711. Cisco documentation says:
"On detection of a fax tone on an established VoIP call, the gateways switch into fax pass-through mode by suspending the voice codec and loading the pass-through parameters for the duration of the fax session. This process, called upspeeding, changes the bandwidth needed for the call to the equivalent of G.711.
With pass-through, the fax traffic is carried between the two gateways in RTP packets using an uncompressed format resembling the G.711 codec. This method of transporting fax traffic takes a constant 64-kbps (payload) stream plus its IP overhead end-to-end for the duration of the call. IP overhead is 16 kbps for normal voice traffic, but when switching to pass-through, the packetization period is reduced from 20 ms to 10 ms, which means that half as much data can be put into each frame. The result is that you need twice as many frames and twice as much IP overhead. For pass-through, the total bandwidth is 64 plus 32 kbps, for a total of 96 kbps. For normal voice traffic, total bandwidth is 64 plus 16 kbps, for a total of 80 kbps."
There is no mention at the Cisco web site that this RTP fax pass-through call can

be compressed with cRTP.
Therefore, 96kbps - 96kbps = 0 kbps left for calls.

QUESTION 67:

How many voice calls could be placed between Redmond and the Certkiller campus, assuming no header compression is used and data consumes 45% of the L3 T1 bandwidth?

A. 9
B. 17
C. 19
D. 24
E. 36
F. 32
G. 35

Answer: F

Explanation:
1500 - (1500 x .45) = 825
825 / 25.6 (Frame Relay) = 32.22

Codec	Ethernet	PPP	ATM	Frame Relay	VPN	V3PN
Header	18 bytes	6 bytes	5 bytes	6 bytes		
G.711 at 50 pps	85.6 kbps	82.4 kbps	106 kbps	81.6 kbps		
G.711 at 33 pps	77.6 kbps	75.5 kbps	84 kbps	75 kbps		
G.729A at 50 pps	29.6 kbps	26.4 kbps	42.4 kbps	25.6 kbps		
G.729A at 33 pps	22.2 kbps	20 kbps	28 kbps	19.5 kbps		

Bandwidth consumption per conversation for various transport types (includes Layer 2 overhead)

QUESTION 68:

There is a concern about voice quality over the link. In reviewing the estimated WAN traffic patterns, you see that voice will be approximately 45% of the traffic and there are no other real-time applications on the link. Which three of the following tools will help to insure that the bandwidth is used as efficiently as possible and the highest voice quality is attainted? (Choose three)

A. CBWFQ
B. cRTP
C. LFI
D. LLQ
E. Traffic policing
F. Traffic shaping

Answer: B, C, D

QUESTION 69:

Which cluster design would provide Certkiller with the highest redundancy and scalability, assuming 1200 users, 100 MoH sessions, and 24 conferencing sessions?

A. A single publisher and TFTP server combined. Two subordinate servers (primary and back-up). One subscriber will support conferencing and the other MoH.
B. A single publisher/TFTP server. Two subscriber servers (one primary and one back-up) that also support conferencing and a dedicated MoH server.
C. A dedicated TFTP server and a dedicated publisher. Two subscriber (one primary and one back-up) that support MoH and conferencing.
D. A dedicated TFTP server and dedicated publisher. Two subscriber servers load balancing 50/50. A dedicated MoH and conferencing server.

Answer: D
Highest redundancy requires a 1:1 scheme. A dedicated TFTP and publisher server are also required for this solution. A co-resident MoH server will not support 100 MoH streams so you must have a standalone MoH server.

QUESTION 70:

What is a major drawback of the 2:1 redundancy scheme?

A. CallManager software upgrades will cause phone outages.
B. Failover time is twice as long as with the 1:1 redundancy scheme.
C. The backup server must have twice the capacity of the primary servers.
D. The processing load doubles on the backup server when a primary server fails.

Answer: A

QUESTION 71:

Including the current 895 phones, what would be the minimum number of DIDs required for all the employees if each new phone has two line appearances?

A. 202
B. 404
C. 895
D. 1097
E. 1790
F. 2194

Answer: D
This question goes with the scenario, there are 895 total phones with DID (users and lobby phones, etc). There are a total of 202 new phones.
895 + 202 = 1097

QUESTION 72:

Certkiller understands that the existing dial plan may not be able to handle the planned growth. The planned growth will push the number of DIDs over one thousand. How should Certkiller handle the planned growth?

A. Certkiller will need to obtain another range of DIDs.
B. By using translation patterns, Certkiller won't need to get an additional range of DIDs.
C. Certkiller will be able to implement route patterns that will allow the company to support the new employees without new DID ranges.
D. Certkiller will need another range of DIDs to support the growth, but can simplify the dial plan with three-digit dialing.

Answer: A
The new phones will require DIDs and since the company only has 1000 available, they will not have enough. They need to obtain more from the LEC (preferably in a range that does not overlap with the current range)

QUESTION 73:

Certkiller has found that the commercial deployment of methane collection systems can sometimes require technical support outside of normal operating hours. Certkiller would like to deploy a call center to develop the skills necessary to provide phone, chat, and e-mail support. The company would like to start with five agents and grow to twenty agents. They would like to be able to identify the caller's name and the agent's phone but they do not plan on ***MISSING***

A. PRI
B. CAS
C. FXO
D. FXS

E. E&M

Answer:

QUESTION 74:

IT wants to minimize the amount of bandwidth consumed by MoH streams. What type of MoH transport mechanism can satisfy these requirements, and what is the maximum number of simultaneous MoH messages that can be streaming at any one time? (Choose two)

A. multicast
B. unicast
C. 1
D. 5
E. 8
F. 13
G. 26

Answer: A, F

Explanation: Multicast requires less bandwidth since it does not create a new stream for each new MoH request.
13 will be required because in the scenario it mentions that there are 12 departments with their own MoH and one generic MoH.

QUESTION 75:

As the new Ethernet network is deployed with QoS, it will be important to assign a trust boundary. Which two are the optimum locations where network devices can be trusted to apply QoS correctly? (Choose two)

A. core layer switch
B. distribution layer switch
C. access layer switch
D. IP phone
E. PC

Answer: C, D
The best place for trust boundary is as close to the edge as possible. The IP Phone is the ideal place for the boundary while the Access Layer switch comes in right after.

QUESTION 76:

Certkiller is currently using four contiguous registered Class C addresses for their network. Certkiller 's service provider said it could give Certkiller more address space, but at quite a high cost, and the new addresses would not be in a contiguous block with the ones Certkiller currently uses. Which option would make the most business sense for Certkiller ?

A. Maintain their current registered address space, obtain one more registered Class C address range for growth, and deploy RFC 1918 addresses for the new IP telephony solution.
B. Keep the current registered address ranges and migrate internal address to RFC 1918 addresses as the company moves off of the Token Ring network to the Ethernet network.
C. Readdress the network with new registered addresses now so that all the IP address issues can be worked out prior to the IP telephony deployment.
D. Keep one or two registered address ranges and migrate the entire network over to an RFC 1918 address range prior to the IP telephony deployment.

