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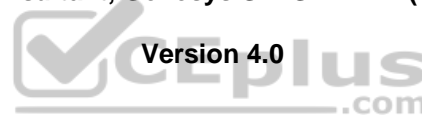
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GE0-807

System Consultant, Genesys SIP SERVER (GCP8 - SIP)



## Exam A

**QUESTION 1** A "Cradle to grave" ("beginning to end") call control is a functionality of which of the following? (Choose 2 answers).

- A. Registrar
- B. Stateless Proxy
- C. Stateful Proxy
- D. B2BUA
- E. Redirect Server

**Correct Answer:** AD

**Section:** (none)

**Explanation**

**Explanation/Reference:**

## QUESTION 2

Which of the following is responsible for sending "302 Moved Temporarily"?

- A. Registrar
- B. Stateless Proxy
- C. Stateful Proxy
- D. B2BUA
- E. Redirect Server

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: <https://tools.ietf.org/html/rfc3261#page-184>

## QUESTION 3

What does 1PCC mean?

- A. All call handling actions (answer, hold, initiate transfer, etc.) are performed on the SIP endpoint
- B. All call handling actions (answer, hold, initiate transfer, etc.) are performed on the same PC (no Hardware endpoint can be used)
- C. A T-lib client such as Agent Desktop can control a call which is actually between other parties such as sip endpoints
- D. Both, SIP endpoint and T-lib client such as Agent Desktop has to be integrated in one application

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 4** Which statement is correct?

- A. Each SIP-capable endpoint can be used in 3PCC, but not IPCC.
- B. SIP Server supports both 1PCC and 3PCC.

**Correct Answer:** B



**Section: (none)**

**Explanation**

**Explanation/Reference:**

**QUESTION 5** Which statement is correct?

- A. When using SIP Server with Framework 8, Genesys recommends that you Install SIP Server and Media Server on the same server to improve performance.
- B. The configuration wizards for all of the Genesys products can be installed from the Framework 8 Installation Media.
- C. Customers who have a legacy PBX deployed and now want to deploy GenesysFramework 8 with Genesys SIP Server will need to dismantle their legacy infrastructure prior to deploying SIP.
- D. Genesys Administrator is used to provision applications in Framework 8.

**Correct Answer: B**

**Section: (none)**

**Explanation**

**Explanation/Reference:**

**QUESTION 6**

What method can be used to send DTMF tones?

- A. UPDATE
- B. C INFO
- C. C MESSAGE
- D. C NOTIFY
- E. C OPTIONS

**Correct Answer: E**

**Section: (none)**

**Explanation**

**Explanation/Reference:**

Reference: <http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/sip/configuration/12-4t/sip-12-4t-book/voi-sip-dtmf.html>

**QUESTION 7** Which of the following is SIP Server deployment modes used with a Framework 8 solution? Choose 3 answers

- A. Stand Alone Mode
- B. Application Server Mode
- C. Multimedia Mode
- D. Customer Side Proxy Mode
- E. CTI Mode

**Correct Answer: ABD**

**Section: (none)**

**Explanation**

**Explanation/Reference:**

Reference: [http://genesyslab.info/wiki/index.php/Special:Repository/81fr\\_dep-sip.pdf?id=2e30d00a-05d6-4c84-a539-eb7ddcbde5f4](http://genesyslab.info/wiki/index.php/Special:Repository/81fr_dep-sip.pdf?id=2e30d00a-05d6-4c84-a539-eb7ddcbde5f4) (page 34)

**QUESTION 8** Which of the following stays in the signaling path during the entire session? Choose 2 answers

- A. Registrar
- B. Stateless Proxy
- C. Stateful Proxy

D. R B2BUA  
E. Redirect Server  
**Correct Answer:** CE  
**Section:** (none)  
**Explanation**

**Explanation/Reference:**

**QUESTION 9** If registering process requires a password and it is not provided, what message will the registrar send?

- A. 400 Bad request
- B. 401 Unauthorized
- C. 402 Payment required
- D. 403 Forbidden
- E. 404 not found

**Correct Answer:** B  
**Section:** (none)  
**Explanation**

**Explanation/Reference:**

Reference: <https://www.ietf.org/rfc/rfc3261.txt> (See 21.4.2 401 Unauthorized).

