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About This Software

This CD contains the following components:

- SIP Server provides access to SIP Communications Networks for Genesys applications. It functions as a SIP-based IP ACD, a SIP-based IP PBX, and as a SIP-based T-Server using SIP to control agents registered to another softswitch.
- Network SIP Server is a component of the Network IP Contact Center that enables Universal Routing Server (URS) to communicate with the network carrier and redirect interactions from the network to the appropriate SIP Server.
- Genesys SIP Proxy provides an interface for SIP communication between SIP devices and SIP Server components. It handles register requests, load-balances SIP transactions, and operates in N+1 high availability.
- sipspan2 is a log navigation tool for contact center troubleshooting, including log analysis and system-wide call tracing.

New Features in 8.1.x

The following new features were introduced in the June 2017 CD update for the 8.1.1 release:

- Enhanced support to SIP Feature Server for assigning calling profiles (such as dial plans) to VOIP Service DNs with the `service-type` option set to `softswitch` and Trunk DNs.
- High-Availability enhacement. The primary and backup SIP Servers now exchange information about calls and trigger synchronization of missing calls to the backup SIP Server after establishing the High-Availability (HA) connection.
- GIR solution support. Remote Recording integrated with Genesys T-Server for Skype for Business version 8.5.001.17 or later.

The following new features were introduced in the December 2016 CD update for the 8.1.1 release:

- Recording an Agent Greeting. Support for recording of the agent call leg during the personal greeting.
- HTTP Live Streaming support. SIP Server must be integrated with MCP version 8.5.161.34 or later.
- Support for the SIP INVITE timeout for individual DNs.

- Customizing music for music-on-hold treatments, which can now be enabled on a Routing Point.
- Controlling CPU usage overload by automatically reducing the server's log level when the CPU usage overload threshold is reached.
- New Standard-level log events to monitor SIP Feature Server availability.
- Support of IVR Recording solution integration with GVP/Media Server.
- Support for SRV FQDN—FQDN resolving to SRV records—received in the Contact or Record-Route headers of a SIP message.
- Support of Call Forwarding Loop with the Unified OpenScape Voice platform enhancement.
- Masking sensitive data in SIP messages contained in SIP Server logs.
- Support for the Red Hat Enterprise Linux AP 64-bit x86 7 operating system.
- Support for the VMware ESXi 6 virtualization platform.
- Support for Siemens OpenScape Voice version 9.

The following new features were introduced in the June 2016 CD update for the 8.1.1 release:

- Mute/Unmute control by the agent from their desktop application in standard 2-party calls.
- Support for Genesys Security Pack 8.5 including support for TLS v1.2.
- Control transition from Early Media to Connected Call from a routing strategy.
- Enhanced Reporting of Multiple Routing Attempts when `divert-on-ringing` is `false`.
- Support for the SIP Feature Server dial plan as an alternative to the internal SIP Server dial plan.
- Instant Messaging support for multi-site calls.
- No-Answer Supervision enhancement. Defining SIP Server's default action for setting the state of an agent who was not able to answer the routed call before the `after-routing-timeout` expired.
- Caller Information Delivery Content for AT&T Trunks enhancement. Support for passing the multipart body content received in INVITE messages to GVP.
- From Header in SIP INVITE. Enhanced ability to set the `From` header in outgoing SIP INVITE messages, with the option to specify different values for the agent call leg and customer call leg to meet billing requirements for some networks.
- HTTP Monitoring Interface to monitor various operational statistics for its internal modules and statistics relating to trunks.
- Support for Cisco UCM v11 and Cisco CUBE v11.
- Support for Genesys/AudioCodes 420HD v2.2.8.