Answer: B

QUESTION 77:

How can high availability be accomplished with a Fast Ethernet or Gigabit Ethernet network?

A. Deploy dual-connected core switches, each with a single connection to each MDF.
B. Deploy dial MDF switches, each with dual connections to the IDF switches.
C. Deploy stacked IDF switches that are dual connected to each core switch in the computer room.
D. Deploy dial MDF switches that each connect to two core switches in the compute room.

Answer: D

QUESTION 78:

Each IDF will function at Layer 2. Which deployment solution will provide the highest availability and still provide for in-line power to the IP phones? (Choose two)

A. Using the current data cable drop, deploy a dial VLAN solution where FLP would make the determination of providing in-line power.
B. Deploy a single chassis-based switch with dual Layer 3 supervisors and a single connection to the MDF.
C. Use the present dual-cable drop to each desktop, providing in-line power on both drops and use FLP to make the determination of providing in-line power.
D. Deploy a stackable switch solution with dial connections from each switch to the MDF.
E. Use the present dual-cable drop to each desktop, providing in-line power on the cable that the current data device is connected to.
F. Deploy a chassis solution with dual connections to the MDF.

Answer: D

QUESTION 79:

You are doing a physical site survey of the Certkiller campus.
You noticed that the facility is divided by Country Line Road, which is the boundary between Cobb and Fulton counties. What issue needs to be addresses for the IP telephony design?

A. Whether separate call data information will need to be kept for both counties.
B. Whether the phones in each county require overlapping extension numbers.
C. Whether calls from the Certkiller buildings in Cobb County incur a toll change when calling the Certkiller buildings in Fulton County.
D. Whether the tax rate for telephony in Cobb County is different from what it is in Fulton County.
E. Whether a PSTN connection in bidg C can route emergency calls to the correct PSAP for the buildings in Cobb County.

Answer: E

QUESTION 80:

You are doing a physical site survey of the Certkiller campus.
What questions will need to be answered about each network closet to insure that the physical network devices can be deployed successfully? (Choose two)

A. Is each cable run to the closet within the maximum distance specification?
B. Is enough electrical power available for each network closet?
C. Are the IDFs stacked on top of each other or are they offset in each building?
D. Are any of the cable runs shared between voice and data (two-pairs for voice and two-pairs for data)?
E. Does Certkiller have an office numbering scheme?
F. Are there any extra cable runs pulled to each office that are not terminated?

Answer: A, B

QUESTION 81:

You are in a meeting with the Certkiller telephony services manager and the data network mananger.
You need information on the PBX and the voice-mail system. What four pieces of information will be of benefit? (Choose four)

A. The manufacturer, model, and capacity of the PBX and voice mail systems.
B. The person who performs the adds, moves, and changes at Certkiller .

C. The number of PBX outages in the last year and if that number violated their update expectations.
D. The current PBX vendors response time for service.
E. The connection type between the PBX and the PSTN.
F. How the voice mailboxes are deployed in the telephone system.
G. The power requirements for the PBX and voice mail systems.

Answer: A, D, E, F

QUESTION 82:

You are in a meeting with the Certkiller telephony services manager and the data network manager.
What four of the following are questions you would ask to obtain the information needed for the design of the IP telephony network? (Choose four)

A. What network access method and topology is deployed?
B. How many cable paths are available to each IDF?
C. Can additional servers or additional domains be added to the existing Certkiller domain?
D. What CDR is deployed in the existing network?
E. Who is responsible for moving telephones within Certkiller ?
F. What is the current traffic level and distribution of applications on the network?

Answer: A, B, C, F

QUESTION 83:

Which of these pieces of information must be addresses in the design of the IP telephony system for Certkiller ? (Choose three)

A. The network is running at 45% of capacity at peak times.
B. They were given a cost estimate of $83,000 00 to upgrade their existing PBX to support IP.
C. They have entered into negotiations to purchase a small, 45 person consulting engineering firm that specializes in design and installing lined lagoons and ponds.
D. The monthly cost of ling distance is $14,000 00.
E. Users complain that they receive a fast busy when trying to dial out to the PSTN during the middle of the day.

Answer: A, C, E

QUESTION 84:

The Certkiller domain coordinator is interested in determining how the CallManager servers can be monitored and also determine if the system has been compromised. Which two methods can be used to support these requirements and still maintain a high degree of security? (Choose two)

A. Provide a uniform level of access to all operators, technicians and system administrators.
B. Turn on event logging in all the CallManagers servers and regularly review event logs for the cluster.
C. Enable SNMP on all CallManager servers and change the default community strings.
D. Enable SNMP for all the devices that support IP telephony in the Certkiller network and change the ***.
E. Send all audit logs to a dedicated log server and filter the reports to view only critical and fatal ***.

Answer: B, C

QUESTION 85:

The Certkiller network coordinator is concerned that the IP phones can be targeted for malicious attack, specifically Ettercrap and VOMIT. When the security is designed for the IP telephony solution, which two methods can be used to break these two malicious attacks? (Choose two)

A. Isolate the IP phone VLAN from the PC.
B. Ensure the PC port is disabled if a PC is not attached.
C. Ensure the CallManager is configured to disable acceptance and GARP on IP phones.
D. Ensure that IP phone firmware is validated with the CallManager.

Answer: A, C

QUESTION 86:

What will you tell the new Certkiller trainee technician to do when you find that voice frames are being trapped by numerous data packets after fragmentation was implemented?

A. Decrease packet size.
B. Increase serialization.
C. Decrease serialization.
D. Implement priority queuing.
E. Implement aggressive fragmentation.

Answer: D

QUESTION 87:

What is the maximum number of users that can be supported by any of the specified sets of platforms, albeit Cisco, Compaq, Dell or IBM, that supports the Cisco Unity application.

A. 1,490
B. 2,400
C. 5,000
D. 7,500
E. 9,500

Answer: D

QUESTION 88:

You have been commissioned to design a WAN solution that will support both voice and data traffic for a Certkiller client, a college that has three remote campuses that are connected via dedicated T-1 respectively wants to extend IP telephony to its remote campuses. Every remote campus has approximately ten PCs and four telephones that are connected to key systems.
Which of the following issues will you take into consideration? (Choose all that apply.)

A. Will 5 digit dialing be supported at the remote campuses?
B. Which WAN access method will be cost efficient as well as provide the best service?
C. Will there be a need for new extension numbers?
D. Can E-91 be supported in a centralized call processing environment?
E. Which codes type will provide the best voice quality as well as not consuming too much WAN bandwidth?

Answer: B, E

QUESTION 89:

What will be your first action when you develop a voice over data migration?