**QUESTION 10** What does 3PCC mean?

- A. All call handling actions (answer, hold, initiate transfer, etc.) are performed on the SIP endpoint
- B. All call handling actions (answer, hold, initiate transfer, etc.) are performed on the multiple PCs (In this case 3 PCs)
- C. T-lib signaling takes over SIP signaling, so third party application such as Agent Desktop can transfer VoIP packets to SIP Server and back using T-lib
- D. A T-lib client such as Agent Desktop can control a call which is actually between other parties such as SIP endpoints

**Correct Answer:** D  
**Section:** (none)  
**Explanation**

**Explanation/Reference:**

**QUESTION 11** Which of the following are valid configurations for Open standard SIP architecture?  
Choose 2 answers

- A. 1 SIP Server, 1 Softswitch
- B. 5 SIP Server, 1 Network SIP Server
- C. 1 SIP Server, 1 T-Server
- D. 3 SIP Server, 5 T-Server, 1 Cisco call Manager

**Correct Answer:** AC  
**Section:** (none)  
**Explanation**

**Explanation/Reference:**

**QUESTION 12** Which protocol is SIP Server using when in communication with Softswitch?

- A. CTI protocol

- B. T-Lib
- C. SIP
- D. Proprietary protocol

**Correct Answer:** C

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 13** Which of these components has the functionality of a redirect server?

Choose 2 answers

- A. T Server
- B. SIP Server
- C. Network SIP Server
- D. Media Server

**Correct Answer:** BD

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 14**

In a SIP Server Stand-alone mode, persistent registrar data can be configured to be held by which of the following?

- A. SIP Server
- B. Softswitch
- C. SIP Proxy
- D. Configuration Server

**Correct Answer:** D

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 15** Which statements are correct?

(Choose 2 answers)

- A. A third-party Softswitch must be integrated with sip server using a CTI link.
- B. If there are multiple outbound dialing rules configured within a single Class of Service (COS), and more than one rule matches the dialed number (containing a similar pattern). SIP Server will choose the outbound dialing rule with the mostmatched digits in it.
- C. When integrating SIP Server with an approved third-party Softswitch, all SIP endpoints must be registered with the particular Softswitch and not the SIP server.
- D. It is possible to configure Auto Agent Logout after a specific period of Agent inactivity on SIP Server 8.

**Correct Answer:** BD

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 16**

How is the Genesys SIP Server licensed?

- A. per seat
- B. per running instance
- C. per any DN
- D. per Agent

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 17** Which of the following types of conference calls can support more than three parties?

- A. First Party Call Control (1PCC)
- B. Third Party Call Control (3PCC)
- C. Both 1pcc and 3PCC
- D. Neither 1PCC or 3PCC

**Correct Answer:** B

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucme/admin/configuration/guide/cmeadm/cmeconf.html](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeadm/cmeconf.html)

**QUESTION 18** In a single site deployment, SIP Server can be deployed in which of the following modes?

Choose 3 answers

- A. Stand alone SIP Server
- B. SIP Server integrated with a Softswitch
- C. Open Standard SIP Architecture
- D. Hybrid architecture
- E. Application Server Mode

**Correct Answer:** ABC

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://www.genesys.com/resources/training/certification/Exam\\_Study\\_Guide\\_GCP8\\_SIP.pdf](http://www.genesys.com/resources/training/certification/Exam_Study_Guide_GCP8_SIP.pdf) (See the Describe SIP Server Deployment modes).

**QUESTION 19** For remote supervision what DN must be configured?

- A. gcti::supervisor
- B. gcti::park
- C. gctl::record
- D. No Additional DNs need to be configured

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://www.cisco.com/c/en/us/td/docs/ios/voice/fxs/configuration/guide/15\\_1/fxs\\_15\\_1\\_cg\\_book/fxsbasic.html](http://www.cisco.com/c/en/us/td/docs/ios/voice/fxs/configuration/guide/15_1/fxs_15_1_cg_book/fxsbasic.html)

**QUESTION 20**

Which of the following are SIP messages that SIP Server can map to T-Library messages? Choose 4 answers

- A. ACK
- B. INVITE
- C. INFO
- D. UPDATE
- E. REFER
- F. ANNC

**Correct Answer:** BCDE

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://docs.genesys.com/Special:Repository/80fr\\_dep-sip.pdf?id=6483fb5b-9ea1-43a0-8aeb-f38b19fb29a7](http://docs.genesys.com/Special:Repository/80fr_dep-sip.pdf?id=6483fb5b-9ea1-43a0-8aeb-f38b19fb29a7) (See the Page #177).

