Configuring Sonus SBC 5000 with Genesys Voice Platform Solutions

Application Notes

Last Updated: July 22, 2014

Sonus Equipment	Type	Version
SBC 5200	SBC 5000	04.00.00R0

3rd Party Equipment	Type	Version
Genesys Management Framework	SIP Server	8.1.100.98
GVP Media Server	Media Server	8.1.504.93

Contents

1 Document Overview

These Application Notes describe the configuration steps required for the Sonus Session Border Controller 5000 series (5100, 5110, 5200, 5210) to interoperate with the Genesys contact center systems. SBC 5000 series functionality was compliance tested using a SIP trunk to Genesys SIP Server from an SBC 5200.

The objective of this document is to describe the procedure to be followed during interoperability testing of SBC 5000 series with Genesys Systems.

The interoperability tested was between SIP clients, Genesys SIP Server, Genesys GVP Media Server, and Sonus SBC 5200.

For additional information on Sonus SBC 5000 series, visit http://www.sonusnet.com.

For additional information on Genesys Systems, visit http://www.genesys.com.

1.1 Glossary

Term	Definition
1pcc	First party Call Control. All telephony commands executed directly from the physical handset.
3pcc	Third party call control. All telephony commands executed on behalf of the physical handset by a computer software or application like SIP Server.
AOC	Advice Of Charge
B2B UA	Back to Back User Agent
CP	Calling Party
CPD	Call Progress Detection
CPE	Customer Premise Equipment – Genesys SIP Server is the CPE device in this case.
CTI	Computer Telephony Integration
DNIS	Dialed Number Identification Service
GVP	Genesys Voice Platform
IP	Internet Protocol
IPXC	IP Transfer Connect
IPTF	IP Toll Free
IW	Interaction Workspace

Term	Definition
MCP	Media Control Platform
MS	Media Server
PBX	Private Branch Exchange
PSX	Policy Server Exchange
RM	Resource Manager
RP	Routing Point (Genesys terminology) / Redirecting Party (AT&T terminology)
RS	Reporting Server
SDOP	Signaled Digits Out-Pulsed
SIP	Session Initiation Protocol
SM	Stream Manager
TP	Target Party
URS	Universal Routing Server
UUI	User to User Information

2 Introduction

This document provides a configuration guide for Sonus SBC 5000 Series (Session Border Controller) when connecting to the Genesys SIP Server and other internal PBX systems.

The Sonus SBC 5200 is a Session Border Controller that connects disparate SIP trunks, SIP PBXs, and communication applications within an enterprise. It can also be used as a SIP routing and integration engine. The Sonus SBC is the point of connection between the Genesys System and local PBX Systems that hosts internal SIP phones. The Sonus SBC is also the point of connection to external carrier SIP trunk providers for SIP Carrier connection. In this case, the internal PBX is out of scope of this document.

2.1 Audience

This technical document is intended for telecommunication engineers for the purpose of configuring both the Sonus SBC 5xx0 and aspects of the Genesys SIP Server products. There will be steps that require navigating the third-party and Sonus SBC Command Line Interface (CLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

Technical support on SBC 5000 can be obtained through the following:

- Phone: (978) 614-8589 or (888) 391-3434 (Toll-free)
- Web: http://sonusnetworks.force.com/PortalLoginPage

2.2 Requirements

The following equipment and software was used for the sample configuration provided:

Sonus Equipment	Type	Version
SBC 5200 BMC BIOS ConnexIP OS SonusDB EMA SBC	SBC 5000	04.00.00R0 V2.4.1 V2.1.2 V02.00.02-R00 sonusdb-V03.01.02-R002 ems-V04.00.00-R000 sbx-V04.00.00-R000

3rd Party Equipment	Type	Version
Genesys Management Framework	SIP Server	8.1.100.98
GVP Media Server	Media Server	8.1.504.93

2.3 Reference Configuration

A simulated enterprise site consists of the following elements: Genesys Administrator, Genesys Management Framework, GVP and Oracle Database. An SBC 5200 system running software version 4.0.0 R0 was used during testing. SIP trunks were used to connect the SBC to Genesys SIP Server.

2.3.1 Network Topology

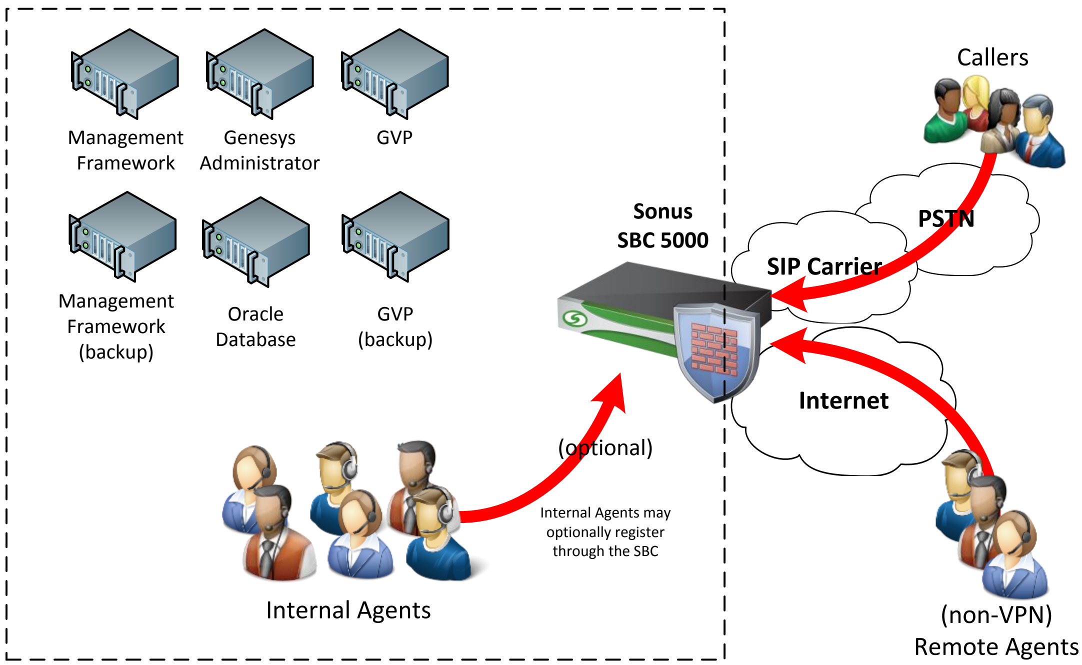


Figure 1 : Network Topology

The figure above represents the equipment that was used for the Genesys integration and certification testing. The Sonus SBC 5200 was used to route the calls to the Genesys SIP Server and SIP endpoints, depending on the test being run.

3 Interworking Architecture

This section discusses some general architectural issues.

3.1 Genesys Functionality

- General Genesys functionality provides the Contact Center application.
- All the Agents (SIP Phones) will Register with Genesys SIP Servers.
    - Remote Agents will always Register through the Sonus SBC 5000 unless using a VPN.
    - Internal Agents, and/or Remote Agents using a VPN, may or may not Register through the Sonus SBC 5000 depending on the customer's desire.
        - Some users are interested in protecting the Genesys SIP Servers (e.g. SIP Registration "avalanche" scenarios), in which case Agents would register through the SBC 5000.

3.2 Genesys Components

3.2.1 Genesys Voice Platform Solution (VPS)

The Genesys Voice Platform, SIP Server, Management Framework, and Genesys Administrator together constitute the Voice Platform Solution (VPS), which integrates voice self-service, agent-assisted service, and application-management functions into a single IP-based contact center solution.

3.2.2 Genesys Customer Interaction Management (CIM)

The Genesys Framework, a mandatory part of any Genesys-based interaction management system, provides the following functions required for the normal operation of any Genesys solution:

Configuration - Centralizes processing and storage of all the data required for Genesys solutions to work within a particular environment.
- **Access Control -** Sets and verifies users' permissions for access to, and manipulation of, solution functions and data.
- **Solution Control -** Starts and stops solutions and monitors their status.
- **Alarm Processing -** Defines and manages conditions critical to the operation of solutions.
- **Troubleshooting -** Hosts a user-oriented, unified logging system with advanced storage, sorting, and viewing capabilities.
- **Fault Management -** Automatically detects and corrects situations that might cause operational problems in solutions.
- **External Interfaces -** Enables communication with a variety of telephony systems and database management systems (DBMS).
- **Attached Data Distribution -** Supports the distribution of business data attached to interactions, within and across solutions.

3.2.3 **Genesys SIP Server**

Genesys SIP Server is the Genesys software component that provides an interface between the telephony hardware and the rest of the Genesys software components. It translates and keeps track of events and requests that come from, and are sent to the telephony device. Genesys SIP Server is a TCP/IP-based server that can also act as a messaging interface between SIP Server clients.

3.2.4 **Genesys Voice Platform (GVP)**

Genesys Voice Platform (GVP) is a software suite that constitutes a robust, carrier-grade voice processing platform. GVP unifies voice and web technologies to provide a complete solution for the customer self-service or assisted service.

GVP 8.1 provides a unified communication layer within the Genesys suite, and offers a robust solution that incorporates all required call control including computer-telephony integration (CTI) and media related functions.

GVP includes a basic test interactive Voice Response (IVR) application to be utilized for the verification of the platform.

This application hosts all the Media. Multiple GVP servers can be deployed. The lab had two GVP servers acting as active/active servers in a load balanced mode.

GVP reporting will be hosted on a Oracle 11g. (Tier 1 DB servers)

4 Configuring Sonus SBC 5000 Series

This section describes how to use the Sonus Command Line Interface (CLI) to configure and manage the SBC 5000 Series.

The SBC can equally be configured and managed by the Embedded Management Application (EMA), which is a Web-based interface management system for the Sonus SBC 5000 Series. However, documentation of the equivalent configuration steps via EMA is beyond the scope of this document.

4.1 SBC Configuration Diagram

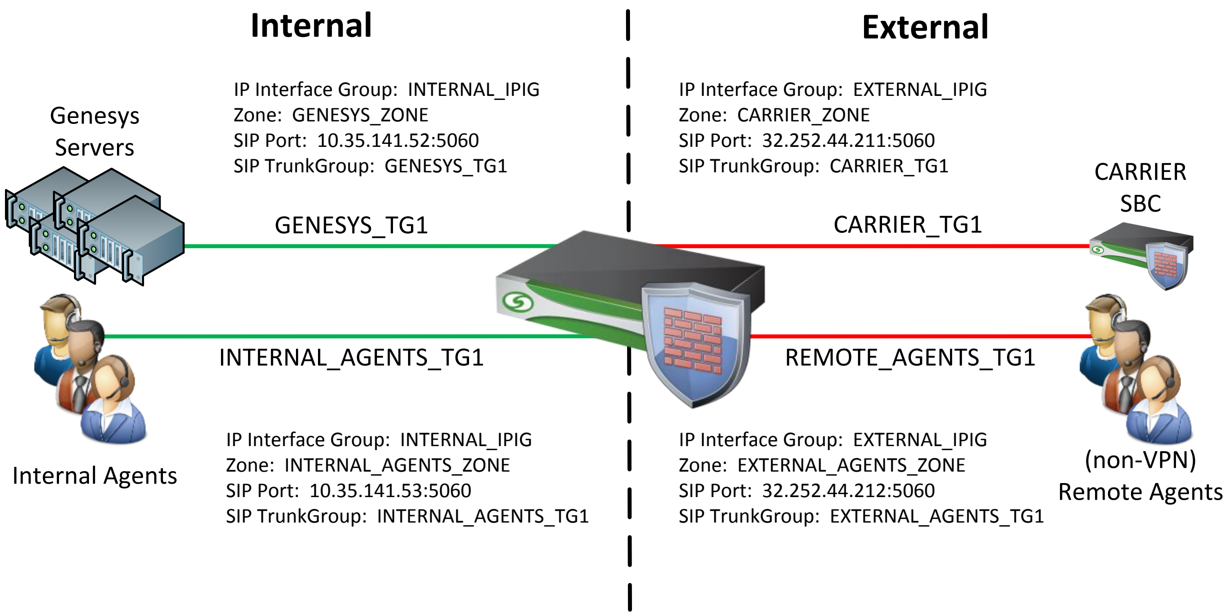


Figure 2: SBC 5000 SIP Trunk Diagram

4.2 SBC Naming Conventions

Unique address contexts are needed only when using overlapping IP address space. This deployment assumes no such overlapping IP space; thus, all configurations are in Address Context “default”.

4.2.1 Internal Side Configuration

Configuration Entity	Genesys	Internal Agents
Address Context	default	default
IP Interface Group	INTERNAL_IPIG	INTERNAL_IPIG
Zone	GENESYS_ZONE	INTERNAL_AGENTS_ZONE
Trunk Group	GENESYS_TG1	INTERNAL_AGENTS_TG1
IP Peer	GENESYS_PEER1 GENESYS_PEER2	None
IP Signaling Profile (IPSP)	GENESYS_IPSP	INTERNAL_AGENTS_IPSP
Packet Service Profile (PSP)	GENESYS_PSP	INTERNAL_AGENTS_PSP
Routing Label	TO_GENESYS_RL	None

4.2.2 External Side Configuration

Configuration Entity	SIP Carrier	Remote Agents (non-VPN)
Address Context	default	default
IP Interface Group	EXTERNAL_IPIG	EXTERNAL_IPIG
Zone	CARRIER_ZONE	REMOTE_AGENTS_ZONE
Trunk Group	CARRIER_TG1	REMOTE_AGENTS_TG1
IP Peer	CARRIER_PEER1 CARRIER_PEER2	None
IP Signaling Profile (IPSP)	CARRIER_IPSP	REMOTE_AGENTS_IPSP
Packet Service Profile (PSP)	CARRIER_PSP	REMOTE_AGENTS_PSP
Routing Label	TO_CARRIER_RL	None

4.3 SBC Configuration Workflow

```
|---- Global Configuration
|                   |----Media Port Range
|                   |---- DSP Resources
|                   |----Codec Entry
|
|---- Internal side Configuration
|                   |----IP Interface and IP Interface Group
|                   |----IP Static Routes
|
|---- Genesys Configuration
|                   |----Configuring Profiles
|                   |          |----Packet Service Profile
|                   |          |----IP Signaling Profile
|                   |----Configuring Address Context
```

```
|                   |         |----Zone
|                   |         |----SIP Signaling Port
|                   |         |----IP Peer
|                   |         |----SIP Trunk Group
|
|---- Internal Agent Side Configuration (optional)
|                   |----Configuring Profiles
|                   |         |----Packet Service Profile
|                   |         |----IP Signaling Profile
|                   |----Configuring Address Context
|                   |         |----Zone
|                   |         |----SIP Signaling Port
|                   |         |----IP Peer
|                   |         |----SIP Trunk Group
|
|---- External side Configuration
|                   |----IP Interface and IP Interface Group
|                   |----IP Static Routes
|
|---- Carrier Configuration
|                   |----Configuring Profiles
|                   |         |----Packet Service Profile
|                   |         |----IP Signaling Profile
|                   |----Configuring Address Context
|                   |         |----Zone
|                   |         |----SIP Signaling Port
|                   |         |----IP Peer
|                   |         |----SIP Trunk Group
|
|---- Remote Agent Configuration
|                   |----Configuring Profiles
|                   |         |----Packet Service Profile
|                   |         |----IP Signaling Profile
|                   |----Configuring Address Context
|                   |         |----Zone
|                   |         |----SIP Signaling Port
|                   |         |----IP Peer
|                   |         |----SIP Trunk Group
|
|---- Global Call Routing Configuration
|                   |----Genesys Side Routing
|                   |         |----Routing Label
|                   |----Carrier Side Routing
|                   |         |----Routing Label
|                   |----Agent Routing
|                   |----Routing
```

4.4 Global Configuration

4.4.1 General Global parameters

Increase the maximum PDU size to accommodate larger SIP packets and ensure that multiple contacts per Address Of Record (AoR) are enabled.

```
set global signaling sipSigControls maxPduSizeValue pdusize60kb
set global signaling sipSigControls multipleContactsPerAor enabled
commit
```

4.4.2 UDP Port Range for RTP (media)

The Sonus SBC 5000 series defaults to using a UDP port range of 1024-65148 for RTP (media) traffic. Many enterprise networking devices, including security devices, may assume a range of 16384-32767. The following configuration modifies the SBC to work within a more limited range with no changes to the existing devices.

This configuration is optional for the SBC and is only needed if the enterprise wishes to limit the RTP port range.

```
set system media mediaPortRange baseUdpPort 16384 maxUdpPort 32767
commit
```

4.4.3 DSP Resources

Ensure that the SBC has DSP resources allocated for compression/transcoding. Packet Service Profiles, configured later in this document, will use Conditional Transcoding, which will only function properly if the SBC has been deployed with DSP resources that have been allocated for transcoding.

```
set system mediaProfile compression 90 tone 10
commit
```

4.4.4 Codec Entry for G.711u SIP INFO

Create a Codec Entry for the G711u codec with DTMF Relay configured for Out of Band so DTMF information will be carried in the signaling protocol (e.g. SIP INFO method).

Parameter	Description
G711U_20_INFO	Name of codec entry
G711	Codec selected
20	Packet size in milliseconds
outOfBand	Type of DTMF Relay chosen: carriers DTMF in signaling protocol

```
set profiles media codecEntry G711U_20_INFO codec g711 packetSize 20
set profiles media codecEntry G711U_20_INFO dtmf relay outOfBand
commit
```

4.4.5 Codec Entry for G.711u RFC2833

Create another Codec Entry for the G711u codec with DTMF Relay configured for RFC2833 so DTMF information will be carried in the audio path as RTP events (e.g. 2833 method).

Parameter	Description
G711U_20_2833	Name of codec entry

G711	Codec selected
20	Packet size in milliseconds
rfc2833	Type of DTMF Relay chosen: carriers DTMF in signaling protocol

```
set profiles media codecEntry G711U_20_2833 codec g711 packetSize 20
set profiles media codecEntry G711U_20_2833 dtmf relay rfc2833
commit
```

4.4.6 Address Reachability Service Profile

Create an Address Reachability Service (ARS) Profile for Genesys servers, which will be applied to the Genesys SIP Trunk Group. ARS allows peers to be blacklisted when unresponsive, which allows faster route-advancing.

Parameter	Description
GENESYS_ARS	Name of Address Reachability Service (ARS) Profile
blkListAlgorithms timeouts,retryafter	Types of algorithms used for blacklisting endpoints: SIP INVITE timeouts and 503 w/retry-after response.
recoveryAlgorithm probe	Type of recovery mechanism for blacklisted endpoints. Probe mechanism is a SIP OPTIONS message.

```
set profiles services sipArsProfile GENESYS_ARS
set profiles services sipArsProfile GENESYS_ARS blkListAlgorithms timeouts,retryafter
set profiles services sipArsProfile GENESYS_ARS blkListAlgRetryAfterType sip-503
set profiles services sipArsProfile GENESYS_ARS blkListAlgTimeoutsType sip-invite
set profiles services sipArsProfile GENESYS_ARS blkListAlgTimeoutsNumTimeouts 4
set profiles services sipArsProfile GENESYS_ARS blkListAlgTimeoutsDuration 120
set profiles services sipArsProfile GENESYS_ARS recoveryAlgorithm probe
set profiles services sipArsProfile GENESYS_ARS recoveryAlgProbeInterval 30
set profiles services sipArsProfile GENESYS_ARS recoveryAlgProbeNumResponses 6
set profiles services sipArsProfile GENESYS_ARS recoveryAlgProbeDuration 240
commit
```

4.5 Internal Side SBC Configuration

4.5.1 IP Interface Group

The below configuration is for a Sonus 52x0 system using both Media 2 and Media 3 ports for Internal connectivity, which is the Sonus convention. A similar configuration (not shown) for the Sonus 51x0 system, which has only two Media ports, would use the Media 1 port for Internal connectivity per Sonus convention. SBC 5000 Media ports do not have dedicated Internal/External roles and, while recommended, the Sonus convention does not need to be followed. For more information on Media port deployment options or other network connectivity queries, refer to the *SBC 5000 Network Deployment Guide* or contact your local Sales team for information regarding the Sonus Network Design (NDAG) professional services offerings.

