

Configuring the Sonus SBC 1000/2000 with Genesys 8.1

Application Notes Rev. 0.1

Last Updated: Jan 27, 2015

Sonus Equipment	Туре	Version		
SBC 1000	SBC 1000	4.1.0 Build 369		

3rd Party Equipment	Туре	Version
Genesys	SIP Server	8.1.100.98
Genesys	GVP_MCP	8.1.504.93
Polycom	500	S6.2119

Copyright © 2014, Sonus and/or its affiliates. All rights reserved.

This document may not be reproduced or transmitted in any form or by any means, electronic or mechanical, for any purpose, without the prior written permission of Sonus.

Table of Contents

1	Do	ocument Overview	4
1.1	C	Dverview	4
2	Int	troduction	5
2.1	A	Audience	5
2.2	F	Requirements	5
2.3	F	Reference Configuration	6
2.3.1		Network Topology	6
3	Сс	onfiguring Sonus SBC 1000 and SBC 2000 Series	7
3.1	S	BC Configuration Diagram	7
3.2	S	BC Default Profiles	8
3.2.1		Default SIP Profile	8
3.2.2		Default Voice Codec Profile	9
3.2.3		Default Media Profile	. 10
3.3	E	External Peer Side SBC Configuration	. 11
3.3.1		Node Interfaces	. 11
3.3.	1.1	Node Ports	. 11
3.3.	1.2	Node Interfaces	
3.3.2		PSTN-Ext-DN's	
3.3.	2.1	SIP Signaling Group	
3.3.	2.2	Call Routing Table	
3.3.		Transformation Tables	
3.3.	2.4	SIP Server Table	
3.3.3		Remote DN's	
3.3.		SIP Signaling Group	
3.3.		Call Routing Table	
3.3.		Transformation Table	
3.3.		SIP Server Table	
3.4	Ir	nternal Side SBC configuration	
3.4.1		Node Interface Ports	
3.4.		Node Ports	
3.4.	1.2	Node Interfaces	
3.4.2		Signaling Group	
3.4.3		Call Routing Table	. 23

SIP Call Routing Table to PSTN	
Call Routing Table To RDN	
Transformation Table	
Passthrough Untouched	
To Remote DN	
Local/Pass-thru Auth Table	
SIP Server Table	
Remote Authorization Table	
enesys configuration	
Accessing Genesys Tools and Interfaces	
Creating SIP Switch in Genesys Administrator	
SIP Server Configuration in Genesys Administrator	
Genesys Media Server Deployment	
Stat Server Configuration	
Universal Routing Configuration in Genesys Administrator	
URS Routing Strategies	
Strategy #1 - Route Call to Available Agent	
Strategy #2 - Play Announcement and Route to Available Agent .	
Strategy #3 – Play Announcement and Collect Seven Digits	
Strategy #4 – Route to External SIP Carrier Number	
Strategy #5 – Route to External SIP Carrier Number	51
xceptions	53
SBC1000/2000 Exceptions	
Call hold using RFC 2543 method	
dix A	
ver and DN configuration	
er standard configuration	
ard configuration	
er and DN non-standard configuration per test case	
VER: sip-hold-rfc3264=false	
ne configuration	
	Call Routing Table To RDN Transformation Table Passthrough Untouched To Remote DN Local/Pass-thru Auth Table SIP Server Table Remote Authorization Table. enesys configuration Accessing Genesys Tools and Interfaces Creating SIP Switch in Genesys Administrator SIP Server Configuration in Genesys Administrator Genesys Media Server Deployment Stat Server Configuration in Genesys Administrator Universal Routing Configuration in Genesys Administrator URS Routing Strategies Strategy #1 - Route Call to Available Agent Strategy #3 – Play Announcement and Route to Available Agent Strategy #4 – Route to External SIP Carrier Number Strategy #5 – Route to External SIP Carrier Number SBC1000/2000 Exceptions Call hold using RFC 2543 method dix A ver and DN configuration er and DN non-standard configuration per test case VER: sip-hold-rfc3264=false

1 Document Overview

These Application Notes describe the configuration steps required for the Sonus Session Border Controller (SBC) 1000 and SBC 2000 to interoperate with the Genesys 8.1 system and a SIP trunk group to PSTN.

The objective of the document is to describe the configuration procedures to be followed during interoperability testing of SBC 1000 and SBC 2000 with a Genesys system and T1 trunk group to PSTN.

For additional information on Sonus SBC 1000 and SBC 2000 series, visit http://www.sonus.net

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

1.1 Overview

The Sonus SBC 1000 and SBC 2000 session border controllers have been designed to use the same application software, boot image and Survivable Branch Appliance software. They differ in the number of physical Ethernet connections and processing power but are otherwise viewed from a software standpoint as being the same. With this in mind, this particular effort was tested with an SBC 1000 but is fully applicable to an SBC 2000.

2 Introduction

This document provides a configuration guide for Sonus SBC 1000 Series (Session Border Controller) when connecting to a SIP PSTN trunk group and a Genesys 8.1 PBX.

The Sonus SBC 1000 and SBC 2000 are Session Border Controllers that connects disparate SIP trunks, SIP PBXs, and communication applications within an enterprise. The SBC can also be used as a SIP routing and integration engine.

The Sonus SBC is the point of connection between the SIP trunk group to PSTN and the Genesys PBX.

2.1 Audience

This technical document is intended for telecommunication engineers with the purpose of configuring the Sonus SBC 1000 and SBC 2000 and aspects of the SIP trunk group together with Genesys 8.1 product. There will be steps that require navigating the third-party and Sonus SBC Command Line Interface (CLI). Understanding the basic concepts of IP/Routing and SIP/RTP is 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.

Technical support on SBC 1000 and SBC 2000 can be obtained through the following:

- Phone: +1 888-391-3434 (Toll-free) or +1 978-614-8589 (Direct)
- Web: http://www.sonus.net/company/maintenance/log-trouble-tickets

2.2 Requirements

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

Sonus Equipment	Туре	Version	
SBC 1000	SBC 1000	4.1.0 Build 369	
3rd Party Equipment	Туре	Version	
Genesys	SIP Server	8.1.100.98	
Genesys	GVP MCP	8.1.504.93	
Polycom SoundPoint IP 501 SIP	SIP Phone	2.1.3	

While the SBC 2000 was not tested, at the same time the results obtained for SBC 1000 would be seem with the SBC 2000 as they feature common code base.