**QUESTION 21** Which statement is true?

- A. Geo-location allows choosing a device at the same premise where the Agent SIP Endpoint is located to minimize network load for RTP traffic.
- B. Geo-location feature is not suitable in selecting the most cost effective gateway for outbound calls.

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://docs.genesys.com/Special:Repository/80fr\\_dep-sip.pdf?id=6483fb5b-9ea1-43a0-8aeb-f38b19fb29a7](http://docs.genesys.com/Special:Repository/80fr_dep-sip.pdf?id=6483fb5b-9ea1-43a0-8aeb-f38b19fb29a7)

**QUESTION 22** Which statements are correct?  
(Choose 2 answers)

- A. It is possible to deploy a combination of standard TDM PBX(s) and sip server(s).
- B. Genesys SIP server is capable of processing SIP signing and RTP/RTCP streams as well.
- C. Customers who have a legacy PBX deployed and want to deploy Genesys SIP Server have to dismantle their legacy infrastructure.
- D. SIP Server supports the exchange of early media before a particular session is accepted.

**Correct Answer:** BD

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 23**

If multiple patterns are matched in a dial plan, which of the following describes how SIP Server could determine which pattern match to use? (Choose 2 answers)

- A. First Valid match
- B. Lowest Dial-plan-rule-(number)
- C. Most specific match
- D. Last Valid match

**Correct Answer:** BC

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 24**

You have 3 separate remote sites using SIP technology without P8X. 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 Network SIP servers needed for this installation?

- A. 0
- B. 1
- C. 2
- D. 3

**Correct Answer:** C  
**Section:** (none)  
**Explanation**

**Explanation/Reference:**

**QUESTION 25**

Which of the following is considered an industry standard as the voice quality measurement approach for VoIP networks?

- A. MOS
- B. PESQ
- C. QRT
- D. PSQM

**Correct Answer:** B  
**Section:** (none)  
**Explanation**

**Explanation/Reference:**  
Reference: <http://www.opticom.de/technology/polqa.php>



**QUESTION 26** If the option 'auto-logout-ready' is set to true, in what state will the Agent be logged-out?

- A. from any state even if Agent is on call
- B. from any state except if Agent is on call
- C. only from not-ready state
- D. only from not-ready state except work-related modes such as Aftercallwork or LegalGuard

**Correct Answer:** A  
**Section:** (none)  
**Explanation**

A. 0	2
B. 1	5
C. 2	7
D. 3	7
<b>Correct Answer:</b>	7

**Explanation/Reference:**

**QUESTION 27**

You have 3 separate remote sites using SIP technology without PBX, One location needs MOH and MCU capability, second location needs MOH and treatment capability and third site needs MOH and recording capability, what is the minimal number of Media Servers you need to install?

**Section:** (none)  
**Explanation**  
**Explanation/Reference:**

**QUESTION 28** Which of the following are valid parameters of the Announcement Treatment?

(Choose 2 answers)

- A. Language
- B. ID
- C. Music\_DN
- D. Duration

**Correct Answer:** AB

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://www.genesyslab.info/repository/PSDK/8.0-Java\\_API\\_Reference/com/genesyslab/platform/voice/protocol/doc-files/TLib%20Datatypes/TTreatmentType.html](http://www.genesyslab.info/repository/PSDK/8.0-Java_API_Reference/com/genesyslab/platform/voice/protocol/doc-files/TLib%20Datatypes/TTreatmentType.html) (See the value pairs for treatment text to speech).

**QUESTION 29**

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

- 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

**Correct Answer:** ABC

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://www.eetimes.com/document.asp?doc\\_id=1203045](http://www.eetimes.com/document.asp?doc_id=1203045)



**QUESTION 30**

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?