A. Do a complete survey of the network topology.
B. Find out from the client what business requirements support a Voice over Data Migration.
C. Determine the existing voice features currently installed that will need migration to the Voice over Data Environment.
D. Determine if the migration of the existing voice features can be done without upgrading obsolete equipment.
E. None of the above.

Answer: C

QUESTION 90:

Why do companies opt to make use of a G.792a rather than a G.729?

A. It receives a higher MOS score.
B. It is riddled with complex algorithms which makes virus attacks less like to occur.
C. It uses less complex algorithms.
D. It receives a lower PSQM score which makes it more user friendly.
E. It samples speech patterns more often.

Answer: C

QUESTION 91:

What advice will you give to the new Certkiller trainee technician as to how much delay should be added to the total delay budget if Voice Activity Detection (VAD) is implemented?

A. 5 ms.
B. 10 ms.
C. 15 ms.
D. 50 ms.

Answer: A

QUESTION 92:

Serialization delays usually occur when voice and data share a single PVC. What can you use to avoid excessive serialization delays when working in a Frame Relay network environment on a VoIP?

A. FRF.12.
B. Multilink PPP.
C. cRTP.
D. WRED.
E. None of the above.

Answer: A

QUESTION 93:

Which of the following can have lost voice packets as a result? (Choose all that apply.)

A. Dejitter buffer overrun.
B. Global synchronization caused by multiple TCP flow simultaneously experiencing TCP slow start.
C. Network congestion/performance.
D. Network architecture (IP best effort).

E. All of the above.

Answer: A, C, D

QUESTION 94:

What is the effect of a SID in a VAD environment?

A. Provides "Noise" to the listener.
B. Provides voice options to the server.
C. Saves bandwidth.
D. Performs the 5 ms look-ahead buffer required.
E. Provides management control of voice clipping.

Answer: C

QUESTION 95:

Which of the following PBX features can be used to do the digit translation that is needed to provide dial plan transparency to a user when migrating from a traditional PBX system to an IP telephony environment?

A. ANI
B. CoS
C. NFAS
D. UDP/CDP.
E. None of the above

Answer: D

QUESTION 96:

Which of the following determines the users that may access a partition?

A. Route lists.
B. Route group.
C. Partition Access Control List (PACL)
D. Calling Search Space.
E. Access Control List (ACL)

Answer: D

QUESTION 97:

The newly appointed Certkiller trainee technician wants to know which of the following models would be most suitable for a Cisco router to use is PBX signaling protocol uses a single channel for information signaling when sending a proprietary signaling protocol across in IP WAN. What will your reply be?

A. Translate model using the proprietary signal.
B. Transport model using the proprietary signal.
C. Translate model using the cross-Connect method.
D. Transport model using the cross-connect method.
E. Translate model using the frame forwarding method.
F. Transport model using the frame forwarding method.

Answer: F

QUESTION 98:

When using a clear-channel TDM Group, how many signaling channels can you cross-connect?

A. 1
B. 24
C. None.
D. Any number.

Answer: D

QUESTION 99:

The new Certkiller trainee wants to know which of the following databases is used by the Public Safety Answering Point (PSAP) in a PBX Enhanced 911 implementation. What will your reply be?

A. PS-ALI Database.
B. ANI-ALI Database.
C. PBX-MLTS Database
D. LEC. ANI Database.
E. PBX-PSAP Database.

Answer: B

QUESTION 100:

Which of the following signaling approaches, on a T1 circuit, is able to use 24 channels for voice as well as carry signaling information on each of those 24 channels?

A. E&M
B. RTP
C. CAS

D. CCS
E. QSIG

Answer: C

QUESTION 101:

Which Cisco CallManager (CCM) deployment option would be the best suited for use by a company that has four site offices that are thousands of miles apart and the WAN currently in use is for purposes of toll bypass which leaves a large amount of bandwidth free?

A. Multiple site deployment
B. WAN isolated deployment
C. WAN distributed deployment
D. Single site deployment
E. WAN centralized deployment

Answer: C

QUESTION 102:

How is bandwidth associated with a location when location-based Call Admission Control is implemented?

A. The gatekeeper uses SAA to maintain bandwidth.
B. The bandwidth is constantly balanced throughout the network.
C. The location has an allocated bandwidth pool.
D. The Cisco CallManager (CCM) uses RSVP to maintain bandwidth usage levels.

Answer: C

QUESTION 103:

What type of DSP resource would you recommend for use when a company wants to deploy centralized voice processing and its branch offices use G.729a over WAN and the Unity Voice Mail server is located in the central office and supports only G.711?

A. MTP
B. codec
C. transcoder
D. RTP
E. conference

Answer: C

QUESTION 104:

Which of the following are characteristics of the Catalyst 4000 gateway? (Choose all that apply.)

A. G.711 and G.729a codes support.
B. Supports 6 ports analog, or 4 ports of analog and 2 TI/EI Ports.
C. Uses the Skinny Gateway Protocol.
D. Provides infrastructure consolidation and Cisco IP telephony support.
E. All of the above.

Answer: ABD.

QUESTION 105:

Cisco CallManager (CCM) V3.0 and V3.1 is capable of registering a maximum of _______ phones per server.

A. 250.
B. 1,000
C. 2,500.
D. 10,000.

Answer: C

QUESTION 106:

One of the Certkiller customers needs to have their legacy SMDI protocol voice mail system supported. Which signaling protocol should the connection gateway support to fully support the customer's legacy voice mail system?

A. SIP.
B. H.323.
C. H.245.
D. SCCP.
E. MGCP.

Answer: E

QUESTION 107:

Which of the following statements regarding the Catalyst 6500 gateways are valid? (Choose all that apply.)

A. It is incapable of supporting Cisco Fax relay.
B. It can supply full out-of-band DTMF and Cisco CallManager (CCM) Failover.

C. It enables the implementation of IP telephony by using existing Cisco Catalyst 6500 routers.
D. They are link cards that can be installed in any 6500 series switch.
E. It provides DID support or calling line identification (CLID) Support.

Answer: B, D

QUESTION 108:

Which of the following services is provided by gateways? (Choose all that apply.)

A. The connection between POTS and an FXS Port.
B. The access point for Cisco CallManager (CCM).
C. Access to IP telephones hassle-free.
D. Protocol conversion between terminals.
E. All of the above.

Answer: A, D

QUESTION 109:

What does one call an endpoint that starts a SIP connection request?

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

Answer: B

QUESTION 110:

A call that was configured for MGCP came through a PSTN gateway. What is required of the gateway to complete the call?

A. It must query the gatekeeper
B. It must query the redirect server
C. It must query the call agent
D. It must query the internal dial peers

Answer: C

QUESTION 111:

Which of the following protocols opens a logical channel between devices in the H.323

protocol suite?

A. H.225
B. H.245
C. H.322
D. H.323

Answer: B

QUESTION 112:

The Common Channel Signaling produces what type of signaling?