Create an IP Interface Group and assign it interfaces, including IP addresses.

Parameter	Description
default	Name of the address context
INTERNAL_IPIG	IP Interface Group name for the internal side of the SBC
SONUSSBC01A	SBC element name
IPIF2	Name for IP Interface (on pkt2)
IPIF3	Name for IP Interface (on pkt3)

pkt2	Gigabit Ethernet port used for internal signaling and media
pkt3	Gigabit Ethernet port used for internal signaling and media
10.35.141.50	IP address for the first internal media port
10.35.141.51	IP address for the second internal media port
27	IP subnet prefix (subnet mask in CIDR format)

```
set addressContext default ipInterfaceGroup INTERNAL_IPIG
commit
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF2 ceName SONUSSBC01A
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF2 portName pkt2
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF2 ipAddress 10.35.141.50
prefix 27
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF2 mode inService state enabled
commit
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF3 ceName SONUSSBC01A
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF3 portName pkt3
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF3 ipAddress 10.35.141.51
prefix 27
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF3 mode inService state enabled
commit
```

4.5.2 IP Static Route

Create a default route to the subnet's IP nexthop for the IP Interface Group and Interfaces.

Parameter	Description
default	Name of the address context
0.0.0.0	Default route
0	IP subnet prefix (subnet mask in CIDR format)
10.35.141.33	IP Nexthop for subnet
INTERNAL_IPIG	IP Interface Group name for the internal side of the SBC
IPIF2	Name for IP Interface (on pkt2)
IPIF3	Name for IP Interface (on pkt3)
100	Preference of the route within the Interface Group

```
set addressContext default staticRoute 0.0.0.0 0 10.35.141.33 INTERNAL_IPIG IPIF2 preference 100
set addressContext default staticRoute 0.0.0.0 0 10.35.141.33 INTERNAL_IPIG IPIF3 preference 100
commit
```

4.6 SBC Configuration for Genesys SIP Server

4.6.1 Profile Configuration

4.6.1.1 Packet Service Profile (PSP)

Create a Packet Service Profile (PSP) for the Genesys SIP trunk with a single codec specified. The PSP will be specified within the SIP Trunk Group configuration.

Parameter	Description
GENESYS_PSP	Name of the Genesys PSP

G711U_2833	Use the codec created earlier (global config section)
Conditional	Only transcode, if certain conditions are met
G711U	Only specify codec on this leg
differentDtmfRelay	Allow transcoding for different DTMF relay behaviors
differentPacketSize	Allow transcoding for different codec packet sizes

```
set profiles media packetServiceProfile GENESYS_PSP
set profiles media packetServiceProfile GENESYS_PSP codec codecEntry1 G711U_2833
set profiles media packetServiceProfile GENESYS_PSP packetToPacketControl transcode conditional
set profiles media packetServiceProfile GENESYS_PSP packetToPacketControl codecsAllowedForTranscoding
thisLeg g711u
set profiles media packetServiceProfile GENESYS_PSP packetToPacketControl
conditionsInAdditionToNoCommonCodec differentDtmfRelay enable differentPacketSize enable
commit
```

4.6.1.2 IP Signaling Profile (IPSP)

Create an IP Signaling Profile (IPSP) for the Genesys SIP trunk. The IPSP will be specified within the SIP Trunk Group configuration.

Parameter	Description
GENESYS_IPSP	Name of the Genesys IPSP

```
set profiles signaling ipSignalingProfile GENESYS_IPSP
set profiles signaling ipSignalingProfile GENESYS_IPSP ipProtocolType sipOnly
commit
```

4.6.2 Address Context Configuration

As mentioned earlier, as no overlapping IP addressing is used on the SBC in this document, all configuration will be done under the “default” Address Context.

4.6.2.1 Zone

Parameter	Description
default	Name of the address context
GENESYS_ZONE	Name of the Genesys Zone
2	A unique numeric identifier (2-2048) for the zone
appServer	The type of device for servicing SIP Registrations in an Access (non-Peering) scenario.

```
set addressContext default zone GENESYS_ZONE id 2 remoteDeviceType appServer
commit
```

4.6.2.2 SIP Signaling Port

A SIP Signaling Port is a logical address permanently bound to a specific zone and is used to send and receive SIP call signaling packets.

Parameter	Description

default	Name of the address context
GENESYS_ZONE	Name of the Genesys Zone
INTERNAL_IPIG	IP Interface Group name for the internal side of the SBC
sip-udp, sip-tcp	Transport protocols allowed for SIP signaling to Genesys SIP
10.35.141.52	IPv4 address for the SIP Signaling Address for the SBC
5060	SIP signaling TCP/UDP port of SBC
26	DiffServ Code Point value for SIP signaling traffic from SBC

```
set addressContext default zone GENESYS_ZONE sipSigPort 2 ipInterfaceGroup INTERNAL_IPIG
set addressContext default zone GENESYS_ZONE sipSigPort 2 transportProtocolsAllowed sip-udp,sip-tcp
set addressContext default zone GENESYS_ZONE sipSigPort 2 ipAddressV4 10.35.141.52
set addressContext default zone GENESYS_ZONE sipSigPort 2 portNumber 5060 dscpValue 26
set addressContext default zone GENESYS_ZONE sipSigPort 2 state enabled mode inService
commit
```

4.6.2.3 IP Peer

Create an IP Peer with the signaling IP addresses of the Genesys SIP Servers and assign it to the Genesys Zone.

The IP Peer entity is used on egress, while the ingressIpPrefix parameter in the sipTrunkGroup entity is used on ingress, for determining the applicable SIP Trunk Group.

These two IP Peers represent an active-standby pair of Genesys SIP Servers for redundancy.

Parameter	Description
default	Name of the address context
GENESYS_ZONE	Name of the Genesys Zone
GENESYS_PEER1	Name of the Genesys IP Peer
10.35.176.111	IP Address of Genesys SIP Server
5060	SIP signaling TCP/UDP port of Genesys SIP Server

Parameter	Description
default	Name of the address context
GENESYS_ZONE	Name of the Genesys Zone
GENESYS_PEER2	Name of the Genesys IP Peer
10.35.176.112	IP Address of Genesys SIP Server
5060	SIP signaling TCP/UDP port of Genesys SIP Server

```
set addressContext default zone GENESYS_ZONE ipPeer GENESYS_PEER1 ipAddress 10.35.176.111 ipPort 5060
set addressContext default zone GENESYS_ZONE ipPeer GENESYS_PEER2 ipAddress 10.35.176.112 ipPort 5060
commit
```

4.6.2.4 SIP Trunk Group

Create a SIP Trunk Group internally for the Genesys SIP Server and assign the corresponding Profiles configured earlier in this document.

Parameter	Description
default	Name of the address context
GENESYS_ZONE	Name of the Genesys Zone
GENESYS_TG1	Name of the SIP Trunk Group for the Genesys SIP
INTERNAL_IPIG	IP Interface Group name for the internal side of the SBC
10.35.176.111 10.35.176.112	IP Address of Genesys SIP Servers (active and standby)
32	IP prefix (subnet mask in CIDR format)
GENESYS_PSP	Earlier created PSP is applied in the Trunk Group
GENESYS_ARS	ARS Profile

```
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 media mediaIpInterfaceGroupName
INTERNAL_IPIG
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 ingressIpPrefix 10.35.176.111 32
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 ingressIpPrefix 10.35.176.112 32
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 policy signaling
ipSignalingProfile GENESYS_IPSP
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 policy media packetServiceProfile
GENESYS_PSP
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 services sipArsProfile
GENESYS_ARS
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 state enabled mode inService
commit
```

4.7 Configuration for Internal Agents (optional)

This section only applies if Internal Agents register to the Company's Genesys servers through the SBC. If Internal Agent phones register directly to the Genesys servers, then this section of configuration is not needed.

4.7.1 Profile Configuration

4.7.1.1 Packet Service Profile (PSP)

Create a Packet Service Profile (PSP) for the SIP trunk with a single codec specified. The PSP will be specified within the SIP Trunk Group configuration.

Parameter	Description
INTERNAL_AGENTS_PSP	Name of the Packet Service Profile (PSP)
G711U_20_2833	Use of codec created earlier (global config section)
Conditional	Only transcode if certain conditions are met
G711U	Only specify codec on this leg
differentDtmfRelay	Allow transcoding for different DTMF relay behaviors
differentPacketSize	Allow transcoding for different codec packet sizes

```
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP codec codecEntry1 G711U_20_2833
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP packetToPacketControl transcode conditional
```

```
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP packetToPacketControl
codecsAllowedForTranscoding thisLeg g711u
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP packetToPacketControl
conditionsInAdditionToNoCommonCodec differentDtmfRelay enable differentPacketSize enable
commit
```

4.7.1.2 IP Signaling Profile (IPSP)

Create an IP Signaling Profile (IPSP) for the SIP trunk to Internal Agents. The IPSP will be specified within the SIP Trunk Group configuration.

Parameter	Description
INTERNAL_AGENTS_IPSP	Name of the IPSP
minimizeRelayingOfMediaChangesFromOtherCallLeg enable	Enabling this flag suppresses redundant Modify offers being sent from SBC.
relayDataPathModeChangeFromOtherCallLeg enable	Enabling this flag in conjunction with minimizeRelayingOfMediaChangesFromOtherCallLeg, the HOLD modify offers (or any modify offer that changes the data-path-mode) will get propagated.
relayFlags refer enable	Relay the REFER from a registered phone to Genesys and not act upon it.
transparencyFlags referredByHeader enable	Referred-By header must be transparent in order for the SBC to pass it to Genesys

```
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP ipProtocolType sipOnly
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLeg enable
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP commonIpAttributes flags
relayDataPathModeChangeFromOtherCallLeg enable
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP commonIpAttributes relayFlags refer
enable
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP commonIpAttributes transparencyFlags
referredByHeader enable
commit
```

4.7.2 Address Context Configuration

As mentioned earlier, as no overlapping IP addressing is used on the SBC in this document, all configuration will be done under the “default” Address Context.

4.7.2.1 Zone

The Internal Agents Zone groups the objects to communicate to the Internal Agents.

Parameter	Description
default	Name of the address context
INTERNAL_AGENTS_ZONE	Name of Zone for Remote (non-VPN) Agents
201	A unique numeric identifier (2-2048) for the zone
accessDevice	The type of device for servicing SIP Registrations in an Access (non-Peering) scenario.

```
set addressContext default zone INTERNAL_AGENTS_ZONE id 3 remoteDeviceType accessDevice
```

```
commit
```

4.7.2.2 SIP Signaling Port

A SIP signaling port is a logical address permanently bound to a specific zone, and is used to send and receive SIP call signaling packets. In this case all the SIP packets for the InternalAgents will be sent and received using this SIP address and port.

Parameter	Description
default	Name of the address context
INTERNAL_AGENTS_ZONE	Name of Zone for Access
4	A unique identifier (1-2048) for the signaling port
INTERNAL_IPIG	IP Interface Group name for the internal side of the SBC
sip-udp,sip-tcp	Transport protocols allowed for SIP signaling to internal agents
10.35.141.53	IPv4 address for the SIP Signaling Address for the SBC
5060	SIP signaling TCP/UDP port of SBC
26	DiffServ Code Point value for SIP signaling traffic from SBC

```
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 ipInterfaceGroup INTERNAL_IPIG
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 transportProtocolsAllowed sip-udp,sip-
tcp
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 ipAddressV4 10.35.141.53
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 portNumber 5060 dscpValue 26
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 state enabled mode inService
commit
```

4.7.2.3 SIP Trunk Group

Create a SIP Trunk Group internally for any Internal Agents not registering directly to the Genesys servers. The ingressIpPrefix specified in the sipTrunkGroup is limited to the 10.0.0.0/8 network which is used by the enterprise in the example.

Assign the corresponding Profiles configured earlier in this document to the Trunk Group.

Parameter	Description
default	Name of the address context
INTERNAL_AGENTS_ZONE	Name of Zone for Internal Agents
INTERNAL_AGENTS_TG1	Name of the SIP Trunk Group for Internal Agents
INTERNAL_IPIG	IP Interface Group name for the internal side of the SBC
lateMediaSupport passthru	Passthru SIP reverse offer rather than converting it
10.0.0.0	IP Prefix
8	IP Mask (subnet mask in CIDR format)
INTERNAL_AGENTS_PSP	Earlier created PSP is applied in the Trunk Group
INTERNAL_AGENTS_IPSP	Earlier created IPSP is applied in the Trunk Group
NANP_ACCESS	Default Numbering Plan
rel100Support disabled	Disables 100rel option tag for provisional messages
relayNonInviteRequest enabled	Enables relaying of NonInvite SIP Request messages

psxRouteForSubscribe enabled	Enable SBC route lookup for SIP messages other than INVITE and REGISTER
requireRegistration required	Registration of endpoints is required on this Trunk Group. Prevents unregistered endpoints from making or receiving any calls
expires 60	The time (in seconds) a SIP endpoint's registration session lasts before requiring re-registration

```
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 media
mediaIpInterfaceGroupName INTERNAL_IPIG lateMediaSupport passthru
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 ingressIpPrefix
10.0.0.0 8
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 policy signaling
ipSignalingProfile INTERNAL_AGENTS_IPSP
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 policy media
packetServiceProfile INTERNAL_AGENTS_PSP
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 policy
digitParameterHandling numberingPlan NANP_ACCESS
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 signaling
rel100Support disabled relayNonInviteRequest enabled psxRouteForSubscribe enabled
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 signaling
registration requireRegistration required expires 60
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 state enabled
mode inService
commit
```

4.8 External Side SBC Configuration

4.8.1 IP Interface Group

The below configuration is for a Sonus 52x0 system using both Media 0 and Media 1 ports for Internal connectivity which is the Sonus convention. A similar configuration (not shown) for the Sonus 51x0 system, which only has a total of two Media ports, would use the Media 0 port for Internal connectivity per Sonus convention. SBC 5000 Media ports do not have dedicated Internal/External roles and, while recommended, the Sonus convention does not need to be followed. For more information on Media port deployment options or other network connectivity queries, refer to the *SBC 5000 Network Deployment Guide* or contact your local Sales team for information regarding the Sonus Network Design Administrator's Guide (NDAG) professional services offerings.

Create an IP Interface Group and assign interfaces; including IP addresses.

Parameter	Description
default	Name of the address context
EXTERNAL_IPIG	IP Interface Group name for the external side of the SBC
SONUSSBC01A	SBC element name
IPIF0	Name for IP Interface (on pkt0)
IPIF1	Name for IP Interface (on pkt1)
pkt0	Gigabit ethernet port used for external signaling and media
pkt1	Gigabit ethernet port used for external signaling and media
32.252.44.209	IP address for the first external media port
32.252.44.210	IP address for the second external media port
29	IP subnet prefix (subnet mask in CIDR format)

```
set addressContext default ipInterfaceGroup EXTERNAL_IPIG
commit
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF0 ceName SONUSSBC01A
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF0 portName pkt0
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF0 ipAddress 32.252.44.209
prefix 29
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF0 mode inService state enabled
commit
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF1 ceName SONUSSBC01A
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF1 portName pkt1
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF1 ipAddress 32.252.44.210
prefix 29
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF1 mode inService state enabled
commit
```

4.8.2 IP Static Route

Create a default route to the subnet's IP nexthop for the Interface and IP Interface Group.

Parameter	Description
default	Name of the address context
0.0.0.0	Default route
0	IP subnet prefix (subnet mask in CIDR format)
32.252.44.214	IP Nexthop for subnet
EXTERNAL_IPIG	IP Interface Group name for the external side of the SBC
IPIF0	Name for IP Interface (on pkt0)
IPIF1	Name for IP Interface (on pkt1)
100	Preference of the route within the Interface Group

```
set addressContext default staticRoute 0.0.0.0 0 32.252.44.214 EXTERNAL_IPIG IPIF0 preference 100
set addressContext default staticRoute 0.0.0.0 0 32.252.44.214 EXTERNAL_IPIG IPIF1 preference 100
commit
```

4.9 SBC Configuration for SIP Carrier

This section only applies if Contact Center callers ingress the enterprise via a SIP Carrier. If callers ingress the enterprise via local PSTN circuits (PRI or CAS trunks), then a SIP Trunk Group would need to be built to whatever Media Gateway is terminating the PSTN circuits (such as a Sonus GSX9000 Media Gateway). This section is also for a generic SIP Carrier. If one is available, refer to the Sonus SBC 5000 Application Note that is specific to your carrier.

4.9.1 Profile Configuration

4.9.1.1 Packet Service Profile (PSP)

Create a Packet Service Profile (PSP) for the SIP trunk with a single codec specified. The PSP will be specified within the SIP Trunk Group configuration.

Parameter	Description

CARRIER_PSP	Name of the PSP for SIP Carrier
G711U_20_2833	Use of codec created earlier (global config section)
Conditional	Only transcode if certain conditions are met
G711U	Only specify codec on this leg
differentDtmfRelay	Allow transcoding for different DTMF relay behaviors
differentPacketSize	Allow transcoding for different codec packet sizes

```
set profiles media packetServiceProfile CARRIER_PSP
set profiles media packetServiceProfile CARRIER_PSP codec codecEntry1 G711U_20_2833
set profiles media packetServiceProfile CARRIER_PSP packetToPacketControl transcode conditional
set profiles media packetServiceProfile CARRIER_PSP packetToPacketControl codecsAllowedForTranscoding
thisLeg g711u
set profiles media packetServiceProfile CARRIER_PSP packetToPacketControl
conditionsInAdditionToNoCommonCodec differentDtmfRelay enable differentPacketSize enable
commit
```

4.9.1.2 IP Signaling Profile (IPSP)

Create an IP Signaling Profile (IPSP) for the SIP Carrier SIP trunk. The IPSP will be specified within the SIP Trunk Group configuration.

Parameter	Description
CARRIER_IPSP	Name of the SIP Carrier IPSP
minimizeRelayingOfMediaChangesFromOtherCallLeg enable	Enabling this flag suppresses redundant Modify offers being sent from SBC.
relayDataPathModeChangeFromOtherCallLeg enable	Enabling this flag in conjunction with minimizeRelayingOfMediaChangesFromOtherCallLeg, the HOLD modify offers (or any modify offer that changes the data-path-mode) will get propagated.

```
set profiles signaling ipSignalingProfile CARRIER_IPSP
set profiles signaling ipSignalingProfile CARRIER_IPSP ipProtocolType sipOnly
set profiles signaling ipSignalingProfile CARRIER_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLeg enable
set profiles signaling ipSignalingProfile CARRIER_IPSP commonIpAttributes flags
relayDataPathModeChangeFromOtherCallLeg enable
set profiles signaling ipSignalingProfile CARRIER_IPSP
commit
```

4.9.2 Address Context Configuration

As mentioned earlier, as no overlapping IP addressing is used on the SBC in this document, all configuration will be done under the “default” Address Context.

4.9.2.1 Zone

This Zone groups the set of objects used for the communication to the SIP Carrier.

Parameter	Description
default	Name of the address context
CARRIER_ZONE	Name of the SIP Carrier Zone

4	A unique numeric identifier (2-2048) for the zone

```
set addressContext default zone CARRIER_ZONE id 4
commit
```

4.9.2.2 SIP Signaling Port

A SIP Signaling Port is a logical address permanently bound to a specific zone, and is used to send and receive SIP call signaling packets. In this case, it is bound to the SIP Carrier zone and will send and receive SIP packets for the SIP Carrier.

