
Sonus SBC 5XX0 5.1.1R0 IOT Genesys Voice Platform 8.1

Application Notes

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Document Overview

This document provides a configuration guide for Sonus SBC 5XX0 Series (Session Border Controller) when connecting to Genesys Voice Platform.

This configuration guide supports features documented in the Genesys Voice Platform guides.


- For additional information on Genesys Voice Platform, please visit <http://www.genesys.com/>
- For additional information on Sonus SBC, please visit <http://sonus.net>.

Introduction

The interoperability compliance testing focuses on verifying inbound and outbound calls flows between Sonus SBC 5XX0 and Genesys Voice Platform.

Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring both the Sonus SBC and the third-party product. There will be steps that require navigating third-party as well as the 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.

 This configuration guide is offered as a convenience to Sonus customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide are believed to be accurate, but are presented without warranty of any kind, express or implied, and are provided "AS IS." Users must take full responsibility for the application of the specifications and information in this guide.

Requirements

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

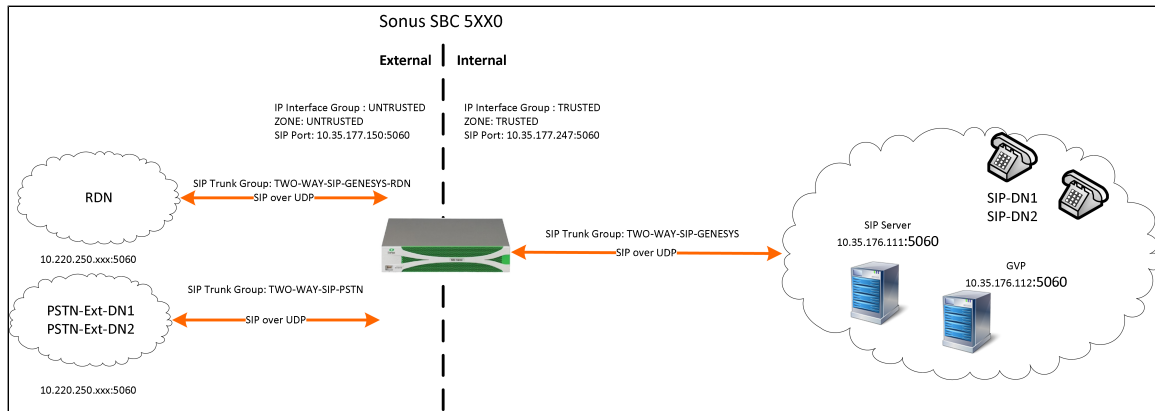
Table 1: Requirements

	Equipment	Software Version
Sonus Networks	Sonus SBC 5200	V05.01.01-R000
	BMC	V02.16.00
	BIOS	V02.06.00
	ConnexIP OS	V03.01.01-R000
	SonusDB	V05.01.01-R000
	EMA	V05.01.01-R000
	SBX	V05.01.01-R000
Third-party Equipment	Genesys SIP Server	8.1.100.98
	Genesys GVP MCP	8.1.504.93
	Polycom SoundPoint IP 501 SIP	2.1.3

Reference Configuration

The following reference configuration shows connectivity between third-party and Sonus SBC 5XX0.

Figure 1: Connectivity Between Third-Party and Sonus SBC 5XX0



Support

For any questions regarding this document or the content herein, please contact your maintenance and support provider.

Third-Party Product Features

The testing was executed with the Genesys test plan. The following features were tested:

Inbound calls

- Inbound Call to Agent released by caller
- Inbound Call to Agent released by agent
- Inbound Calls rejected
- Inbound Call abandoned
- Inbound Call to Route Point with Treatment
- Interruptible Treatment
- IVR (Collect Digit) Treatment
- Inbound Call routed by using 302 out of SIP Server signaling path

Outbound Calls

- 1PCC Outbound Call from SIP Endpoint to external destination
- 3PCC Outbound Call to external destination
- 1PCC Outbound Call Abandoned

Music on hold

- Caller is put on hold and retrieved by using RFC 2543 method
- T-Lib-Initiated Hold/Retrieve Call with MOH using RFC 3264 method

Call Transfers

- 3PCC 2 Step Transfer to internal destination by using re-INVITE method
- 3PCC Alternate from consult call to main call
- 1PCC Unattended (Blind) transfer using REFER
- 1PCC Attended Transfer to external destination
- 3PCC Two Step Conference to external party
- 3PCC (same as 1PCC) Single-Step Transfer to another agent
- 3PCC Single Step Transfer to external destination using REFER
- 3PCC Single Step Transfer to internal busy destination using REFER

Other scenarios

- Early Media for Inbound Call to Route Point with Treatment
- Early Media for Inbound Call with Early Media for Routed to Agent
- Inbound call routed outbound (Remote Agent) using INVITE without SDP
- Call Progress Detection
- Out of Service detection. Checking MGW live status
- SIP Authentication for outbound calls
- SIP Authentication for incoming calls

Remote Agent behind SBC scenarios

- T-Lib-Initiated Answer/Hold/Retrieve Call for Remote SIP endpoint which supports the BroadSoft SIP Extension Event Package
- 3PCC Outbound Call from Remote SIP endpoint to external destination

- 3PCC 2 Step Transfer from Remote SIP endpoint to internal destination
- 1PCC Attended Transfer from Remote SIP endpoint to external destination

Verify License

SBC-POL-RTU

Genesys Voice Platform Configuration

The following new configurations are included in this section:

1. [Accessing Genesys Tools and Interfaces](#)
2. [Creating SIP Switch in Genesys Administrator](#)
3. [SIP Server Configuration in Genesys Administrator](#)
4. [Genesys Media Server Deployment](#)
5. [Stat Server Configuration](#)

1. Accessing Genesys Tools and Interfaces

This section provides the configuration required for the Genesys components.

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.

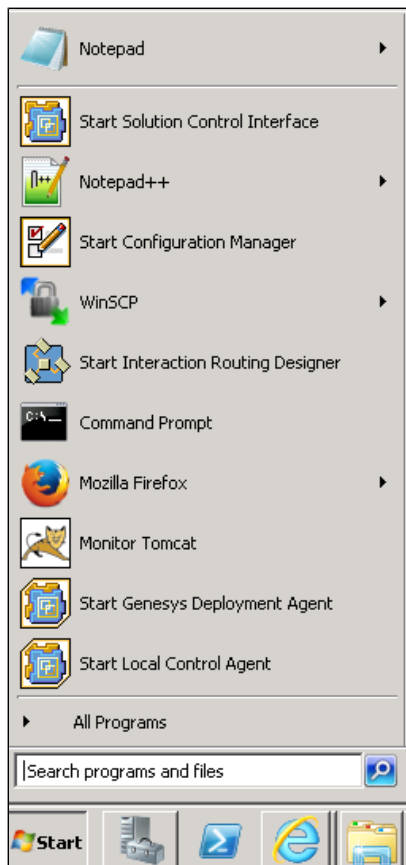
Once logged onto the Genesys system, click the Start button and look for the installed applications shown below. If the applications are not visible on the Start Menu, find them using the search box just above the Start button.

- The Solution Control Interface is used to start/stop the various applications, as well as identify the configuration of each application.
- The Configuration Manager is a tool 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.
- Internet Explorer (not shown in the startup menu) is used to access the Genesys Administrator. The URI should be available from those who installed the platform.