A. In-band.
B. Shared-band.
C. Out-of-band.
D. Diffused-band
E. Separate-band.

Answer: C

QUESTION 113:

If one neglect to check LAN/WAN utilization before design one can expect to encounter the following.

A. Improper route plan.
B. Inappropriate code selection.
C. A current data link being oversubscribed.
D. An increase in call blocking
E. A decrease in the probability of call blocking.

Answer: C

QUESTION 114:

Which of the following does not form part of the projected monthly voice expense model?

A. New circuits to support growth.
B. New voice applications made possible by the migration.
C. Voice ports to support current voice traffic error-free.
D. New data network needs to support possible future growth.

Answer: B

QUESTION 115:

What must be fully comprehended by a network designer so as to enable him to provide practical, proper capacity for IP telephony networks?

A. WAN link costs.
B. Number of IP phones.
C. The bandwidth of each VoIP call.
D. Number of outgoing calls per day
E. Router CPUS clock speeds.

Answer: C

QUESTION 116:

The new Certkiller trainee is curious as to which group of devices have similar dialing accessibility. What will your reply be?

A. CoS groups
B. route groups
C. partitions
D. voice port groups
E. Calling Search Spaces

Answer: E
Group of devices (ip-phones) that have similar dialing-accessibility are associated with Calling search spaces, not to partitions (Not C).

QUESTION 117:

Which of the following represents a component of the H.323 specification?

A. RTP
B. H.322.
C. RAS
D. G.711
E. T.120

Answer: C

QUESTION 118:

Why will you advice the new Certkiller trainee technician NOT to use CBWFQ for voice traffic?

A. No FIFO

B. Permits queue starvation.
C. Guarantees bandwidth but not delay.
D. Prohibits bandwidth.
E. Guarantees delay but not bandwidth.

Answer: C

QUESTION 119:

Which of the following issues can you treat by using QoS Tools in a voice over data traffic network? (Choose all that apply.)

A. Scheduled access.
B. Outages of no more than a few microseconds.
C. Chronological data flow
D. Isochronous data flows.
E. First come, first served access.
F. Fixed data rate (Irrespective of network conditions).

Answer: D, E
D:
Consecutive packets that experience different amounts of delay have experienced jitter. In a packet network, with variable delay components, jitter always occurs—the question is whether the jitter impacts the application enough to degrade the service. Typically, data applications expect some jitter, and do not degrade. However, some traffic, such as digitized voice, requires that the packets be transmitted in a consistent, uniform manner (for instance, every 20 ms). The packets should also arrive at the destination with the same spacing between them. (This type of traffic is called *isochronous traffic*.)
Not B: To avoid outages of more than a few micro seconds by using qos tools) must be wrong!
It is impossible to avoid an outage by using qos tools. Rerouting is possible but not in a few microseconds.

QUESTION 120:

Which types of network equipment will you tell the Certkiller trainee technician to replace when checking whether IP Telephony is supported in an existing data network? (Choose all that apply.)

A. All Catalyst 1900 Platforms.
B. All Catalyst 1500 platforms.
C. All hubs.
D. All switch ports.
E. All switches that do not support Layer 3.

Answer: A, C

QUESTION 121:

What is usually the first step that is performed when migrating a traditional telephony system to a VoIP system?

A. Determine new applications to be integrated.
B. Determine the costs associated with existing PSTN Services.
C. Determine which of the existing voice features require migration.
D. Determine the projected future growth of the company.
E. Determine the voice and data requirements for the LAN/WAN.

Answer: C

QUESTION 122:

64 kbps bandwidth for voice payload and no compression on voice signal is needed by which of the following coding algorithms?

A. G.726.
B. G.729a
C. G.711
D. G.728.
E. G.729

Answer: C

QUESTION 123:

In which way does Cisco Routers accommodate jitter?

A. It can use cRTP to specify interpacket delay.
B. It can use WRED to specify interpacket delay.
C. It can provide an adaptable play out buffer.
D. It can use RAS to eliminate jitter.
E. It can use LLQ to eliminate jitter for traffic placed in a priority traffic class.

Answer: C

QUESTION 124:

What reason can one say is it that echo does not hamper operations on a conventional circuit-switched telephone network?

A. Circuit switched networks do not have 2 Wire to 4 wire hybrid circuits.
B. The round-trip delays through a circuit switched telephony network are typically less than 50 ms.

C. Circuit switched networks do not use the codes found in the IP telephony environment.
D. Round-trip delays on a circuit switched telephone network is more than 50 ms.
E. Echo is suppressed in a circuit switched network due to echo cancellers at the central office.

Answer: B

QUESTION 125:

Which of the following error detection schemes are obsolete in the voice world? (Choose all that apply.)

A. Playout buffers.
B. UDP Checksums.
C. Sequence numbers.
D. Retransmission.
E. All of the above.

Answer: B, D

QUESTION 126:

When migrating a traditional PBX system to an IP Telephony system, which factors will you take into consideration? (Choose all that apply.)

A. Scalability.
B. Accessibility
C. Cost
D. Transportability.
E. Functionality.
F. Compatibility.
G. Referencability.

Answer: A, C, E, F

QUESTION 127:

Which of the following has the ability to transparently route a call to a dialed destination no matter the available physical path?

A. A route Table
B. Class of service.
C. A dial plan.
D. Number expansion.

Answer: C

QUESTION 128:

On what is the router-to-router signaling based on when networking PBXs over an IPWAN?

A. CAS
B. H.322
C. CCS
D. H.323.
E. QSIG

Answer: D

QUESTION 129:

Which of the following Enhanced 911 components is capable of identifying a dialable telephone number associated with a 911 call?

A. LEC
B. PS-ALI
C. ANI.
D. PSAP.
E. MLTS.

Answer: C

QUESTION 130:

Which of the following clients warrants the invoking of an MTP?

A. clients that support supplementary call services (like call hold, and conference bridge) with H.232 clients
B. clients that do not support the H.323v2 features of OpenLogicalChannel and CloseLogicalChannel with the EmptyCapabilitiesSet
C. clients that do not need supplementary call services (like call hold and conference bridge) with H.323 clients
D. clients that support H.323v2 features of OpenLogicalChannel and CloseLogicalChannel with the EmptyCapabilitiesSet

Answer: B

QUESTION 131:

In which of the following ways does QoS ensure voice quality? (Choose all that apply.)

A. by preventing voice from oversubscribing the link
B. by giving voice and data separate but equal access to the link
C. by alternating voice and data packets to provide fair access to the link
D. by giving voice priority over data
E. by preventing data from oversubscribing the link
F. all of the above

Answer: A, D

QUESTION 132:

Which signaling technique is used in a situation where calls are made between Cisco CallManager (CCM) V3.0 and V3.1 clusters across a WAN?

A. SS7
B. not-so-stubby
C. H.323
D. MGCP
E. Skinny

Answer: C

QUESTION 133:

How may devices does three IP phones and two gateways represent in a Cisco CallManager (CCM) V3.1 environment?