2.3 Reference Configuration

A simulated enterprise site consisting of a Genesys 8.1 PBX and a SIP trunk group to PSTN to connect to the SBC 1000. The SBC 1000 was running software version 4.1.0 Build 396 during testing.

2.3.1 Network Topology



Figure 1: Network Topology

The figure above represents the equipment used for the integration and certification testing. The SBC 1000 is used to route and facilitate calls between the PSTN and the Genesys system.

The SBC 1000 under test has 2 Ethernet ports configured. For more information on Media port deployment options or other network connectivity queries, refer to the SBC 1000 Network Deployment Guide or contact your local Sales team for information regarding the Sonus Network Design professional services offerings.

3 Configuring Sonus SBC 1000 and SBC 2000 Series

3.1 SBC Configuration Diagram



Figure 2: SBC 1000 SIP Trunk Diagram

3.2 SBC Default Profiles

3.2.1 Default SIP Profile

SIP Profiles control the how the Sonus SBC 1000/2000 communicates with SIP devices. They control important characteristics such as: session timers, SIP header customization, SIP timers, MIME payloads, and option tags. Below is the default SIP profile used for the SBC 1000 for this testing effort.

Description Default SIP Profile	
Session Timer	MIME Payloads
Session Timer Disable	ELIN Identifier LOC PIDF-LO Passthrough Enable Unknown Subtype Passthrough Disable
Header Customization	Options Tags
UA Header Sonus SBC Subscription State Passthrough Enable FQDN in From Header Disable Send Assert Header Trusted Only Trusted Interface Enable Calling Info Source RFC Standard Diversion Header Selection Last	100rel Supported Update Supported
Timers	SDP Customization
Transport Timeout Timer 5000 Maximum Retransmissions RFC Standard	Send Number of Audio Channels True Connection Info in Media Section True Origin Field Username SBC
RFC timers	Session Name VoipCall
Timer T1 500 Timer T2 4000 Timer T4 5000 Timer D 32000 Timer B 32000 ms Timer F 32000 ms Timer H 32000 ms (64*TimerT1) Timer J 32000 ms (64*TimerT1)	

Figure 3: SIP Profile

3.2.2 Default Voice Codec Profile

Below are the default voice codec profiles used for the SBC 1000 in this testing effort.

Voice Codec Configuration				
Description Default G711A				
Codec	G.711 A-Law			
Payload Size	20			

Figure 4: G.711A Codec Profile

Voice Codec Configuration				
Description	Default G711u			
Codec	G.711 μ-Law			
Payload Size	20			

Figure 5: G.711U Codec Profile

3.2.3 Default Media Profile

Media Profiles allow you to specify the individual voice and fax compression codecs and their associated settings, for inclusion in a Media List. Different codecs provide varying levels of compression, allowing one to reduce bandwidth requirements at the expense of voice quality. Below is the default media profile used for the SBC 1000 and is for reference only.

[
Description	Default Media List Default G711A Default G711u
Media Profiles List	*
Crypto Profile ID	None
Media DSCP	46
RTCP Mode	RTCP
Dead Call Detection	Disabled
Silence Suppression	Enabled
Gain Control	Digit Relay
Receive Gain 0	Digit (DTMF) Relay Type RFC 2833
Transmit Gain 0	Digit Relay Payload Type 101
Pa	assthrough/Tone Detection
Modem Passthrough	Enabled
Fax Passthrough	Enabled
CNG Tone Detection	Disabled

Figure 6: Default Media List

3.3 External Peer Side SBC Configuration

3.3.1 Node Interfaces

The Sonus SBC 1000 allows you to configure the Identification information, Physical Data Layer, and Networking Layer for the Ethernet ports. If you want to change the IP Address, you must configure the associated Logical Interface or use the Modify Ethernet IP task found under the Tasks tab.

Below are the settings for the Ethernet connection between the Sonus SBC 1000 and trunks acting as PSTN.

3.3.1.1 Node Ports



Figure 7 : Node Port

3.3.1.2 Node Interfaces

		dentification/Status		
Interface Name	Ethernet 1 IP			
I/F Index	2			
Alias				
Description				
Admin State	Enabled			
		Networking		
	MAC Address	00:10:23:e0:01:47	ACL In	None
	IP Address	10.35.177.226	ACL Out	None
	IP Netmask	255.255.255.192	ACL Forward	None
	IP Assign Method	Static		
	Primary Address	10.35.177.226		
	Primary Netmask	255.255.255.192		
Configure Se	condary Interface	Disabled		
s	econdary Address			
	condary Netmask			



3.3.2 PSTN-Ext-DN's

3.3.2.1 SIP Signaling Group

Signaling groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which Call Routes are selected. They are also the location from which Tone Tables and Action Sets are selected. In the case of SIP, they specify protocol settings and link to server, media and mapping tables.

Description PSTN-Ext-DN1 Admin State Enabled Service Status Up

	SIP Channel	s and Routing			
			Media	Informatio	on
	Action Set Table	None			
0	Call Routing Table	Calls From PSTN-Ext-DNs	Audio/Fax Stream I	Proxy Mode	Enabled
	No. of Channels	60	Audio/Fax Stream	DSP Mode	Enabled
	SIP Profile	Default SIP Profile	Video/Application Stream I	Proxy Mode	Disabled
	SIP Mode	Basic Call	м	edia List ID	Default Media List
	Agent Type	Back-to-Back User Agent	Pla	y Ringback	Auto
	SIP Server Table	PSTN-Ext-DN1		Tone Table	Default Tone Table
	Load Balancing	Round Robin		Early 183	Disable
	Channel Hunting	Most Idle	Mu	isic on Hold	Disabled
Notify	Lync CAC Profile	Disable			
C	hallenge Request	Disable	Mapr	ing Tables	
	Outbound Proxy		mapp	ing rubice	
Out	bound Proxy Port	5060	SIP To Q.850 Overri	de Table [Default (RFC4497)
o Channel A	vailable Override	34: No Circuit/Channel Available	Q.850 To SIP Overri	de Table [Default (RFC4497)
Call Setup	o Response Timer	255	Pass-thru Peer SIP Response Code Enable		
			SIP	IP Details	
			NAT Traversal	None	
			Signaling/Media Source IP	Ethernet 1	IP (10.35.177.226)
			Signaling DSCP	40	
	Liste	n Ports	Federa	ated IP/FQ	ON
Total 2 SIP L	isten Port Rows		Total 1 SIP Federated IP Row	_	_
Port	Protocol	TLS Profile ID	IP/FQDN	Netmas	k
5060	UDP	N/A	10.220.250.55	255.255	.255.255
5060	тср	N/A			

Figure 9: SIP Signaling Group to PSTN-Ext-DN'

3.3.2.2 Call Routing Table

Call Routing allows calls to be carried between signalling groups, thus allowing calls to be carried between ports, and between protocols (like ISDN to SIP). Routes are defined by Call Routing Tables, which allow for flexible configuration of which calls are carried, and how they are translated. These tables are one of the central connection points of the system, linking Transformation Tables, Message translations, Cause Code Reroute, Tables, Media Lists and the three types of Signaling Groups (ISDN, SIP and CAS).

