Free Questions for GE0-807 by certsdeals

Shared by Dotson on 24-05-2024

For More Free Questions and Preparation Resources

Check the Links on Last Page

Question 1

Question Type: **MultipleChoice**

Which of the statements about gcti DNs is correct? (Choose 2 answers)

Options:

A- GctiDN must have a name in the following format: 'gcti: :XXX' where XXX is the type of functionality presented by this DN

B- Gctl DN must have in a Annex tab 'TServer' Section an option service-type with following value: gcti::XXX' where XXX is the type of functionality presented by this DN

C- Gctl DN must be a VoIP Service DN type only

D- Gcti DN can be a VoIP Service DN type

E- Gcti DN can be a Trunk

Answer:

B, D

Question 2

Question Type: **MultipleChoice**

When using the REFER method, where would you specify the party the call is being transferred to?

Options:

A- Refer header URI

B- To: header

C- contact: header

D- REFER To: header

E- Is not specified in the REFER messages

Answer:

A

Question 3

Question Type: **MultipleChoice**

Which Genesys component is used to load balance between multiple SIP Servers?

Options:

A- Load Distribution Server

B- SIP Server

C- Network SIP server

D- Media Server

Answer:

C

Explanation:

http://www.genesys.com/resources/events/2012_gforce_barcelona_ed_day_sip_server_8_1.pdf (See the Page #15).

Question 4

Question Type: **MultipleChoice**

If you need to choose a gateway at the same premise where the agent SIP endpoint is located to minimize network load for RTP traffic a3 for VoIP media services, such as MCU, MOH or voice recording, which of the following is the best solution?

Options:

A- Selection based on prefix match

B- Selection based on best availability

C- Selection based on geo-location

D- Selection based on priority settings

Answer:

C

Question 5

Question Type: **MultipleChoice**

You would like to record what the customer hears and says from the very beginning of the call to when it is terminated, regardless of whether or not an agent transferred the call, or created a conference, and so on. Which ON type should you use to enable recording?

Options:

A- Extension

B- RP

C- Trunk

D- VOIP port

Answer:

A

Question 6

Question Type: **MultipleChoice**

Which header in the INVITE message will SIP Server analyze in order to select the proper geo-location for an inbound call?

Options:

A- From

B- To

C- via

D- contact

Answer:

A

Explanation:

http://docs.genesys.com/Special:Repository/sip81rn.html?id=c4d480e4-94a2-4cf8-ac24-5ccf3ccb7759

Question 7

Question Type: MultipleChoice

You want to install one SIP Server and 3 Media Servers. You want to communicate with them directly using NET ANN. You also want each Media Server to support MOH, treatment and MCU. What will be the minimum number of VoIP Service DNs you have to configure to support this request?

Options:

A- 1

B- 3

C- 6

D- 9

Answer:

A

Question 8

Question Type: MultipleChoice

You have 3 separate remote sites using SIP technology without PBX. One location has one MOH and one MCU server, second location has a one MOH and one treatment server and third site has one MOH and one recording Media Server installed. The whole contact center has a rate of 30calls/sec. What is the minimal number of SIP servers to be installed?

Options:

A- 0

B- 1

C- 2

D- 3

Answer:

C

Question 9

Question Type: **MultipleChoice**

Which of the features below does a B2BUA deliver to a SIP-based VoIP architecture? (Choose 3 answers)

Options:

A- centralized call management

B- SIP-based VoIP interworking between LAN and WAN

C- management and monitoring of the entire call state

D- Provide conversion of TDM to DIP

E- connect legacy phones

Answer:

A, B, C

Explanation:

http://www.eetimes.com/document.asp?doc_id=1203045