A. 5
B. 9
C. 12
D. 18

Answer: B

QUESTION 134:

How many subscribers are needed to support 10,000 IP phones and 20,000 device units when working in a Cisco CallManager (CCM) 3.3 environment with 1.1 redundancy that supports 2500 phones?

A. 4
B. 8
C. 16
D. 32

Answer: B

QUESTION 135:

Which of the following statements regarding Cisco VG 200 is valid? (Choose all that apply).

A. The Cisco VG200 is not intended, tested, or supported for H.323 toll bypass, or as a session initiation protocol (SIP) gateway.
B. The Cisco VG200 is equipped with digital signal processors (DSPs) that convert analog and digital voice into IP packets using standard codes G.711 and G.723.1 but must be upgraded to use G.729 a or other.
C. The Cisco VG200 provides an autosensing 10/100 Ethemet Port.
D. The Cisco VG200 connects a Cisco IP telephony network to the PSTN, but not directly to an existing PBX.
E. The Cisco VG200 connects a Cisco IP telephony network to legacy telephones. Fax Machines, and voice conference units.
F. All of the above.

Answer: A, C, E.

QUESTION 136:

Which of the following has a stateless endpoint architecture, akin to SCCP, insofar as that there is no real call routing intelligence on an endpoint?

A. SIP.
B. MGCP.
C. RTP.
D. H.323.

Answer: B

QUESTION 137:

Which statement regarding a gatekeeper for H.323 call setup is valid?

A. It makes call setup optional.
B. It is not used for call setup.
C. If present in the network, it must be used for call setup.
D. If present in the network it may be used for call setup.
E. It must be present in the network and used for call setup.

Answer: C

QUESTION 138:

Which of the following albeit not a component of H.232 protocol, must be supported by H.323 devices in a Voice over Data network?

A. RAS.
B. STP
C. RTP.
D. T.120
E. H.245

Answer: C

QUESTION 139:

Which of the following formulae will you use to determine the required bandwidth for voice?

A. Code bandwidth x 4.
B. Numbers of simultaneous calls x code bandwidth.
C. Code bandwidth x analog lines.
D. Code bandwidth x 16
E. Number of simultaneous calls x 24 kbps.

Answer: B

QUESTION 140:

What percentage of capital cost would be estimated for cutover cost from a traditional telephony to an IP telephony network when making use of the Standard Cost-Saving Model to estimate implementation costs and no detailed analysis is available?

A. 2 To 5%
B. 5 To 10%
C. 10 To 15%
D. 25 To 30%.

Answer: C

QUESTION 141:

The new Certkiller trainee wants to know which network protocols is capable of supporting fax transmission over a data network. What will your reply be? (Choose all that apply.)

A. T.120
B. FRF.11
C. T.10
D. T.30
E. T.38

Answer: B, E

QUESTION 142:

What can you use to avoid large data packet from delaying voice packets on a Frame Relay network?

A. FRF-5.
B. FRF-12
C. FRF-8
D. FRF-10

Answer: B

QUESTION 143:

Which queuing technique would you recommend to the Certkiller trainee technician when handling voice traffic?

A. PQ
B. WFQ
C. LLQ
D. ARQ.
E. CBWFQ.

Answer: C

QUESTION 144:

Certkiller has three remote campus locations. The remote campuses are located in offices, each with only four POTS connections, one of which also functions as a Fax connection.
A key system at each location has a dedicated T-1 circuit connecting back to the main campus and two POTS connections to the local PSTN CO.
Your CEO wants a new IP telephony system address in a centralized call processing environment.
Which design issues must the new IP telephony system address?

A. In the event of WAN failure, will Cisco CallManager (CCM) failover be required?
B. In the event of WAN failure, will there be a need to provide SRST functionality?
C. Can the new IP telephony solution provide the same features as their existing key

systems?
D. Will there be a need to provide local Cisco CallManager (CCM) support at each location to replace the key systems?

Answer: B

QUESTION 145:

Below are various types of coding algorithms. Which of them resort under the "Hybrid Coders" category of speech coding schemes?

A. LPC
B. PCM
C. HELP
D. CELP.
E. ADPCM

Answer: D

QUESTION 146:

Which of the following are regarded as variable delays when calculating delay for voice? (Choose all that apply.)

A. Queuing delay.
B. Serialization.
C. Dejitter buffer.
D. Propagation delay
E. All of the above.

Answer: A, C

QUESTION 147:

What is the purpose of a dejitter buffer?

A. A dejitter buffer replaces dropped packets with randomly generated comfort noise.
B. A dejitter buffer uses random queuing strategy to reduce variance on a first come first serve basis.
C. A dejitter buffer uses a priority queuing strategy to reduce variance in voice packet arrival times.
D. A dejitter buffer induces delay and smoothes out the variance in voice packet arrival times.
E. A dejitter buffer uses a look ahead algorithm to predict the next voice packet, thus reducing variable delay.

Answer: D

QUESTION 148:

In what instances is translation utilized? (Choose all that apply.)

A. When routing all external inbound calls to an IVR System).
B. When routing a call to a recorded message if the caller tries to reach an unassigned DID number.
C. When restricting calls between partitions when Calling Search Space do not exist.
D. When rerouting a call to an attendant at an extension (e.g IIII) whenever a user dials 0.
E. All of the above.

Answer: A, B, D

QUESTION 149:

You are a network engineer at Certkiller . You want to send a proprietary PBX signaling protocol across an IP WAN. The PBX signaling protocol on the Cisco router uses two channels for signaling information.
Which model would a Cisco router use in this scenario?

A. Translate model using the proprietary signal.
B. Transport model using the proprietary signal.
C. Translate model using the cross-connect method.
D. Transport model using the cross-connect method.
E. Translate model using the frame forwarding method.
F. Transport model using the frame forwarding method.

Answer: D

QUESTION 150:

Your newly appointed Certkiller trainee technician wants to know what the major difference is between a PBX E911 solution and a Cisco CallManager (CCM) E911 solution. What will your reply be?

A. The CCM solution requires a PS-ALI database.
B. The AS5300 gateway is required for states with ISDN PRI Services.
C. The AS5300 gateway is required for states with CAMA trunks.
D. With the CCM solution, the PSAP can directly query the CCM.
E. None of the above.

Answer: B

QUESTION 151:

Which Cisco CallManager (CCM) deployment option would be the best suited for use by a company that has four site offices that are thousands of miles apart with three of these site offices having difficulty in hiring system administrative personnel and the WAN currently in use has free bandwidth.

A. single site deployment
B. WAN centralized deployment
C. WAN isolated deployment
D. multiple site deployment
E. WAN distributed deployment

Answer: B

QUESTION 152:

A phone that is moved from site A to site B, requests 16K for calls.
What is the resulting bandwidth when a call request is made?