The following new features were introduced in the December 2015 CD update for the 8.1.1 release:

- On-Hold Privacy during a Conference mutes a customer if an agent in the conference places the call on hold (preventing other conference participants such as a supervisor from hearing the customer).
- Enhanced Geo-location Support. SIP Server sends geo-location in a MIME format compliant with RFC 6442 when a call is routed to an external destination that requires the information.
- Find Me Follow Me. Call forwarding with sequential and/or simultaneous ringing, plus optional control based on time/day. Note: Support is limited to single-site deployments at this time.
- Shared Call Appearance is supported in SIP Business Continuity environments.
- Enhanced Presentation of Supervisor Presence & Conference Participants. SIP Server adds new information for Supervisor presence and Conference participants in the events distributed to clients, such as an agent

desktop application. This information, in LCTParty and LCTSupervisor fields of call events, is required for optimum presentation to the agent during complex or multi-site call flows.

- Enhanced Alternate Routing when URS/ORS are non-operational (or out-of-service) or unresponsive. SIP Server will route to one or more Default DN destinations in round-robin fashion, and can use `route` or `direct-uui` ISCC transaction types with out-of-signal-path transfer, or it can use ISCC Call Overflow and stay in-the-path.
- Private Conversations During a Conference. SIP Server supports an agent suspending/disconnecting one party temporarily during a conference using the TListenDisconnect request. Most commonly, the customer is suspended to allow a private consultation between the agent and another agent or expert.
- Enhanced Security for Agent Assisted Card Transactions Multi-site Support (DTMF Clamping in a Conference). Genesys SIP enables an agent to conference an IVR application, which collects customer-entered DTMF digits while masking the digits from the agent for purposes of enhanced security and PCI compliance. The agent is able to hear everything except the digits, and can provide some assistance. This feature is now supported in multi-site deployments with multiple SIP switches.
- Origination DN name and location provided as new key-value pairs in EventRinging. The agent desktop can use this information to collect extended data about the originating party, such as the agent name, and present it to the destination party while the phone is ringing.
- Support for Siemens OpenScape Voice version 8, including support for Business Continuity deployments.
- Active Recording support for IMS deployments (with minor limitations).
- Support for AudioCodes/Genesys 420HD IP phone v2.2.2, including agent login from the phone.

The following new features were introduced in the June 2015 CD update for the 8.1.1 release:

- Support for Caller Information Delivery body for AT&T trunks.
- Support for SIP Proxy deployment using Transport Layer Security (TLS).
- Geo-location for MSML-based services: strict matching enhancement.
- Geo-location support by GVP enhancement.
- Enhancement for nailed-up connection on Agent Login or Ready state.
- Support of Cisco UCM v10.5 in both modes: as a parking platform in front of Cisco UCM and as an application server.
- VoiceXML support for agent greeting in multi-site and Business Continuity deployments.
- Support for TDeleteFromConference requests in multi-site deployments.
- Shared Call Appearance (SCA) support.
- Phone-based agent login and state update enhancement.
- Support for TSingleStepConference requests to an external destination.

The following new features were introduced in the December 2014 CD update for the 8.1.1 release:

- Attach Call Progress Detection (CPD) results for Trunk Group-based calls to UserData and CallState of EventEstablished/EventReleased.
- Delay media connection between caller and agent until after personal greetings are applied.
- Hunt Groups with simultaneous ringing support in Business Continuity deployments.
- Support for TMuteTransfer requests.
- Strict matching of the geo-location when selecting an MSML-based media service.
- Enhancements for TCP connections: keep alive support and improved handling of broken socket connections.
- Support for reliability and voicemail session recovery.
- Support for AfterCallWork (ACW) completion in Business Continuity deployments.

- Enhanced support for switching between call supervision modes in MSML-based call monitoring.

The following new features were introduced in the July 2014 CD update for the 8.1.1 release:

- Hunt Groups enhancement to support sequential ringing.
- VXML support for agent greeting.
- IMS integration enhancement to route calls parked on a Media Server using a call-terminating leg.
- Nailed-up connections enhancement to support in Business Continuity deployments.
- Support for call participant locations in multi-site scenarios.
- ISCC Path Optimization support.
- Support of Windows 2012 64-bit.