- A. 0
- B. 1
- C. 2
- D. 3

**Correct Answer:** C

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 31**

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?

- A. 1
- B. 3
- C. 6
- D. 9

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 32**

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

- A. From
- B. To
- C. via
- D. contact

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

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

**QUESTION 33**

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?

- A. Extension
- B. RP
- C. Trunk
- D. VOIP port

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**



**QUESTION 34**

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 a£3 for VoIP media services, such as MCU, MOH or voice recording, which of the following is the best solution?

- A. Selection based on prefix match
- B. Selection based on best availability
- C. Selection based on geo-location
- D. Selection based on priority settings

**Correct Answer:** C

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 35** Which Genesys component is used to load balance between multiple SIP Servers?

- A. Load Distribution Server
- B. SIP Server
- C. Network SIP server
- D. Media Server

**Correct Answer:** C

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://www.genesys.com/resources/events/2012\\_gforce\\_barcelona\\_ed\\_day\\_sip\\_server\\_8\\_1.pdf](http://www.genesys.com/resources/events/2012_gforce_barcelona_ed_day_sip_server_8_1.pdf) (See the Page #15).

**QUESTION 36** When using the REFER method, where would you specify the party the call is being transferred to?

- A. Refer header URI
- B. To: header
- C. contact: header
- D. REFER To: header
- E. Is not specified in the REFER messages

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 37** Which of the statements about gcti DN is correct?

(Choose 2 answers)

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



**Correct Answer:** BD

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 38** What is the SIP Server supported methods for initiating two-step call transfer?

(Choose 2 answers)

- A. INVITE
- B. PUBLISH
- C. INITIATE
- D. REFER
- E. MESSAGE

**Correct Answer:** AD

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://docs.genesys.com/Special:Repository/AudioCodes-SIP-Phone-SIP-Server-Application\\_Note.pdf?id=658f5674-f029-4d77-949b-cd705abba1a4](http://docs.genesys.com/Special:Repository/AudioCodes-SIP-Phone-SIP-Server-Application_Note.pdf?id=658f5674-f029-4d77-949b-cd705abba1a4)

**QUESTION 39**

Which statement is true?

- A. Recording is not possible from a routing strategy.
- B. If busy treatments are played from a routing strategy while Customer is waiting for an available Agent, as soon as a target is available, the busy treatment is stopped.

**Correct Answer:** B

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 40** Which of the following options can be configured on a trunk DN to enable Active Out-of-Service detection? (Choose 2 answers)

- A. 'recovery-timeout'
- B. 'oos-check'
- C. 'oos-force'
- D. 'priority'

**Correct Answer:** BC

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: <http://docs.genesys.com/index.php?title=Documentation:SIPS:SIPProxyDeployment:SIPProxyOverview:8.1.1&action=pdfbook> (page 6)

**QUESTION 41**

What is the response to REFER when the REFER method is used?

- A. 200 OK
- B. 202 Accepted
- C. 183 session in Progress
- D. C ACK
- E. C NOTIFY

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 42**

If you wish to allow recording on a call, and recording is configured for two devices on the call. Which one of the following scenarios is possible?

- A. Call recording will be started when both of the devices signal EventEstablished.
- B. Call recording will be started only on the device that first signals EventEstablished.
- C. Call recording will only be started when either one of the devices first signals Eventreleased.
- D. Call will not be recorded

**Correct Answer:**

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 43**

When the internal registrar is enabled and SIP Server is also configured to store registration information in the configuration database, where, in CME, can you find the contact option specifying current location?

- A. corresponding DN
- B. corresponding Agent Login
- C. corresponding Voice over IP Service
- D. corresponding SIP Server
- E. It is not visible in CME

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

#### QUESTION 44

Which of the following are methods supported by SIP Server for detecting whether a particular device is unavailable and needs to be placed in Out of Service state? (Choose 2 answers)

- A. Active out of service detection
- B. Passive out of service detection
- C. Auto out of service detection
- D. Presence subscription
- E. Registration
- F. class of service