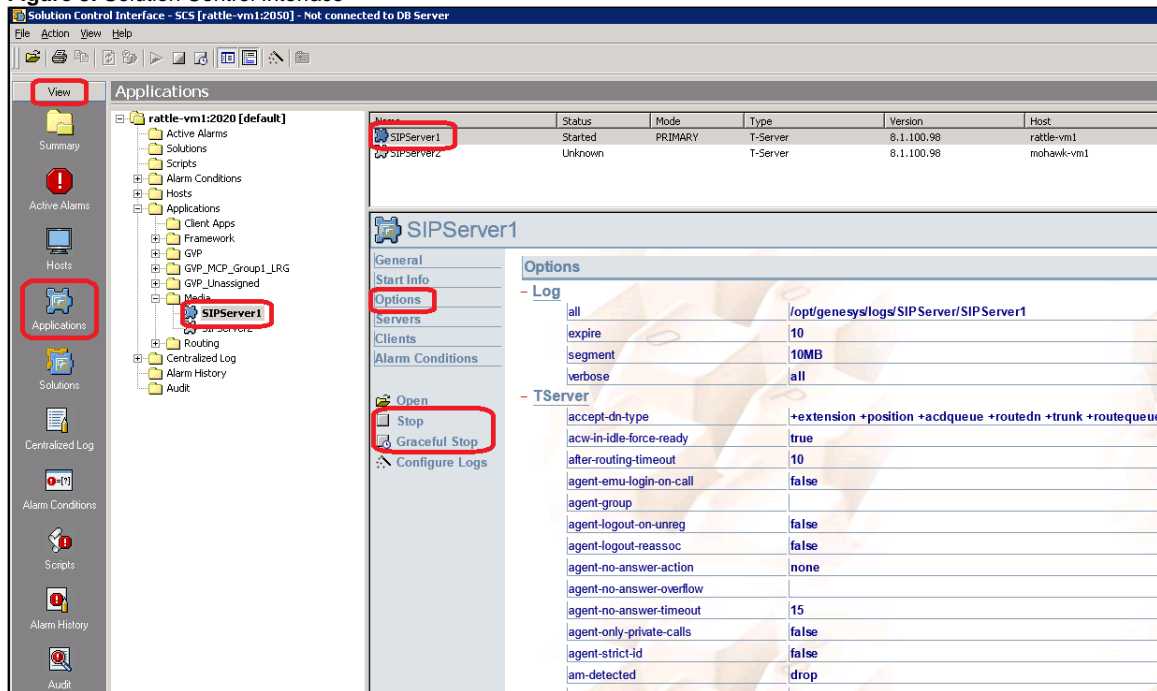
Steps can be performed using multiple tools. For example, starting or stopping an application can be performed in the Genesys Administrator as well as the Solution Control Interface.

Figure 2: Access Solution Control Interface



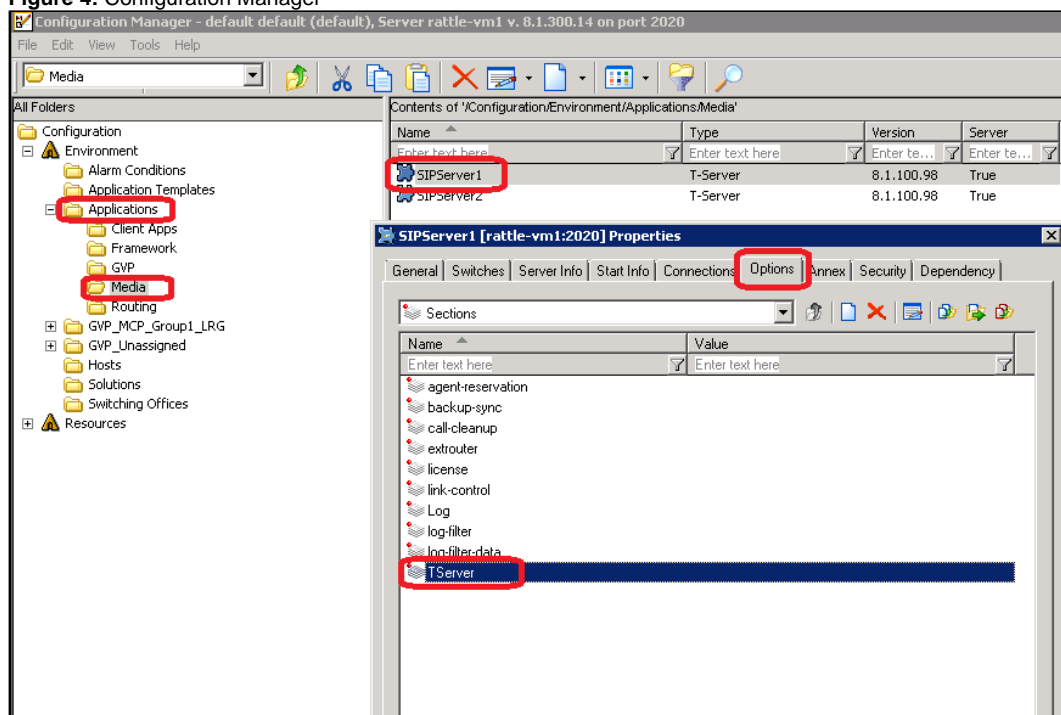
- Below is a screenshot of the Solution Control Interface. In this application, click **View**, top left, to gain access to the area of interest. Click **Applications** and expand the folders of interest. In this example, SIPServer1 properties are displayed. Clicking the various links will display the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

Figure 3: Solution Control Interface



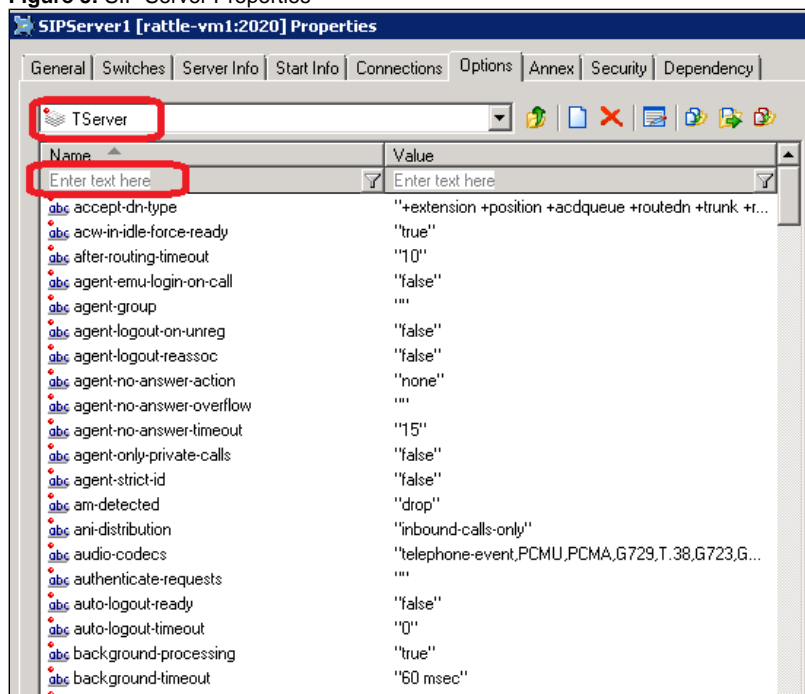
- The Configuration Manager is used to configure the platform and its applications. Once opened, click **Applications/Media** (it's possible that your SIP Server is in a different folder under Applications) and the upper left pane will display both SIP Servers. Double-click **SIPServer1** and a dialog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