A. decrement only Site B by 16K
B. decrement only Site A by 16K
C. decrement Sites A and by 8K
D. decrement Sites A and 2 by 16K

Answer: B

QUESTION 153:

How many Cisco CallManager (CCM) servers will you need when you want to build up a 2:1 redundant cluster supporting up to 7500 phones. (A Cisco CallManager (CCM) server is capable of supporting 7500 phones.)

A. 1
B. 2
C. 3
D. 6
E. 9

Answer: C

QUESTION 154:

Which of the following is supported by the Catalyst 4000 Access Gateway Module?

A. Digital DID with no digital CLID offered.

B. Analog DID with no digital CLID offered.
C. Digital DID or digital CLID only at one time.
D. Digital DID and digital CLID simultaneously.
E. Analog DID and digital CLID only at one time.

Answer: C

QUESTION 155:

The new Certkiller trainee wants to know what the necessary functions necessary on a voice gateway is. What will you reply? (Choose all that apply.)

A. Out-of-band DTMF Relay.
B. Measurement based CAC.
C. E & M services.
D. Supplementary services.
E. Cisco CallManager (CCM) Redundancy.
F. All of the above.

Answer: A, D, E

QUESTION 156:

Which of the following statements regarding MGCP is valid? (Choose all that apply.)

A. It is TCP-based, not UDP.
B. It involves a small set of simple transactions.
C. Its main emphasis is simplicity and reliability:
D. It is a peer-to-peer protocol.
E. Endpoints can be mass-produced cheaply.
F. It is a server to peer protocol.

Answer: B, C, E

QUESTION 157:

What is the transport protocol used by H.245?

A. ARP
B. TCP
C. IPX
D. PBX
E. UDP

Answer: B

QUESTION 158:

The new Certkiller trainee technician wants to know what type of signaling scheme Channel Associated Signaling is. What can you tell her?

A. Byte-oriented.
B. Call-oriented
C. Bit-oriented.
D. Protocol-oriented.
E. message oriented.

Answer: C

QUESTION 159:

As a Certkiller network planner, what should you clarify with the client in the planning stage?

A. What are the equipment order dates?
B. Who owns the network?
C. Who is upgrading existing systems?
D. Who controls the network?
E. Who is configuring the systems?

Answer: C

QUESTION 160:

Certkiller has five PBXs spread throughout its campus network. One of the five PBXs is designated as the control cabinet, the other four feed back to the control cabinet for call processing. The PBXs use nine T-1 circuits to connect to the PSTN. The five PBX nodes service approximately 6,000 analog telephones, as well as fax machines and conference speakerphones.
Which two design issues must the new IP telephony system address (Choose all that apply.)

A. How old are the correct PBXs?
B. How many users are connected to each PBX?
C. How many PBX technicians are employed by Certkiller ?
D. How many analog ports will be required to support the existing POTS devices?
E. Has a traffic engineering study been done to determine if the number T.Ist is correct?

Answer: D, E

QUESTION 161:

To how many devices units are one voice gateway and one IP phone equivalent to?

A. 2
B. 3
C. 4
D. 6

Answer: C

QUESTION 162:

You are a network engineer at Certkiller . Your newly appointed Certkiller trainee wants to know which QoS functions are used to prioritize Voice over Data.
What will your reply be? (Choose all that apply).

A. Policing.
B. Queuing.
C. Encryption.
D. Classification.
E. Fragmentation.

Answer: B, D

QUESTION 163:

You are the network engineer at Certkiller . You are applying QoS in the Certkiller LAN. You want voice traffic to be prioritized over all other traffic so it can maintain low latency and high quality.
Which tools can you use to accomplish this? (Choose all that apply.)

A. CoS.
B. DSCP.
C. CBWFQ.
D. Fragmentation.
E. Traffic shaping.

Answer: A, B

QUESTION 164:

You are a network engineer at Certkiller . The newly appointed Certkiller trainee wants to know if there are any reasons to disable VAD.
What will your reply be? (Choose all that apply.)

A. For Fax Calls.
B. For Voice Calls.

C. For Modem Calls.
D. For a higher MOS score.
E. To eliminate clipped speech.

Answer: A, C, E

QUESTION 165:

Which of the following CODECs has the highest Mean Opinion Score (MOS)?

A. G.711
B. G.726.
C. G.729.
D. G.729a.

Answer: A

QUESTION 166:

When you are calculating delay for voice, what would be considered a fixed delay?

A. Dejitter buffer.
B. Queuing delay.
C. Propagation delay.
D. Network congestion.

Answer: C

QUESTION 167:

Which of the following methods can you use to play out a lost packet in a VoIP environment? (Choose all that apply.)

A. Repeal the previous packet.
B. Retrieve the lost packet from the upstream router cache.
C. Interpolate the sound based on the previous and following sound packet characteristics.
D. Add in some predetermined, calculated noise.
E. Add in new interference to intercept the lost packets from the cache.

Answer: A, C, D

QUESTION 168:

Which of the following add to the effects of jitter? (Choose all that apply.)

A. Propagation delay.
B. Large Packets on slow links.
C. Queuing delay.
D. Oversubscription.
E. Repeal of lost packets.

Answer: B, C, D

QUESTION 169:

Which of the following will you take into consideration when you need to calculate a delay budget? (Choose all that apply.)

A. MTU Buffer.
B. Processing
C. Queuing delay.
D. Impedance.
E. Propagation
F. All of the above.

Answer: B, C, E

QUESTION 170:

You are a network engineer at Certkiller . Your newly appointed Certkiller trainee wants to know which industry specification is used to internet work or exchange messages between disparate voice messaging systems.
What will your reply be?

A. CTI.
B. CAS
C. AMIS.
D. SMDI.
E. DTMF.

Answer: C

QUESTION 171:

Which of the following statements are valid when one considers that the H.225 gatekeeper controlled trunk will permit communication between Cisco CallManager (CCM) and other CCMs and H.323 devices that are registered to a single gatekeeper. (Choose all that apply.)

A. Calls are lead balanced across the registered trunks in the CCM cluster.
B. The H.225 gatekeeper controlled trunk is not configured in the CCM cluster.

C. The gatekeeper has a gatekeeper zone configured for each site supporting CCM or voice gateways.
D. The interzone bandwidth command on the gatekeeper is used to control bandwidth between CCM clusters and H.323 devices registered directly with the gatekeeper.

Answer: A, C

QUESTION 172:

You are a network engineer at Certkiller . Your newly appointed Certkiller trainee wants to know what mechanism is used in Cisco CallManager (CCM) to limit the bandwidth used between sites for Call Admission Control.
What will your reply be?

A. Sites
B. regions
C. locations
D. partitions
E. device pools

Answer: C

QUESTION 173:

You are a network engineer at Certkiller . Your newly appointed Certkiller trainee wants to know what the purpose of intercluster communication is. What will your reply be? (Choose all that apply.)