**Correct Answer:** DE

**Section:** (none)

**Explanation**

**Explanation/Reference:**



**QUESTION 45** Which of the following are SIP methods that can be used to pass DTMF tones?  
(Choose 3 answers)

- A. In-band method by encoding DTMF tone as regular RTP packets
- B. In-band method by encoding DTMF tone as in specific RTP packets (RFC 2833)
- C. Out-of-band method by encoding DTMF tone in SIP message (UPDATE method)
- D. Out-of-band method by encoding DTMF tone in SIP message (MESSAGE method)
- E. Out-of-band method by encoding DTMF tone in SIP message (INFO method)

**Correct Answer:** BDE

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: <http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/sip/configuration/12-4t/sip-12-4t-book/voi-sip-dtmf.html#GUID-0E228A53-4405-4730-8D54-B37979E69A8B>

#### QUESTION 46

Which is the valid value from the SDP portion of the re-INVITE message if a call is requested to be put on hold?

- A. m=audio 0 RTP/AVP hold
- B. a=rtpmap:0 hold
- C. a=sendrecv
- D. c=IN IP 0.0.0.0
- E. C=INIP 192.168.1.199

**Correct Answer:** D

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://www.cisco.com/c/en/us/td/docs/routers/asr1000/configuration/guide/sbcu/2\\_xe/sbcu\\_2\\_xe\\_book/sbc\\_shd.html#pgfId-1015129](http://www.cisco.com/c/en/us/td/docs/routers/asr1000/configuration/guide/sbcu/2_xe/sbcu_2_xe_book/sbc_shd.html#pgfId-1015129)

**QUESTION 47** SIP Server 8 provides support for which of the following operating systems?

(Choose 3 answers)

- A. HP-UX 11/V3
- B. MAC OS 10.5
- C. Real Hat Enterprise Linux 5-4
- D. MS Windows Server 2003
- E. MS Windows Server 2008

**Correct Answer:** CDE

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 48** What is the meaning of the option 'sip-port' in the Tserver" section in 'options' tab of the SIP server?

- A. It is the port used by SIP Server to communicate with other T-lib clients such as URS, StatServer, etc
- B. It is the port used by SIP Server to communicate with other SIP devices such as Media Server, SIP Endpoints, Gateways etc. using SIP signaling only
- C. It is the port used by SIP Server to communicate using SNMP protocol with a NMS software
- D. It is the port used by SIP Server to communicate with other SIP devices such as Media Server, SIP Endpoints, gateways etc. using RTP/RTCP protocolE. It is the port used by SIP Server to communicate with other SIP devices such as Media Server, SIP Endpoints,
- F. Gateways etc. using both RTP/RTCP protocol and SIP signaling.

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 49** What is the default timeout defined in the SIP specification after which SIP Server will put a DN not responding to INVITE in Out of Service?

- A. 2 Seconds B. 4 Seconds
- C. 8 Seconds
- D. 16 Seconds
- E. 32 SecondsF. 64 Seconds

**Correct Answer:** E

**Section:** (none)

**Explanation**

**Explanation/Reference:****QUESTION 50**

Which of the following SIP Server features requires an MCU? (Choose 5 answers)

- A. 3 PCC conference - Single Step
- B. 3 PCC conference - Two Step
- C. Call Supervision
- D. Push video

E. Emergency recording  
F. voicemail  
G.Email

**Correct Answer:** ABDFG

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 51** Sip Server 8.x introduced the overload control feature, which allows the specification of a value to limit the number of interactions SIP Server handles. At what percentage of this value does SIP Server reject all traffic?

- A. 101%
- B. 120%
- C. 130%
- D. 150%

**Correct Answer:** B

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 52**

SIP Server can set Quality-of-Service bits to a user-defined value to prioritize SIP signaling traffic, which SIP Server option enables this configuration?

- A. 'sip-ip-tos'
- B. 'rtp-ip-tos'
- C. 'set-qos-ip'
- D. 'set-quality'

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 53** If you want to represent a Gateway in Configuration Layer, what type of DN do you need to create?

- A. ACD Queue
- B. Extension
- C. Routing Point
- D. Trunk
- E. VoIP Service

**Correct Answer:** E

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 54** Which statements are correct?  
(Choose 2 answers)

- A. With SIP Server, it is possible to use its internal registrar functionality and to instruct SIP server to contact an external registrar for registration process.
- B. In a sip server stand-alone mode installation without a softswitch integration, it is not necessary to create a 'switch' object in CME.C. If an initiator of an S-participant conference on an MCU drops the call, the conference will soil continue without the dropped party.
- D. SIP Server supports 3PCC with REFER method for two-step-transfer but not for single-step transfer.