Figure 4: Configuration Manager



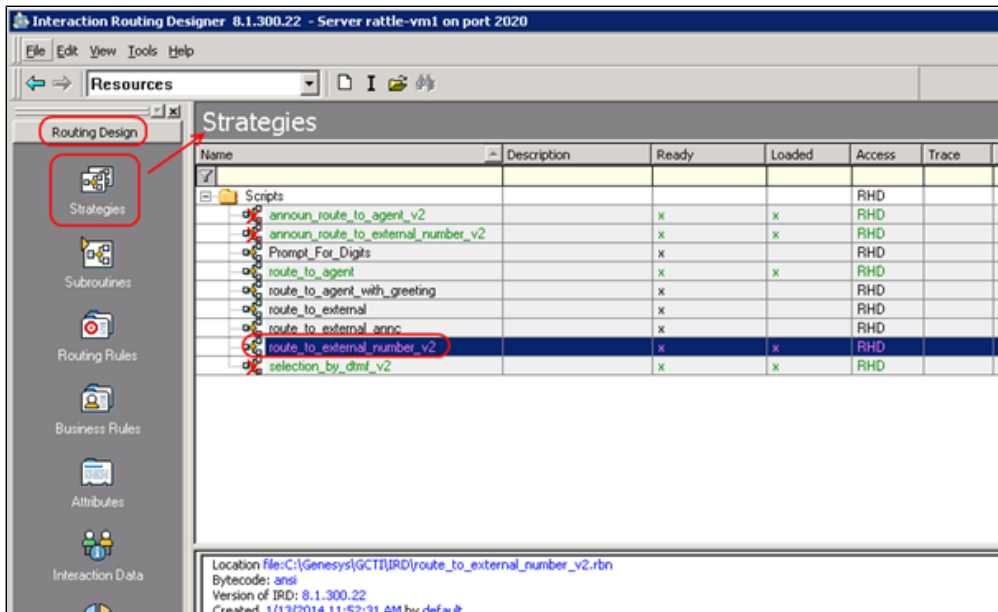
Note that there are many option parameters. Type the name of the option in the filter "notice that it filters in real time". Some options can be set at both Application and Switch/DN levels. The option setting at the DN level takes precedence over the Application-level setting. See the *Genesys SIP Server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

Figure 5: SIP Server Properties



- The Interaction Routing Designer is a tool used to create Route Points and/or Strategies. Access the Strategies by clicking **Routing Design>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



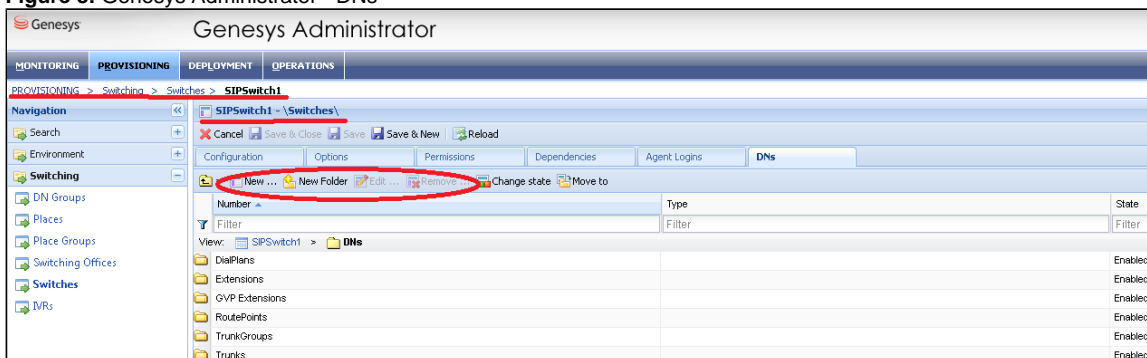
- Access the Genesys Administrator with a web browser. Contact the administrator or person(s) who performed the install of the system to determine the URL. Once opened, click the **Provisioning** tab and under the Navigation area click Switching. Under **Switching**, click **Switches** to display the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



2. Creating SIP Switch in Genesys Administrator

- Within Genesys Administrator, create a Switching Office named SIPServer Switching Office.

Figure 9: Genesys Administrator - Creating SIP Switch

The screenshot shows the Genesys Administrator interface. The top navigation bar includes tabs for MONITORING, PROVISIONING (selected), DEPLOYMENT, and OPERATIONS. Below the navigation bar, the breadcrumb trail reads: PROVISIONING > Switching > Switching Offices > SIPOffice. The left sidebar contains a 'Navigation' pane with a tree view showing 'Switching' expanded, with sub-items: DN Groups, Places, Place Groups, Switching Offices (selected), Switches, and IVRs. The main content area is titled 'SIPOffice - \Switching Offices\' and contains a toolbar with 'Cancel', 'Save & Close', 'Save', 'Save & New', and 'Reload' buttons. Below the toolbar are four tabs: 'Configuration' (selected), 'Options', 'Permissions', and 'Dependencies'. The 'Configuration' tab displays the following fields: '* Name:' with the value 'SIPserver', '* Switch Type:' with the value 'SIP Switch', and 'State:' with a checked 'Enabled' checkbox.

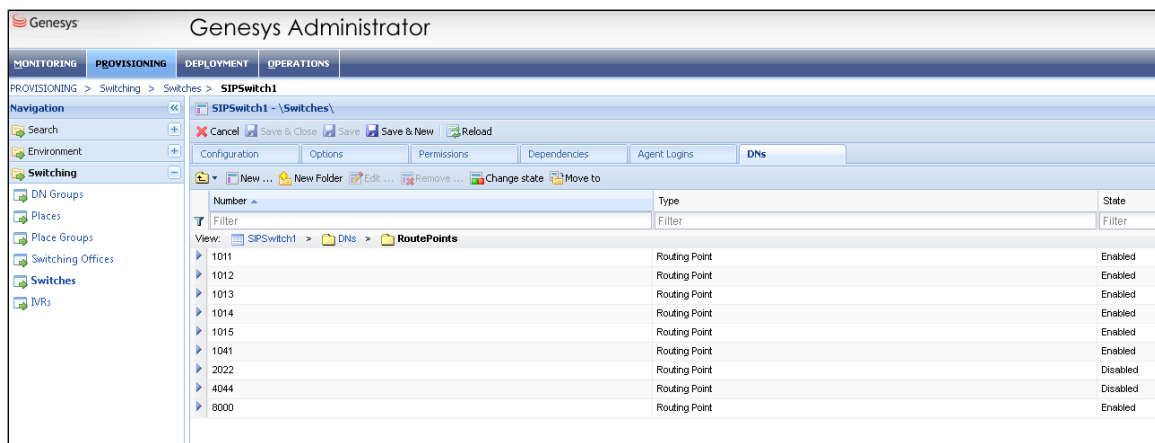
- Within Genesys Administrator, create a SIP Server Switch and associate the Switching Office created in the previous step with this switch.

Figure 10: Genesys Administrator - SIP Switch Association

This screenshot is identical to Figure 9, showing the Genesys Administrator interface for creating a new SIP Switch. The 'Configuration' tab is active, and the fields are: '* Name:' (SIPserver), '* Switch Type:' (SIP Switch), and 'State:' (Enabled checkbox checked).