A. It controls call routing
B. It stores standard report data
C. It passes device status messages
D. It distributes database information
E. It propagates run-time information

Answer: C, E

QUESTION 174:

What is the maximum number of servers in cluster that Cisco CallManager (CCM) V3.1 supports?

A. 4
B. 5
C. 6
D. 7
E. 8

Answer: E

QUESTION 175:

Which of the following does VG-248 include? (Choose all that apply.)

A. 24 digital ports.
B. 48 digital ports.
C. 24 analog ports.
D. 48 analog ports.
E. Low density gateways.
F. Support for analog telephones fax machines, modems, voice mail systems and speakerphones.

Answer: D, F

QUESTION 176:

The SIP protocol is used for communication between which devices in a Cisco IP telephony network?

A. CCM to MCU.
B. CCM to gateway.
C. IP Phone to CCM.
D. IP Phone to MCU.
E. IP Phone to IP Phone.

Answer: E

QUESTION 177:

What was the major reason for the development of revision 2 of the H.323 standard?

A. To overcome the product incompatibility that marred the original H.323 standard.
B. New packet based PSTN service initiative started to encroach on new VolP markets.
C. To overcome the product incompatibility that resulted from an absence of a standard for H.323 QoS.
D. New PSTN capabilities rendered the existing H.323 standard ineffective in delivering quality VolP services.

Answer: A

QUESTION 178:

Which of the following protocols has Cisco implemented in VoIP networks? (Choose all that apply.)

A. SIP.
B. DCS.
C. CDP.
D. H.323.
E. MGCP.
F. MCDN.

Answer: A, D, F

QUESTION 179:

Which of the following are integral components necessary for the successful deployment of voice of data networks, especially in the Planning stage? (Choose all that apply.)

A. Dial-plan.
B. MPLS frame tags.
C. Telephone numbers.
D. IP addressing.
E. All of the above.

Answer: A, C, D

QUESTION 180:

The Deployment model has certain stages. During which stage would you discuss the capture or estimate cost with the customer?

A. Planning.
B. Implement.
C. Design.
D. Operate.

Answer: C

QUESTION 181:

You are a network engineer at Certkiller . Your newly appointed Certkiller trainee wants to know what he should use when given the instruction to implement QoS tools to protect voice from voice. What will your reply be?

A. Prioritization tools.
B. Call admission tools.
C. Link efficiency tools

D. Traffic regulating tools
E. Traffic shaping tools

Answer: B

QUESTION 182:

Certkiller has three remote offices that are connected to the head office via dedicated T-1 respectively. The CEO wants to extend this connectivity by means of IP telephony. Every remote office has between five and ten PCs and four telephones that are connected to key systems.
As the network designer, which issues will you take into consideration when designing the WAN solution that will be capable of supporting both voice and data traffic?
(Choose all that apply)

A. What IP Telephone features are required at each location?
B. How many PBX technicians are employed by Certkiller ?
C. How much WAN bandwidth is currently being used by the data traffic?
D. Can the remote campuses be accommodated in the correct dial plan?
E. How much bandwidth is expected to be consumed by each IP Phone call?
F. Will there be a need for new extension numbers?

Answer: C, E

QUESTION 183:

Which codec amounts to the least delay when calculating the overall delay budget?

A. G.729a.
B. G.711
C. G.726.
D. G.728
E. G.729.

Answer: B

QUESTION 184:

In which of the following would you most probably encounter echo?

A. In a PBX.
B. In a router
C. In the central Office
D. In a two wire bi-directional transmission medium

Answer: D

QUESTION 185:

Which of the following statements regarding the way in which Cisco CallManager (CCM) can use translation patterns are valid? (Choose all that apply.)

A. Digit translate is only possible on external calls.
B. Digit translate can be used on internal and external calls.
C. Digit translation is only possible on outbound calls.
D. Digit translation can be used on inbound or outbound calls.
E. All of the above.

Answer: B, D

QUESTION 186:

Which two approaches can be used for signaling purposes between legacy voice mail systems and PBXs? (Choose all that apply.)

A. Out-of-band SMDI.
B. In-Band SMDI.
C. In band MGCP
D. In band DTMF.
E. Out-of-Band DTMF.
F. Out-of-band MGCP.

Answer: A, D

QUESTION 187:

What would be the appropriate Cisco CallManager (CCM) deployment option for a customer who has four geographically dispersed offices that are experiencing heavy traffic on its current data WAN, but does not want to upgrade?

A. single site deployment
B. WAN distributed deployment
C. WAN distributed deployment
D. Multiple site deployment
E. WAN centralized deployment

Answer: C

QUESTION 188:

What action will an IP phone perform when the primary Cisco CallManager (CCM) server is brought back online?

A. re-establishes a connection to the primary CCM server after a wait period
B. re-establishes a connection to the primary CCM server
C. maintains connectivity with the secondary CCM server after device reboot
D. replaces an old primary CCM server
E. establishes the old primary CCM server as the new secondary CCM server

Answer: B

QUESTION 189:

Which of the following represents characteristics of the Catalyst 4000 gateway? (Choose all that apply.)

A. Makes use of the not-so-stubby Gateway Protocol.
B. Capable of supporting 6 ports analog, or 4 ports of analog and 2 TI/EI Ports.
C. Provides infrastructure consolidation in addition to Cisco IP telephony support for the remote office.
D. Uses the Skinny Gateway Protocol.
E. Supports the G.711 and G.729a codes.
F. All of the above.

Answer: B, C, E

QUESTION 190:

SIP is characteristic of which type of protocol?

A. Client peer.
B. Server-to-server
C. Server-peer.
D. Peer-to-Peer.
E. Client-server.

Answer: D

QUESTION 191:

What is the function of H.225 RAS when used by H.323?

A. register with the MCU.
B. request extension numbers
C. register with the gatekeeper
D. request extension resources from CFB

E. request conference resources from the CFB

Answer: C

QUESTION 192:

What is the function of a data network assessment?

A. Determining current voice traffic cost.
B. Determining PBX protocol requirements.
C. Determining IP telephony considerations.
D. Determining delay costs.
E. Determining voice circuits cost between all offices.

Answer: C

QUESTION 193:

Which of the following can you expect to encounter if you neglect to check LAN/WAN utilization before design?

A. Improper route plan.
B. Inappropriate code selection.
C. A current data link being oversubscribed.
D. An increase in call blocking
E. A decrease in the probability of call blocking.

Answer: C

QUESTION 194:

How many subscribers are required to support 10,000 IP phones and 20,000 device units when working in a Cisco CallManager (CCM) 3.3 environment with 1.1 redundancy, supporting 2500 phones?

A. 3
B. 8
C. 9
D. 15

Answer: B

QUESTION 195:

What feature does G.729b add to G.729?

A. Increases sampling rate.
B. Lower Complexity Algorithm.
C. Higher Complexity algorithm.
D. Voice Activity Detection (VAD).