	Route	e Details			
Description Admin State Enabled Route Priority 1 Call Priority Normal Number/Name Transformation Table Match 10 digits to Genesys					
	Destinatio	n Information			
Destination Type Normal Message Translation Table None Cause Code Reroutes None Cancel Others upon Forwarding Disabled Fork Call No Destination Signaling Groups (SIP) To/From Genesys *					
Media		Quality of Service			
Audio/Fax Stream Mode Video/Application Stream Mode Media Transcoding Media List	DSP Disabled Enabled None	Quality Metrics Number of Calls Quality Metrics Time Before Retry Min. ASR Threshold Enable Max. R/T Delay Max. R/T Delay Enable Max. Jitter Max. Jitter	10 10 0 Enabled 65535 Enabled 3000		

Figure 10: Call Routing Table

3.3.2.3 Transformation Tables

Transformation Tables facilitate the conversion of names, numbers and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number, or into a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table, and they are selected from there. In addition, Transformation tables will be configurable as a reusable pool that Action Sets can reference.



Figure 11: Transformation Table matching Genesys extensions



Figure 12: Transformation Table Matching all

3.3.2.4 SIP Server Table

SIP Server Tables contain information about the SIP devices connected to the Sonus SBC 1000/2000. The entries in the tables provide information about the IP Addresses, ports, and protocols used to communicate with each server. The Table Entries also contain links to counters that are useful for troubleshooting.

Server Host	Transport
Server Lookup IP/FQDN Priority 1 Host 10.220.250.55 Port 5060 Protocol UDP	Monitor None
Remote Authorization and Contacts	
Remote Authorization Table None Contact Registrant Table None	

Figure 13: SIP Server Table

3.3.3 Remote DN's

3.3.3.1 SIP Signaling Group

Signaling groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which Call Routes are selected. They are also the location from which Tone Tables and Action Sets are selected. In the case of SIP, they specify protocol settings and link to server, media and mapping tables.

Description Genesys-RDN Admin State Enabled Service Status Up

SIP Channel	s and Routing	Media Information
Action Set Table	None	Media Information
Call Routing Table	Calls from Genesys RDN	Audio/Fax Stream Proxy Mode Enabled
No. of Channels	60	Audio/Fax Stream DSP Mode Enabled
SIP Profile	Default SIP Profile	Video/Application Stream Proxy Mode Disabled
SIP Mode	Local Registrar	Media List ID Default Media List
Registrar	Local Registrar #2	Play Ringback Auto
Agent Type	Access Mode	Tone Table Default Tone Table
Interop Mode	BroadSoft Extension	Early 183 Disable
Load Balancing	Round Robin	Music on Hold Disabled
Channel Hunting	Most Idle	
Notify Lync CAC Profile	Disable	Manning Tables
Outbound Proxy		Mapping Tables
Outbound Proxy Port		SIP To Q.850 Override Table Default (RFC4497)
Io Channel Available Override	34: No Circuit/Channel Available	Q.850 To SIP Override Table Default (RFC4497)
Call Setup Response Timer	255	Pass-thru Peer SIP Response Code Enable
		· · · · · · · · · · · · · · · · · · ·
		SIP IP Details
		NAT Traversal None
		Signaling/Media Source IP Ethernet 1 IP (10.35.177.226)
		Signaling DSCP 40
Liste	n Ports	Federated IP/FQDN
Total 2 SIP Listen Port Rows		Total 1 SIP Federated IP Row
Port Protocol	TLS Profile ID	IP/FQDN Netmask
5060 UDP	N/A	10.220.250.0 255.255.255.0
5060 TCP	N/A	

Message Manipulation Disabled

Figure 14: SIP Signaling Group to Remote DN's

3.3.3.2 Call Routing Table

Call Routing allows calls to be carried between signalling groups, thus allowing calls to be carried between ports, and between protocols (like ISDN to SIP). Routes are defined by Call Routing Tables, which allow for flexible configuration of which calls are carried, and how they are translated. These tables are one of the central connection points of the system, linking Transformation Tables, Message translations, Cause Code Reroute, Tables, Media Lists and the three types of Signaling Groups (ISDN, SIP and CAS).



Figure 15: Call Routing Table

3.3.3.3 Transformation Table

Transformation Tables facilitate the conversion of names, numbers and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number, or into a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table, and they are selected from there. In addition, Transformation tables will be configurable as a reusable pool that Action Sets can reference.

Description Match 10 digits to Genesys Admin State Enabled	
Match Type Optional	
Input Field	Output Field

Figure 16: Transformation Table matching Genesys extensions



Figure 17: Transformation Table matching all

3.3.3.4 SIP Server Table

SIP Server Tables contain information about the SIP devices connected to the Sonus SBC 1000/2000. The entries in the tables provide information about the IP Addresses, ports, and protocols used to communicate with each server. The Table Entries also contain links to counters that are useful for troubleshooting.

Server Host	Transport
Server Lookup IP/FQDN Priority 1 Host 0.0.0.0 Port 5060 Protocol UDP	Monitor None
Remote Authorization and Contacts	
Remote Authorization Table Authorization Genesys Contact Registrant Table None	

Figure 18: SIP Server Table

3.4 Internal Side SBC configuration

3.4.1 Node Interface Ports

The Sonus SBC 1000 allows you to configure the Identification information, Physical Data Layer, and Networking Layer for the Ethernet ports. If you want to change the IP Address, you must configure the associated Logical Interface or use the Modify Ethernet IP task found under the Tasks tab.

Below are the settings for the Ethernet connection between the Sonus SBC 1000 and trunks acting as Internal.