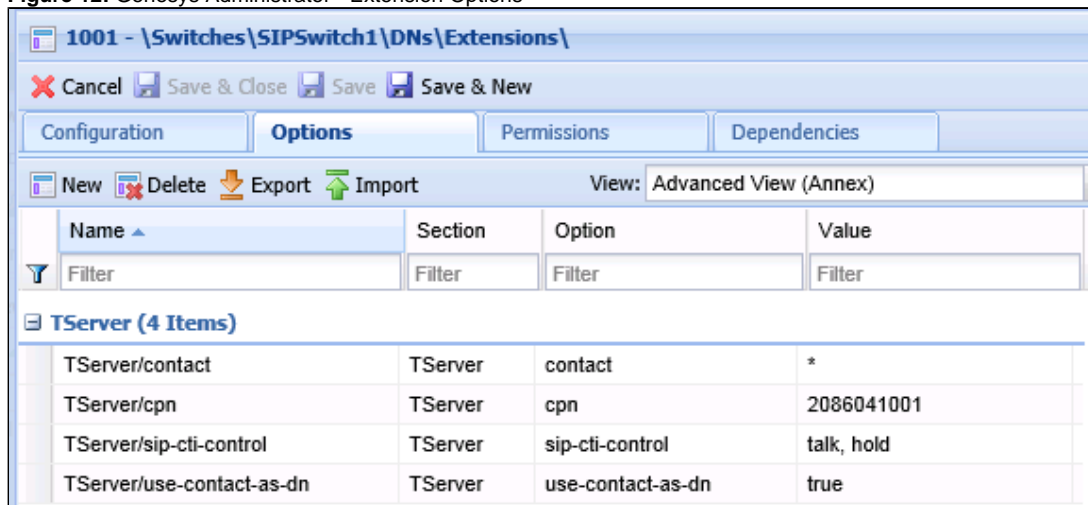
- Under the SIPSwitch created in the above step, define Routing Points to run URS strategies from, the SIP trunk representing connection of SIP Server to Sonus, and a "msml" VoIP service DN required to integrate SIP Server with Media Server to support call hold and conferencing functionalities.

Figure 11: Genesys Administrator - Define Routing Points



- Define DNs of type Extension under SIPSwitch with the following options in the TServer section for various SIP endpoints that will register to the 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



- Define a SIP trunk DN to represent all SIP calls arriving from the Sonus SBC internal interface to the 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 1000/2000 through which the SIP REGISTER message was received by SIP Server.

Figure 13: Genesys Administrator - SIP Trunk Options

Bock-SBC-Priv - \Switches\SIPSwitch1\DNs\Trunks

Cancel Save & Close Save Save & New

Configuration **Options** Permissions Dependencies

New Delete Export Import View: Advanced View (Annex)

Name	Section	Option	Value
Filter	Filter	Filter	Filter

TServer (7 Items)

TServer/contact	TServer	contact	10.35.141.52:5060
TServer/cpd-capability	TServer	cpd-capability	mediaserver
TServer/dial-plan	TServer	dial-plan	DialPlanInbound
TServer/prefix	TServer	prefix	214340

- Define an 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

MediaServer - \Switches\SIPSwitch1\DNs\Trunks

Cancel Save & Close Save Save & New

Configuration **Options** Permissions Dependencies

New Delete Export Import View: Advanced View (Annex)

Name	Section	Option	Value
Filter	Filter	Filter	Filter

TServer (6 Items)

TServer/contact-list	TServer	contact-list	10.35.176.112:5060, 10.3...
TServer/oos-check	TServer	oos-check	15
TServer/oos-force	TServer	oos-force	20
TServer/prefix	TServer	prefix	msml=
TServer/service-type	TServer	service-type	msml
TServer/subscription-id	TServer	subscription-id	Resources

- 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

GVP_RMPair - \Switches\SIPSwitch1\DNs\Trunks\

Cancel Save & Close Save Save & New

Configuration Options Permissions Dependencies

New Delete Export Import View: Advanced View (Annex)

Name	Section	Option	Value
Filter	Filter	Filter	Filter

TServer (9 Items)

TServer/contact-list	TServer	contact-list	10.35.176.112:5060, 10.35....
TServer/make-call-rtc3725-flow	TServer	make-call-rtc3725-...	1
TServer/oos-check	TServer	oos-check	15
TServer/oos-force	TServer	oos-force	20
TServer/prefix	TServer	prefix	msml=
TServer/refer-enabled	TServer	refer-enabled	false
TServer/ring-tone-on-make-call	TServer	ring-tone-on-make...	false
TServer/service-type	TServer	service-type	msml
TServer/subscription-id	TServer	subscription-id	Resources

- 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

1011 - \Switches\SIPSwitch1\DNs\RoutePoints\

Cancel Save & Close Save Save & New

Configuration Options Permissions Dependencies

General Advanced Routing & Orchestration Cost Based Routing Default DNS

General

Number: 1011

Type: Routing Point

Switch: SIPSwitch1

Association:

Register: True

State: ☒ Enabled

Advanced

Alias: 1011_SIPSwitch1

Route Type: Default

Group: [Unknown Group]

Use Override: ☒ True

Override:

DN Login ID:

Switch-specific Type: 1

Trunks: 0

Figure 17: Genesys Administrator - Routing Strategies

Routing & Orchestration

Orchestration Application:

Routing Strategies:

Add Edit Remove

Routing Strategy	Router
route_to_agent	URS1
route_to_agent	URS2

Cost Based Routing

- The 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.

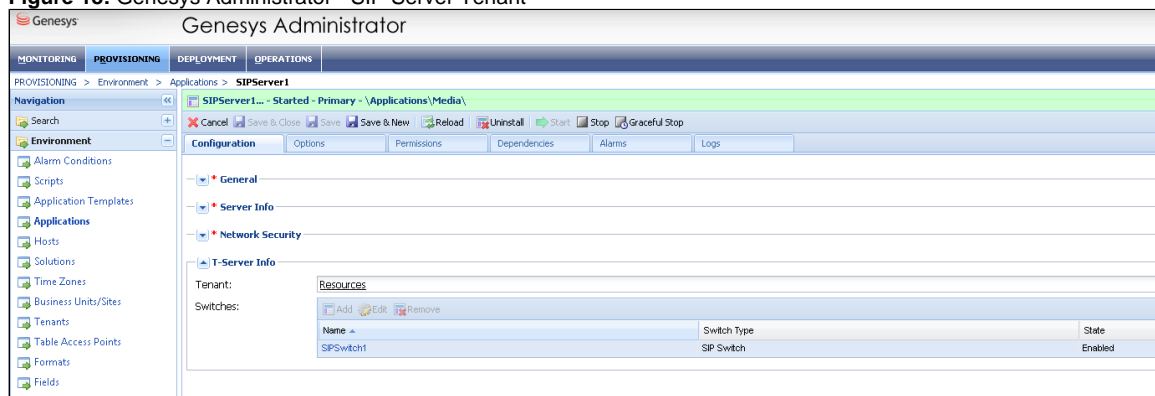
With this full control access, the SIP Server Switch grants DNs like Extension to update their options like “contact” when a new SIP register message is received from end points moving to a new IP location

3. SIP Server Configuration in Genesys Administrator

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

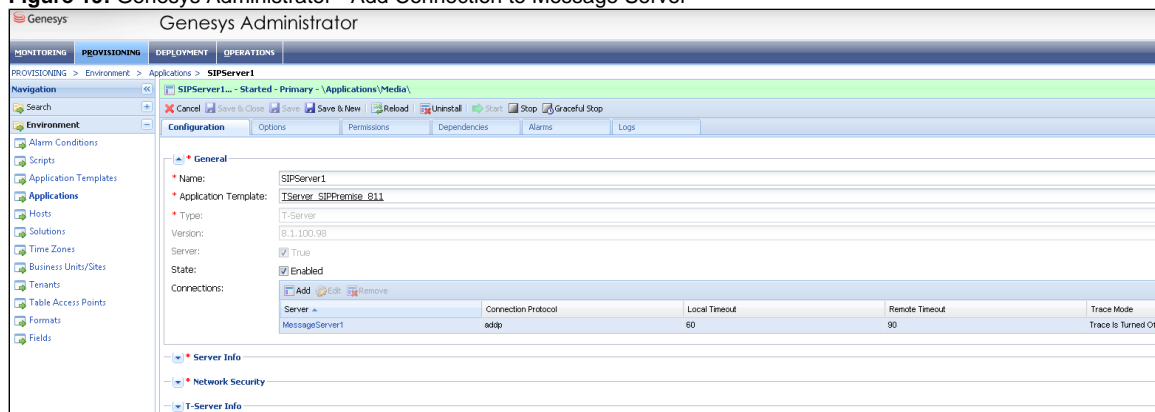
- Install and configure SIP Server as per *Genesys Framework SIP Server Deployment Guide*.
- Add a connection to the SIP Server Switch created above, to monitor all the resources under this switch: **Genesys Administrator->Provisioning->Environment->Applications->SIP Server Application**. Also, SIP Server should add a connection to the tenant.