Parameter	Description
default	Name of the address context
CARRIER_ZONE	Name of the SIP Carrier Zone
4	A unique numeric identifier (1-2048) for the signaling port
EXTERNAL_IPIG	IP Interface Group name for the external side of the SBC
sip-udp	Transport protocols allowed for SIP signaling to SIP Carrier
32.252.44.211	IPv4 address for the SIP Signaling Address for the SBC
5060	SIP signaling TCP/UDP port of SBC
26	DiffServ Code Point value for SIP signaling traffic from SBC

```
set addressContext default zone CARRIER_ZONE sipSigPort 4 ipInterfaceGroup EXTERNAL_IPIG
set addressContext default zone CARRIER_ZONE sipSigPort 4 transportProtocolsAllowed sip-udp
set addressContext default zone CARRIER_ZONE sipSigPort 4 ipAddressV4 32.252.44.211
set addressContext default zone CARRIER_ZONE sipSigPort 4 portNumber 5060 dscpValue 26
set addressContext default zone CARRIER_ZONE sipSigPort 4 state enabled mode inService
commit
```

4.9.2.3 IP Peer

Create an IP Peer with the signaling IP address of the SIP Carrier peer and assign it to the SIP Carrier zone.

The IP Peer entity is used on egress. The ingressIpPrefix parameter in the sipTrunkGroup object is used on ingress for determining the applicable SIP Trunk Group.

Parameter	Description
default	Name of the address context
CARRIER_ZONE	Name of the SIP Carrier Zone
CARRIER_PEER1	Name of the SIP Carrier IP Peer
12.194.20.88	IP Address of SIP Carrier SIP Server
5060	SIP signaling TCP/UDP port of SIP Carrier SIP Server

Parameter	Description
default	Name of the address context
CARRIER_ZONE	Name of the SIP Carrier Zone
CARRIER_PEER2	Name of the SIP Carrier IP Peer

12.194.18.88	IP Address of SIP Carrier SIP Server
5060	SIP signaling TCP/UDP port of SIP Carrier SIP Server

```
set addressContext default zone CARRIER_ZONE ipPeer CARRIER_PEER1 ipAddress 12.194.20.88 ipPort 5060
set addressContext default zone CARRIER_ZONE ipPeer CARRIER_PEER2 ipAddress 12.194.18.88 ipPort 5060
commit
```

4.9.2.4 SIP Trunk Group

Create a SIP Trunk Group externally for the SIP Carrier and assign the corresponding Profiles configured earlier in this document.

Parameter	Description
default	Name of the address context
CARRIER_ZONE	Name of the SIP Carrier Zone
CARRIER_TG1	Name of the SIP Trunk Group for SIP Carrier
EXTERNAL_IPIG	IP Interface Group name for the external side of the SBC
12.194.16.0	IP Address of SIP Carrier SIP Server
21	IP prefix (subnet mask in CIDR format)
CARRIER_PSP	Earlier created PSP is applied in the Trunk Group

```
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 media mediaIpInterfaceGroupName
EXTERNAL_IPIG
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 ingressIpPrefix 12.194.16.0 21
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 policy signaling
ipSignalingProfile CARRIER_IPSP
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 policy media packetServiceProfile
CARRIER_PSP
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 policy digitParameterHandling
numberingPlan NANP_ACCESS
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 state enabled mode inService
commit
```

4.10 SBC Configuration for Remote Agents

This section applies only if Remote Agents are not using a VPN to register to the Company's Genesys servers. If agent remote access for is provided by a VPN, then this section does not apply and the agents should be treated as Internal Agents for the purposes of the SBC configuration.

4.10.1 Profile Configuration

4.10.1.1 Packet Service Profile (PSP)

Create a Packet Service Profile (PSP) for the Access SIP trunk with a single codec specified. The PSP will be specified within the SIP Trunk Group configuration.

Parameter	Description
REMOTE_AGENTS_PSP	Name of the Packet Service Profile (PSP)

G711U_20_2833	Use of codec created earlier (global config section)
Conditional	Only transcode if certain conditions are met
G711U	Only specify codec on this leg
differentDtmfRelay	Allow transcoding for different DTMF relay behaviors
differentPacketSize	Allow transcoding for different codec packet sizes

```
set profiles media packetServiceProfile REMOTE_AGENTS_PSP
set profiles media packetServiceProfile REMOTE_AGENTS_PSP codec codecEntry1 G711U_20_2833
set profiles media packetServiceProfile REMOTE_AGENTS_PSP packetToPacketControl transcode conditional
set profiles media packetServiceProfile REMOTE_AGENTS_PSP packetToPacketControl
codecsAllowedForTranscoding thisLeg g711u
set profiles media packetServiceProfile REMOTE_AGENTS_PSP packetToPacketControl
conditionsInAdditionToNoCommonCodec differentDtmfRelay enable differentPacketSize enable
commit
```

4.10.1.2 IP Signaling Profile (IPSP)

Create an IP Signaling Profile (IPSP) for the SIP trunk to Remote Agents. The IPSP will be specified within the SIP Trunk Group configuration.

Parameter	Description
REMOTE_AGENTS_IPSP	Name of the IPSP
minimizeRelayingOfMediaChangesFromOtherCallLeg enable	Enabling this flag suppresses redundant Modify offers being sent from SBC.
relayDataPathModeChangeFromOtherCallLeg enable	Enabling this flag in conjunction with minimizeRelayingOfMediaChangesFromOtherCallLeg, the HOLD modify offers (or any modify offer that changes the data-path-mode) will get propagated.
relayFlags refer enable	Relay the REFER from a registered phone to Genesys and not act upon it.
transparencyFlags referredByHeader enable	Referred-By header must be transparent in order for the SBC to pass it to Genesys

```
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP ipProtocolType sipOnly
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLeg enable
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP commonIpAttributes flags
relayDataPathModeChangeFromOtherCallLeg enable
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP commonIpAttributes relayFlags refer enable
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP commonIpAttributes transparencyFlags
referredByHeader enable
commit
```

4.10.2 Address Context Configuration

As mentioned earlier, as no overlapping IP addressing is used on the SBC in this document, all configuration will be done under the “default” Address Context.

4.10.2.1 Zone

The Remote Agents Zone groups the objects to communicate to the Remote (non-VPN) Agents.

Parameter	Description
default	Name of the address context
REMOTE_AGENTS_ZONE	Name of Zone for Remote (non-VPN) Agents
5	A unique numeric identifier (2-2048) for the zone
accessDevice	The type of device for servicing SIP Registrations in an Access (non-Peering) scenario.

```
set addressContext default zone REMOTE_AGENTS_ZONE id 5 remoteDeviceType accessDevice
commit
```

4.10.2.2 SIP Signaling Port

A SIP signaling port is a logical address permanently bound to a specific zone, and is used to send and receive SIP call signaling packets. In this case all the SIP packets for the Remote Agents will be sent and received using this SIP address and port.

Parameter	Description
default	Name of the address context
REMOTE_AGENTS_ZONE	Name of Zone for Access
5	A unique identifier (1-2048) for the signaling port
EXTERNAL_IPIG	IP Interface Group name for the external side of the SBC
sip-udp,sip-tcp	Transport protocols allowed for SIP signaling to Access
32.252.44.212	IPv4 address for the SIP Signaling Address for the SBC
5060	SIP signaling TCP/UDP port of SBC
26	DiffServ Code Point value for SIP signaling traffic from SBC

```
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 ipInterfaceGroup EXTERNAL_IPIG
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 transportProtocolsAllowed sip-udp,sip-
tcp
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 ipAddressV4 32.252.44.212
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 portNumber 5060 dscpValue 26
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 state enabled mode inService
commit
```

4.10.2.3 SIP Trunk Group

Create a SIP Trunk Group externally for any Remote Agents that are not using a VPN. The ingressIpPrefix specified in the sipTrunkGroup is completely wild-carded, so that Remote Agents may use IP addressing from any provider. If Remote Agents can be assumed to only come from specific IP ranges, then the ingressIpPrefix may be configured for those more specific ranges.

Assign the corresponding Profiles configured earlier in this document to the Trunk Group.

Parameter	Description
default	Name of the address context
REMOTE_AGENTS_ZONE	Name of Zone for Remote Agents
REMOTE_AGENTS_TG1	Name of the SIP Trunk Group for Remote Agents
EXTERNAL_IPIG	IP Interface Group name for the external side of the SBC

lateMediaSupport passthru	Passthru SIP reverse offer rather than converting it
sourceAddressFiltering disabled	Disable sourceAddressFiltering when NAT support is enabled.
0.0.0.0	IP Prefix
0	IP Mask (subnet mask in CIDR format)
REMOTE_AGENTS_PSP	Earlier created PSP is applied in the Trunk Group
REMOTE_AGENTS_IPSP	Earlier created IPSP is applied in the Trunk Group
NANP_ACCESS	Default Numbering Plan
signalingNat enabled mediaNat enabled	Enable media and signaling NAT support for Remote Agents
rel100Support disabled	Disables 100rel option tag for provisional messages
relayNonInviteRequest enabled	Enables relaying of NonInvite SIP Request messages
psxRouteForSubscribe enabled	Enable SBC route lookup for SIP messages other than INVITE and REGISTER
requireRegistration required	Registration of endpoints is required on this Trunk Group. Prevents unregistered endpoints from making or receiving any calls
expires 60	The time (in seconds) a SIP endpoint's registration session lasts before requiring re-registration

```
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 media
mediaIpInterfaceGroupName EXTERNAL_IPIG lateMediaSupport passthru sourceAddressFiltering disabled
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 ingressIpPrefix
0.0.0.0 0
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 policy signaling
ipSignalingProfile REMOTE_AGENTS_IPSP
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 policy media
packetServiceProfile REMOTE_AGENTS_PSP
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 policy
digitParameterHandling numberingPlan NANP_ACCESS
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 services natTraversal
signalingNat enabled mediaNat enabled
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 signaling
rel100Support disabled relayNonInviteRequest enabled psxRouteForSubscribe enabled
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 signaling
registration requireRegistration required expires 60
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 state enabled mode
inService
commit
```

4.11 Global Call Routing Configuration

A Routing Label (RL) is a user-named object that contains a list of one or more nexthop peers - defined as Routing Label Routes - that can reach a specified destination.

A Routing Label Route (RLR) defines a single peer (Trunk Group + IP Peer) to which the call can be delivered. There may be many Routing Label Routes (1 to n) in a Routing Label.

For each call placed to a destination Routing Label, the SBC will advance through the list of peers (RLRs) until the call is completed or the list is exhausted. The RL's Prioritization Type determines the order in which the list will be processed.

Routing Labels are then assigned within the Route entity.

4.11.1 **Element Routing Priority**

The Element Routing Priority (ERP) Profile determines the priority or precedence for criteria used for call routing. An ERP profile is then applied in the Trunk Group entity.

When providing support for SIP Registration-based endpoints (sometimes referred to as an "Access" environment to differentiate it from "Trunking"), the SBC must have an ERP profile which prioritizes the Trunk Group entity above others. This allows routing of traffic from the ingress Trunk Group to the Trunk Group of the SIP Server for messages lacking called/calling numbers (e.g. SIP Registrations, etc…)

This prioritization can be accomplished by creating a new ERP profile and applying it on all the Access-related Trunk Groups (i.e. INTERNAL_AGENTS_TG1 and REMOTE_AGENTS_TG1) or by simply modifying the default ERP profile (DEFAULT_IP). This document modifies the default ERP profile.

Parameter	Description
DEFAULT_IP	Name of default object for the profile; in this case, the ERP.

```
set profiles callRouting elementRoutingPriority DEFAULT_IP entry nationalType 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry nationalType 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry _private 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry _private 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry nationalOperator 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry nationalOperator 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry transit 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry transit 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry trunkGroupCutThrough 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry trunkGroupCutThrough 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry localOperator 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry localOperator 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry userName 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry userName 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry internationalOperator 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry internationalOperator 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry longDistanceOperator 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry longDistanceOperator 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry othertrunkGroupChosen 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry othertrunkGroupChosen 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry internationalType 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry internationalType 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry mobile 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry mobile 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry test 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry test 1 entityType trunkGroup
commit
```

4.11.2 **Genesys Routing**

4.11.2.1 Routing Label

Create a Routing Label with a single Routing Label Route to bind the Genesys SIP Trunk Group with the Genesys SIP IP Peer.

Parameter	Description

TO_GENESYS_RL	Name of the Routing Label for Genesys SIP
Sequence	The prioritization of Routing Label Routes within a Routing Label
1	The first Routing Label Route within the Routing Label
GENESYS_TG1	Trunk Group for Genesys SIP
GENESYS_PEER1	IP Peer for Genesys SIP

Parameter	Description
TO_GENESYS_RL	Name of the Routing Label for Genesys SIP
Sequence	The prioritization of Routing Label Routes within a Routing Label
2	The second Routing Label Route within the Routing Label
GENESYS_TG1	Trunk Group for Genesys SIP
GENESYS_PEER2	IP Peer for Genesys SIP

```
set global callRouting routingLabel TO_GENESYS_RL routePrioritizationType sequence action routes
routingLabelRoute 1 trunkGroup GENESYS_TG1 ipPeer GENESYS_PEER1 inService inService
set global callRouting routingLabel TO_GENESYS_RL routePrioritizationType sequence action routes
routingLabelRoute 2 trunkGroup GENESYS_TG1 ipPeer GENESYS_PEER2 inService inService
commit
```

4.11.3 **SIP Carrier Routing**

4.11.3.1 Routing Label

Create a Routing Label with a single Routing Label Route to bind the SIP Carrier Trunk Group with the SIP Carrier IP Peers.

Parameter	Description
TO_CARRIER_RL	Name of the Routing Label for SIP Carrier
Sequence	The prioritization of Routing Label Routes within a Routing Label
1	The first Routing Label Route within the Routing Label
CARRIER_TG1	Trunk Group for SIP Carrier
CARRIER_PEER1	IP Peer for SIP Carrier

Parameter	Description
TO_CARRIER_RL	Name of the Routing Label for SIP Carrier
Sequence	The prioritization of Routing Label Routes within a Routing Label
2	The first Routing Label Route within the Routing Label
CARRIER_TG1	Trunk Group for SIP Carrier
CARRIER_PEER2	IP Peer for SIP Carrier

```
set global callRouting routingLabel TO_CARRIER_RL routePrioritizationType sequence action routes
routingLabelRoute 1 trunkGroup CARRIER_TG1 ipPeer CARRIER_PEER1 inService inService
set global callRouting routingLabel TO_CARRIER_RL routePrioritizationType sequence action routes
routingLabelRoute 2 trunkGroup CARRIER_TG1 ipPeer CARRIER_PEER2 inService inService
commit
```

4.11.4 **Agent Routing**

No Routing Labels need to be provisioned for Remote Agents or Internal Agents. The SBC will use a combination of a dynamically created entry for the endpoint upon a successful SIP Registration along with a Destination Trunk Group (DTG) parameter in the SIP message sent by the Genesys SIP Server to properly route traffic to a registered endpoint.

4.11.5 **Routing**

Routing is the final step in the SBC configuration which must be provisioned in order to send calls to the correct destination. Within the Route entity, all available Route Match Criteria are used to determine the most specific match which is linked to Routing Labels (destinations). The result of a Route match is a Routing Label. Routing Labels were created in earlier sections of this document.

For the purposes of this Application Note, routing was kept simple and limited to lab scenarios.

Trunk Group routing was required for SIP Registration-based endpoints. Both REMOTE_AGENTS_TG1 and INTERNAL_AGENTS_TG1 route to the Genesys servers (via the TO_GENESYS_RL Route Label which maps to the GENESYS_TG1 Trunk Group and Genesys IP Peers). Trunk group routing can also be used for non-registration-based endpoints in simple scenarios (e.g. a single PBX and a single SIP carrier), but that scenario was not applicable to this Application Note.

Called Number routes were also created for the lab to enable call testing. Called numbers beginning with the digits “68” were routed to the carrier, while called numbers beginning with the digits “21443268” were routed to the Genesys servers. In practice, call routing configuration will likely revolve around the enterprise’s dial plan. For specific help in planning and/or implementing your routing contact your local Sales team for information regarding the Sonus Routing Design (SSDB) professional services offerings.

Route Match Criteria	Route Label
Called Number begins with 68	Match Criteria with Destination of TO_CARRIER_RL
Called Number begins with 21443268	Match Criteria with Destination of TO_GENESYS_RL
Originating Trunk Group = REMOTE_AGENTS_TG1	Match Criteria with Destination of TO_GENESYS_RL
Originating Trunk Group = INTERNAL_AGENTS_TG1	Match Criteria with Destination of TO_GENESYS_RL

Create Route entries for non-Trunk Group routing with Matching Criteria and a Route Label destination.

Parameter	Description
none	The entity type for the route
Sonus_NULL	elementId1 – for an entityType of None, value is Sonus_NULL
Sonus_NULL	elementId2 – for an entityType of None, value is Sonus_NULL
standard	The type of routing for the route.
68 / 21443268	Destination national number
1	Destination country number
all	Call Type
all	Digit Type

ALL	Time Range Profile (note the capitalization)
none	Call Parameter Filter Profile
Sonus_NULL	Destination Domain Name
TO_CARRIER_RL / TO_GENESYS_RL	Destination Routing Label

Create Route entries for Trunk Group routing with Matching Criteria and a Route Label destination.

Parameter	Description
trunkGroup	The entity type for the route
REMOTE_AGENTS_TG1 / INTERNAL_AGENTS_TG1	elementId1 – for an entityType of trunkGroup, value is the ingress trunk group
SONUSSBC01	elementId2 – for an entityType of trunkGroup, value is the SBC System Name in all upper case (not hostname / element name)
standard / username	The type of routing for the route
Sonus_NULL	Destination national number
Sonus_NULL	Destination country number
all	Call Type
all	Digit Type
ALL	Time Range Profile (note the capitalization)
none	Call Parameter Filter Profile
Sonus_NULL	Destination Domain Name
TO_GENESYS_RL	Destination Routing Label

```
set global callRouting route none Sonus_NULL Sonus_NULL standard 68 1 all all ALL none Sonus_NULL
routingLabel TO_CARRIER_RL
commit
set global callRouting route none Sonus_NULL Sonus_NULL standard 21443268 1 all all ALL none Sonus_NULL
routingLabel TO_GENESYS_RL
commit
set global callRouting route trunkGroup REMOTE_AGENTS_TG1 SONUSSBC01 standard Sonus_NULL Sonus_NULL all
all ALL none Sonus_NULL routingLabel TO_GENESYS_RL
commit
set global callRouting route trunkGroup REMOTE_AGENTS_TG1 SONUSSBC01 username Sonus_NULL Sonus_NULL all
all ALL none Sonus_NULL routingLabel TO_GENESYS_RL
commit
set global callRouting route trunkGroup INTERNAL_AGENTS_TG1 SONUSSBC01 standard Sonus_NULL Sonus_NULL
all all ALL none Sonus_NULL routingLabel TO_GENESYS_RL
commit
set global callRouting route trunkGroup INTERNAL_AGENTS_TG1 SONUSSBC01 username Sonus_NULL Sonus_NULL
all all ALL none Sonus_NULL routingLabel TO_GENESYS_RL
commit
```

5 Genesys configuration

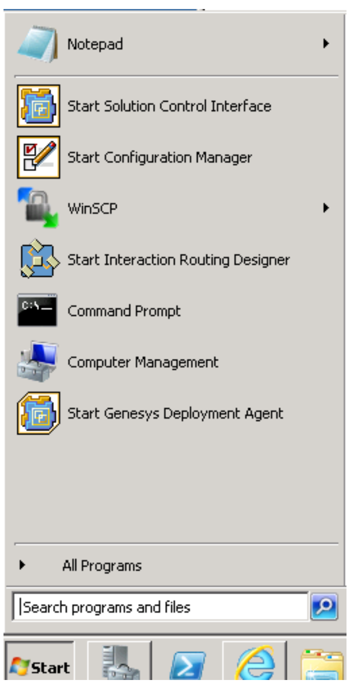
This section provides the configuration required for the Genesys components.

