System Consultant, Genesys SIP SERVER (GCP8 - SIP)

Genesys GE0-807

Version Demo

Total Demo Questions: 11

Total Premium Questions: 119

Buy Premium PDF

QUESTION NO: 1

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. B2BUA

E. Redirect Server

ANSWER: C D

Explanation:

Stateful Proxy is correct because it maintains transaction state while proxying SIP messages and can establish itself in the dialog's route set by adding a Record-Route header. Endpoints then use the resulting Route information for subsequent in-dialog requests, such as re-INVITE, UPDATE, BYE, and REFER, ensuring that the proxy continues to receive and forward signaling for the life of that dialog. SIP specifies proxy processing and Record-Route behavior in RFC 3261, section 16 and describes dialog route sets in section 12.1.

B2BUA is also correct because it terminates the SIP dialog from one leg and creates a separate dialog toward the other leg. As the logical endpoint of both call legs, it must remain involved whenever signaling is exchanged during the session. This architecture allows the B2BUA to apply call control, routing, policy, interoperability handling, and session-related services consistently throughout the call. In a Genesys SIP environment, this ongoing signaling control is fundamental to components that act as call-control intermediaries rather than merely providing registration or a one-time routing response.

QUESTION NO: 2

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. INVIT

D. INVITE, 200 OK and 200 ACCEPTED

ANSWER: A

Explanation:

INVITE, 200 OK and ACK is the core SIP signaling sequence for successfully establishing a dialog and session. The user agent client initiates the request by sending an INVITE, which carries the session-establishment request and commonly includes SDP media-negotiation information. Once the called endpoint accepts the session, it returns a final `200 OK` response. The originating endpoint must then send an ACK to confirm receipt of that successful final response. This ACK completes the three-way exchange for an accepted INVITE and allows both endpoints to treat the dialog as established.

This behavior is especially important in SIP Server deployments because call-control components must process the INVITE transaction statefully, correlate the final success response with the originating request, and ensure that the ACK is routed correctly within the established dialog. SIP can also use provisional responses while a call is progressing, but the required successful call-establishment handshake represented here is INVITE, followed by 200 OK, followed by ACK. RFC 3261 defines INVITE transaction processing and the ACK required for final INVITE responses: RFC 3261: SIP. Genesys SIP documentation is available at Genesys SIP Server documentation.

QUESTION NO: 3

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. Red Hat Enterprise Linux 5.4

D. MS Windows Server 2003

E. MS Windows Server 2008

ANSWER: C D E

Explanation:

SIP Server 8 supports the Windows Server and Red Hat Enterprise Linux platforms identified here: Red Hat Enterprise Linux 5.4, Microsoft Windows Server 2003, and Microsoft Windows Server 2008. These operating systems were included in the supported-platform matrix for the SIP Server 8 release and provide the operating-system prerequisites on which the SIP Server application, its required runtime components, and its associated configuration files can be deployed. Platform qualification is important for a SIP infrastructure component because the supported operating system determines the tested combinations of networking behavior, process management, libraries, and system administration tools used by the server. The Linux entry is properly named Red Hat Enterprise Linux 5.4; the original wording contained a typographical error. For implementation planning, administrators should also verify the precise service-pack, architecture, and patch requirements for the particular SIP Server maintenance release, because support is defined at that detailed level rather than solely by the operating-system family name. Genesys product documentation and release-specific deployment guidance are available through the Genesys Documentation portal and the Genesys Documentation site.

QUESTION NO: 4

A SIP endpoint sends an INVITE and receives a 486 Busy Here response. What does this response indicate?

A. The destination user is busy.

B. The destination user has moved temporarily.

C. The request requires authentication.

D. The destination does not exist.

ANSWER: A

Explanation:

486 Busy Here indicates that the called user agent was contacted successfully but is currently unable or unwilling to accept the call because it is busy. The response is generated by the user agent server for the specific destination that received the INVITE. SIP Server can use this final response when applying configured call handling, routing, or treatment logic.

This differs from a network-wide busy indication, such as 600 Busy Everywhere, which states that the callee is busy at all known locations. See RFC 3261, section 21.4.24.

QUESTION NO: 5

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

ANSWER: A

Explanation:

6 seconds is correct. In the default Genesys SIP Server endpoint-monitoring behavior, an endpoint that does not provide a response to an INVITE within the configured no-response interval is treated as unavailable and is marked out of service after six seconds. This fast endpoint-state transition allows routing components to stop selecting an endpoint that is not responding, rather than continuing to offer calls to a potentially unreachable destination. The setting concerns Genesys SIP Server's endpoint availability handling after an INVITE is sent; it is not simply the general SIP transaction timeout defined by RFC SIP timer behavior. The effective value can be changed through SIP Server configuration when deployment requirements call for a longer or shorter failure-detection interval, but the question explicitly asks for the default setting. For configuration context, consult the official Genesys SIP Server documentation and the Genesys Documentation portal, selecting the documentation version that matches the deployed GCP environment.

QUESTION NO: 6

For remote supervision what DN must be configured?

A. gcti::supervisor

B. gcti::park

C. gctl::record

D. No AdditionalDNs need to be configured

ANSWER: A

Explanation:

`gcti::supervisor` is the special DN that must be configured for Genesys SIP Server remote supervision. It is used as an AdditionalDN to identify the supervisor side of a remote-supervision operation and lets SIP Server apply the expected supervision behavior when a supervisor monitors, coaches, or otherwise supervises an agent interaction remotely. The DN is a reserved Genesys CTI service DN name, so it must be entered exactly as shown, including the `gcti::` prefix. Its configuration is part of enabling the remote-supervision feature rather than a normal agent or routing DN assignment. In practice, confirm that the DN is provisioned in the appropriate SIP Server configuration context and that the involved applications and endpoints are configured for the supervision capabilities required by the deployment. Genesys documentation for SIP Server configuration and feature behavior is available in the Genesys SIP Server documentation. For the broader configuration model used to define DNs and application objects, consult the Genesys Framework documentation.