Identificatio	on/Status		Networking
Primary Key Port ID Hardware Type I/F Index Port Alias Description Admin State Operational Status Up/Down Since	Ethernet 1 Ethernet 22 Enabled Up	ACL In ACL Forward Frame Type Default Untagged VLAN Tagged VLANs	None All Ethernet 1 VLAN
Physical/Da	ta Layer		Spanning Tree
Configured Spee Negotiated Spee Configured Duplexit Negotiated Duplexit	d 1000 Mbps y Auto	MSTP State Protocol BPDU Version Ra Protocol BPDU Version Ta	x None

Figure 19: Internal Node Interface Port

3.4.1.2 Node Interfaces



Figure 20: Internal Node Logical Interfaces

3.4.2 Signaling Group

Signaling groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. They are the entity to which calls are routed, as well as the location from which Call Routes are selected. They are also the location from which Tone Tables and Action Sets are selected. In the case of SIP, they specify protocol settings and link to server, media and mapping tables.



Figure 21: Signaling Group

3.4.3 Call Routing Table

Call Routing allows calls to be carried between signalling groups, thus allowing calls to be carried between ports, and between protocols (like ISDN to SIP). Routes are defined by Call Routing Tables, which allow for flexible configuration of which calls are carried, and how they are translated. These tables are one of the central connection points of the system, linking Transformation Tables, Message translations, Cause Code Reroute, Tables, Media Lists and the three types of Signaling Groups (ISDN, SIP and CAS).

3.4.3.1 SIP Call Routing Table to PSTN

	Rout	e Details			
Description Admin State Enabled Route Priority 1 Call Priority Normal Number/Name Transformation Table Passthrough Untouched					
	Destinatio	n Information			
Destination Type Message Translation Table Cause Code Reroutes Cancel Others upon Forwarding Fork Call Destination Signaling Groups	Normal None Disabled No (SIP) PSTN-Ext	-DN1 *			
Media		Quality of Service			
Audio/Fax Stream Mode Video/Application Stream Mode Media Transcoding Media List	DSP Disabled Enabled None	Quality Metrics Number of Calls Quality Metrics Time Before Retry Min. ASR Threshold Enable Max. R/T Delay Max. R/T Delay Enable Max. Jitter Max. Jitter	10 10 0 Enabled 65535 Enabled 3000		

Figure 22: SIP Call Routing Table To PSTN

3.4.3.2 Call Routing Table To RDN

	Route	e Details	
Descrip Admin S Route Pric Call Pric Number/Name Transformation Ta	tate Enabled prity 1 prity Normal	s-RDN	
	Destination	n Information	
Destination Type Message Translation Table Cause Code Reroutes Cancel Others upon Forwarding Fork Call Destination Signaling Groups	Normal None Disabled No (SIP) Genesys-F	RDN *	
Media		Quality of Service	
Audio/Fax Stream Mode Video/Application Stream Mode Media Transcoding Media List	DSP Disabled Enabled None	Quality Metrics Number of Calls Quality Metrics Time Before Retry Min. ASR Threshold Enable Max. R/T Delay Max. R/T Delay Enable Max. Jitter Max. Jitter	10 10 0 Enabled 65535 Enabled 3000

Figure 23: ISDN Call Routing Table To/From PSTN

3.4.4 Transformation Table

Transformation Tables facilitate the conversion of names, numbers and other fields when routing a call. They can, for example, convert a public PSTN number into a private extension number, or into a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table, and they are selected from there. In addition, Transformation tables will be configurable as a reusable pool that Action Sets can reference.

3.4.4.1 **Passthrough Untouched**



Figure 24: Transformation Table to match all

3.4.4.2 To Remote DN



Figure 25: Transformation Table to Remote DN

3.4.5 Local/Pass-thru Auth Table

Local Pass-through Tables contain entries with information about SIP endpoints, The Sonus SBC 1000/2000 uses this information to challenge SIP request messages such as REGISTER. It is used in the SIP Signaling Group when the Challenge Request is enabled.

Autho	rization Parameters
Type of Address of Record	Remote
Address of Record URI	sip:2221234567@10.35.177.226:5060
User Name Password Setting	uthentication genesys Use Current

Figure 26: Local/Pass-thru table

3.4.6 SIP Server Table

SIP Server Tables contain information about the SIP devices connected to the Sonus SBC 1000/2000. The entries in the tables provide information about the IP Addresses, ports, and protocols used to communicate with each server. The Table Entries also contain links to counters that are useful for troubleshooting.

	Server Host	Transport
Server Lookup Priority Host Port Protocol	IP/FQDN 1 10.35.176.111 5060 UDP	Monitor None
Remote Authoriz	Authorization and Contacts zation Table Authorization Genesys strant Table None	

Figure 3: SIP Server Table

3.4.7 Remote Authorization Table

Remote Authorization Tables and their entries contain information used to respond to request message challenges by an upstream server. The Remote Authorization tables defined in this page appear as options in the Remote Authorization and Contacts Panel for SIP Servers.



Figure 28: Remote Authorization Table

4 Genesys configuration

This section provides the configuration required for the Genesys components.

Accessing Genesys Tools and Interfaces 4.1

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 4: 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.

🎾 Media 📃 🗾		🗋 • 📖 • 💝 🔎		
I Folders	Contents of '/Configuration/Environ	ment/Applications/Media'		
Configuration	Name 👚	Туре	Version	Server
Environment	Enter text here	Enter text here	Y Enter te Y	Enter t
Conditions	SIPServer1	T-Server	8.1.100.98	True
Application Templates	SIPServer2	T-Server	8.1.100.98	True
	Sections		Annex Security Depe	
GVP Media Routing The GVP_Group1_LRG GVP_Unassigned Hosts	Name A	<u> </u>		
GVP G Media G Routing C GVP_MCP_Group1_LRG G GVP_Unassigned Hosts Solutions	Sections Name Enter text here w agent-reservation	Value		0>
GVP GVP Media Routing C GVP_MCP_Group1_LRG GVP_Unassigned Hosts Solutions Solutions Switching Offices	Sections Name Enter text here agent-reservation backup-sync	Value		0>
GVP G Media G Routing C GVP_MCP_Group1_LRG G GVP_Unassigned Hosts Solutions	Sections Name Enter text here agent-reservation backup-sync w call-cleanup	Value		0)
GVP GVP Media Routing Comparison GVP_MCP_Group1_LRG GVP_Unassigned Hosts Solutions Solutions Switching Offices	Sections Name Enter text here agent-reservation backup-sync call-cleanup extrouter	Value		0)
GVP Media Routing GVP_Group1_LRG GVP_Unassigned Hosts Solutions Solutions Switching Offices	Sections Name Enter text here Sections Sectors agent-reservation Sector-sync Sector-sync	Value		0)
GVP Media Routing GVP_Group1_LRG GVP_Unassigned Hosts Solutions Solutions Switching Offices	Sections Name Enter text here agent-reservation backup-sync call-cleanup sextrouter sectores icense ink-control	Value		0)
GVP Media Routing GVP_Group1_LRG GVP_Unassigned Hosts Solutions Solutions Switching Offices	Sections Name Enter text here Sections Sectors agent-reservation Sector-sync Sector-sync	Value		0

Figure 5: 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.

