Exam Code: 642-414
Telephony Design Exam (IPTD)

<u>Demo Version</u>

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QUESTION 1
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 voice calls communication over the WAN network. If you want to implement external calls or call remote sites, you can use PSTN.

QUESTION 2
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
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 3
Which of these pieces of information most be addresses in the design of the IP telephony system for Itexamworld .com? (Choose three)

A. Itexamworld .com is using a public IP addressing scheme. They have four contiguous Class C addresses.
B. Itexamworld .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 Itexamworld .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
Itexamworld .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 4
You are in a meeting with the Itexamworld .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 Itexamworld .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 Itexamworld .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 5
You are in a meeting with the Itexamworld .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 Itexamworld .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 recommended because the voice-mail/e-mail server is running additional services that are required in the data segment.

QUESTION 6
You are doing a physical site survey of the Itexamworld .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 Itexamworld .com buildings in Cobb County will incur a toll charge when calling the Itexamworld .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 7
You are doing a physical site survey of the Itexamworld .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.

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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 power, lighting, and

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QUESTION 8
To provide the fastest response to an outage in the connection between the access layer, distribution layer and core of the Itexamworld .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 9
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 10

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 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 11
Itexamworld .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 12
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 13
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 14
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 15
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 16
Itexamworld .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 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 17
In their business day, Itexamworld .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 Itexamworld .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 18
In their busiest day, Itexamworld .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 Itexamworld .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 19
Itexamworld .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 Itexamworld .com with an additional range of numbers, 556-0000 through 556-3999. The LEC is currently sending Itexamworld .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 20
Currently, Itexamworld .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 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_ch
apter0918
6a00802c37f9.h