The following new features were introduced in the March 2014 CD update for the 8.1.1 release:

- Trunk Capacity Control to restrict the number of both incoming and outgoing calls (previously only outgoing calls could be restricted).
- Enhanced media filtering to optionally block video, while allowing voice and Instant Messaging.
- Allow a routing strategy to prevent call recording by overriding the `record=true` option on a DN.
- Support for TLS connections to Resource Manager in active-active HA mode.
- Use the Request-URI of a SIP INVITE as the DNIS based on the `init-dnis-by-ruri` configuration option.
- Dial Plan enhancements:
    - Limit the length of the DNIS, and place extra “overdial” digits in an AttributeExtensions parameter.
    - Apply dial plan to outbound calls initiated with TMakePredictiveCall.

The following new features were introduced in the November 2013 CD update for the 8.1.1 release:

- Support for HTTP Digest authentication challenges for predictive calls.
- Ability to suppress agent greetings for different call types.
- Provision of Advice of Charge notifications for established calls.
- Provision of an original destination number in T-Events before the dial plan is applied.
- Support for 3pcc and 1pcc blind transfer operations.
- Ability to record a call without recording a music-on-hold treatment when a call is placed on hold.
- Call recording alarms enhancement.
- SIP-to-T-Library mapping enhancement using the `EXTRACT_SIP_HEADERS` extension key.

The following new features were introduced in the May 2013 CD update for the 8.1.1 release:

- Support for SIP Voicemail in a Business Continuity deployment. Requires 8.1.2 version of SIP Feature Server.
- Enhanced Group Mailbox functionality. Requires 8.1.2 version of SIP Feature Server.
- Enhanced capacity on Linux for up to 15,000 connected agents.

The following new features were introduced in the initial (March 2013) 8.1.1 release:

- Support for a restricted release of SIP Cluster. For information about Genesys SIP Cluster technology, contact your Genesys representative.
- Support for Genesys SIP Proxy.

Call Park/Retrieve support.
- Call Pickup support.
- Hunt Group support: Simultaneous ringing on Hunt Group members.
- IPv6 support.
- Support for Resource Manager in active-active HA mode.
- High-Availability enhancements:
    - NIC status monitoring
    - Recovery after network failure
    - Primary/backup sync improvement

  NOTE: HA improvements depend on 8.1.2+ Management Framework components.
- Support for Red Hat Enterprise Linux 6 64-bit native.
- Support for Cisco UCM v9.0.
- Support for Siemens OpenScape Voice version 7, in standard and SIP Business Continuity environments.

The following new features were introduced in the January 2013 CD update for the 8.1.0 release:

- Support for REFER authentication.
- Support for TAnswerCall when integrating with Microsoft Lync.
- Support of the `Answer-Mode` SIP header in `Auto` mode as described in RFC 5373.
- Support of beep detection for an outbound campaign executed in `Transfer` mode.

The following new features were introduced in the August 2012 CD update for the 8.1.0 release:

- Call Divert Destination. Ability to divert a call upon agent or IVR hangup.
- Support of the `divert-on-ringing` configuration option at a DN level.
- Support of geo-location functionality for Active Recording.
- Support for setting Not Ready state based on corresponding SIP response codes by using the `set-notready-on-busy` configuration option.

The following new features were introduced in the June 2012 CD update for the 8.1.0 release:

- Call Completion Feature support.
- Enabling additional parameters in Request-URI.
- Support for `Server` and `User-Agent` headers.
- Enhanced MWI support.

The following new features were introduced in the March 2012 CD update for the 8.1.0 release:

- Network Attended Transfer support.
- Siemens OpenScape Voice version 6 support.
- Improved 503 error response handling.

The following new features were introduced in the December 2011 CD update for the 8.1.0 release:

- Enhanced Network Asserted Identity functionality.
- Enhanced SIP error code mapping.
- Disconnect on remote agent logout.
- Call release tracking.