Figure 18: Genesys Administrator - SIP Server Tenant



- Add a connection to Message Server.

Figure 19: Genesys Administrator - Add Connection to Message Server



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

4. Genesys Media Server Deployment

Follow these steps to configure a Media Server deployment.

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.

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

3. Verify VoIP service DN of type=msml as specified in section Creating SIP Switch in Genesys Administrator to support SIP Server-Media Server MSML interactions to support treatments and conferencing capabilities.
4. 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 0.

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

5. 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.

6. 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.

7. 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>

8. 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).

9. 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.

1. On the **Provisioning** tab, click **Voice Platform>Resource Groups**.
2. On the **Details** pane tool bar, click **New**.
3. The Resource Group Wizard opens to the Welcome page.
4. 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.
5. 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.

6. 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*.

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

1. Log in to Genesys Administrator.
2. On the **Provisioning** tab, select **Voice Platform>IVR Profiles**.
3. In the **Tasks** panel, click **Define New IVR Profile**. The IVR Profile Wizard opens to the Welcome page.
4. On the Service Type page, enter the name of the default IVR Profile, IVR_App_Default.
5. Select either **Conference** or **Announcement** from the drop-down list. (Only one service type per IVR Profile is supported.)
6. If you selected Conference, on the Service Properties page, enter a conference ID number.
7. If you selected Announcement, on the Service Properties page, enter the URL of the announcement, for example, http://webserver/hello.wav.
8. 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

9. In the gvp.general section of the Tenant's Annex tab, set the default-application to this default IVR Profile name – IVR_App_Default.

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

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 per the *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

The screenshot shows the Genesys Administrator interface for configuring a Stat Server. The left sidebar contains navigation links for Environment, Scripts, Application Templates, Applications, Hosts, Solutions, Time Zones, Business Units/Sites, Tenants, Table Access Points, Formats, and Fields. The main pane is titled 'StatServer1' and shows the following configuration:

- General:**
 - Name: StatServer1
 - Application Template: Stat_Server_012
 - Type: Stat Server
 - Version: 6.1.200.24
 - Server: ☒ True
 - State: ☒ Enabled
- Connections:**

Server	Connection Protocol	Local Timeout	Remote Timeout	Trace Mode
MessageServer1	adtp	60	90	Trace is Turned Off
SIPServer1	adtp	60	90	Trace is Turned Off

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 per the *Genesys Universal Routing Deployment Guide*.
2. Add connections to Message Server, Stat Server, and SIP Server.
3. Add connection to SIP Server to monitor events received by SIP Server for various route points and extensions on the SIP Server Switch.
4. Add connection to Stat Server to query Stat Server for routing calls to available and ready agents.

Figure 21: Genesys Administrator - URS Configuration

The screenshot shows the Genesys Administrator interface for configuring a Universal Routing Server (URS). The left sidebar is the same as in Figure 20. The main pane is titled 'URS1' and shows the following configuration:

- General:**
 - Name: URS1
 - Application Template: UR_Server_013
 - Type: Universal Routing Server
 - Version: 6.1.200.27
 - Server: ☒ True
 - State: ☒ Enabled
- Connections:**

Server	Connection Protocol	Local Timeout	Remote Timeout	Trace Mode
MessageServer1	adtp	60	90	Trace is Turned Off
SIPServer1	adtp	60	90	Trace is Turned Off
StatServer1	adtp	60	90	Trace is Turned Off

Use any of the strategies below to test your configuration.

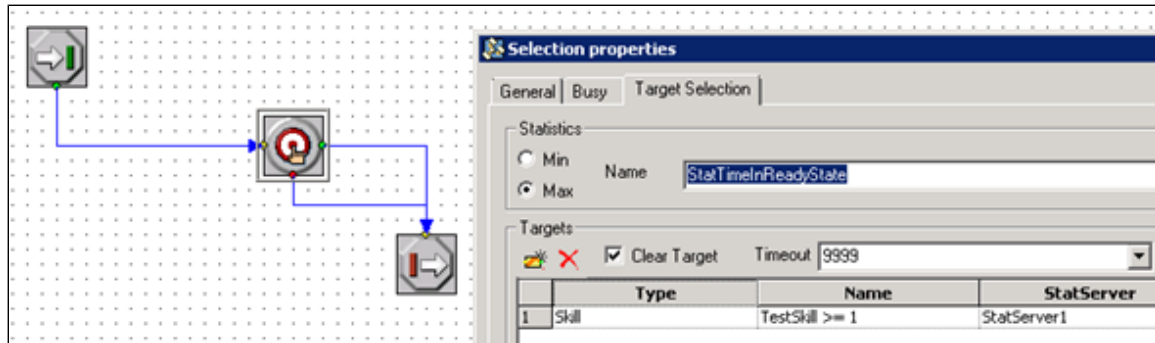
7. URS Routing Strategies

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

- 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



- 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

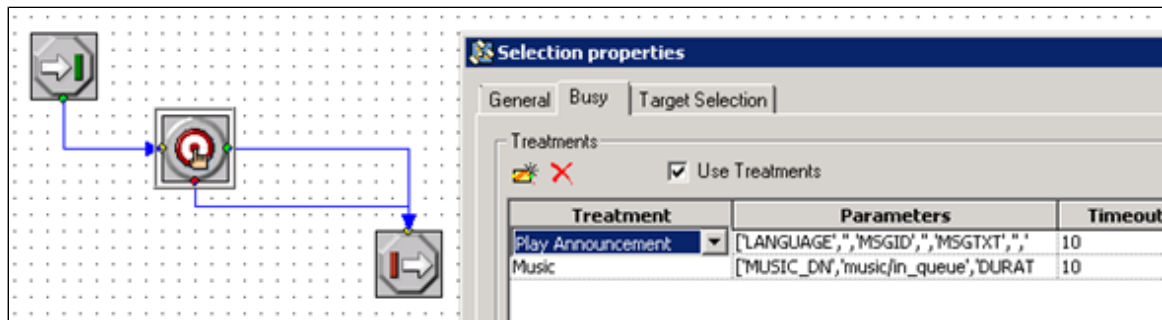
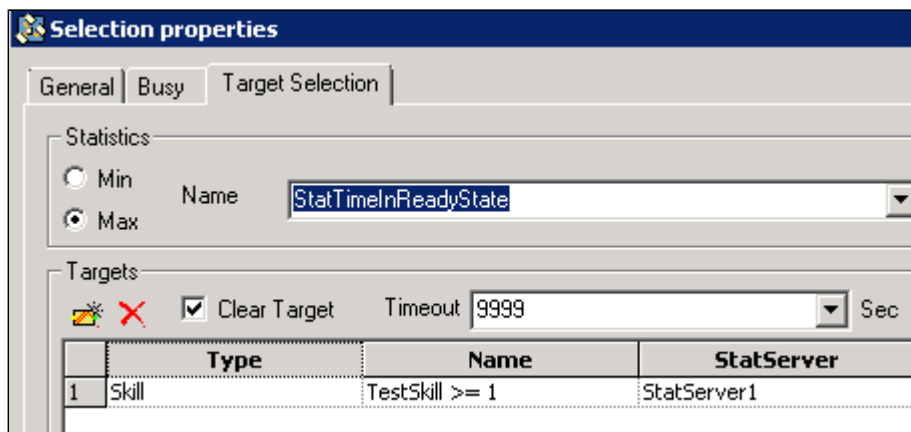