Server TServer	💌 🦻 🗋	🗙 🖾 🔊 🕼 🕸
Name A	Value	
Enter text here	Enter text here	7
abs accept-dn-type	"+extension +position +acdqueue +route	dn +trunk +routequeue"
abc acw-in-idle-force-ready	"true"	
abc after-routing-timeout	"10"	
abs agent-emu-login-on-call	"Talse"	
abs agent-group		
abc agent-logout-on-unreg	"false"	
abc agent-logout-reassoc	"false"	
abc agent-no-answer-action	"none"	
abc agent-no-answer-overflow		
abc agent-no-answer-timeout	"15"	
abc agent-only-private-calls	"false"	
abs agent-strict-id	"Talse"	

Figure 6: 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.

File Edit View Tools Help							
(Routing Design)	Strategies						
	Name	 Description 	Ready	Loaded	Access	Trace	
	7						
<u> <u> </u></u>	⊡				RHD		
Strategies	announ_route_to_agent_v2		×	×	RHD		
		per_v2	×	×	RHD		
<u>⊷</u>			x		RHD		
Subroutines	Provide_to_agent		×	×	RHD		
Subroutines			×		RHD		
<u>s</u>			×		RHD		
<u>o</u>	route to external anno		×		RHD		
Routing Rules	route_to_external_number_v2		×	×	RHD		
	www.selection_by_dtmf_v2		×	×	RHD		
2							
Business Rules							
Attributes							
88							
	Location file:C:\Genesys\GCTI\IRD\route	to external number v2 rbn					

Figure 7: 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.

Senesys Genesys Administrator										
MONITORING PROVISIONING DEPLOYMENT OPERATIONS										
PROVISIONING > Switching > 9	PROVISIONING > Switching > Switches									
Navigation	Navigation 🛞 Switches									
潯 Search	+	💼 🔻 🗊 New 🍐 New Folder 📝 Edit 🙀 Remove 🔂 Change state 🗟 Move to								
Environment	+	Name 🔺 🔻 Switch Type								
a Switching		T Filter								
DN Groups		View: Root > C Switches								
Places		SIPSwitch1 SIP Switch								
Place Groups										
Switching Offices										
Switches										
IVRs										
I										

Figure 8: 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.

Senesys	G	enes	ys Adm	inistra	for				
	DEP	LOYMENT	OPERATIONS						
PROVISIONING > Switching >	Switches	> SIPSwi	itch1						
Navigation	«	SIPSwitch	n1 - \Switches\						
潯 Search	+	Cancel 🚽	Save & Close 🛃	Save 🛃 Save	& New 🛛 🛃 R	eload			
🙀 Environment	+ (Configuration	Option	IS	Permissions	2	Dependencies	Agent Logins	DNs
潯 Switching	Ξ.	New	💁 New Fold	er 📝 Edit 🛛	Remove	Change	state 📑 Move to		\smile
DN Groups		Number -			~	Туре	_		State
Places	T	Filter				Filter			Filter
Place Groups	Vi	ew: 🔄 SIP	Switch1 > 🛅	DNs					
Switching Offices		DialPlans							Enabled
		Extensions							Enabled
		GVP Exten	sions						Enabled
IVRs		RoutePoint	s						Enabled
		Trunks							Enabled

Figure 9: Genesys Administrator - DNs

4.2 Creating SIP Switch in Genesys Administrator

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

	Genesys /	Administrat	for	Tenant: Environment	P Ner	w Window Log out 💮					
MONITORING PROVISIONING	DEPLOYMENT OPER	ATIONS									
ROVISIONING > Switching > Swit	ching Offices > SIPser	ver									
lavigation 《	SIPserver - \Sw	itching Offices\									
🗟 Search 🛛 🛨	+ 💢 Cancel 🚽 Save & Close 🚽 Save 🚽 Save & New 🛛 🔀 Reload										
🗟 Environment 🛛 🛨	Configuration	Options	Permissions	Dependencies							
🗟 Switching 📃											
📑 DN Groups	* Name:	SIPserver									
📑 Places	* Switch Type:	SIP Switch				~					
Place Groups	State:	I Enabled									
Switching Offices											
Switches											
IVRs											
Routing/eServices +											
Desktop											
Accounts +	-										
Voice Platform +											
Outbound Contact											

Figure 10: 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.

Senesys		Genesys Administrator New Window Log out									
MONITORING PROVIS		DEPLOYMENT OPERA	TIONS								
PROVISIONING > Switchi	ing > Swit	ches > SIPSwitch1									
Navigation	«	SIPSwitch1 - \Sw	vitches\								
😛 Search 🕘 🔀 Cancel 🙀 Save & Close 🚽 Save & New 🔀 Reload											
😝 Environment	+	Configuration	Options	Permissions	Dependencies	Agent Logins	DNs				
潯 Switching	Ξ	-						General Access Co			
DN Groups		- 🔺 * General									
Places		* Name:	SIPSwitch1					×			
Place Groups		* Switching Office:	SIPOffice					م ×			
Switching Offices		* Switch Type:	SIP Switch					~			
Switches		T-Server:	SIPServer1					Q ×			
🕞 IVRs		DN Range:									
		State:	Chabled								
		State.	M Enabled								
		Access Codes -									
Routing/eServices	+										
Desktop	+										
Accounts	+										
Voice Platform	(+) (+)										
潯 Outbound Contact	+										

Figure 11: 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.