The following new features were introduced in the initial 8.1.0 release:

- SIP Business Continuity architecture provides full disaster recovery and site maintenance capabilities with near real-time failover. This feature is licensed by the “SIP Business Continuity” sellable item.

- Integration with Genesys SIP Voicemail provides full support for integration to voicemail, including deposit, retrieve, and Message Waiting Indication (MWI) capabilities; support for “group voicemail” and voicemail for individual agents. Note that Genesys SIP Voicemail must be purchased separately.
- Active Call Recording. SIP Server and Genesys Media Server form the core of an architecture which supports media stream replication for voice calls, eliminating the need for network SPAN ports. Media will be replicated to an external recording server. Full-Time, Selective, and Dynamic recording, plus screen captures of the agent desktop are supported.
- Multiple SIP-related enhancements:
  - Support for the Diversion header (RFC 5806).
  - Control over the SIP Alert-Info header via T-Library requests and DN configuration.
  - Configurable domain in the Refer-To header of a REFER message.
  - RFC 3263 support – highlighted by resolution of FQDNs via DNS SRV records.
  - Mapping T-Library events to SIP INFO messages.
  - Authentication of outbound calls to a Trunk DN, when challenged by destination.
- Advice of Charge (per TS 29.658). SIP Server works with Orchestration Server to provide billing/charging functionality known as “Advice of Charge." This feature is built upon the new T-Library to SIP INFO functionality.
- Enhanced Integration with Siemens OpenScape Voice version 5:
  - First-party call control supported for agents behind a Siemens OpenScape Voice switch (some limitations regarding allowed mix of 1pcc and 3pcc).
  - Support for “Split Node” deployments.
- Automatic call deletion after many failed routing attempts.
- Support for Alcatel-Lucent OmniPCX Spatial Redundancy (v9.x and v10.x).
- Support for “Silence” result from Call Progress Detection during Genesys Outbound or Proactive campaigns.
- Call Supervision for consultation calls.
- Support for 64-bit on Red Hat Linux 5 and Windows 2008 R2.
- Relay of NETANN (RFC 4240) requests to Genesys Media Server.
- Enhanced logging capabilities.
- Support for FlexNet Publisher 11.9.
- Support for Graceful Shutdown.
- Updated support for third-party endpoints and gateways. See the *Genesys Supported Media Interfaces Reference Manual* for a complete list of supported products and versions.

Directories on This CD

documentation
Contains the ReadMe file, the graphics for the ReadMe, and the versions.html file.

media_layer
Contains switch-specific software.

templates
Contains the application templates used for installation.

Documentation

Product documentation is provided on the Genesys Documentation website, and the Documentation Library DVD.

Any information regarding this release that was discovered too late to be included in the documentation is available in the Release Advisory.

In addition to an updated library of product documentation, the Genesys Customer Care website also contains product advisories that describe recently discovered issues related to Genesys products.

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Technical Support

Contacting

Genesys provides technical support to customers worldwide through Customer Care centers in eastern Canada, the United Kingdom, Australia, India, and Japan. You can contact Genesys Customer Care by telephone, e-mail, or on the World Wide Web.

For complete information on how and when to contact Customer Care, read the *Genesys Care Support Guide for On-Premises*. Please tell the Customer Care representative that you are a SIP Server 8.1 customer.

Licensing

Along with its software, Genesys supplies its customers with software licenses. Licenses manifest the customers' legal rights to use the features that Genesys software provides. To obtain the necessary product licenses, you will need to complete an order form, which has detailed information to assist you in placing an order. For complete information on obtaining licenses, refer to the *Genesys Licensing Guide* on the Genesys Documentation website and the licensing section of the *Genesys Migration Guide*.

Supported Operating Environment Information

Information on supported hardware and third-party software is available on the Genesys Customer Care website in the following documents:

- *Genesys Supported Operating Environment Reference Guide*
- *Genesys Supported Media Interfaces Reference Manual*

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Your Responsibility for Your System

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