Figure 24: URS Strategy #2 - Target Selection



- 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

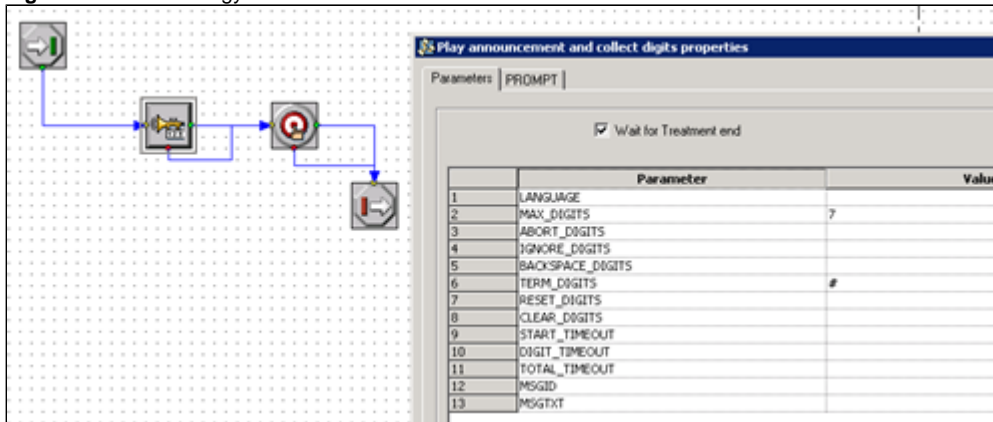


Figure 26: URS Strategy #3 - Prompt Tab

Play announcement and collect digits properties						
Parameters: PROMPT						
	Interruptable	ID	Digits	User_Ann_ID	Text	User_ID
1	<input checked="" type="checkbox"/>	1002				

Figure 27: URS Strategy #3 - Busy Tab

Selection properties		
General Busy Target Selection		
Treatments <input checked="" type="checkbox"/> Use Treatments		
Treatment	Parameters	Timeout
Music	[MUSIC_DW, 'music/in_queue', DURATION	10

Figure 28: URS Strategy #3 - Target Selection

Selection properties

General | **Busy** | Target Selection

Statistics

☐ Min
 ☒ Max
 Name:

Targets

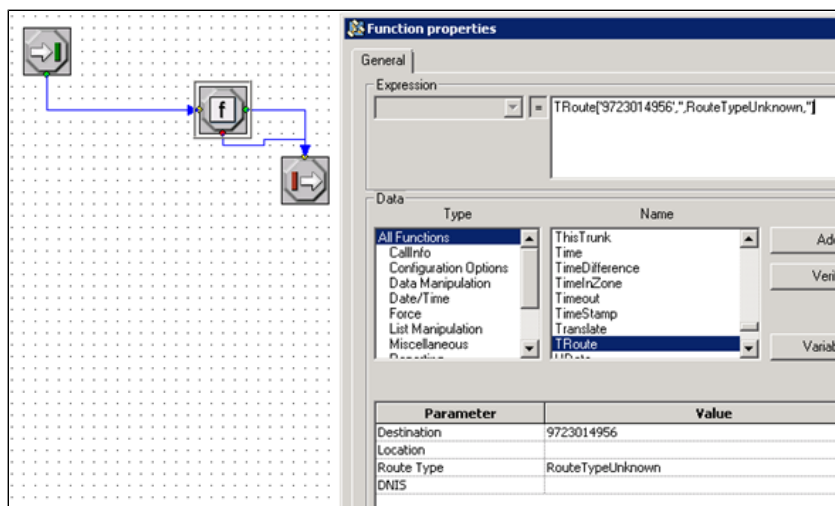
☒ Clear Target
 Timeout: Sec

	Type	Name	StatServer
1	Skill	TestSkill >= 1	StatServer1

- 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



- 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

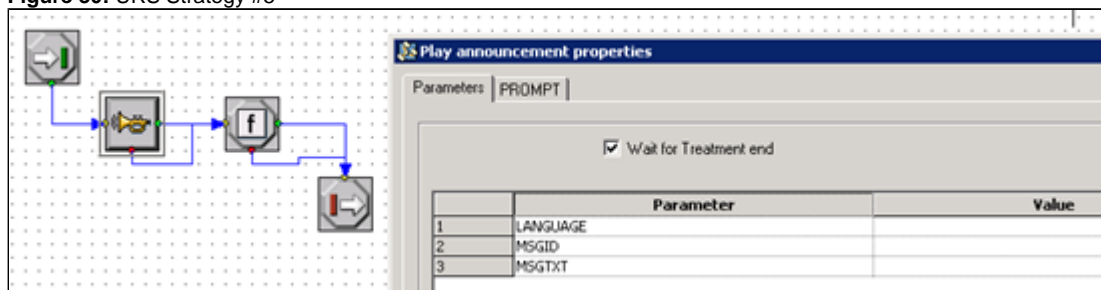


Figure 31: URS Strategy #5 - Prompt Tab

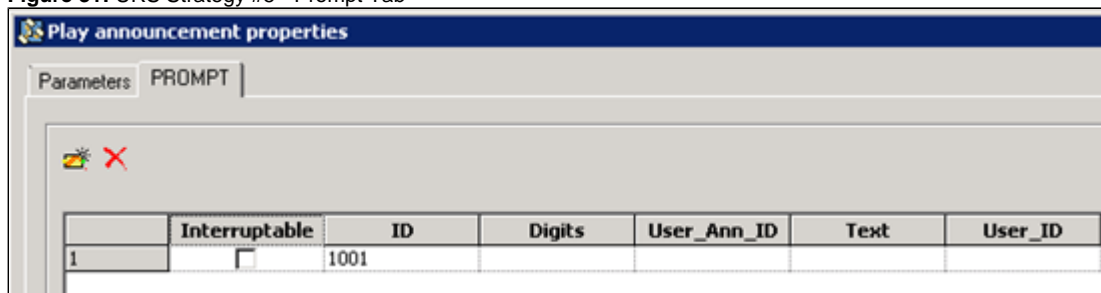
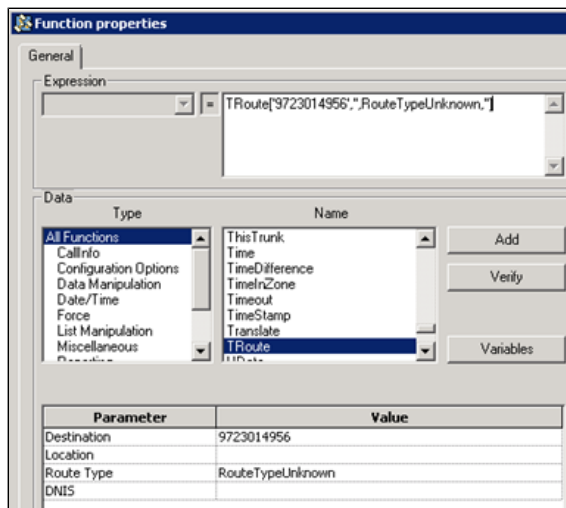