**Correct Answer:** CD

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 55** SIP Server supports active out-of-service detection for which of the following types of ONs?

(Choose 2 answers)

- A. VoIP Service (MCU, treatment, etc.)
- B. Extension
- C. Trunk
- D. Route Point

**Correct Answer:** AB

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 56** Which statements are correct?

- A. The Genesys Media Server and Genesys Stream Manager are the only music sources available when using Genesys SIP Server 8.
- B. Genesys Media Server or Stream Manager performance is not affected by the codec selected for use.
- C. Media Server and Stream Manager require that URS make requests for media to be streamed to a caller.
- D. Genesys Media Server supports Treatments

**Correct Answer:** B

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 57** Which statements are correct?

(Choose 2 answers)

- A. You need to install a Media Server In order to use Instant Messaging to play an Announcement.
- B. SIP Server supports the Integration with Genesys Media Server with both NETANN and MSML.
- C. Media Server can, if needed, convert a codec to another format.
- D. Genesys Media Server is based on the MGCP protocol for enhanced media functionality.

**Correct Answer:** BC

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://docs.genesys.com/Special:Repository/81gvp\\_dep-gms.pdf?id=af7ab4db-d7d8-46b3-85f1-decbaa5e90eb](http://docs.genesys.com/Special:Repository/81gvp_dep-gms.pdf?id=af7ab4db-d7d8-46b3-85f1-decbaa5e90eb) (page 53, real-time transcoding)

**QUESTION 58**

When comparing the network bandwidth and performance (jitter, packet loss, latency) between G.711 and 6.729 codes, which one of the statements is true?

- A. G.729 and G.711 have the same bandwidth and network performance requirements  
B. G.729 requires less bandwidth than G.711, but G.711 requires better network performance  
C. G.729 requires less bandwidth than G.711, and there are no differences between G.711 and G.729 in their required network performance  
D. G.729 requires less bandwidth than G.711, but G.729 requires better network performance

**Correct Answer:** B

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Explanation:

G.711, with a sample size of 64kbit/s, achieves a maximum MOS of 4.1, whereas G.729, with a much smaller sample size of 8kbit/s, can achieve a MOS of 3.9. G.729 is "compressed eight times smaller than G.711 while sounding almost as good."

**QUESTION 59** To connect to Media Server or Stream Manager, what type of DN must be created?

- A. Email  
B. Position  
C. voice treatment port  
D. Voice over IP Service

**Correct Answer:** D

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 60**

Which of the following is a correct description of how Media Server or Stream Manager can be configured with SIP Server in a multi-tenant environment?

- A. One Stream Manager can be associated with only one SIP Server  
B. One Stream Manager can be associated with multiple SIP Servers only under the same Tenant  
C. One Stream Manager can be associated with multiple SIP Servers and Tenants

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://docs.genesys.com/Special:Repository/76fr\\_dep\\_sm.pdf?id=af8f39b5-a029-4cb9-a509-770b43ce75f2](http://docs.genesys.com/Special:Repository/76fr_dep_sm.pdf?id=af8f39b5-a029-4cb9-a509-770b43ce75f2)

**QUESTION 61** Which protocol is SIP Server using in communication with Media Server?

- A. CTI protocol  
B. T-Lib  
C. SIP  
D. H.323  
E. voice XML

**Correct Answer:** D

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://en.wikipedia.org/wiki/List\\_of\\_SIP\\_software](http://en.wikipedia.org/wiki/List_of_SIP_software)

**QUESTION 62** When configuring a Voice over IP Service ON in Genesys Administrator to enable treatment capability of Media Server using NET ANN, which options are mandatory in the TServer section of the Annex Tab of this DN? (Choose 2 answers)

- A. contact = host and port of the SIP server handling request for treatment
- B. contact = host and port of the Media Server handling request for treatment
- C. service-type = treatment
- D. request-uri = URI to be sent by SIP server to Media Server to play proper treatment (e.g. Filename to be played)
- E. name = name of the treatment to be played

**Correct Answer:** BD

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 63**

We want to use a treatment 'PlayAnnouncement' from routing strategy to play following file:

C:\ProgramFiles\Mediaserver\Announcement\l\_mulaw.wav, How would the URI in the INVITE message sent to Media Server look like?