Senesys	Genesys Administrator										
MONITORING PROVISIONING	DEPLOYMENT OPERATIONS										
PROVISIONING > Switching > Swit	itches > SIPSwitch1										
Navigation 🔍	SIPSwitch1 - \Switches\										
🔉 Search 🛨 🔀 Cancel 🚽 Save & Close 🚽 Save & New 🛛 🔀 Reload											
😝 Environment 🛛 🕂	Configuration Options Permissions Dependencies Agent Logins DNs										
Switching –	🕒 🔻 🕅 New 🔥 New Folder 🍞 Edit 🙀 Remove 📷 Change state 🔂 Move to	đb									
DN Groups	Number A Type State										
Places	T Filter Filter										
Place Groups	View: SIPSwitch1 > DNs > RoutePoints										
Switching Offices	▶ 1011 Routing Point Enabled										
Switches	International	Enabled									
Contraction of the second s	1013 Routing Point Enabled										
🕞 IVRs	1014 Routing Point Enabled										
	1015 Routing Point Enabled										
	8000 Routing Point Enabled										
Routing/eServices +											
Routing/eServices + Desktop + Accounts +											
😝 Accounts 🛛 🕂											
Solution Voice Platform											
🙀 Outbound Contact 🛛 🕂		Displaying objects 1 - 6 of 6									

Figure 12: 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.

1001 - \Switches\SIPSwitch1\DNs\Extensions\											
×	Cancel 🚽 Save & Close 🚽 Save 🚽	🚽 Save & New									
C	Configuration Options Permissions Dependencies										
	New 🙀 Delete 生 Export 🚡 Impo	ort	View: Advan	ced View (Annex)							
	Name 🔺	Section	Option	Value							
T	Filter	Filter	Filter	Filter							
81	lServer (4 Items)										
	TServer/contact	TServer	contact	ż							
	TServer/cpn	TServer	cpn	208604100	11						
	TServer/sip-cti-control	TServer	sip-cti-control	talk, hold							
	TServer/use-contact-as-dn	TServer	use-contact-as-dn	true							

Figure 13: 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 1000/2000 through which the SIP REGISTER message was received by SIP Server.
| | Bock-SBC-Priv - \Switches\SIPS | witch1\DNs | \Trunks\ | | |
|----|--------------------------------|--------------|----------------|-------------------|---|
| X | Cancel 🚽 Save & Close 🚽 Save 🍃 | 🚽 Save & New | | | |
| C | onfiguration Options | Per | missions | Dependencies | |
| | New 🙀 Delete 生 Export 🚡 Impo | ort | View: Advan | ced View (Annex) | ~ |
| | Name 🔺 | Section | Option | Value | |
| T | Filter | Filter | Filter | Filter | |
| 31 | [Server (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 | |

Figure 14: 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.

	MediaServer - \Switches\SIPSwitch1\DNs\Trunks\													
X	Cancel 🚽 Save & Close 🚽 Save 🚽	🚽 Save & New												
C	onfiguration Options	Perr	missions Depend	lencies										
	New 🙀 Delete 生 Export 🚡 Impo	ort	View: Advanced View	(Annex)										
e	Name 🔺	Section	Option	Value										
T	Filter	Filter	Filter	Filter										
81	l Server (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										

Figure 15: 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.

Configuration Optio	Born	nissions	pendencies
New 🙀 Delete 👱 Export	삼 Import	View: Advanced V	lew (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-rfc3725-f	ow TServer	make-call-rfc3725	. 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

Figure 16: 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.

Configuration	Options	Permissions	Dependencies		
	General Adv	anced Routing & Orch	estration Cost Based	d Routing De	efault DI
A General					
* Number:	1011				
* Type:	Routing Point			*	
* Switch:	SIPSwitch1			X X	
Association:					
* Register:	True			~	
State:	Enabled				0
Advanced —					
Alias:	1011_SIPSwite	h1			
0	1011_SIPSwitc Default	h1		¥]
Alias: * Route Type:				× م	
Alias: * Route Type: Group:	Default				\$
Alias: * Route Type: Group: Use Override:	Default [Unknown Gro				\$
Alias:	Default [Unknown Gro				\$
Alias: * Route Type: Group: Use Override: Override:	Default [Unknown Gro				\$

Figure 17: Genesys Administrator - Create DN of Type Routing Point

Orchestration Application:			Q
Routing Strategies:	📰 Add 🎡 Edit 🙀 Remov	/e	
	Routing Strategy 🔺	Router	
	route_to_agent	URS1	
	route_to_agent	URS2	

Figure 18: 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.

4.3 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.

Senesys	Genesys A	dministrat	or				New Window Log out	⇔ • @ •
MONITORING PROVISIONING	DEPLOYMENT OPERA	TIONS						_
PROVISIONING > Environment >	Applications > SIPServe	r1						
Navigation	SIPServer1 St	arted - Primary - \App	lications\Media\					
Search 🛨	🗙 Cancel 🛃 Save & C	lose 🛃 Save 🛃 Save 8	k New 📑 Reload 📷	Uninstall 📄 Start 🏾	Stop 🛃 Graceful Stop			
😝 Environment 📃	Configuration	Options	Permissions	Dependencies	Alarms	Logs		
Alarm Conditions	—(▼) * General					General Ser	ver Info Network Security	T-Server Info
Application Templates	— (▼) * Server Info —							
Hosts	- 💌 * Network Secu	rity						
Solutions	T-Server Info-							
Time Zones	Tenant:	Resources					× P	
Business Units/Sites	Switches:	🖬 Add 🎲 Edit	t Remove					
📑 Tenants		Name 🔺		Switch Type		State		
Table Access Points		SIPSwitch1		SIP Switch		Enabled		
📑 Formats								
🕞 Fields								
Switching +								
Routing/eServices +								
😝 Desktop 🕴								
Accounts +								
Voice Platform +								
Goutbound Contact								