QUESTION NO: 7

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

ANSWER: B

Explanation:

Media Gateway is not registered with SIP Server is the condition that matches both symptoms. SIP Server can complete its own process startup while still being unable to provide the telephony/media service expected by its clients. A registered Media Gateway is required for SIP Server to maintain the service link used for media-related call processing. When that gateway registration is absent, SIP Server regards the required event link as unavailable, reports its service state to Solution Control Server as *Service Unavailable*, and notifies connected clients through the `EventLinkDisconnected` event.

This distinction is important operationally: a successfully started process only confirms that the SIP Server executable initialized; it does not prove that all required external service dependencies have registered and are ready. The appropriate troubleshooting path is therefore to verify the Media Gateway process and its connectivity, configuration, registration parameters, and any relevant SIP Server logs, then confirm that the gateway is registered with the intended SIP Server instance. Genesys product documentation for SIP Server architecture and deployment is available through the Genesys SIP Server documentation; the broader documentation portal is available at Genesys Documentation.

QUESTION NO: 8

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

ANSWER: B C

Explanation:

Genesys Media Server supports NETANN and MSML as media-service interfaces. NETANN is Genesys's network-announcement interface, used by call-control applications to request media resources such as playing announcements, collecting digits, recording, conferencing, and related treatment functions. It is designed for integration with Genesys call-control components, including SIP Server, using Genesys-defined signaling and control conventions.

MSML, or Media Server Markup Language, is a standards-based XML control language for media servers. It lets an application control media operations through structured requests, including prompts, digit collection, recording, conferencing, and dialog management. Supporting MSML enables interoperability with components that use this media-control standard while retaining the media-processing functions supplied by Genesys Media Server.

These two interfaces identify how external call-control or application components request and manage media services; they are not physical connectivity types. For product documentation and deployment materials, see the Genesys Media Server documentation and the Genesys documentation portal.

QUESTION NO: 9

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. Out-of-band method by encoding DTMF tone 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)

ANSWER: A B E

Explanation:

DTMF can be conveyed for a SIP call using audio-band signaling, RTP telephone-event signaling, or SIP INFO signaling. “In-band method by encoding DTMF tone as regular RTP packets” represents ordinary audible DTMF tones sent within the negotiated audio media stream. This method is applicable when the media path preserves the tones sufficiently for the receiving endpoint or media resource to detect them.

“Out-of-band method by encoding DTMF tone in specific RTP packets (RFC 2833)” represents the commonly deployed RTP telephone-event approach. Rather than relying on audible tones in the voice payload, the endpoint sends a defined RTP event payload for each digit. RFC 2833 has been obsoleted by RFC 4733, whose telephone-event payload specification retains this approach: RFC 4733.

“Out-of-band method by encoding DTMF tone in SIP message (INFO method)” uses the SIP INFO request to carry mid-dialog signaling information such as a DTMF digit. SIP INFO is explicitly defined for application-layer session information that does not alter the SIP session state; see RFC 2976. The selected methods therefore cover the standard in-band RTP, RTP telephone-event, and SIP INFO approaches used with SIP endpoints and media servers.

QUESTION NO: 10

Which of the following statements about SIP Server and RTP media are correct? (Choose 2 answers)

A. SIP Server controls SIP signaling for call establishment and call control.

B. RTP media normally flows directly between the negotiated media endpoints.

C. SIP Server always relays RTP packets between all endpoints.

D. SIP Server performs codec transcoding for every call.

E. T-Library is used to transport RTP audio.

ANSWER: A B

Explanation:

SIP Server controls SIP signaling for call establishment and call control. It processes SIP requests and responses and coordinates the related Genesys telephony events. **RTP media normally flows directly between the negotiated media endpoints**, such as SIP phones, gateways, or Media Server resources, when the network topology and media negotiation permit it.

SIP Server is not normally an RTP media relay or a transcoding resource. Where media processing, recording, conferencing, or codec conversion is required, a suitable media component such as Genesys Media Server is used. RTP is specified in RFC 3550; SIP session establishment is specified in RFC 3261.

QUESTION NO: 11

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

ANSWER: C

Explanation:

2 is the minimum because SIP Server sizing is driven primarily by the engineered call-processing rate, not by a one-server-per-media-resource or one-server-per-site rule. The Music on Hold, Media Control Unit, treatment, and recording resources are SIP-connected media components; their distribution among the three remote locations does not by itself require a dedicated SIP Server at every location. SIP Server can provide call-control functions for SIP endpoints and media resources across reachable network sites.

For this sizing scenario, the required aggregate rate is 30 calls per second. Using the standard SIP Server capacity figure of 15 calls per second per server, two SIP Server instances are required to supply 30 calls per second of call-processing capacity. This calculation establishes the minimum capacity deployment: 30 divided by 15 equals 2. The deployment must also provide the required network connectivity and appropriate high-availability design for the production environment, but those considerations do not change the capacity-based minimum asked for here.

Genesys documentation describes SIP Server as the component that provides SIP call-control capabilities and integrates SIP endpoints and media resources. See the Genesys SIP Server documentation and the Genesys Documentation portal for SIP Server deployment and capacity-planning guidance.