- A. sip:annc@MediaServer:port;play=ftle://announcement/l;repeat=l
- B. sip:annc@MediaSen/er:port;plav=c:\ProgramFiles\MediaServer\announcement\l;repeat=l
- C. sip:annc@MediaServer:porc;play=announcement/'i\_mulaw.vvav;repeat=l
- D. sip:annc@MediaServer:port;play=l;repeat=l
- E. sip:annc@MediaServer:port;play=l\_mulaw.wav;repf^t

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**



**QUESTION 64**

What does the parameter 'confrole=monitor' in the INVITE message sent to Media Server indicate about this participant?

- A. is a customer
- B. is an agent
- C. is a supervisor doing silent monitoring
- D. is a supervisor doing whisper coaching
- E. has to be monitored by recording

**Correct Answer:** D

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 65**

In the case where MSML is configured between SIP Server and Media Server, what is the SIP method used to carry information between them both?

- A. XML
- B. OPTION
- C. T-Lib (Events/Requests)
- D. voice XML
- E. INFO
- F. MESSAGE

**Correct Answer:** E

**Section: (none)**

**Explanation**

**Explanation/Reference:**

Reference: <https://tools.ietf.org/html/rfc5707> (See SIP INFO).

**QUESTION 66** What Type of DN has to be created for SIP Server or Media Server integration using MSML?

- A. Trunk Group
- B. Voice over IP Service
- C. Routing Point
- D. Communication DN
- E. Voice over IP Port
- F. No DN is necessary. Just the SIP Server option "msml-support=true" needs to be set

**Correct Answer: B**

**Section: (none)**

**Explanation**

**Explanation/Reference:**

**QUESTION 67** When using Genesys Media Server, which of the following are the two types of supported media interfaces?  
(Choose 2 answers)

- A. ADSL
- B. NETANN
- C. MSML
- D. KVM / USB



**Correct Answer: BC**

**Section: (none)**

**Explanation**

**Explanation/Reference:**

**QUESTION 68** Which two ways can Genesys Media Server integrate with SIP Server?

Choose 2 answers

- A. Directly
- B. Through GVP
- C. Through Proxy Manager
- D. Through Resource Manager

**Correct Answer: BD**

**Section: (none)**

**Explanation**

**Explanation/Reference:**

Reference: [http://genesyslab.info/wiki/index.php/Special:Repository/81fr\\_dep-sip.pdf?id=2e30d00a-05d6-4c84-a539-eb7ddcbde5f4](http://genesyslab.info/wiki/index.php/Special:Repository/81fr_dep-sip.pdf?id=2e30d00a-05d6-4c84-a539-eb7ddcbde5f4)

**QUESTION 69** In SIP Server, how would you link an emulated ACD Queue to an associated RP?

- A. RP Default DN set to Queue

- B. Queue object Association set to RP
- C. RP Annex Tab option identifying Queue
- D. Same name suffix, prefixed by Q\_ and RP\_ respectively

**Correct Answer:** B

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 70** Noting the subtle differences, which of the following is the correct format for the SIP INVITE as per RFC 4240?

- A. sip:annc@;play=[:] B. slp:annc@:play=[:]
- C. sip:annc@:play=[:] D. sip:annc@ play=[:]

**Correct Answer:** D

**Section:** (none)

**Explanation**

**Explanation/Reference:**

**QUESTION 71** In a SIP direct call setup, what message does the originating UAC send to the UAS of the recipient?

- A. INVITE
- B. Ringing
- C. ACK
- D. OK



**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: file:///C:/Users/AbDullah/Downloads/CCVP%20VOICE%20Self-Study%20Guide%20Chapter%205.pdf (See the Page #54, 1st Point).

**QUESTION 72** Which of the following choices below contains all the messages that SIP is required to act on during the request/response handshake?