Figure 19: Genesys Administrator - SIP Server Tenant

3. Add a connection to Message Server.

Senesys		Genesys A	dn	ninistrat	or							New Wi	ndow Log out	@ • @ •
MONITORING PROVISIONING	D	DEPLOYMENT OPERAT	IONS											
PROVISIONING > Environment	> Ap	plications > SIPServer	1											
Navigation	«	SIPServer1 Sta	rted -	Primary - \App	lications\M	ledia \								
🙀 Search	•	🔀 Cancel 🛃 Save & Cl	lose 🖌	Save 🚽 Save 8	New 🛛 😹 I	Reload 🛛 📷	Uninstall 🛛 📫 Sta	art 📓 Ste	op 🔣 Graceful Stop					
潯 Environment	Θ	Configuration	Optio	ons	Permission	ıs	Dependencies		Alarms	Logs				
Alarm Conditions	~										General	Server Info	Network Security	T-Server Info
🕞 Scripts		- * General												
Application Templates		* Name:		SIPServer1										
Applications		* Application Templa	ate:	TServer SIPPre	emise 811								× P	
		* Type:		T-Server									~	
Solutions		Version:		8.1.100.98										
📑 Time Zones		Server:		True										
Business Units/Sites		State:		Enabled										
🕞 Tenants		Connections:		Edit	Remove	e								
Table Access Points				Server 🔺		Connection P	rotocol Loc	al Timeou	ut Remote	Timeout		Trace Mode		
G Formats	~			MessageServer	1 a	addp	60		90		1	Trace Is Turn	ed Off	
🕞 Fields														
潯 Switching	+	- 💌 * Server Info												
😝 Routing/eServices	+	(1 ** ** 1 * *												
潯 Desktop	+	- → * Network Secu	rity											
😝 Accounts	+	- T-Server Info -												
Solution Voice Platform	+													
🥁 Outbound Contact	+													

Figure 20: 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 Serverbased 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.

4.4 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.

- Verify VoIP service DN of type=msml as specified in section <u>4.2, Creating SIP Switch in Genesys</u> <u>Administrator</u> 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** = <u>file://\$InstallationRoot\$</u> (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 4.7.

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 multitenant 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, <u>http://webserver/hello.wav</u>.
 - 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.

4.5 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.

Genesys	Genesys Ac	dministrat	or				New Window Lo	ig out	•	
MONITORING PROVISIONING	DEPLOYMENT OPERATIO	ONS								
PROVISIONING > Environment > A	Applications > StatServer1	i.								
Navigation 🔍	StatServer1 Star	ted - Primary - \App	lications\Routing\							
😝 Search 🛛 🛨	🔀 Cancel 🛃 Save & Clos	se 🚽 Save 🛃 Save 8	k New 📑 Reload 🙀	Uninstall 📫 Start 📓	Stop 🛃 Graceful Stop					
🕞 Environment 📃	Configuration	Options	Permissions	Dependencies	Alarms	Logs				
Alarm Conditions	- (▲) * General						General Server	Info	Network	Securi
Application Templates	* Name:	StatServer1						×	1	
Applications	* Application Template	: Stat Server 81	12					×P	A	
Hosts	* Type:	Stat Server						~		
Solutions	Version:	8.1.200.24								
📑 Time Zones	Server:	True								
Business Units/Sites	State:	C Enabled								
Tenants	Connections:	T Add 🌼 Edi	t 🔜 Remove							
Table Access Points		Server .	Connection Pr	rotocol Local Time	out Remote	Timeout Ti	ace Mode			
🕞 Formats		MessageServer	1 addp	60	90	Tr	ace Is Turned Off			
🛃 Fields		SIPServer1	addp	60	90	Tr	ace Is Turned Off			
💫 Switching 🛛 🕂										
Routing/eServices +	— 💌 * Server Info —									
🕞 Desktop 🛛 🛨	* Network Securit	hr								
😝 Accounts 🛛 🛨	- Wetwork Securit	- 7								
🙀 Voice Platform 🛛 🕂										
😝 Outbound Contact 🛛 🕒										

Figure 21: Genesys Administrator - Stat Server Connections

4.6 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.

Senesys:	Genesys Ad	dministrat	or				New Window Log ou	t 🍈 🕶 🔞
MONITORING PROVISIONING	DEPLOYMENT OPERATI	ONS						_
PROVISIONING > Environment >	Applications > URS1							
Navigation 🔍	URS1 - Started - Pri	imary - \Application	s\Routing\					
🙀 Search 🛛 🕂	💢 Cancel 🛃 Save & Clo	se 🛃 Save 🛃 Save 8	& New 📑 Reload 📑	Uninstall 📫 Start 📓	Stop 🔣 Graceful Stop			
🙀 Environment 📃	Configuration	Options	Permissions	Dependencies	Alarms	Logs		
Alarm Conditions						General	Server Info Network S	ecurity Advanc
Scripts	- A General							
Application Templates	* Name:	URS1					×	:
Applications	* Application Templat	e: UR Server 813	3				×	C
Hosts	* Type:	Universal Routi	ing Server					~
Solutions	Version:	8.1.300.27						
📑 Time Zones	Server:	True						
Business Units/Sites	State:	Enabled						
Tenants	Connections:	Add 🌼 Edi	t Remove					
Table Access Points		Server	Connection F	Protocol Local Time	out Remote	Timeout Tra	ace Mode	
Formats		MessageServer		60	90		ce Is Turned Off	
Fields		SIPServer1	addp	60	90	Tra	ce Is Turned Off	
Switching +		StatServer1	addp	60	90	Tra	ce Is Turned Off	
🙀 Routing/eServices 🛛 🕂								
Desktop +	─ ▼ * Server Info —							
Accounts +	* Network Securi	by .						
Voice Platform +	- Hetwork Securi	.,						
💫 Outbound Contact 🛛 🕂	- Advanced							

Figure 22: 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.

4.7 URS Routing Strategies

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

4.7.1 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.

	• •									•						• •																		• •			 			• •		
÷		2	h	L :	÷		÷	 ÷	÷		÷	÷		:	÷			÷					<u>, 8</u>	Sele	ecti	ion	pro	operti	ies	;							 					
:	1	Ŷ	/		:				:		 :	:			÷							•	ŀ	Gene	ral	Bu	usy	Tar	rgel	et Sel	lection	ηÌ										
·											ſ	1	1		:			2				•		_ Sta	atist	ics-		_														
:		:	:			:	:		:		 []	0		/	.			:		·	•	•		0			١	Name		Sta	atTime	elnF	Reac	yStal	e		_				_	_
·			:	· ·	:	:		 :	:		 1	ł			:		1			I	:				rgel												 					
											 -	:						- 10	ŀ	÷		:		Z	ř	×	5	🗹 Clea	ar T	Targ	et	Ti	meo	ut [99	199		_					•
	• •	·	·			·	•		·	•	 ·	·	• •	•	·	• •				_		•						Ту	pe	•					Nam	e			Sta	atSei	rver	•
·								:	:		ł	÷			:		-	÷			·	•		1	S	ikill						Т	estS	kill >∶	= 1		 S	tatSe	rver1			