5.1 Accessing Genesys Tools and Interfaces

1. Genesys is configured using several different tools and interfaces. The tools and interfaces used in this document are shown below to include their location and method of access.

   To access these items, a Remote Desktop Connection (RDC) to the Genesys server is required. The username, password, and IP address of the system to be accessed should be provided by the person(s) installing the Genesys system.

2. Once logged onto the Genesys system, click the Start button and look for the installed applications shown below. If the applications are not visible on the Start Menu, find them using the search box just above the Start button.



The Solution Control Interface can be used to start/stop the various applications as well as identify the configuration of each application.

The Configuration Manager is a tool that is used to configure and verify the many settings on the different applications.

The Interaction Routing Designer is used to create and configure Route Points and strategies.

Use Internet Explorer (not shown in the startup menu) to access the Genesys Administrator. The URI should be available from those who installed the platform.

It is important to know that certain steps can be performed using multiple tools. For example, starting or stopping an application can be performed in the Genesys Administrator as well as the Solution Control Interface.

3. Below is a snapshot of the Solution Control Interface. In this application, click View, top left, to gain access to the area of interest. Click Applications and expand the folders of interest. In this example, SIPServer1 properties are displayed. Clicking the various links will display the appropriate property windows. You can Start/Stop/GracefulStop any application from within this tool.

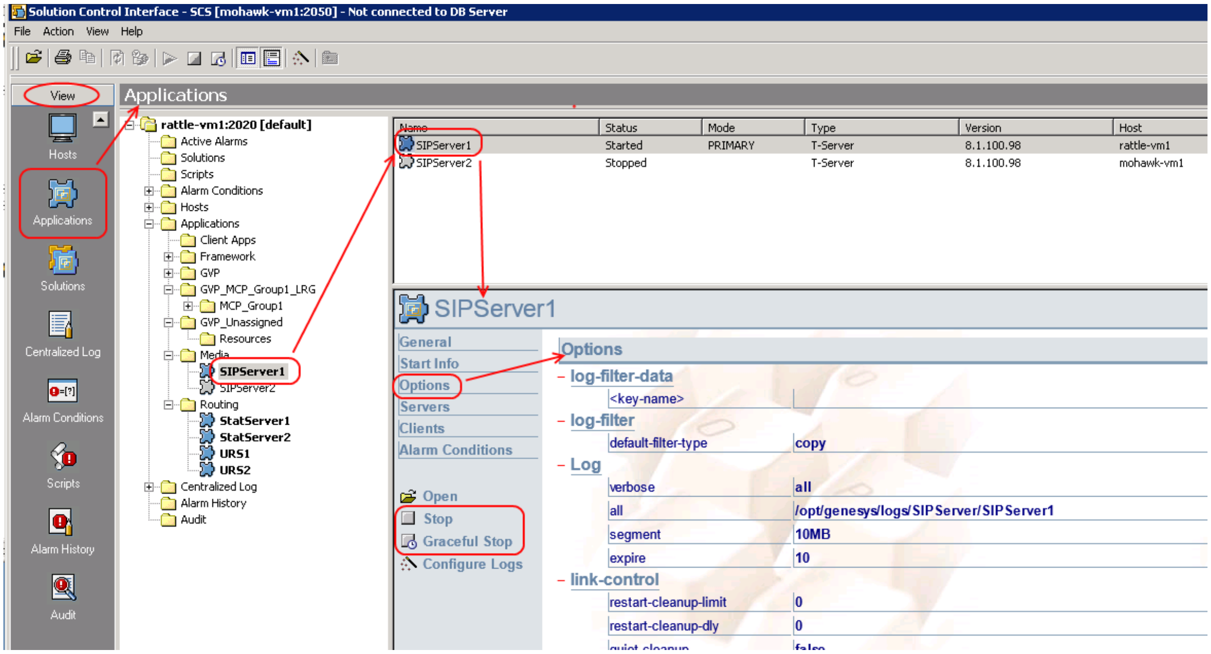


Figure 3: Solution Control Interface

4. The Configuration Manager is used to configure the platform and its applications. Once opened, click Applications/Media (it’s possible that your SIP Server is in a different folder under Applications) and the upper left pane will display both SIP Servers. Double-click SIPServer1 and a dialog box will open. To view all options for this server, click the Options tab, and then click the TServer section.

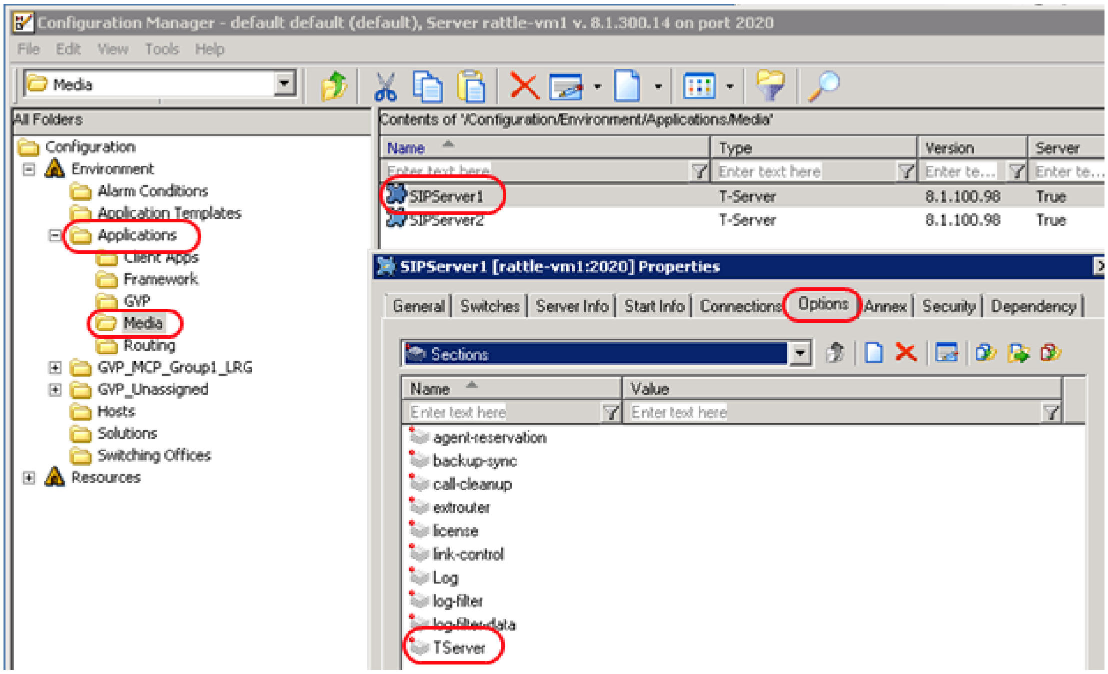


Figure 4: Configuration Manager

Note that there are many option parameters. Type the name of the option in the filter and it will filter in real time. Some options can be set at both Application and Switch/DN levels. The option setting at the DN level takes precedence over the Application-level setting. See the *Genesys SIP Server Deployment Guide* for details.

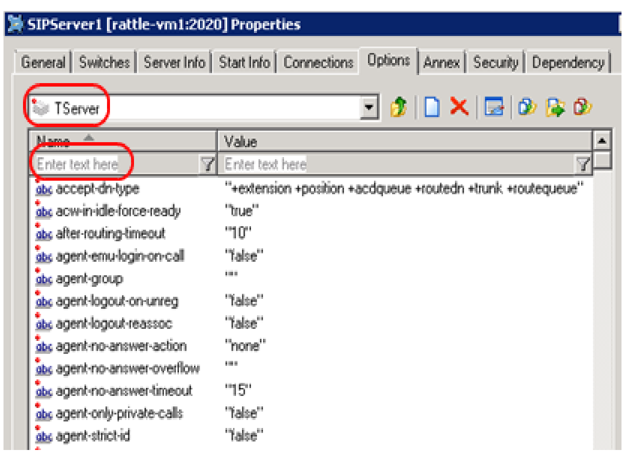


Figure 5: SIP Server Properties

5. The Interaction Routing Designer is a tool used to create Route Points and/or Strategies. Access the Strategies by clicking "Routing Design" in the upper left and then select Strategies. Double-clicking any

strategy will bring up a second window (not shown here). This second window is where Strategies can be created and modified.

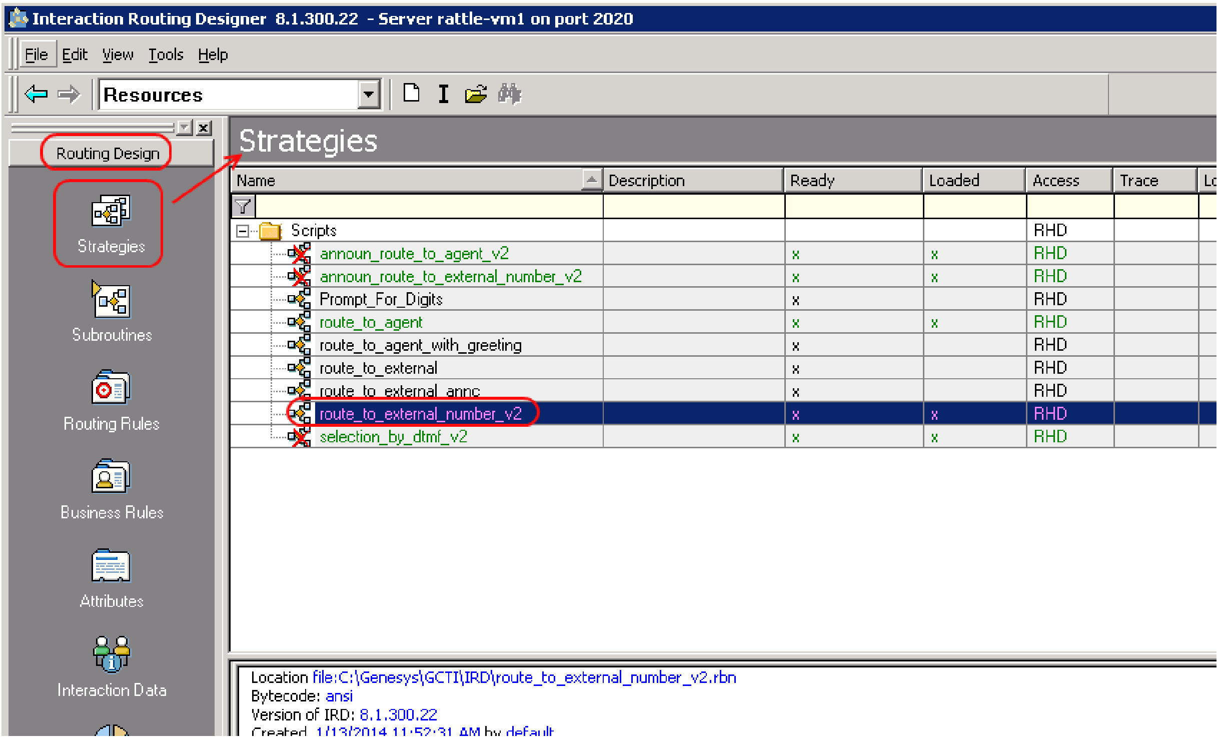


Figure 6: Interaction Routing Designer

6. Access the Genesys Administrator with a web browser. Contact the administrator or person(s) who performed the install of the system to determine the URL. Once opened, click the Provisioning tab and under the Navigation area click Switching. Under Switching, click Switches to display the names of the Switch objects.

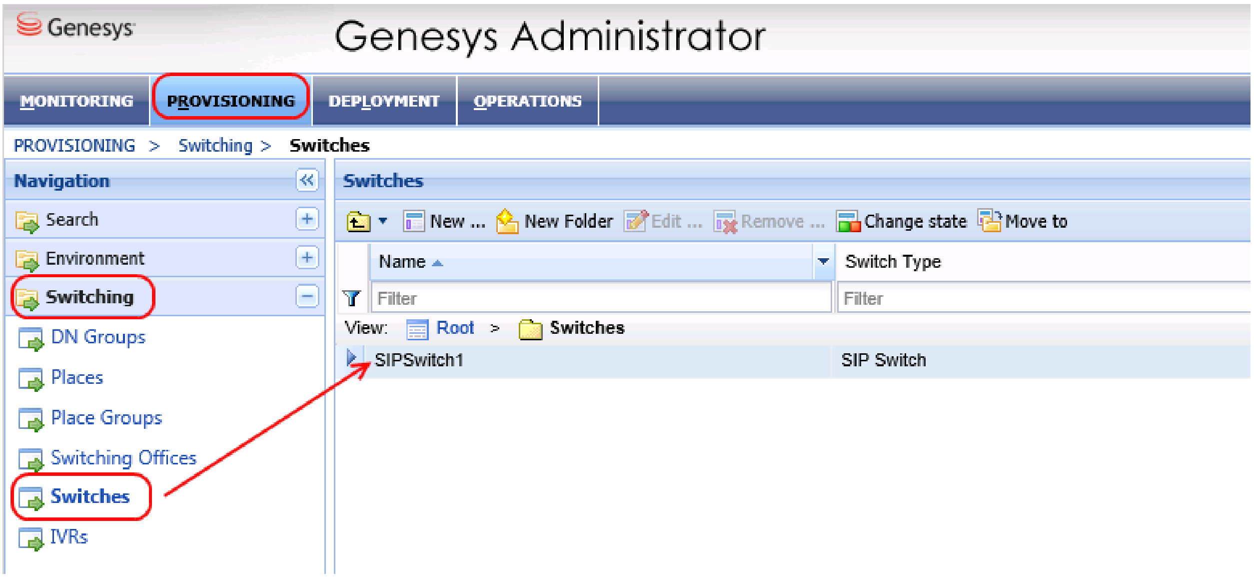


Figure 7: Genesys Administrator - Switches

Double-click the switch name and then click the DNs tab. The DNs for your SIP Switch will be shown. Each folder can be double-clicked to access the contents. Underlined are the bread crumbs for navigation. Circled is the icon area to add new DNs or delete existing ones.

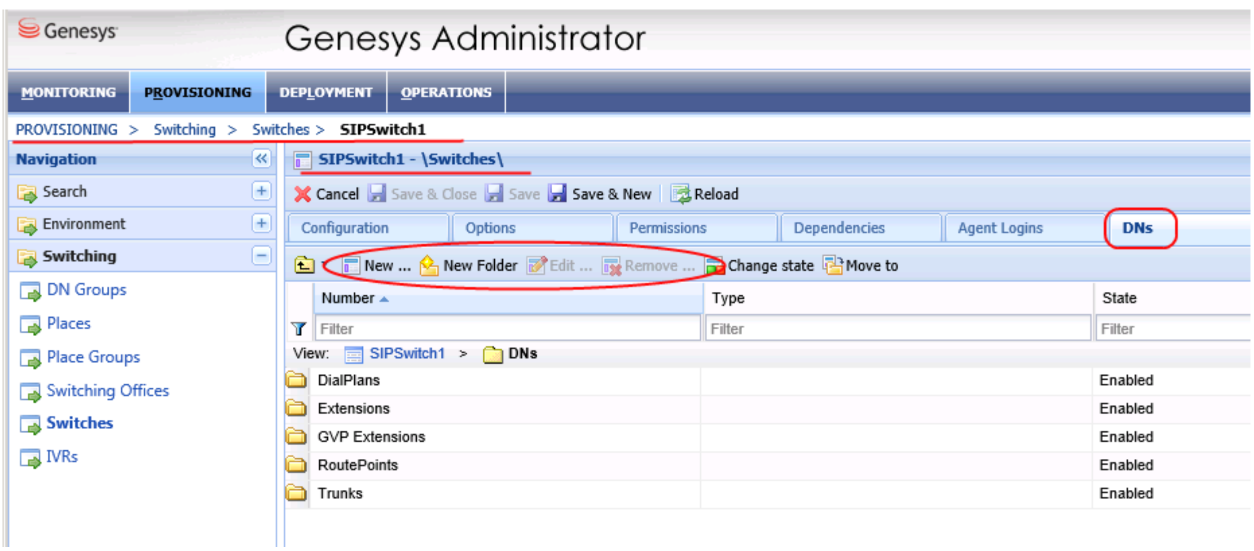


Figure 8: Genesys Administrator - DNs

5.2 Creating SIP Switch in Genesys Administrator

1. Within Genesys Administrator, create Switching Office –> SIPServer Switching Office.

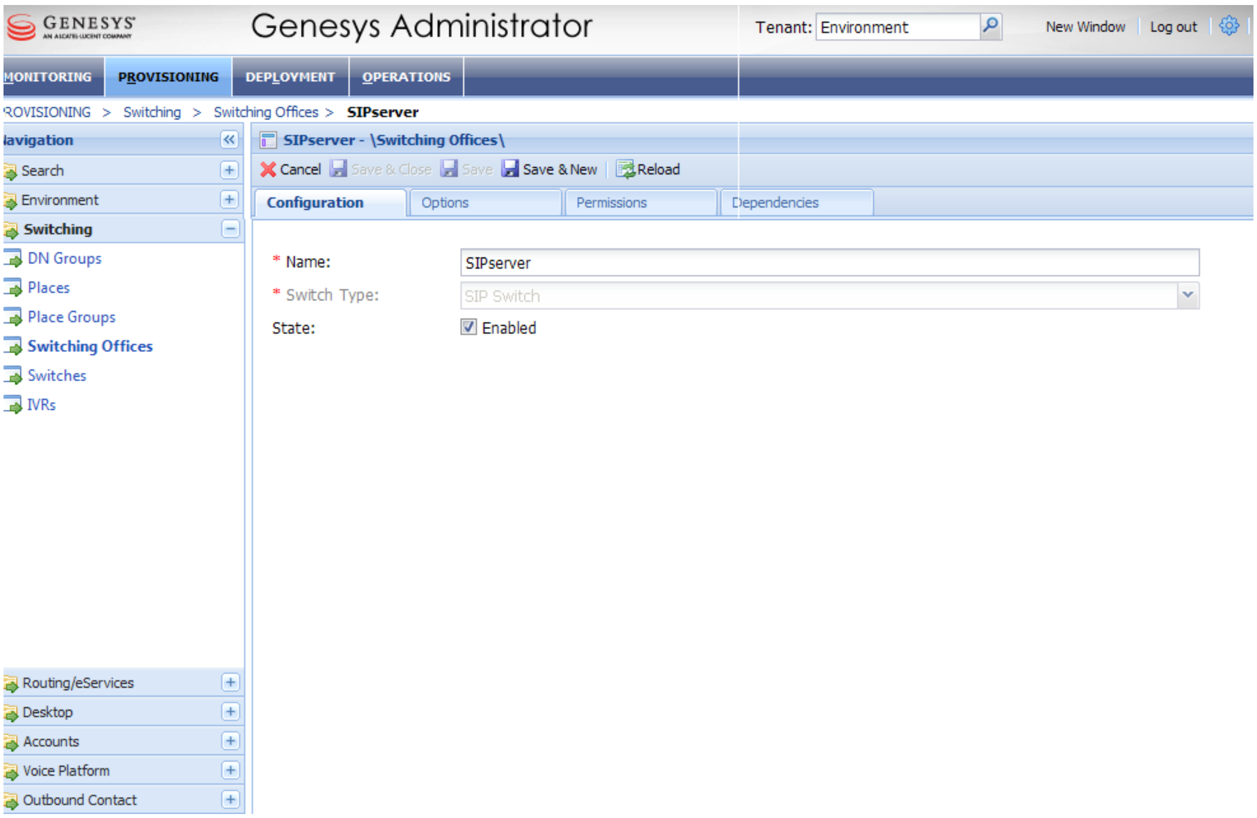


Figure 9: Genesys Administrator - Creating SIP Switch

2. Within Genesys Administrator, create a SIP Server Switch and associate the Switching Office created in the previous step with this switch.