Figure 32: URS Strategy #5 - Function Properties



Sonus SBC 5xx0 Series Configuration

Complete Configuration

```
configure
#DSP Resources
set system mediaProfile compression 90 tone 10
commit
#Element Routing Priority profile
set profiles callRouting elementRoutingPriority TG_ERP entry localOperator 0 entityType trunkGroup
set profiles callRouting elementRoutingPriority TG_ERP entry nationalType 0 entityType trunkGroup
set profiles callRouting elementRoutingPriority TG_ERP entry internationalType 0 entityType trunkGroup
commit
#Ip Signaling Profiles
set profiles signaling ipSignalingProfile GENESYS_IPSP commonIpAttributes flags endToEndBye enable
set profiles signaling ipSignalingProfile GENESYS_IPSP commonIpAttributes relayFlags dialogEventPackage enable
set profiles signaling ipSignalingProfile GENESYS_IPSP commonIpAttributes relayFlags info enable
set profiles signaling ipSignalingProfile GENESYS_IPSP commonIpAttributes relayFlags notify enable
set profiles signaling ipSignalingProfile GENESYS_IPSP commonIpAttributes relayFlags refer enable
set profiles signaling ipSignalingProfile GENESYS_IPSP commonIpAttributes relayFlags statusCode3xx enable
set profiles signaling ipSignalingProfile GENESYS_IPSP commonIpAttributes relayFlags statusCode4xx6xx enable
set profiles signaling ipSignalingProfile GENESYS_IPSP egressIpAttributes flags disable2806Compliance enable
commit
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes flags disableMediaLockDown enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes flags includeTransportTypeInContactHeader
enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes flags reQueryPsxOnRegisterRefresh enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes flags endToEndBye enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes flags
minimizeRelayingOfMediaChangesFromOtherCallLegAll enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes flags
relayDataPathModeChangeFromOtherCallLeg enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes relayFlags dialogEventPackage enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes relayFlags info enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes relayFlags notify enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes relayFlags options enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes relayFlags refer enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes relayFlags statusCode3xx enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes relayFlags statusCode4xx6xx enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes relayFlags updateWithoutSdp enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes transparencyFlags authcodeHeaders enable
set profiles signaling ipSignalingProfile ACCESS_IPSP egressIpAttributes flags disable2806Compliance enable
set profiles signaling ipSignalingProfile ACCESS_IPSP egressIpAttributes sipHeadersAndParameters flags endToEndAck
enable
set profiles signaling ipSignalingProfile ACCESS_IPSP egressIpAttributes flags sameCallIdForRequiredAuthorization
enable
```

```

set profiles signaling ipSignalingProfile ACCESS_IPSP egressIpAttributes privacy transparency enable
set profiles signaling ipSignalingProfile ACCESS_IPSP ingressIpAttributes flags sendSdpInSubsequent18x enable
set profiles signaling ipSignalingProfile ACCESS_IPSP ingressIpAttributes flags suppress183WithoutSdp enable
set profiles signaling ipSignalingProfile ACCESS_IPSP ingressIpAttributes flags suppress183For3xxRedirectResponse
enable sendSdpIn200OkIf18xReliable enable
set profiles signaling ipSignalingProfile ACCESS_IPSP commonIpAttributes flags noPortNumber5060 enable
commit
#Trusted Side Configuration
#IP Interface Group
set addressContext default ipInterfaceGroup TRUSTED ipInterface TRUSTED portName pkt0
set addressContext default ipInterfaceGroup TRUSTED ipInterface TRUSTED ipAddress 10.35.177.246
set addressContext default ipInterfaceGroup TRUSTED ipInterface TRUSTED prefix 26
set addressContext default ipInterfaceGroup TRUSTED ipInterface TRUSTED state enabled
commit
#IP Static Route
set addressContext default staticRoute 0.0.0.0 0 10.35.177.193 TRUSTED TRUSTED preference 100
commit
#zone
set addressContext default zone TRUSTED id 3
commit
#SIP signaling port
set addressContext default zone TRUSTED sipSigPort 3 ipInterfaceGroupName TRUSTED
set addressContext default zone TRUSTED sipSigPort 3 ipAddressV4 10.35.177.247
set addressContext default zone TRUSTED sipSigPort 3 portNumber 5060
set addressContext default zone TRUSTED sipSigPort 3 transportProtocolsAllowed sip-udp
set addressContext default zone TRUSTED sipSigPort 3 mode inService
set addressContext default zone TRUSTED sipSigPort 3 state enabled
commit
#IP Peer
set addressContext default zone TRUSTED ipPeer GENESYS ipAddress 10.35.176.111
set addressContext default zone TRUSTED ipPeer GENESYS ipPort 5060
set addressContext default zone TRUSTED ipPeer GENESYS authentication intChallengeResponse enabled
set addressContext default zone TRUSTED ipPeer GENESYS authentication incInternalCredentials enabled
commit
#SIP trunk group
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS policy callRouting
elementRoutingPriority TG_ERP
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS policy signaling ipSignalingProfile
GENESYS_IPSP
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS media mediaIpInterfaceGroupName TRUSTED
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS ingressIpPrefix 10.35.176.111 32
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS mode inService
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS state enabled
commit
#Untrusted Side Configuration
set addressContext default ipInterfaceGroup UNTRUSTED ipInterface UNTRUSTED ceName LITTLE
set addressContext default ipInterfaceGroup UNTRUSTED ipInterface UNTRUSTED portName pkt2
set addressContext default ipInterfaceGroup UNTRUSTED ipInterface UNTRUSTED ipAddress 10.35.177.150
set addressContext default ipInterfaceGroup UNTRUSTED ipInterface UNTRUSTED prefix 26
set addressContext default ipInterfaceGroup UNTRUSTED ipInterface UNTRUSTED mode inService
set addressContext default ipInterfaceGroup UNTRUSTED ipInterface UNTRUSTED state enabled
commit
#IP Static Route
set addressContext default staticRoute 0.0.0.0 0 10.35.177.129 UNTRUSTED UNTRUSTED preference 100
commit
#zone
set addressContext default zone UNTRUSTED id 4
#SIP signaling port
set addressContext default zone UNTRUSTED sipSigPort 4 ipInterfaceGroupName UNTRUSTED
set addressContext default zone UNTRUSTED sipSigPort 4 ipAddressV4 10.35.177.151
set addressContext default zone UNTRUSTED sipSigPort 4 portNumber 5060
set addressContext default zone UNTRUSTED sipSigPort 4 transportProtocolsAllowed sip-udp,sip-tcp
set addressContext default zone UNTRUSTED sipSigPort 4 mode inService
set addressContext default zone UNTRUSTED sipSigPort 4 state enabled
commit
#IP Peer
set addressContext default zone TRUSTED ipPeer PSTN ipAddress 10.35.8.146
set addressContext default zone TRUSTED ipPeer PSTN ipPort 5060
set addressContext default zone TRUSTED ipPeer PSTN surrogateRegistration authUserName 2086041020
set addressContext default zone TRUSTED ipPeer PSTN surrogateRegistration regAuthPassword 123456
set addressContext default zone TRUSTED ipPeer PSTN surrogateRegistration sendCredentials challengeForAnyMessage
set addressContext default zone TRUSTED ipPeer PSTN authentication intChallengeResponse enabled