Answer: D

QUESTION 196:

In which way does tandem switching affect voice quality?

A. Quality is decreased due to transcoding.
B. Quality is decreased due to dual use of PCM
C. Quality is decreased due to multiple decoding encodings.
D. Quality is decreased due to optimal pathing.
E. Quality is decreased due to the use of PCM.

Answer: C

QUESTION 197:

What type of call is made when one IP phone calls another IP phone that is registered to the same Cisco CallManager (CCM)?

A. Local call.
B. Internal Call.
C. Cached call.
D. International Call
E. External Call.

Answer: B

QUESTION 198:

What signaling approach is capable of using 23 channels for voice and one for data on a TI circuit?

A. CAS
B. CES
C. E&M
D. CCS
E. QSIG

Answer: D

QUESTION 199:

Why is it beneficial to a company to make use of the 1.1 redundancy model?

A. Lower deployment cost.
B. Low labor intensity.
C. High Availability.
D. Increased server count.
E. Lower maintenance cost.

Answer: C

QUESTION 200:

Which of the following are SIP components? (Choose all that apply.)

A. User Agent.
B. Call Agent.
C. Proxy Server.
D. Direct Server
E. Redirect Server.

Answer: A, C, E

QUESTION 201:

A gatekeeper must be capable to provide a number of setup/control services. Which of the following services should a H.323 gatekeeper provide? (Choose all that apply.)

A. bandwidth control
B. address translation
C. routing control
D. call routing
E. admission control

Answer: A, B, E

QUESTION 202:

Why does it make economic sense to make a transition to a Voice over Data network?

A. Reduced hardware cost.
B. Reduced call connection time.
C. Reduced complexity.
D. Reduced long distance call expenses.

E. Reduced WAN bandwidth requirements.

Answer: D
One argument for Ip telephony is toll by-pass whitch must be very significant i long distance calls.

QUESTION 203:

Which of the following types of switches do you have to upgrade when you survey an existing data network to see if it can support IP telephony?

A. That does not support STP.
B. That does not support Layer 3 Services.
C. That does not support in-line power.
D. That does not support TCP/IP
E. That does not support Gigabit Ethernet.

Answer: C

QUESTION 204:

Which of the following are components of a delay budget in an IP telephony environment? (Choose all that apply.)

A. Variable delay.
B. Dynamic delay
C. Fixed delay.
D. Static delay
E. All of the above

Answer: A, C

QUESTION 205:

Which of the following will have echo as a result?

A. Oversubscription.
B. Propagation delay.
C. 2-Wire to 4 Wire conversion
D. Translation delay
E. Large Packets on slow links.

Answer: C

QUESTION 206:

What component of Enhanced 911 will provide specific location data, such as a street address as well as subaddress (e.g. on office number)?

A. LEC
B. RST
C. ALI
D. PSAP
E. MLTS

Answer: C

QUESTION 207:

DSPs can provide which of the following resources? (Choose all that apply.)

A. transcoding
B. transponding
C. conferencing
D. translating
E. Music On Hold (MOH)

Answer: A, C

QUESTION 208:

Which of the following can be regarded as an advantage of the MGCP architecture?

A. Stateful endpoints for rapid failover
B. Gateways can route calls in the absence of a call agent.
C. Centralized dial plan
D. Stateless endpoints for rapid failover
E. Mixed vendor networks are possible as MGCP is an approved standard.

Answer: C

QUESTION 209:

You are a network engineer at Certkiller . Your newly appointed Certkiller trainee wants to which of the following are used in current cost savings discovery. What will your reply be? (Choose all that apply.)

A. Toll/LEC/Circuit costs for services.
B. Ongoing implementation costs of new equipment.
C. Staffing costs to maintain separate networks.
D. Costs associated with plans for future growth.
E. Site/Power/and cable requirements for existing offices.

Answer: A, C, E

QUESTION 210:

Which of the following codecs is capable of compressing voice down to a rate of 6.5 kbps?

A. G.711
B. G.729a
C. G.723.1
D. G.726.
E. G.729.

Answer: C

QUESTION 211:

Which of the following represents the standard analog Fax protocol?

A. T.37
B. T.38
C. T.30
D. FRF. 11.

Answer: C

QUESTION 212:

What can you use to transfer signaling information across a WAN when a PBX makes use of a serial control channel? (Choose all that apply.)

A. CES (For an ATM WAN).
B. OOB-SS7 (For an ATM WAN).
C. NFAS (For a Frame Relay WAN).
D. STUN (For a Frame Relay WAN).

Answer: A, D

QUESTION 213:

What type of deployment would be best suited for a company that has several buildings on a single site and all its network connections are within a Local Area Network (LAN)?

A. LAN isolated deployment
B. single site deployment

C. LAN distributed deployment
D. Multiple site deployment
E. WAN centralized deployment

Answer: B

QUESTION 214:

What is the maximum amount of users that Cisco CallManager (CCM) V3.1 clusters can accommodate?

A. 2.500
B. 5,000
C. 10,000
D. 20,000
E. 30,000

Answer: E

QUESTION 215:

Which of the following are supported by Catalyst 6000 Voice TI and Services Module? (Choose all that apply.)

A. Conference bridging.
B. Supplementary services.
C. Transcoding.
D. Digital TI PSTN and PBX gateway.
E. All of the above.

Answer: A, C, D

QUESTION 216:

Which of the following protocols are capable of transporting voice packets in a VoIP network?

A. CAS.
B. CCS.
C. RTP.
D. H.323.
E. RAS
F. MGCP.

Answer: C

QUESTION 217:

The new Certkiller trainee wants to know what an Erlang is since she was asked to measure traffic load. What will your reply be?

A. 1000 busy hour call attempts.
B. The total number of calls in a single circuit for one hour.
C. Amount of traffic to load a circuit for one hour.
D. The total number of calls on a circuit.
E. 1000 seconds of calls on the same circuit.

Answer: C

QUESTION 218:

AVVID designs require several critical gateway features. Which of the following are such gateway features? (Choose all that apply.)

A. support supplementary services
B. support in-band DTMF signaling for voice mail systems
C. support for DTMF relay.
D. support failover to a redundant Cisco CallManager (CCM)
E. all of the above.

Answer: A, C, D

QUESTION 219:

Which of the following gateways are used to provide connectivity to a legacy voice mail system (SMDI) protocol? (Choose all that apply.)

A. Cisco 2600.
B. Catalyst 6000
C. VG200.
D. Cisco 7200.
E. Catalyst 4000.

Answer: A, C

QUESTION 220:

You are a network engineer at Certkiller . Certkiller has a Cisco CallManager (CCM) 3.3 server that supports 5,000 phones. The company wants a 1:1 deployment option and purchase six new CCM 3.3 servers.
Once the new servers are added, what will be the maximum number of phones that the network can support?

A. 10,000
B. 15,000
C. 20,000
D. 25,000
E. 35,000

Answer: A