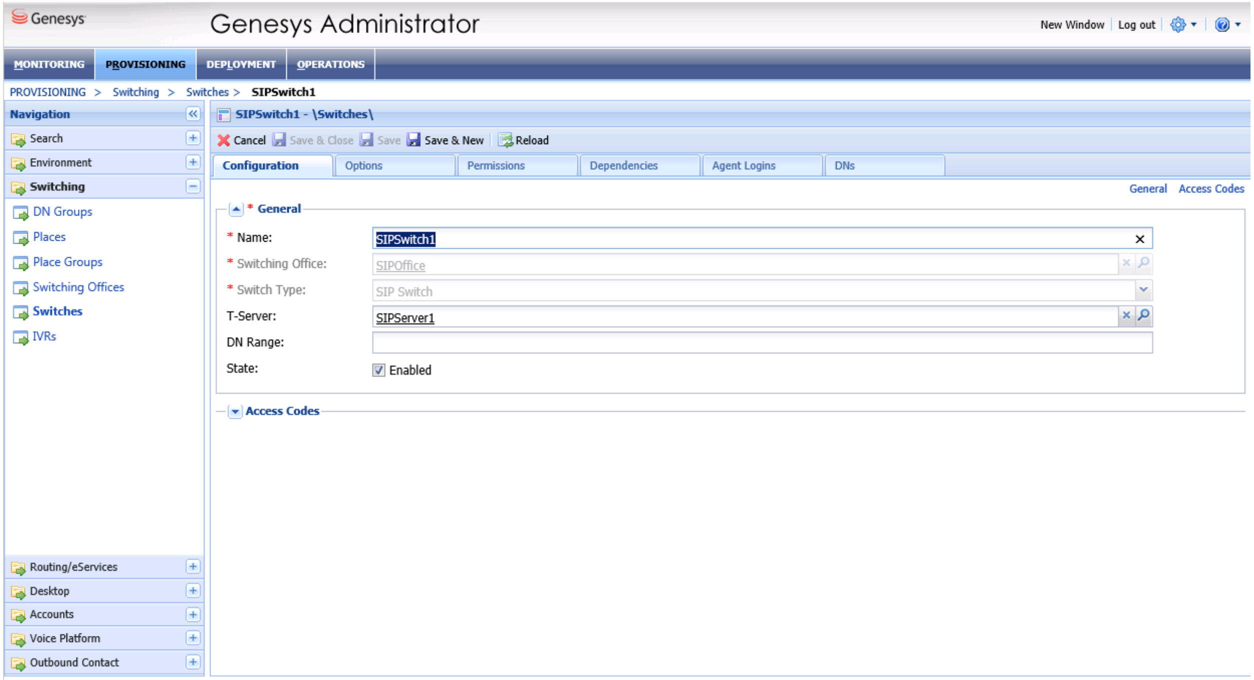


Figure 10: Genesys Administrator - SIP Switch Association

3. Under the SIPSwitch created in the above step, define Routing Points to run URS strategies from, the SIP trunk representing connection of SIP Server to Sonus, and a “msml” VoIP service DN required to integrate SIP Server with Media Server to support call hold and conferencing functionalities.

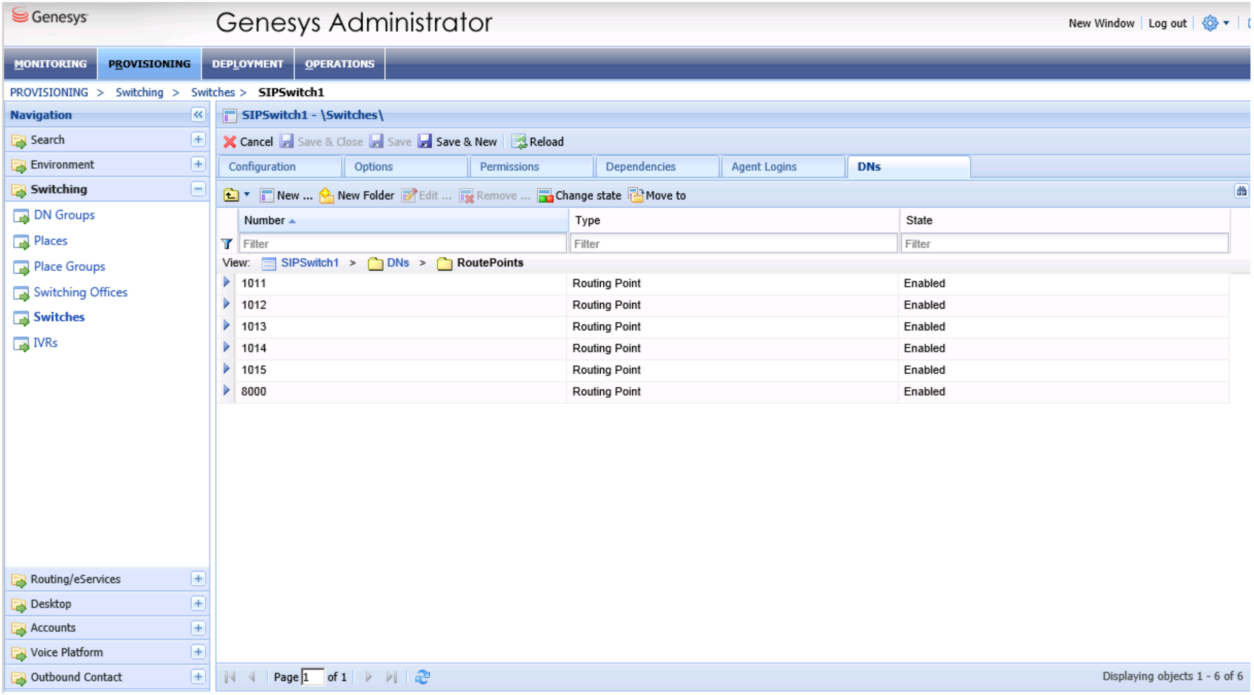


Figure 11: Genesys Administrator - Define Routing Points

Subsequent steps of this section (see Step #8) provide additional details required to configure these DNs.

4. Define DNs of type Extension under SIPSwitch with the following options in the TServer section for various SIP end points that will register to SIP Server.

- **use-contact-as-dn=true –** Specifies whether SIP Server will use the username of the Contact header as ThisDN.
- **contact=* –** Specifies the contact address of the extension DN to which SIP Server should send the SIP call. Here the Contact option value is the IP address of the internal interface of Sonus through which the SIP REGISTER message was received by SIP Server.
- **cpn=<2086041001> –** SIP Server uses the value of this option as the user part of the SIP URI in the From header of the INVITE message that it sends from this DN to the destination DN. Since this option is used to provide customized caller-ID information to the destination, this option must be configured in the originating DN.
- **sip-cti-control=talk,hold –** The SIP method NOTIFY (event talk) or NOTIFY (event hold) is used to request the end point to answer or place a call on hold, respectively.

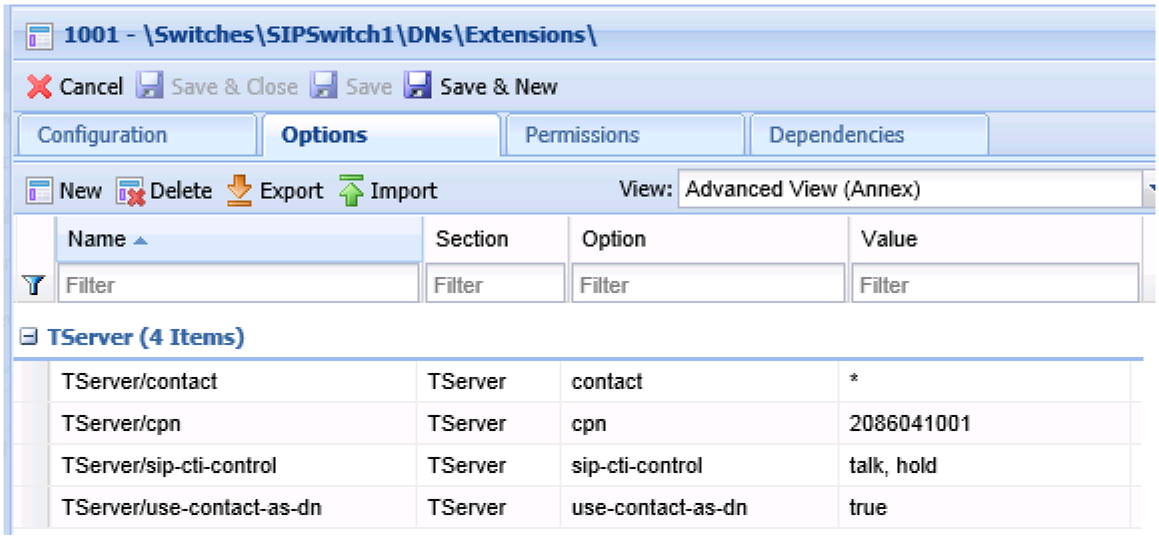


Figure 12: Genesys Administrator - Extension Options

5. Define a SIP trunk DN to represent all SIP calls arriving from the Sonus NBS internal interface to SIP Server. Configure the following options under the TServer section of the Trunk DNs.

Trunk DN:

- **contact=<10.35.141.52:5060> –** IP address and TCP/UDP port number of the SIP Signaling Port of the Sonus SBC 5000 configured for Genesys. The SIP Signaling Port IP address is used by SIP Server to route or receive calls from test phones through this interface.
- **cpd-capability=mediaserver –** Specifies whether SIP Server will use the username of the Contact header as ThisDN.
- **dial-plan=DialPlanInbound –** Specifies which dial-plan DN will be applied to calls
- **prefix=<214340> –** (NPANXX) Specifies the contact address of the extension DN to which SIP Server should send the SIP call. Here the Contact option value is the IP address of the SIP Signaling Port of the Sonus SBC 5000 through which the SIP REGISTER message was received by SIP Server.

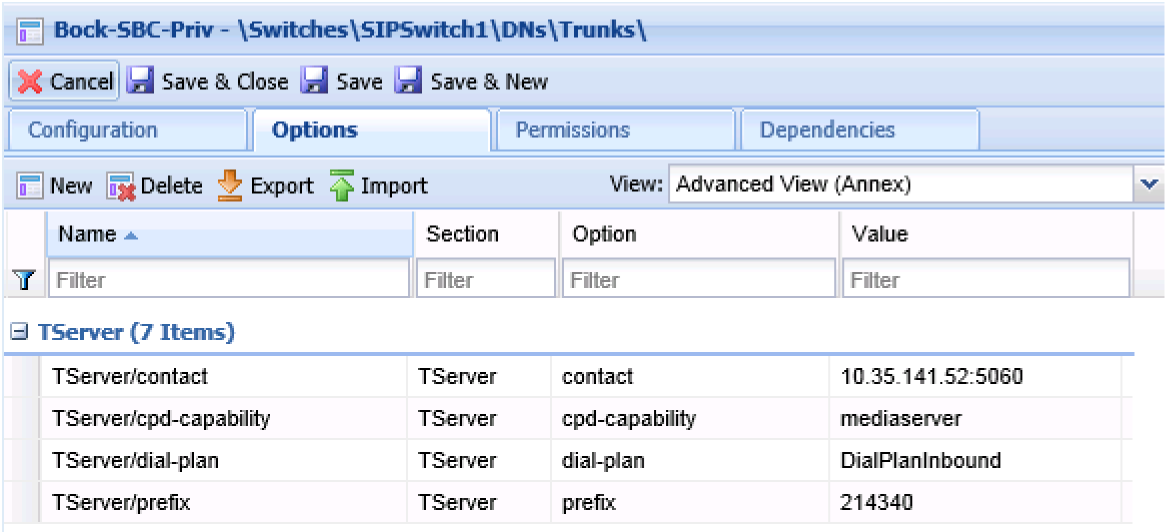


Figure 13: Genesys Administrator - SIP Trunk Options

6. Defining a MSML voice over IP service DN with the following options in the TServer section:

- **contact-list=<IP Address:Port> –** SIP IP address and listening port for Resource Manager.
- **oos-check=15 –** Specifies how often (in seconds) SIP Server checks a device for out-of-service status.
- **oos-force=20 –** Specifies the time interval (in seconds) that SIP Server waits before placing a device that does not respond in out-of-service state when the oos-check option is enabled.
- **prefix=msml= –** Required for conference and monitoring services only.
- **service-type=msml –** Specifies the configured SIP device type or service.
- **subscription-id=Resources –** Specifies the type of subscription ID.

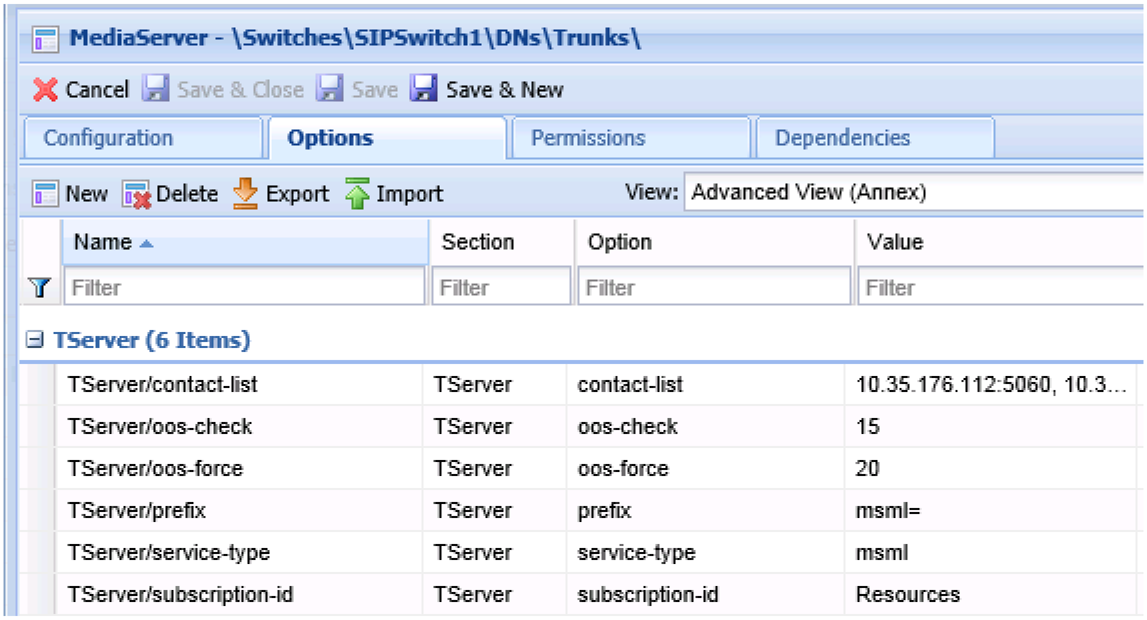


Figure 14: Genesys Administrator - Define MSML

7. Verify GVP_RM Pair settings:

- **prefix=msml= –** Required for conference and monitoring services only.
- **refer-enabled=false –** Specifies the configured SIP device type or service.
- **ring-tone-on-make-call=false –** Affects the TMakeCall request when using the re-INVITE procedure. When the ring-tone-on-make-call option is set to false, there is no ring tone.
- **make-call-rfc3725-flow=1 –** Setting this option to 1 instructs SIP Server to use the 3pcc call flow as defined in the RFC 3725.

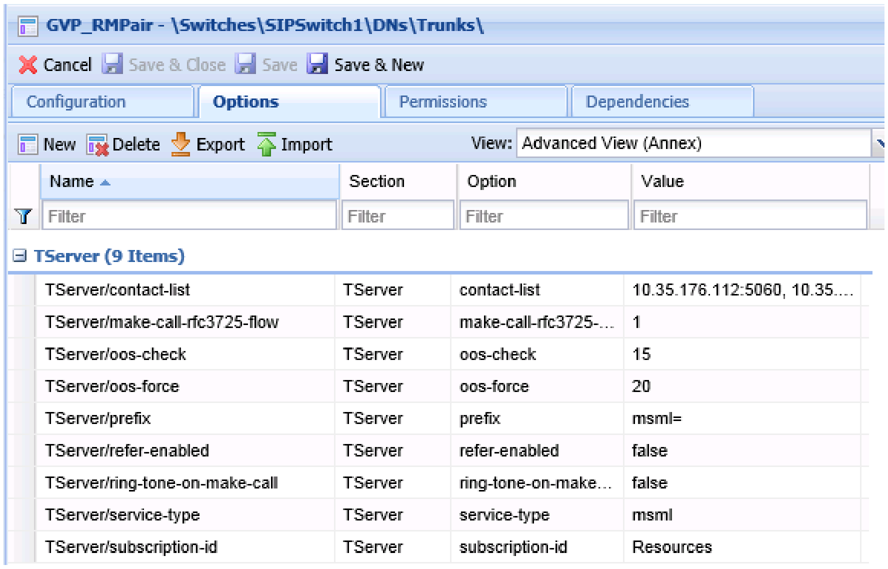


Figure 15: Genesys Administrator - GVP_RM Pair Settings

8. Create DNs of type Routing Point in the SIPSwitch which should match the Request URI user part. In this instance it was extensions 1011-1015.

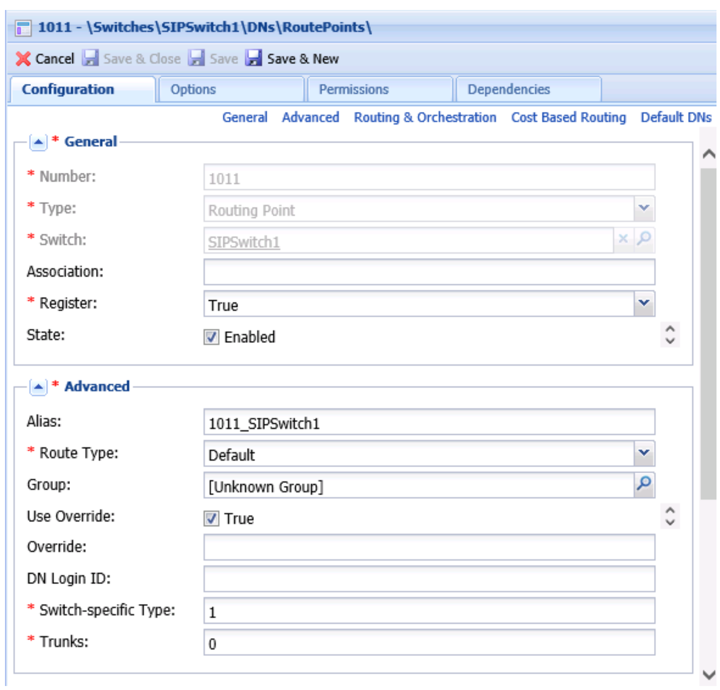


Figure 16: Genesys Administrator - Create DN of Type Routing Point

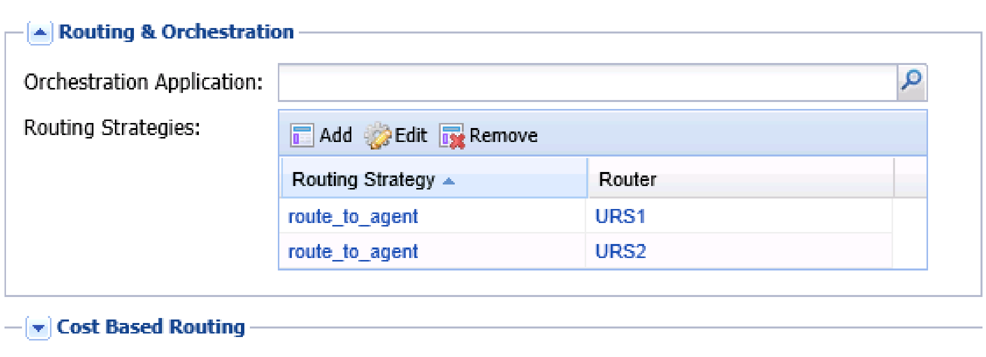


Figure 17: Genesys Administrator - Routing Strategies

9. SIP Server must have Full Control permission for the DN objects under the SIP Server Switch, in order to update various configuration objects under it, such as the Extension DNs.

   By default, it does not have this permission. You must grant “Full Control” permission for the System account for the all DNs on the corresponding switch. It is done for all DNs at once by changing the permissions for the system account on the DN folder in the switch object. Or, you can start SIP Server under another account that has change permission on the necessary DNs.

   With this full control access, the SIP Server Switch grants DNs like Extension to update their options like “contact” when a new SIP register message is received from end points moving to a new IP location.

5.3 SIP Server Configuration in Genesys Administrator

Follow these steps to configure SIP Server to monitor SIP Server Switch resources, such as SIP extensions/SIP end points registered to SIP Server. SIP Server also monitors various route points and notifies URS whenever the call arrives on the Route Point.

1. Install and configure SIP Server as per *Genesys Framework SIP Server Deployment Guide*.

2. Add a connection to the SIP Server Switch created above, to monitor all the resources under this switch: Genesys Administrator-> Provisioning->Environment->Applications->SIP Server Application.

   Also, SIP Server should add a connection to the tenant.

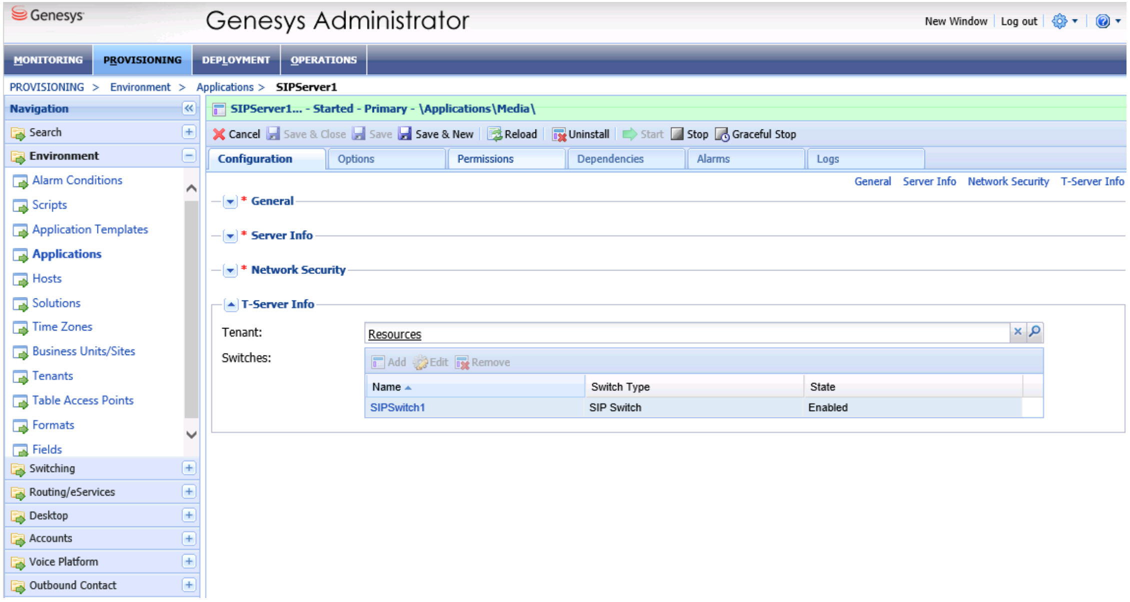


Figure 18: Genesys Administrator - SIP Server Tenant

3. Add a connection to Message Server.

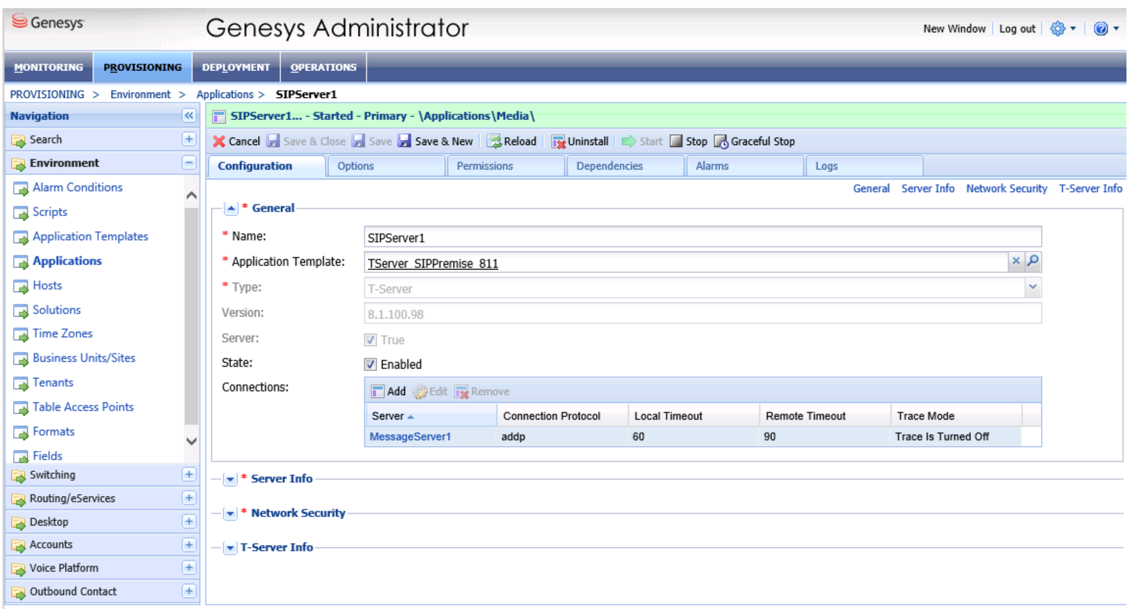


Figure 19: Genesys Administrator - Add Connection to Message Server

4. Configure the following options in the TServer section in the SIP Server Application object using Genesys Administrator or Configuration Manager:

- **sip-enable-moh=true –** Enables music on hold.
- **msml-support: true –** This option and sip-enable-moh (above) allow SIP Server to integrated with Genesys Media server to provide msml/moml-based media services.
- **map-sip-errors=false** – Genesys Universal Router makes the routing decision for the SIP Server-based solution. If a call fails to route properly, the SIP Server generates an appropriate T-Library error message to inform the router. With this new parameter, SIP Server can now propagate SIP error messages to the router. Setting **map-sip-errors=false** triggers this functionality in SIP Server.
- **internal-registrar-persistent=true –** Enables SIP Server to update the DN attribute contact in the configuration database. When an endpoint registers, SIP Server takes the contact information from the REGISTER request and updates or creates a key called contact in the Annex tab of the corresponding DN.
- **sip-dtmf-send-rtp=true –** In order to support DTMF tone generation on behalf of a 3pcc-based SIP end point application such as Interaction Workspace SIP end point. When this option is set to true, SIP Server requests Media server to generate RFC 2833 DTMF tones on behalf of the end point.
- **after-routing-timeout –** Set to 10 seconds. If SIP Server does not get a response on routing a call to SIP agent/Extension DN, it will attempt to route the call to another DN (or default-dn) on expiration of this timer. You must set this timer to be less than the parameter **rq-expire-tmout** value of 32000 (32 seconds).

5. SIP Server is able to start properly with the proper FlexLM license installed.

5.4 Genesys Media Server Deployment

Follow these steps to configure a Media Server deployment.

1. Media Server platform consists of Resource Manager and Media Control Platform applications in the Genesys Voice Platform product suite. To deploy, Media Control Platform and optionally a Resource Manager are required to be installed. When installed, Resource Manager serves as the ingress point to Media services and provides a MCP resource as a media service to the network/calling side.

2. Install and configure MCP (Media control Platform) using the *Genesys Media Server Deployment Guide*.

3. Within the MCP application's Connections tab, add connections to SNMP Master Agent, Message Server, and Reporting Server (optional).

   The connections to applications are added for the following reasons:

Message Server - To ensure that component log information reaches the Log database and can be viewed in the Solution Control Interface (SCI)

Reporting Server - (Optional) To ensure that these components detect the Reporting Server to which they are sending reporting data.

SNMP Master Agent - To ensure that alarm and trap information is captured.

4. Verify VoIP service DN of type=msml as specified in section 5.2, Creating SIP Switch in Genesys Administrator to support SIP Server-Media Server MSML interactions to support treatments and conferencing capabilities.

5. To play music on hold (MOH) and music treatments, verify the following options are set in MCP and SIP Server:

   MCP->msml-> **play.basepath** = file://$InstallationRoot$ (this is the installation folder of Media Server. After this is, it will automatically look for the music sub folder).

   "MOH" and music treatments are located in the "music" folder.

   The 'announcement" folder should contain 'prompt' files with proper IDs to support. Used in the URS Routing Strategies as mentioned in chapter 5.7.

   SIP Server->TServer->**msml-support**=true

6. Install and configure Resource Manager as per *Genesys Media Server Deployment Guide*.

   Note: If SIP Server and Resource Manager are on the same machine and within the Resource Manager application, then the default SIP listening port number should be increased by 100 so the Resource Manager listening port is set to 5160 and the SIP Server application listens on port 5060. Make the necessary port changes within Resource Manager's sip, proxy, register, subscription, and monitor sections.

7. Within the Resource Manager application's Connections tab, add connections to SNMP Master Agent, Message Server, and Reporting Server (optional).

   The connections to applications are added for the following reasons:

   Message Server - To ensure that component log information reaches the Log database and can be viewed in the Solution Control Interface (SCI).

   Reporting Server - To ensure that these components detect the Reporting Server to which they are sending reporting data. (Optional).

   SNMP Master Agent - To ensure that alarm and trap information is captured.

8. **Within the Integrating Media Control Platform with the Resource Manager,** click the Media Control Platform Application object. The Configuration tab appears.

   Click the Options tab, and use the View drop-down list to select Show options in groups...

Select sip to find the routeset option.

In the Value field, type the following:

```
<sip:IP_RM:SIPPort_RM;lr>
```

Where IP_RM is the IP address of the Resource Manager, and SIPPort_RM is the SIP port of the Resource Manager—typically, 5060.

Note: You must include the angle brackets in the Value field in the sip.routeset and sip.securerouteset parameters.

In the Value field of the securerouteset option, type the following:

```
<sip:IP_RM:SIPSecurePort_RM;lr>
```

9. G.729 media codec is not configured by default as a supported codec or as a codec that can be transcoded. This support can be enabled by adding “g729” as one of the values to the mpc.codec and mpc.transcoders space separated list. The G.729 media codec was not provisioned in this Genesys deployment and is only mentioned here for completeness.

   Example:

```
mpc.transcoders=PCM GSM G726 G729
mpc.codec=g729 pcmu pcma g726 gsm h263 h263-1998 h264 telephone-event
```

   Alternately Media Server (specifically MCP component) can be configured to respond a multiple codec offer request with a single codec response. This feature support is available starting with MCP 8.1.4 release.

   This setting can be enabled by setting **mpc.answerwithonecodec=1** (Default=0 – MCP responds to multiple codec offer with a multiple codec response list).

10. This step is optional and is only required if multiple media control platform (MCP) instances are deployed and need to be controlled by Resource Manager for load balancing.

    Log in to Genesys Administrator.

- On the Provisioning tab, click Voice Platform > Resource Groups.
- On the Details pane tool bar, click New.
- The Resource Group Wizard opens to the Welcome page.
- On the Resource Manager Selection page, add the Resource Manager Application object for which you want to create the group. On the Group Name and Type page: enter MCPGroup or any custom name without spaces. Select type as Media Control Platform.
- On the Tenant Assignments page, add the child tenant to which the Resource Group will be assigned.
  - **Note:** -The above bullet item is required only if you are creating the Resource Group in a multi-tenant environment.
- On the Group Properties page, enter the information as specified below for the Resource Group that you are configuring.

  - **Monitoring Method:** Retain the default value: SIP OPTIONS.

- **Load Balance Scheme:** Select round-robin.
- **Port Usage Type:** Select in-and-out.
- **Maximum Conference Size:** Enter -1.
- **Maximum Conference Count:** Leave blank.

Note: For the Media Control Platform group, the Max.Conference Size and Max.Conference Count, and the Geo-location options are optional.

For a complete list of resource-group options and their descriptions, refer to the *Genesys Voice Platform User's Guide*.

11. In this step, you create a default IVR Profile that can be used to accept calls other than those specified in the dialing plans.

- Log in to Genesys Administrator.
- On the Provisioning tab, select Voice Platform > IVR Profiles.
- In the Tasks panel, click Define New IVR Profile. The IVR Profile Wizard opens to the Welcome page.
- On the Service Type page, enter the name of the default IVR Profile, IVR_App_Default.
- Select either Conference or Announcement from the drop-down list. (Only one service type per IVR Profile is supported.)
- If you selected Conference, on the Service Properties page, enter a conference ID number.
- If you selected Announcement, on the Service Properties page, enter the URL of the announcement, for example, http://webserver/hello.wav.
- Click Finish.
    - **Note:** When you use the IVR Profile Wizard to create the default profile, the gvp.general and gvp.service-prerequisites sections are created for you and include the required parameters
- In the gvp.general section of the Tenant's Annex tab, set the default-application to this default IVR Profile name – IVR_App_Default.

12. This completes installation and configuration of Media Server. Make sure Resource Manager and MCP are started successfully.

5.5 Stat Server Configuration

This section explains configuration of Stat Server that connects with T-Servers/SIP Servers and maintains agent and/or extension status which is used by URS during call routing.

1. Install and configure Stat Server as per *Genesys Framework Stat Server Deployment Guide*.

2. Add connections to SIP Server, Message Server to perform real-time monitoring of the SIP agent status.

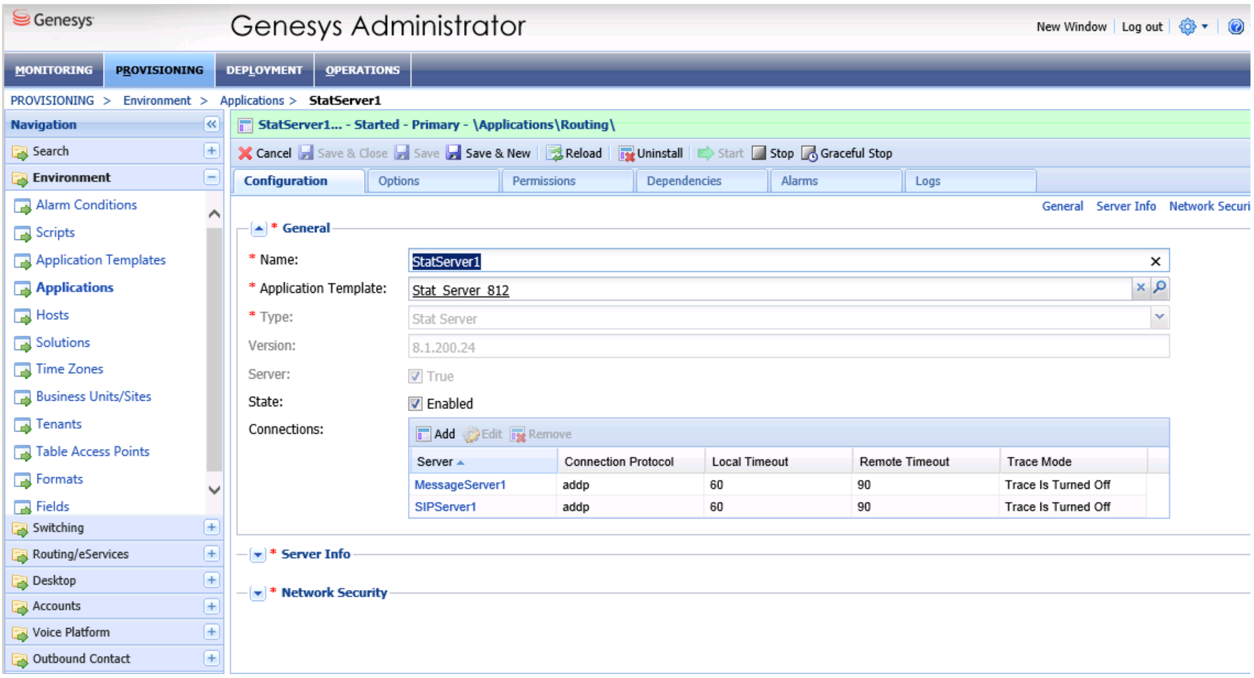


Figure 20: Genesys Administrator - Stat Server Connections

5.6 Universal Routing Configuration in Genesys Administrator

This section explains how to configure a Universal Routing Configuration (URS) to support execution of call routing on SIP Server.

1. Install and configure Universal Routing Server as per *Genesys Universal Routing Deployment Guide*.

2. Add connections to Message Server, Stat Server, and SIP Server.

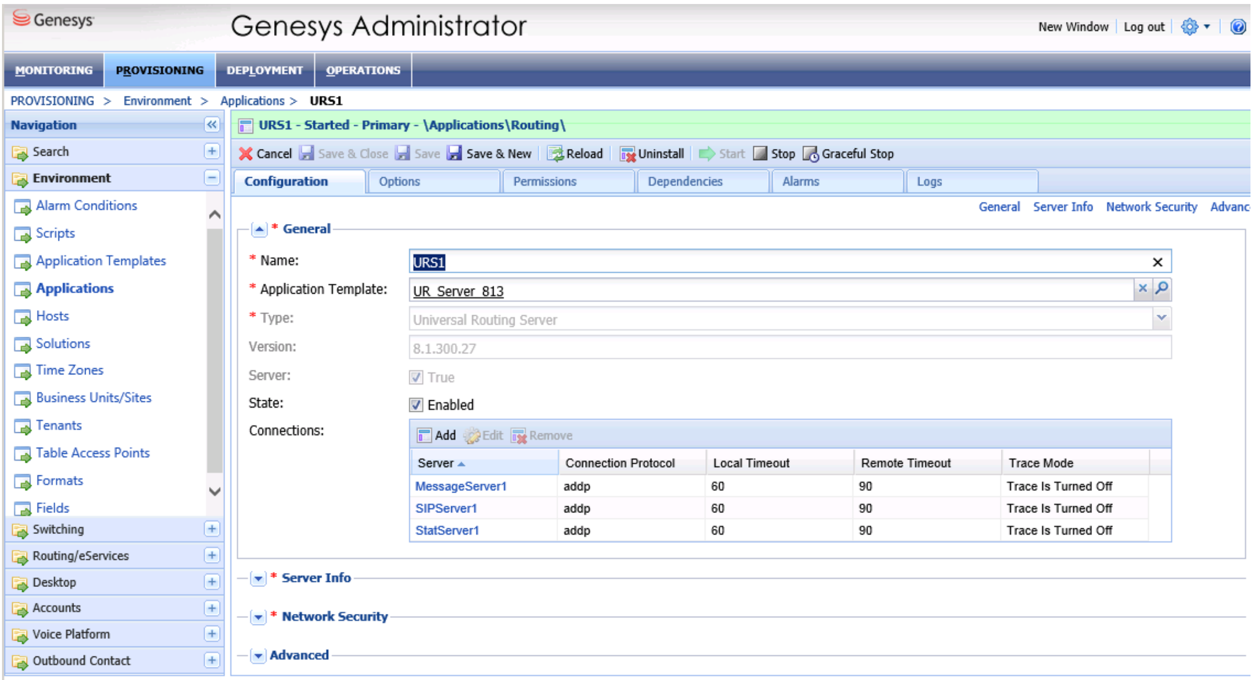


Figure 21: Genesys Administrator - URS Configuration

Add connection to SIP Server to monitor events received by SIP Server for various route points and extensions on the SIP Server Switch.

Add connection to Stat Server to query Stat Server for routing calls to available and ready agents.

Use any of the strategies below to test your configuration.

5.7 URS Routing Strategies

This section shows examples of five URS routing strategies used during testing.

5.7.1 Strategy #1 - Route Call to Available Agent

When RP 1011 is invoked, this strategy routes the call to the next available agent. If no agent is available, it plays MOH until one becomes available.

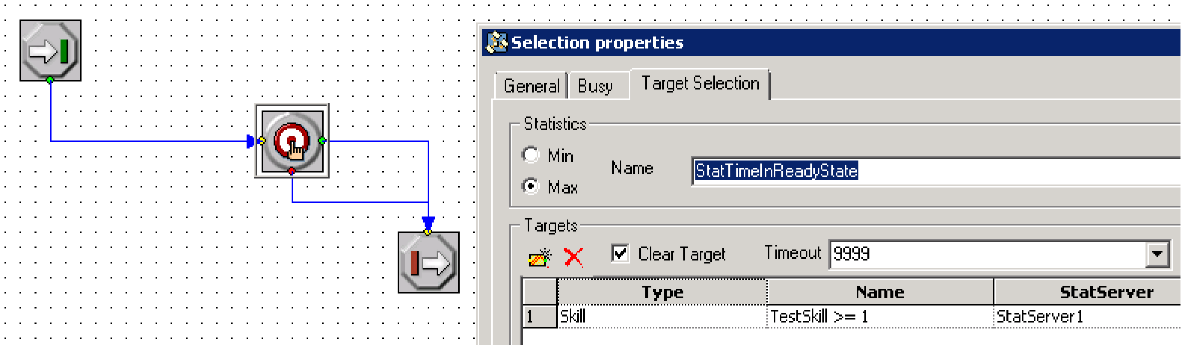


Figure 22: URS Strategy #1

5.7.2 Strategy #2 - Play Announcement and Route to Available Agent

When RP 1012 is invoked, this strategy plays an announcement and routes the call to the next available agent. If no agent is available, it plays MOH until an agent becomes available.

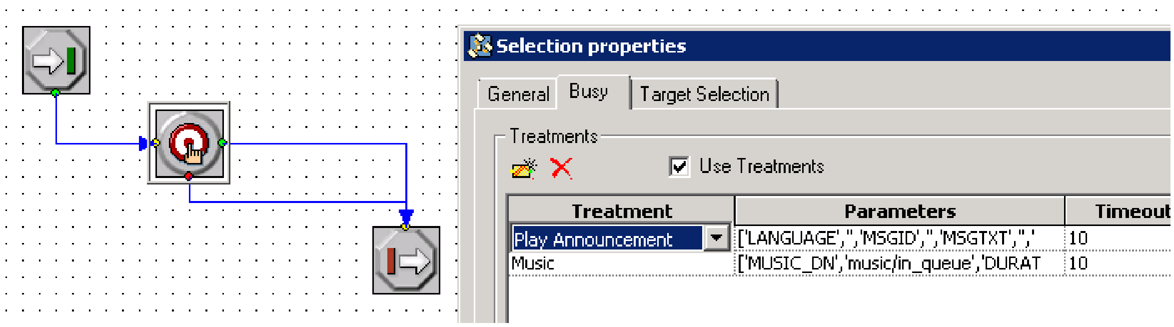


Figure 23: URS Strategy #2

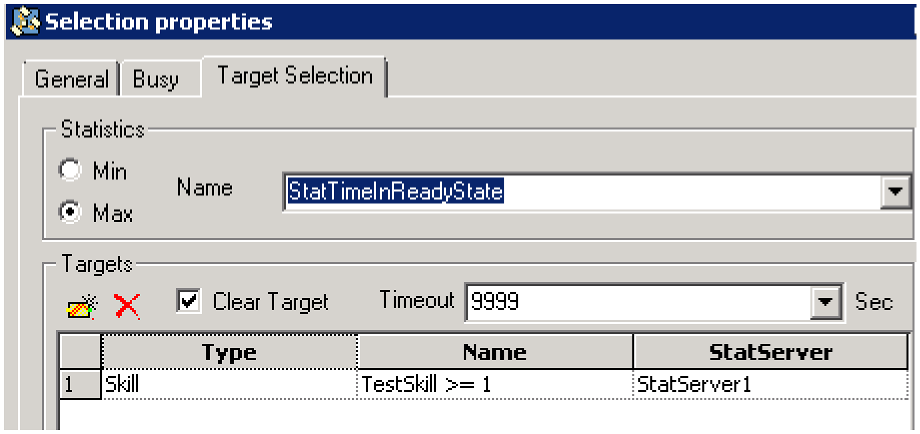


Figure 24: URS Strategy #2 - Target Selection

5.7.3 Strategy #3 – Play Announcement and Collect Seven Digits

When RP 1013 is invoked, this strategy verifies that any seven digits can be collected and then routed to an available agent. If no agent is available, it plays MOH until an agent becomes available.

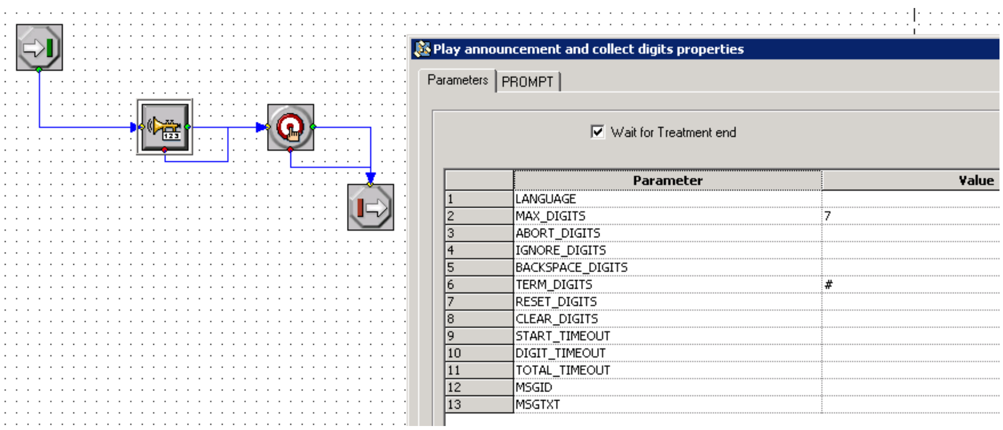


Figure 25: URS Strategy #3

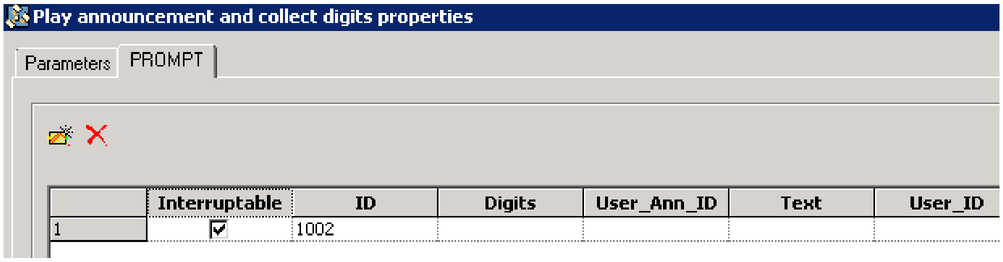


Figure 26: URS Strategy #3 - Prompt Tab

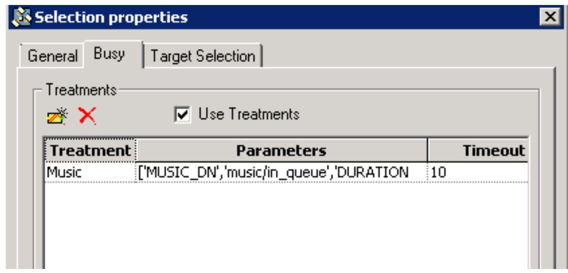


Figure 27: URS Strategy #3 - Busy Tab

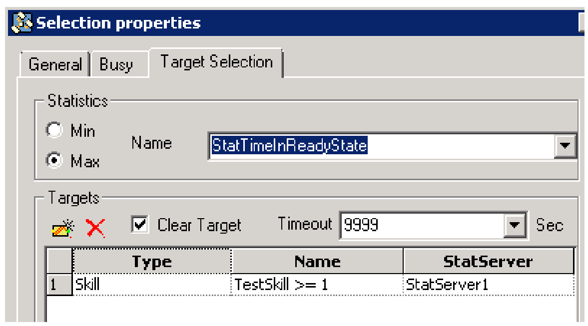


Figure 28: URS Strategy #3 - Target Selection

5.7.4 Strategy #4 – Route to External SIP Carrier Number

When RP 1014 is invoked, this strategy immediately routes the call to an external SIP Carrier number.

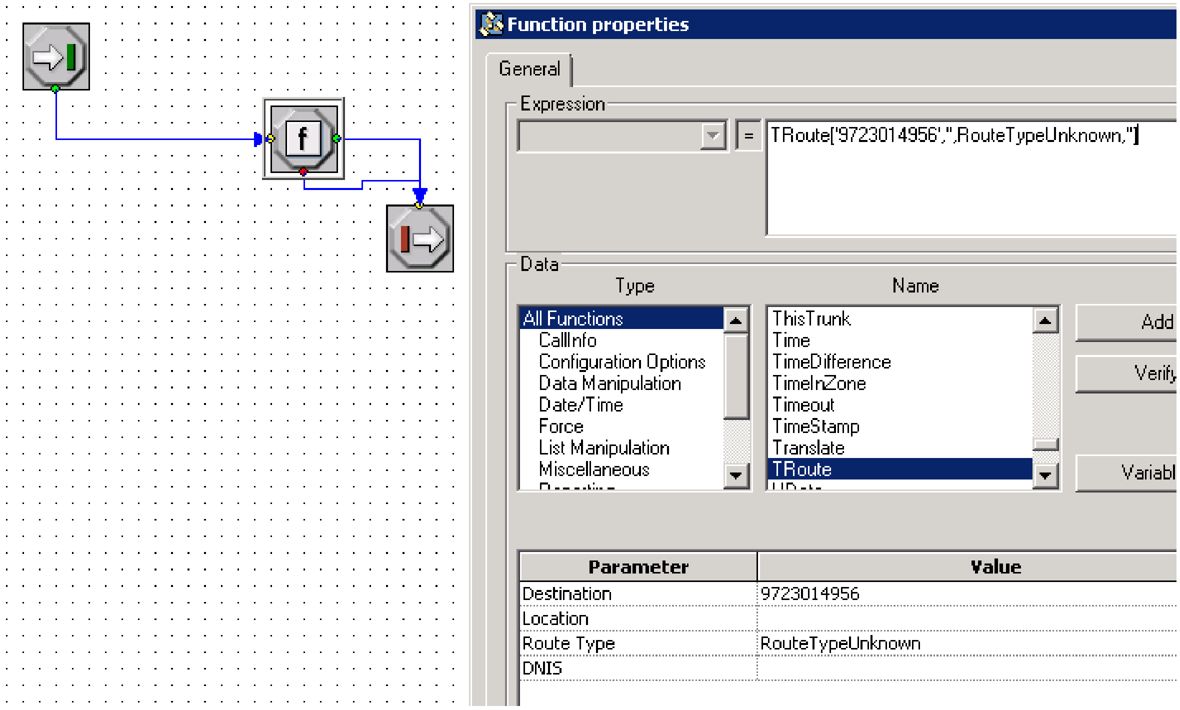


Figure 29: URS Strategy #4

5.7.5 Strategy #5 – Route to External SIP Carrier Number

When RP 1015 is invoked, this strategy plays an announcement and then immediately routes the call to an external SIP Carrier number.

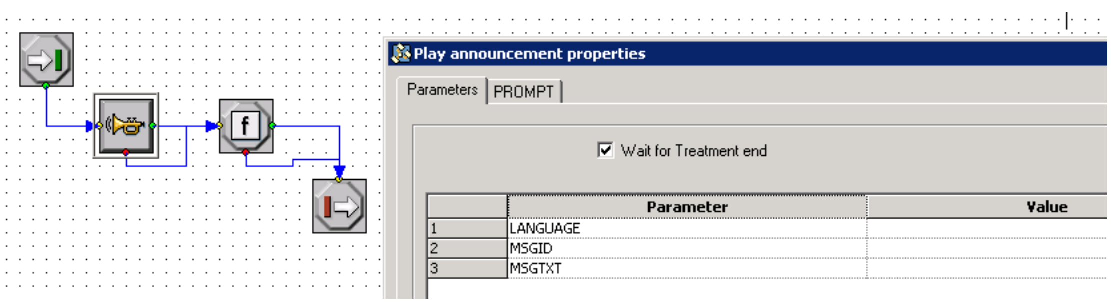


Figure 30: URS Strategy #5

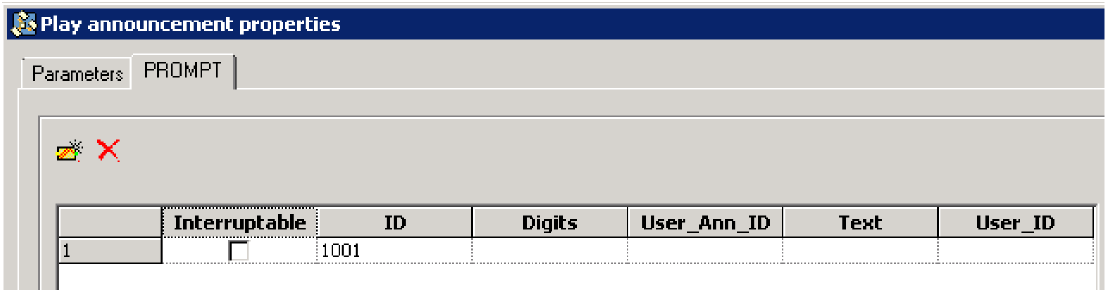


Figure 31: URS Strategy #5 - Prompt Tab

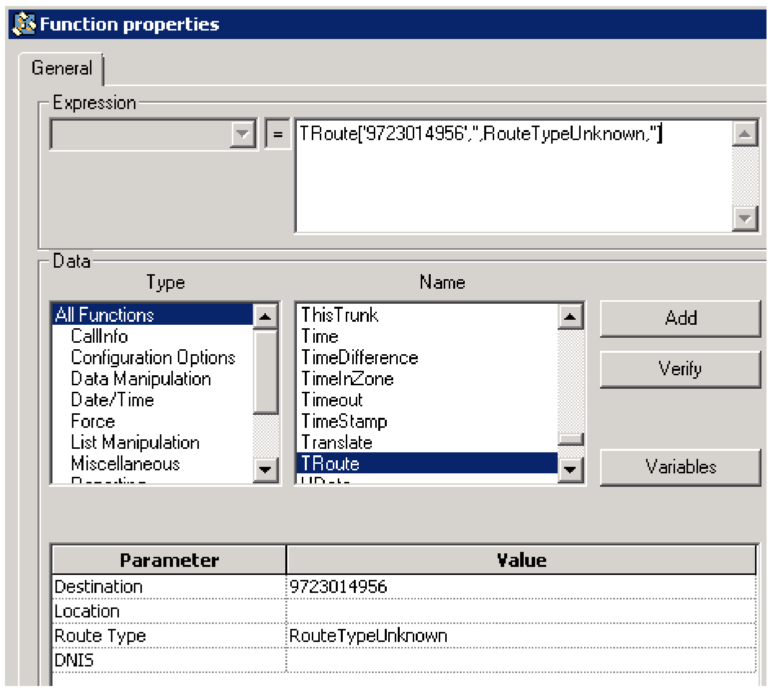


Figure 32: URS Strategy #5 - Function Properties

6 SBC and Genesys-Specific Configurations

Depending upon the type of call scenario desired will determine what settings are required for the SBC and Genesys product. Only scenarios that were tested are listed below.

Additional information regarding the environment setup for Genesys can be found in the Genesys documentation.

6.1 Main Genesys Configuration Settings

6.1.1 Option Priorities

The various options available to the Genesys platform may have multiple “levels” where they can be enabled. It is important to know that if a particular option is set at one level but the behavior doesn’t change, there may be the same option with a different value set at an overriding level. In other words, some options can be set at both Application and DN levels. The option setting at the DN level takes precedence over the Application-level setting. Refer to the *Genesys Framework SIP Server Deployment Guide* for more information.

6.1.2 Configuration Options

The configuration settings used in this testing, with their definitions, are indicated below. The desired outcome of the test will depend on the value of the option.

refer-enabled - Specifies whether the REFER method is sent to an endpoint. When set to true, the REFER method is sent to:

- The call party that originates a TMakeCall request.
- The call party that initiates a consultation call.
- The call party that is transferred to another destination during a single-step transfer.

When set to false, SIP Server uses the re-INVITE method instead.

oosp-transfer-enabled - When set to true, SIP Server puts itself in the Out Of Signaling Path (OOSP) after the single-step transfer or routing to the external destination has been completed.

sip-replaces-mode - When set to 2, SIP Server will use the REFER method with Replaces to process TCompleteTransfer. The Allow and Supported headers will not be analyzed.

6.2 Initiating Transfers with Re-INVITEs to External Destination

Use the Trunk DN configurations below so the Genesys platform will utilize a re-INVITE method to initiate a transfer to an external destination. Because the options are set at a trunk level, the destination are the phones on the external side of the SBC.

In the Genesys configuration environment, Switch->Trunk DN->Annex Tab->TServer Section, set the following options on the Trunk pointed to the SBC:

refer-enabled=false

oosp-transfer-enabled=true

sip-replaces-mode=2

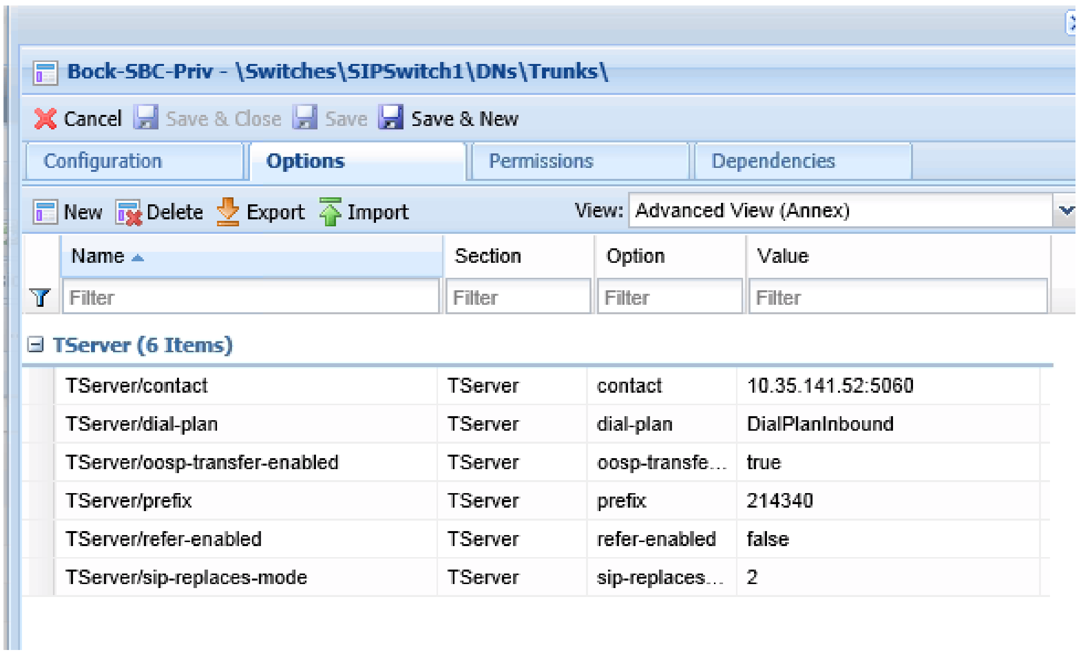


Figure 33: Genesys Administrator – Re-INVITE-based Transfer

NOTE: Be aware that settings on a different level have an effect on behavior. Option settings on the Switch/DN level take precedence over the Application-level setting.

6.3 Initiating Transfers with REFERs to External Destination

Use the Trunk DN configurations below so the Genesys platform will utilize a REFER method to initiate a transfer to an external destination. Because the options are set at a trunk level, the destination are the phones on the external side of the SBC.