```

```

set addressContext default zone TRUSTED ipPeer PSTN authentication incInternalCredentials enabled
commit
#SIP trunk groups
#PSTN trunk
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-PSTN policy callRouting elementRoutingPriority
TG_ERP
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-PSTN policy signaling ipSignalingProfile
GENESYS_IPSP
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-PSTN media mediaIpInterfaceGroupName UNTRUSTED
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-PSTN ingressIpPrefix 10.35.8.146 32
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-PSTN mode inService
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-PSTN state enabled
commit
#RDN trunk
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS-RDN policy callRouting
elementRoutingPriority TG_ERP
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS-RDN policy signaling ipSignalingProfile
ACCESS_IPSP
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS-RDN signaling registration
requireRegistration required
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS-RDN media mediaIpInterfaceGroupName
UNTRUSTED
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS-RDN ingressIpPrefix 0.0.0.0 0
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS-RDN mode inService
set addressContext default zone TRUSTED sipTrunkGroup TWO-WAY-SIP-GENESYS-RDN state enabled
commit
#Global Configuration
#Route Labels
set global callRouting routingLabel TO_TWO_WAY_GENESYS routingLabelRoute 0 routeType trunkGroup
set global callRouting routingLabel TO_TWO_WAY_GENESYS routingLabelRoute 0 trunkGroup TWO-WAY-SIP-GENESYS
set global callRouting routingLabel TO_TWO_WAY_GENESYS routingLabelRoute 0 ipPeer GENESYS
commit
set global callRouting routingLabel TO_TWO_WAY_PSTN routingLabelRoute 0 routeType trunkGroup
set global callRouting routingLabel TO_TWO_WAY_PSTN routingLabelRoute 0 trunkGroup TWO-WAY-SIP-PSTN
set global callRouting routingLabel TO_TWO_WAY_PSTN routingLabelRoute 0 ipPeer PSTN
commit
#Routes
set global callRouting route trunkGroup TWO-WAY-SIP-GENESYS LITTLE standard Sonus_NULL Sonus_NULL all all ALL none
Sonus_NULL routingLabel TO_TWO_WAY_PSTN
set global callRouting route trunkGroup TWO-WAY-SIP-GENESYS-RDN LITTLE standard Sonus_NULL Sonus_NULL all all ALL
none Sonus_NULL routingLabel TO_TWO_WAY_GENESYS
set global callRouting route trunkGroup TWO-WAY-SIP-PSTN LITTLE standard Sonus_NULL Sonus_NULL all all ALL none
Sonus_NULL routingLabel TO_TWO_WAY_GENESYS
commit
exit

```

Test Results

Table 2: Test Results

S.No	Procedure	Observation	Result	Comment
Inbound calls				
1	Inbound Call to Agent released by caller		Pass	
2	Inbound Call to Agent released by agent		Pass	
3	Inbound Calls rejected		Pass	
4	Inbound Call abandoned		Pass	
5	Inbound Call to Route Point with Treatment		Pass	
6	Interruptible Treatment		Pass	
7	IVR (Collect Digit) Treatment		Pass	
8	Inbound Call routed by using 302 out of SIP Server signaling path		Pass	

Outbound Calls				
9	1PCC Outbound Call from SIP Endpoint to external destination		Pass	
10	3PCC Outbound Call to external destination		Pass	
11	1PCC Outbound Call Abandoned		Pass	
Music on hold				
12	Caller is put on hold and retrieved by using RFC 2543 method		Pass	
13	T-Lib-Initiated Hold/Retrieve Call with MOH using RFC 3264 method		Pass	
Call Transfers				
14	3PCC 2 Step Transfer to internal destination by using re-INVITE method		Pass	
15	3PCC Alternate from consult call to main call		Pass	
16	1PCC Unattended (Blind) transfer using REFER		Pass	
17	1PCC Attended Transfer to external destination		Pass	
18	3PCC Two Step Conference to external party		Pass	
19	3PCC (same as 1PCC) Single-Step Transfer to another agent		Pass	
20	3PCC Single Step Transfer to external destination using REFER		Pass	
21	3PCC Single Step Transfer to internal busy destination using REFER		Pass	
Other scenarios				
22	Early Media for Inbound Call to Route Point with Treatment		Pass	
23	Early Media for Inbound Call with Early Media for Routed to Agent		Pass	
24	Inbound call routed outbound (Remote Agent) using INVITE without SDP		Pass	
25	Call Progress Detection		Pass	
26	Out of Service detection. Checking MGW live status		Pass	
27	SIP Authentication for outbound calls		Not Supported	
28	SIP Authentication for incoming calls		Pass	
Remote Agent behind SBC scenarios				
29	T-Lib-Initiated Answer/Hold/Retrieve Call for Remote SIP endpoint which supports the BroadSoft SIP Extension Event Package		Pass	
30	3PCC Outbound Call from Remote SIP endpoint to external destination		Pass	
31	3PCC 2 Step Transfer from Remote SIP endpoint to internal destination		Pass	
32	1PCC Attended Transfer from Remote SIP endpoint to external destination		Pass	

Conclusion

These Application Notes describe the configuration steps required for Sonus SBC 5XX0 to successfully interoperate with Genesys Voice Platform. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in [Test Results](#).

Appendix A

The following new configurations are included in this section:

1. [SIP Server standard configuration](#)
2. [DN Configuration](#)

3. EpiPhone configuration

1. SIP Server standard configuration

Information below displays how objects are configured by default.

Table 3: SIP Server standard configuration

SIP Server Application CME Options TServer section
SIP-hold-rfc3264=true
router-timeout=30
default-dn=
blind-transfer-enabled=true
resource-management-by-rm=true
msml-support=true
sip-enable-moh=true

2. DN Configuration

Table 4: DN Configuration

Name	Number	Name in CME	CME Options TServer section	Comment
MGW-TRUNK	MGW-TRUNK	MGW-TRUNK	refer-enabled=true contact=<TSE_CONTACT> oos-check=10 oos-force=5 oosp-transfer-enabled=true sip-replaces-mode=2	TSE
Ext-DN1	21001	N/A	N/A	
Ext-DN2	21002			
SIP-DN1	7101	7101	refer-enabled=false	
SIP-DN2	7102	7102	ring-tone-on-make-call=false make-call-rfc3725-flow=1 contact=*	
SIP-RDN	7200	7200	refer-enabled=true ring-tone-on-make-call=false make-call-rfc3725-flow=1 contact=*	SIP endpoint which supports the BroadSoft SIP Extension Event Package.
SIP-UNKN	7777	N/A	N/A	
RP	5000	5000		
RP1	5001	5001		
RP2	5002	5002		
SVC_MSML	SVC_MSML	SVC_MSML	prefix=msml=	MS

			contact=<MS_CONTACT> service-type=msml subscription-id= Environment	
--	--	--	---	--

3. EpiPhone configuration

Content of configuration file esttt.conf:

Table 5: EpiPhone configuration

```
[TcCM]

site1 = UTE_HOME

connect-on-startup = true

open-log-on-startup = false

log-to-file = epi-phone.log

#-----

[UTE_HOME]

server = (host=<SIP_SERVER_HOST_IP>,port=<SIP_SERVER_TLIB_PORT>)

sip-register = false

dn1 = 7101,name="Alice",mkcall="7102"

dn2 = 7102,name="Bob"

dn3 = 7200,name="John"

dn4 = 5000,pool="shared"

dn5=5001,pool="shared",script="annc=(PROMPT=("\1"=(INTERRUPTABLE=1,ID=1)))"

dn6=5002,pool="shared",script="collect=(MAX_DIGITS=4,RESET_DIGITS=11,BACKSPACE_DIGITS=22,TOTAL_TIMEOUT=100)"
```