Figure 23: URS Strategy #1

4.7.2 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 24: URS Strategy #2

🔉 Selection properties		
General Busy Target S	election	
C Min		
Max Name	StatTimeInReadyState	_
Targets		
📸 🗙 🔽 Clear Tar	rget Timeout 9999	💌 Sec
Туре	Name	StatServer
1 Skill	TestSkill >= 1	StatServer1

Figure 25: URS Strategy #2 - Target Selection

4.7.3 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 26: URS Strategy #3

🔉 Play announ	cement and coll	ect digits proper	ties			
Parameters PF	ROMPT					
🔺 🔀						
	Interruptable	ID	Digits	User_Ann_ID	Text	User_ID
1		1002				

Figure 27: URS Strategy #3 - Prompt Tab

Parameters Timeout Music ['MUSIC_DN','music/in_queue','DURATION 10	neral Busy Treatments—	Target Selection	
Music ['MUSIC_DN','music/in_queue','DURATION 10			Timeout
	Music	['MUSIC_DN','music/in_queue','DURATION	10

Figure 28: URS Strategy #3 - Busy Tab

🔀 Selection p	Selection properties				
General Bu	sy Target Sele	ction			
Statistics - C Min © Max	Name Stat	TimeInReadyState	<u> </u>		
Targets Z 🗙	🔽 Clear Targel	t Timeout 9999	▼ Sec		
	Туре	Name	StatServer		
1 Skill	1	TestSkill >= 1	StatServer1		

Figure 29: URS Strategy #3 - Target Selection

4.7.4 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 30: URS Strategy #4

4.7.5 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 31: URS Strategy #5

🔉 Play anno	Play announcement properties					
Parameters	PROMPT					
* ×						
	Interruptable	ID	Digits	User_Ann_ID	Text	User_ID
1		1001				

Figure 32: URS Strategy #5 - Prompt Tab

unction properties	
eneral	
Expression	
<u> </u>	= TRoute['9723014956',",RouteTypeUnknown,"]
Data Type	Name
All Functions CallInfo Configuration Options Data Manipulation Date/Time Force List Manipulation Miscellaneous	Add ThisTrunk Time TimeDifference TimeDifference Timeout TimeStamp Translate TRoute Variables
Parameter	Yalue
Destination	9723014956
Location	
Route Type	RouteTypeUnknown
DNIS	

Figure 33: URS Strategy #5 - Function Properties

5 Exceptions

5.1 SBC1000/2000 Exceptions

5.1.1 Call hold using RFC 2543 method

This method is not supported by the SBC1000/2000 because its obsolete and replaced by RFC 3264. The workaround is to create Message Manipulation Rules applied on Internal Signaling Group that will match connect IP 0.0.0.0 and replace a=sendrecv with a=inactive.

SIP Message Manipulation feature is used by a SIP Signaling Group to manipulate the incoming or outgoing messages. This feature is intended to enhance interoperability with different vendor equipment and applications, and for correcting any fixable protocol errors in SIP messages on fly without any changes to firmware/software.

Description	Change sendrecv to inactive
Applicable Messages	All Messages
Table Result Type	Optional



Figure 46: Message Rule Table

Figure 60: Message Manipulation Rule 1

Descript Condition Express Admin St Result Ty	sion tate	change sendrecv to inactive Enabled Mandatory
Result Ty Match Regex		Mandatory
Replace Regex	a=in	active *

Figure 61: Message Manipulation Rule 2

Appendix A

SIP Server and DN configuration

SIP Server standard configuration

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

DN standard 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</tse_contact>	TSE

			oosp-transfer-enabled=true sip-replaces-mode=2	
Ext-DN1 Ext-DN2	2221234567 2221234568	N/A	N/A	
SIP-DN1 SIP-DN2	2086041020 2086041021	1020 1021	refer-enabled=false ring-tone-on-make- call=false make-call-rfc3725-flow=1 contact=*	
SIP-RDN	2076041025	1025	refer-enabled=true ring-tone-on-make- call=false make-call-rfc3725-flow=1 contact=* sip-cti-control=talk,hold	SIP endpoint which supports the BroadSoft SIP Extension Event Package.
SIP-UNKN	2086041025	N/A	N/A	
RP	2086041011	1011		
RP1	2086041012	1012		
RP2	2086041013	1013		
SVC_MSML	SVC_MSML	SVC_MSML	prefix=msml= contact= <ms_contact> service-type=msml subscription-id= Environment</ms_contact>	MS

SIP Server and DN non-standard configuration per test case

12: Caller is put on hold and retrieved by using RFC 2543 method	SIP SERVER: sip-hold-rfc3264=false
15: 3PCC Alternate from consult call to main call SIP-DN1	refer-enabled=true
17: 1PCC Attended Transfer to external destination: MGW-TRUNK	refer-enabled=false, oosp-transfer-enabled=true
21: 3PCC Single Step Transfer to internal busy destination using REFER	MGW-TRUNK: refer-enabled=true; sip-busy-type=2
22: Early Media for Inbound Call to Route Point with Treatment	MGW-TRUNK: sip-early-dialog-mode=1
23: Early Media for Inbound Call with Early Media for Routed to Agent	MGW-TRUNK: sip-early-dialog-mode=1
24: Inbound call routed outbound (Remote Agent) using INVITE without SDP	MGW-TRUNK: oosp-transfer-enabled=false
25: Call Progress Detection: MGW-TRUNK	cpd-capability = mediaserver; refer-enabled=false
27: SIP Authentication for outbound calls	MGW-TRUNK: on the Annex tab configure

	AuthClient section with options username= <username> password=<password></password></username>
28: SIP Authentication for incoming calls	MGW-TRUNK: authenticate-requests=invite ;password=1234
29: T-Lib-Initiated Answer/Hold/Retrieve Call for Remote SIP endpoint which supports the BroadSoft SIP Extension Event Package	SIP-RDN: sip-cti-control=talk,hold; authenticate- requests=REGISTER; password=1234
32: 1PCC Attended Transfer from Remote SIP endpoint to external destination	MGW-TRUNK: refer-enabled=false

EpiPhone configuration

Content of configuration file esttt.conf:

[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_SERVR_TLIB_PORT>) sip-register = false dn1 = 7101,name="Alice",mkcall="7102" dn2 = 7102,name="Bob" dn3 = 7200,name="John" dn50 = 5000,script=",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,T OTAL_TIMEOUT=100)"