In the Genesys configuration environment, Switch->Trunk DN->Annex Tab->TServer Section, remove the following options on the Trunk pointed to the SBC, if set:

oosp-transfer-enabled=true

sip-replaces-mode=2

Set the following option on the Trunk pointed to the SBC:

refer-enabled=true

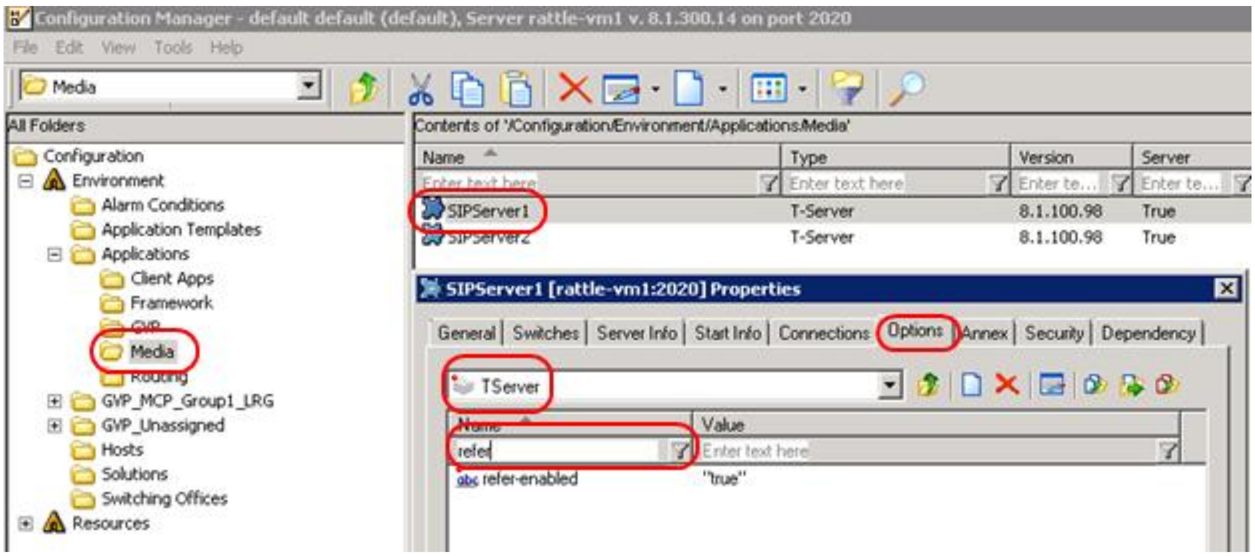


Figure 34: Genesys Administrator – REFER-based Transfer

6.4 Call Hold Using RFC 2543 Method

Use the Trunk DN configurations below so the Genesys platform will utilize RFC 2543 methods of placing a caller on hold (i.e. a=sendonly, a=recvonly, and a=inactive). Because the options are set at a trunk level, the destination are the phones on the untrusted side of the SBC.

In the Genesys configuration environment, Switch->Trunk DN->Annex Tab->TServer Section, set the following option on the Trunk pointed to the SBC:

sip-hold-rfc3264=false

6.5 Call Hold Using RFC 3264 Method

Use the Trunk DN configurations below so the Genesys platform will utilize RFC 3264 methods of placing a caller on hold (i.e. c=0.0.0.0)). Because the options are set at a trunk level, the destination are the phones on the untrusted side of the SBC.

In the Genesys configuration environment, Switch->Trunk DN->Annex Tab->TServer Section, set the following option on the Trunk pointed to the SBC:

sip-hold-rfc3264=true

6.6 Call Progress Detection by Genesys Media Server

Call Progress Detection (CPD) is performed by the Genesys Media Server. For voice, the call will be redirected to an available agent. If the called party is not a voice (silence, fax), the call will be terminated by SIP Server.

In the Genesys configuration environment, Switch->Trunk DN->Annex Tab->TServer Section, set the following option on the Trunk pointed to the SBC:

cpd-capability = mediaserver

In the Test Phone application, go to DataTab-> Extensions Tab->right click Root->Add->String.
Enter the call_answer_type_recognition key with the positive_am_detection value, right click Root->Add->String.

Enter 5 into the call_timeguard_timeout key.

Check the Set as default data checkbox.

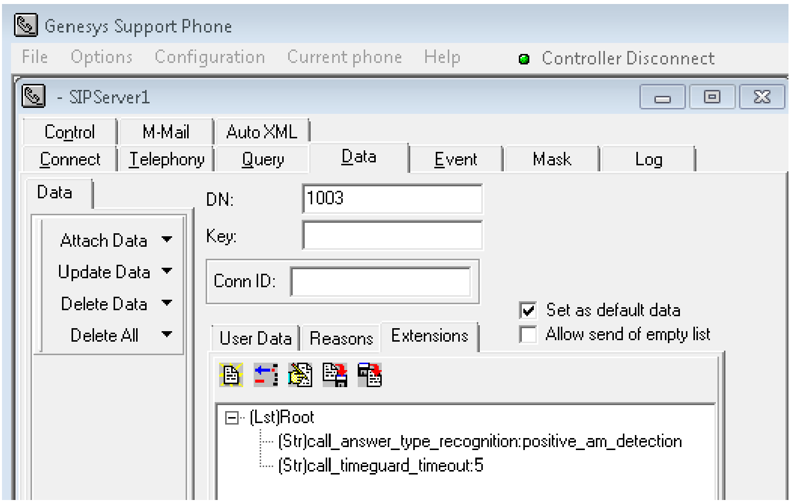


Figure 35: Genesys Testphone - Timeguard

7 Sonus SBC 5000 CLI Configuration Synopsis

This section is a synopsis of all the Sonus SBC 5000 CLI commands used throughout the document, in a single section for easier review and usage.

```
configure
set global signaling sipSigControls maxPduSizeValue pdusize60kb
set global signaling sipSigControls multipleContactsPerAor enabled
commit
set system media mediaPortRange baseUdpPort 16384 maxUdpPort 32767
commit
set system mediaProfile compression 90 tone 10
commit
set profiles media codecEntry G711U_20_INFO codec g711 packetSize 20
set profiles media codecEntry G711U_20_INFO dtmf relay outOfBand
commit
set profiles media codecEntry G711U_20_2833 codec g711 packetSize 20
set profiles media codecEntry G711U_20_2833 dtmf relay rfc2833
commit
set profiles services sipArsProfile GENESYS_ARS
set profiles services sipArsProfile GENESYS_ARS blkListAlgorithms timeouts,retryafter
set profiles services sipArsProfile GENESYS_ARS blkListAlgRetryAfterType sip-503
set profiles services sipArsProfile GENESYS_ARS blkListAlgTimeoutsType sip-invite
set profiles services sipArsProfile GENESYS_ARS blkListAlgTimeoutsNumTimeouts 4
set profiles services sipArsProfile GENESYS_ARS blkListAlgTimeoutsDuration 120
set profiles services sipArsProfile GENESYS_ARS recoveryAlgorithm probe
set profiles services sipArsProfile GENESYS_ARS recoveryAlgProbeInterval 30
set profiles services sipArsProfile GENESYS_ARS recoveryAlgProbeNumResponses 6
set profiles services sipArsProfile GENESYS_ARS recoveryAlgProbeDuration 240
commit
set addressContext default ipInterfaceGroup INTERNAL_IPIG
commit
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF2 ceName SONUSSBC01A
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF2 portName pkt2
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF2 ipAddress 10.35.141.50
prefix 27
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF2 mode inService state enabled
commit
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF3 ceName SONUSSBC01A
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF3 portName pkt3
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF3 ipAddress 10.35.141.51
prefix 27
set addressContext default ipInterfaceGroup INTERNAL_IPIG ipInterface IPIF3 mode inService state enabled
commit
set addressContext default staticRoute 0.0.0.0 0 10.35.141.33 INTERNAL_IPIG IPIF2 preference 100
set addressContext default staticRoute 0.0.0.0 0 10.35.141.33 INTERNAL_IPIG IPIF3 preference 100
commit
set profiles media packetServiceProfile GENESYS_PSP
set profiles media packetServiceProfile GENESYS_PSP codec codecEntry1 G711U_2833
set profiles media packetServiceProfile GENESYS_PSP packetToPacketControl transcode conditional
set profiles media packetServiceProfile GENESYS_PSP packetToPacketControl codecsAllowedForTranscoding
thisLeg g711u
set profiles media packetServiceProfile GENESYS_PSP packetToPacketControl
conditionsInAdditionToNoCommonCodec differentDtmfRelay enable differentPacketSize enable
commit
set profiles signaling ipSignalingProfile GENESYS_IPSP
set profiles signaling ipSignalingProfile GENESYS_IPSP ipProtocolType sipOnly
commit
set addressContext default zone GENESYS_ZONE id 2 remoteDeviceType appServer
commit
set addressContext default zone GENESYS_ZONE sipSigPort 2 ipInterfaceGroup INTERNAL_IPIG
set addressContext default zone GENESYS_ZONE sipSigPort 2 transportProtocolsAllowed sip-udp,sip-tcp
set addressContext default zone GENESYS_ZONE sipSigPort 2 ipAddressV4 10.35.141.52
```

```
set addressContext default zone GENESYS_ZONE sipSigPort 2 portNumber 5060 dscpValue 26
set addressContext default zone GENESYS_ZONE sipSigPort 2 state enabled mode inService
commit
set addressContext default zone GENESYS_ZONE ipPeer GENESYS_PEER1 ipAddress 10.35.176.111 ipPort 5060
set addressContext default zone GENESYS_ZONE ipPeer GENESYS_PEER2 ipAddress 10.35.176.112 ipPort 5060
commit
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 media mediaIpInterfaceGroupName
INTERNAL_IPIG
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 ingressIpPrefix 10.35.176.111 32
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 ingressIpPrefix 10.35.176.112 32
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 policy signaling
ipSignalingProfile GENESYS_IPSP
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 policy media packetServiceProfile
GENESYS_PSP
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 services sipArsProfile
GENESYS_ARS
set addressContext default zone GENESYS_ZONE sipTrunkGroup GENESYS_TG1 state enabled mode inService
commit
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP codec codecEntry1 G711U_20_2833
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP packetToPacketControl transcode conditional
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP packetToPacketControl
codecsAllowedForTranscoding thisLeg g711u
set profiles media packetServiceProfile INTERNAL_AGENTS_PSP packetToPacketControl
conditionsInAdditionToNoCommonCodec differentDtmfRelay enable differentPacketSize enable
commit
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP ipProtocolType sipOnly
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLeg enable
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP commonIpAttributes flags
relayDataPathModeChangeFromOtherCallLeg enable
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP commonIpAttributes relayFlags refer
enable
set profiles signaling ipSignalingProfile INTERNAL_AGENTS_IPSP commonIpAttributes transparencyFlags
referredByHeader enable
commit
set addressContext default zone INTERNAL_AGENTS_ZONE id 3 remoteDeviceType accessDevice
commit
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 ipInterfaceGroup INTERNAL_IPIG
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 transportProtocolsAllowed sip-udp,sip-
tcp
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 ipAddressV4 10.35.141.53
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 portNumber 5060 dscpValue 26
set addressContext default zone INTERNAL_AGENTS_ZONE sipSigPort 3 state enabled mode inService
commit
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 media
mediaIpInterfaceGroupName INTERNAL_IPIG lateMediaSupport passthru
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 ingressIpPrefix
10.0.0.0 8
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 policy signaling
ipSignalingProfile INTERNAL_AGENTS_IPSP
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 policy media
packetServiceProfile INTERNAL_AGENTS_PSP
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 policy
digitParameterHandling numberingPlan NANP_ACCESS
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 signaling
rel100Support disabled relayNonInviteRequest enabled psxRouteForSubscribe enabled
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 signaling
registration requireRegistration required expires 60
set addressContext default zone INTERNAL_AGENTS_ZONE sipTrunkGroup INTERNAL_AGENTS_TG1 state enabled
mode inService
commit
set addressContext default ipInterfaceGroup EXTERNAL_IPIG
commit
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF0 ceName SONUSSBC01A
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF0 portName pkt0
```

```
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF0 ipAddress 32.252.44.209
prefix 29
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF0 mode inService state enabled
commit
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF1 ceName SONUSSBC01A
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF1 portName pkt1
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF1 ipAddress 32.252.44.210
prefix 29
set addressContext default ipInterfaceGroup EXTERNAL_IPIG ipInterface IPIF1 mode inService state enabled
commit
set addressContext default staticRoute 0.0.0.0 0 32.252.44.214 EXTERNAL_IPIG IPIF0 preference 100
set addressContext default staticRoute 0.0.0.0 0 32.252.44.214 EXTERNAL_IPIG IPIF1 preference 100
commit
set profiles media packetServiceProfile CARRIER_PSP
set profiles media packetServiceProfile CARRIER_PSP codec codecEntry1 G711U_20_2833
set profiles media packetServiceProfile CARRIER_PSP packetToPacketControl transcode conditional
set profiles media packetServiceProfile CARRIER_PSP packetToPacketControl codecsAllowedForTranscoding
thisLeg g711u
set profiles media packetServiceProfile CARRIER_PSP packetToPacketControl
conditionsInAdditionToNoCommonCodec differentDtmfRelay enable differentPacketSize enable
commit
set profiles signaling ipSignalingProfile CARRIER_IPSP
set profiles signaling ipSignalingProfile CARRIER_IPSP ipProtocolType sipOnly
set profiles signaling ipSignalingProfile CARRIER_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLeg enable
set profiles signaling ipSignalingProfile CARRIER_IPSP commonIpAttributes flags
relayDataPathModeChangeFromOtherCallLeg enable
set profiles signaling ipSignalingProfile CARRIER_IPSP
commit
set addressContext default zone CARRIER_ZONE id 4
commit
set addressContext default zone CARRIER_ZONE sipSigPort 4 ipInterfaceGroup EXTERNAL_IPIG
set addressContext default zone CARRIER_ZONE sipSigPort 4 transportProtocolsAllowed sip-udp
set addressContext default zone CARRIER_ZONE sipSigPort 4 ipAddressV4 32.252.44.211
set addressContext default zone CARRIER_ZONE sipSigPort 4 portNumber 5060 dscpValue 26
set addressContext default zone CARRIER_ZONE sipSigPort 4 state enabled mode inService
commit
set addressContext default zone CARRIER_ZONE ipPeer CARRIER_PEER1 ipAddress 12.194.20.88 ipPort 5060
set addressContext default zone CARRIER_ZONE ipPeer CARRIER_PEER2 ipAddress 12.194.18.88 ipPort 5060
commit
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 media mediaIpInterfaceGroupName
EXTERNAL_IPIG
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 ingressIpPrefix 12.194.16.0 21
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 policy signaling
ipSignalingProfile CARRIER_IPSP
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 policy media packetServiceProfile
CARRIER_PSP
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 policy digitParameterHandling
numberingPlan NANP_ACCESS
set addressContext default zone CARRIER_ZONE sipTrunkGroup CARRIER_TG1 state enabled mode inService
commit
set profiles media packetServiceProfile REMOTE_AGENTS_PSP
set profiles media packetServiceProfile REMOTE_AGENTS_PSP codec codecEntry1 G711U_20_2833
set profiles media packetServiceProfile REMOTE_AGENTS_PSP packetToPacketControl transcode conditional
set profiles media packetServiceProfile REMOTE_AGENTS_PSP packetToPacketControl
codecsAllowedForTranscoding thisLeg g711u
set profiles media packetServiceProfile REMOTE_AGENTS_PSP packetToPacketControl
conditionsInAdditionToNoCommonCodec differentDtmfRelay enable differentPacketSize enable
commit
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP ipProtocolType sipOnly
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLeg enable
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP commonIpAttributes flags
relayDataPathModeChangeFromOtherCallLeg enable
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP commonIpAttributes relayFlags refer enable
```

```
set profiles signaling ipSignalingProfile REMOTE_AGENTS_IPSP commonIpAttributes transparencyFlags
referredByHeader enable
commit
set addressContext default zone REMOTE_AGENTS_ZONE id 5 remoteDeviceType accessDevice
commit
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 ipInterfaceGroup EXTERNAL_IPIG
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 transportProtocolsAllowed sip-udp,sip-
tcp
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 ipAddressV4 32.252.44.212
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 portNumber 5060 dscpValue 26
set addressContext default zone REMOTE_AGENTS_ZONE sipSigPort 5 state enabled mode inService
commit
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 media
mediaIpInterfaceGroupName EXTERNAL_IPIG lateMediaSupport passthru sourceAddressFiltering disabled
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 ingressIpPrefix
0.0.0.0 0
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 policy signaling
ipSignalingProfile REMOTE_AGENTS_IPSP
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 policy media
packetServiceProfile REMOTE_AGENTS_PSP
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 policy
digitParameterHandling numberingPlan NANP_ACCESS
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 services natTraversal
signalingNat enabled mediaNat enabled
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 signaling
rel100Support disabled relayNonInviteRequest enabled psxRouteForSubscribe enabled
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 signaling
registration requireRegistration required expires 60
set addressContext default zone REMOTE_AGENTS_ZONE sipTrunkGroup REMOTE_AGENTS_TG1 state enabled mode
inService
commit
set profiles callRouting elementRoutingPriority DEFAULT_IP entry nationalType 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry nationalType 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry _private 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry _private 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry nationalOperator 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry nationalOperator 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry transit 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry transit 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry trunkGroupCutThrough 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry trunkGroupCutThrough 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry localOperator 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry localOperator 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry userName 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry userName 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry internationalOperator 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry internationalOperator 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry longDistanceOperator 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry longDistanceOperator 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry othertrunkGroupChosen 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry othertrunkGroupChosen 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry internationalType 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry internationalType 1 entityType
trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry mobile 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry mobile 1 entityType trunkGroup
set profiles callRouting elementRoutingPriority DEFAULT_IP entry test 2 entityType none
set profiles callRouting elementRoutingPriority DEFAULT_IP entry test 1 entityType trunkGroup
commit
set global callRouting routingLabel TO_GENESYS_RL routePrioritizationType sequence action routes
routingLabelRoute 1 trunkGroup GENESYS_TG1 ipPeer GENESYS_PEER1 inService inService
```

```
set global callRouting routingLabel TO_GENESYS_RL routePrioritizationType sequence action routes
routingLabelRoute 2 trunkGroup GENESYS_TG1 ipPeer GENESYS_PEER2 inService inService
commit
set global callRouting routingLabel TO_CARRIER_RL routePrioritizationType sequence action routes
routingLabelRoute 1 trunkGroup CARRIER_TG1 ipPeer CARRIER_PEER1 inService inService
set global callRouting routingLabel TO_CARRIER_RL routePrioritizationType sequence action routes
routingLabelRoute 2 trunkGroup CARRIER_TG1 ipPeer CARRIER_PEER2 inService inService
commit
set global callRouting route none Sonus_NULL Sonus_NULL standard 68 1 all all ALL none Sonus_NULL
routingLabel TO_CARRIER_RL
commit
set global callRouting route none Sonus_NULL Sonus_NULL standard 21443268 1 all all ALL none Sonus_NULL
routingLabel TO_GENESYS_RL
commit
set global callRouting route trunkGroup REMOTE_AGENTS_TG1 SONUSSBC01 standard Sonus_NULL Sonus_NULL all
all ALL none Sonus_NULL routingLabel TO_GENESYS_RL
commit
set global callRouting route trunkGroup REMOTE_AGENTS_TG1 SONUSSBC01 username Sonus_NULL Sonus_NULL all
all ALL none Sonus_NULL routingLabel TO_GENESYS_RL
commit
set global callRouting route trunkGroup INTERNAL_AGENTS_TG1 SONUSSBC01 standard Sonus_NULL Sonus_NULL
all all ALL none Sonus_NULL routingLabel TO_GENESYS_RL
commit
set global callRouting route trunkGroup INTERNAL_AGENTS_TG1 SONUSSBC01 username Sonus_NULL Sonus_NULL
all all ALL none Sonus_NULL routingLabel TO_GENESYS_RL
commit
exit
```