- A. INVITE, 200 OK and ACK
- B. INVITE, 200 OK. ACK and 200 ACCEPTED
- C. INVITE. 200 OK
- D. INVITE, 200 OK and 200 ACCEPTED

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://www.brocade.com/downloads/documents/html\\_product\\_manuals/SI\\_12502\\_ASDBG/wwhelp/wwhimpl/common/html/wwhelp.htm#context=SI\\_12502\\_ASLB\\_Guide&file=sip.3.2.html](http://www.brocade.com/downloads/documents/html_product_manuals/SI_12502_ASDBG/wwhelp/wwhimpl/common/html/wwhelp.htm#context=SI_12502_ASLB_Guide&file=sip.3.2.html)

**QUESTION 73** T-Lib messages can be mapped to all of the SIP messages listed below except which of the following?

- A. INVITE
- B. UPDATE
- C. REFER
- D. REGISTER

**Correct Answer:** D

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://docs.genesys.com/Special:Repository/80fr\\_dep-sip.pdf?id=6483fb5b-9ea1-43a0-8aeb-f38b19fb29a7](http://docs.genesys.com/Special:Repository/80fr_dep-sip.pdf?id=6483fb5b-9ea1-43a0-8aeb-f38b19fb29a7)

**QUESTION 74** Which of the following methods does SIP Server use to create a new SIP dialog?  
(Choose 2 answers)

- A. INVITE
- B. re-INVITE
- C. REFER
- D. REGISTER

**Correct Answer:** AC

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: <https://tools.ietf.org/html/rfc5057>

**QUESTION 75** Which of the following events occur when SIP Server receives INFO and UPDATE messages?

- A. EventRouteRequest
- B. EventAddressInfo
- C. EventEstablished
- D. EventAttachedDataChanged



**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: <https://genesys.secure.force.com/support/servlet/fileField?id=0BEU0000000TOjd>

**QUESTION 76**

When SIP Server is behind the Softswitch, a DN is considered out of service if it receives which of the following messages?

- A. 408 Request Timeout
- B. 404 Busy
- C. 482 Gone
- D. 486 Forbidden

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

Reference: [http://docs.genesys.com/Special:Repository/80fr\\_dep-sip.pdf?id=6483fb5b-9ea1-43a0-8aeb-f38b19fb29a7](http://docs.genesys.com/Special:Repository/80fr_dep-sip.pdf?id=6483fb5b-9ea1-43a0-8aeb-f38b19fb29a7) (See Page #167).

**QUESTION 77**

You have enabled the internal registrar. However, SIP Server does not update the key called contact of the corresponding DN when the endpoint successfully registers with SIP Server. Which two actions are required to address this issue?

(Choose 2 answers)

- A. Set the option 'external registrar' to false
- B. Set the option 'use-register-for-service-state' to true
- C. Set the 'internal-registrar-persistent' to true
- D. Grant 'change permissions' for the System account for the all DNs on the corresponding switch

**Correct Answer:** CD

**Section:** (none)

**Explanation**

**Explanation/Reference:**

#### QUESTION 78

SIP Server starts successfully. However, its status on SCS is "Service Unavailable" and SIP Server distributes EventLinkDisconnected to clients. Which of the following circumstances would be responsible for the status and EventLinkDisconnected message?

- A. Stream Manager is not registered with SIP Server
- B. Media Gateway is not registered with SIP Server
- C. SIP Server fails to open the SIP port
- D. The network is down

**Correct Answer:** B

**Section:** (none)

**Explanation**

**Explanation/Reference:**



#### QUESTION 79

Using SIP default settings, after how many seconds will an endpoint will be marked "out of service" without a response following an INVITE message?

- A. 6 seconds
- B. 10 seconds
- C. 24 seconds
- D. 32 seconds

**Correct Answer:** A

**Section:** (none)

**Explanation**

**Explanation/Reference:**

#### QUESTION 80

Which SIP Method is used by Active out of Service Detection to check a device for out of service status?

- A. INFOK
- B. CHECK
- C. MESSAGE
- D. OPTIONS
- E. SUBSCRIBE
- F. PUBLISH

**Correct Answer:** D

**Section:** (none)

**Explanation**

Explanation/Reference:

