

Verint SIPREC and Genesys Contact Center using AudioCodes Mediant™ SBC

Version 7.2



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Notice

This document describes how to connect the Verint SIPREC recording system and Genesys Contact Center using AudioCodes Mediant SBC product series.

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Documentation Feedback

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1 Introduction

This Configuration Note describes an example implementation of AudioCodes Session Border Controller (hereafter, referred to as *SBC*) for interworking between Genesys's Contact Center and Verint SIPREC recording system.

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and Genesys Partners who are responsible for installing and configuring Genesys's Contact Center and AudioCodes SBC for enabling recording VoIP calls using Verint SIPREC recording system.

1.2 About AudioCodes SBC Product Series

AudioCodes' family of SBC devices enables reliable connectivity and security between the Enterprise's and the service provider's VoIP networks.

The SBC provides perimeter defense as a way of protecting Enterprises from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP-PBX to any service provider; and Service Assurance for service quality and manageability.

Designed as a cost-effective appliance, the SBC is based on field-proven VoIP and network services with a native host processor, allowing the creation of purpose-built multiservice appliances, providing smooth connectivity to cloud services, with integrated quality of service, SLA monitoring, security and manageability. The native implementation of SBC provides a host of additional capabilities that are not possible with standalone SBC appliances such as VoIP mediation, PSTN access survivability, and third-party value-added services applications. This enables Enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.

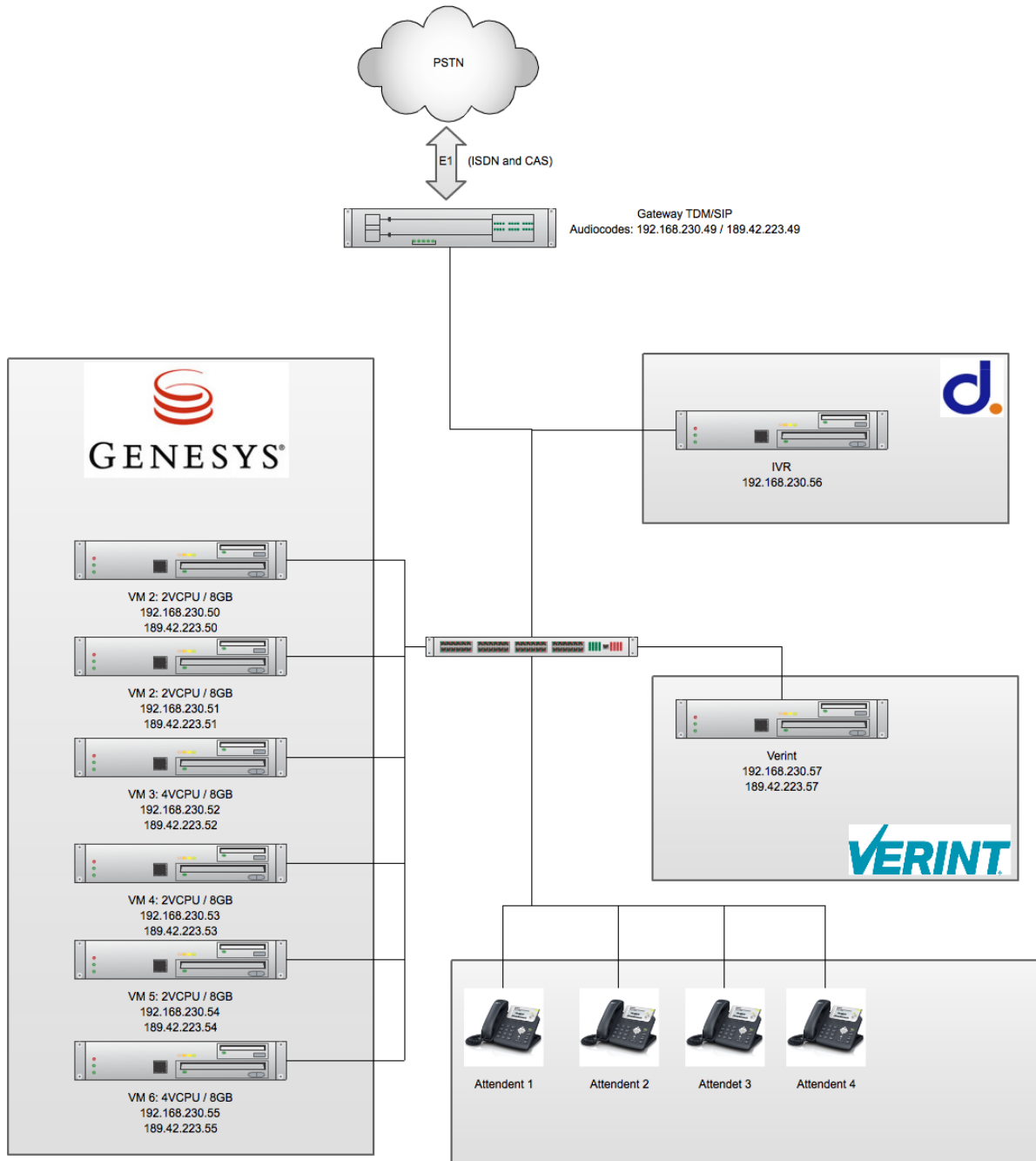
AudioCodes SBC is available as an integrated solution running on top of its field-proven Mediant Media Gateway and Multi-Service Business Router platforms, or as a software-only solution for deployment with third-party hardware.

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1.3 Homologation Test Network Topology

The following figure illustrates the network topology for homologation testing between Verint SIPREC recording system and Genesys Contact Center through AudioCodes Gateway and SBC:

Figure 1-1: Network Topology used in the Homologation



1.3.1 Known Limitations

There were no limitations observed in the homologation tests run between Verint SIPREC recording system and Genesys's Contact Center through AudioCodes SBC.

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2 Configuring Verint Recording System

This section describes an example of the Verint Recording System settings for integrating with Genesys Contact Center and creating Custom Data.

Table 2-1: Configured Servers

cpspaavcdb01	Data Center	Interaction Data Warehouse, Contact Database, Framework Applications, Framework Reports, QM Database, Framework Database, Archive Database, Interaction Reports, Contact OLTP Database, Interaction Applications, Framework Integration Service, Interaction Flow Manager
cpspaavrec01	Recorder	IP Recorder, Content Server, Recorder Integration Service, Central Archive
cpspaavrec02	Recorder	Recorder Integration Service, Content Server, IP Recorder, Central Archive

Figure 2-1: Servers

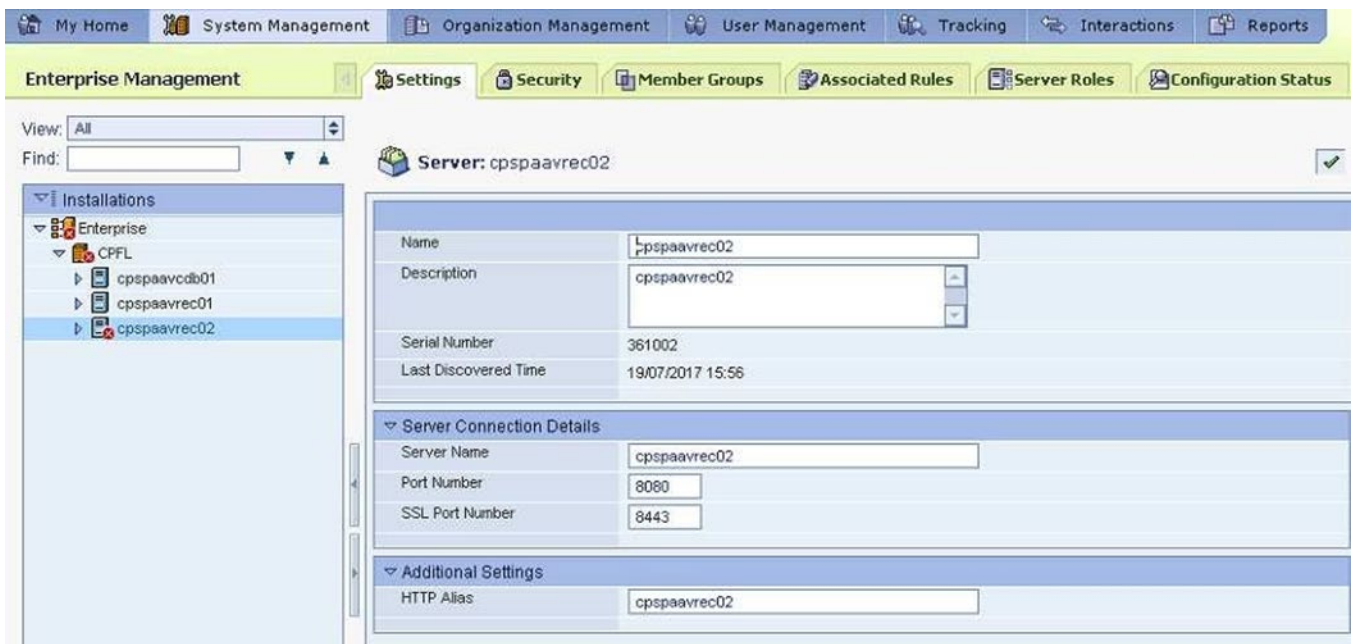


Figure 2-2: Server Roles

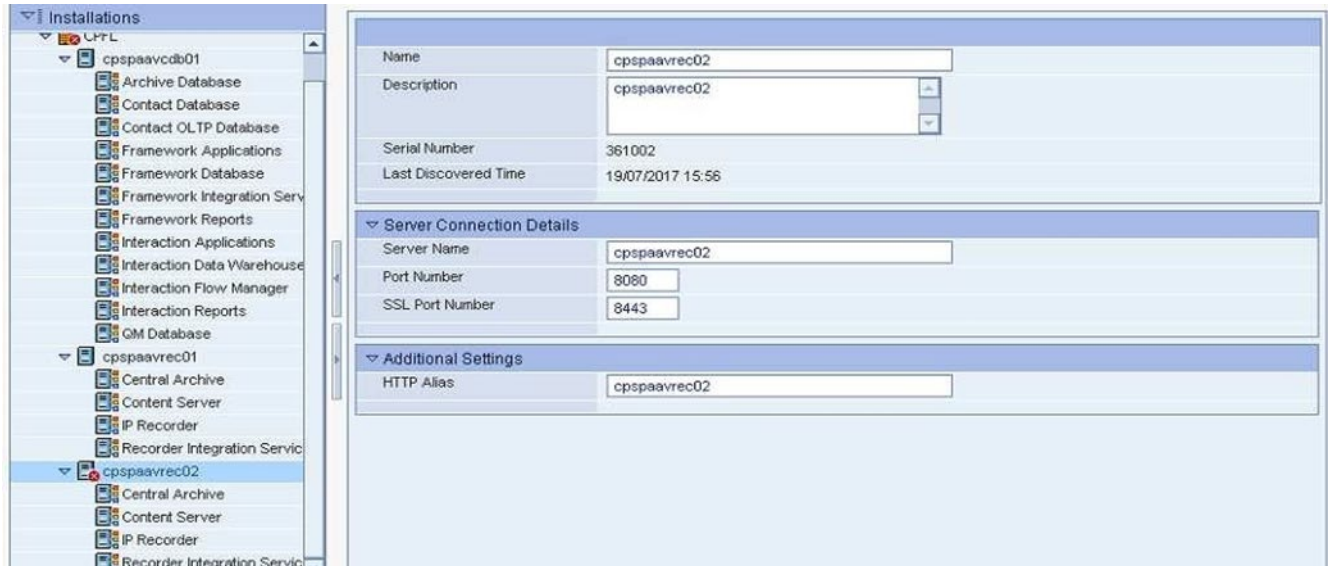


Figure 2-3: Load-balance Configuration – Rec01



Figure 2-4: Load-balance Configuration – Rec02

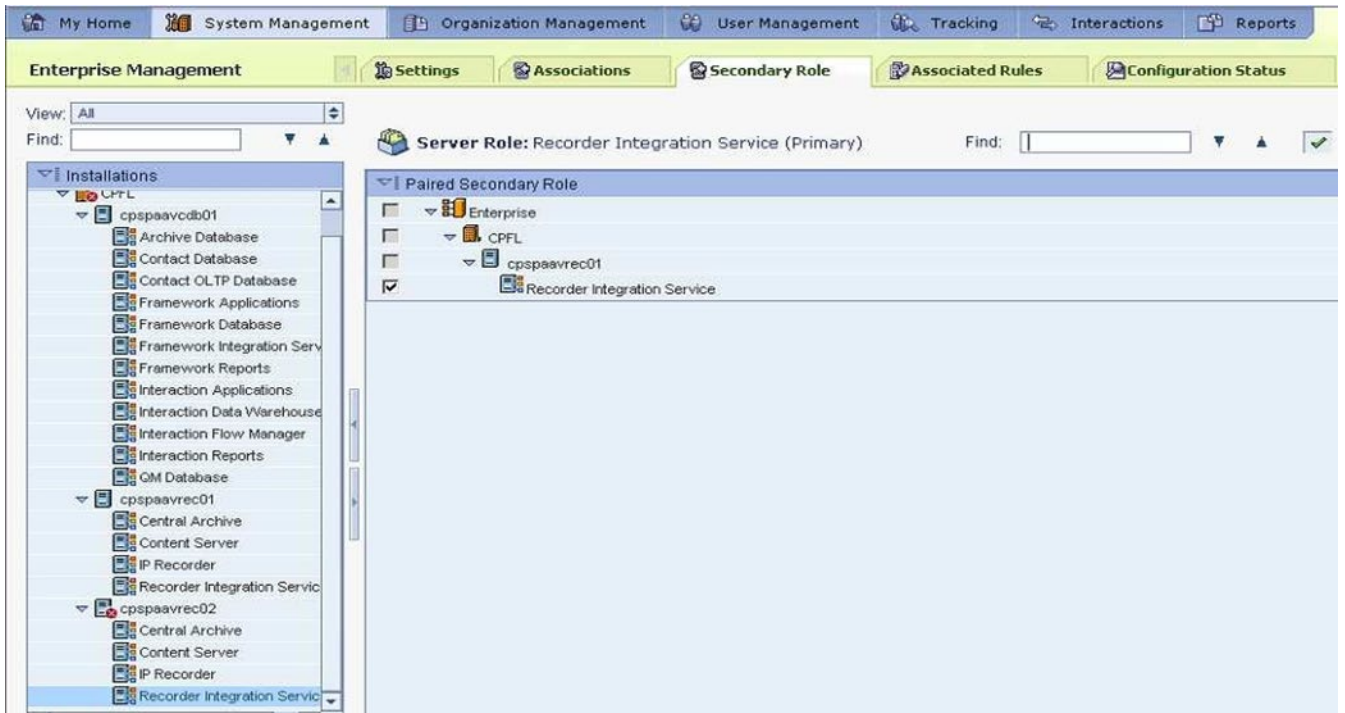


Figure 2-5: Data Source

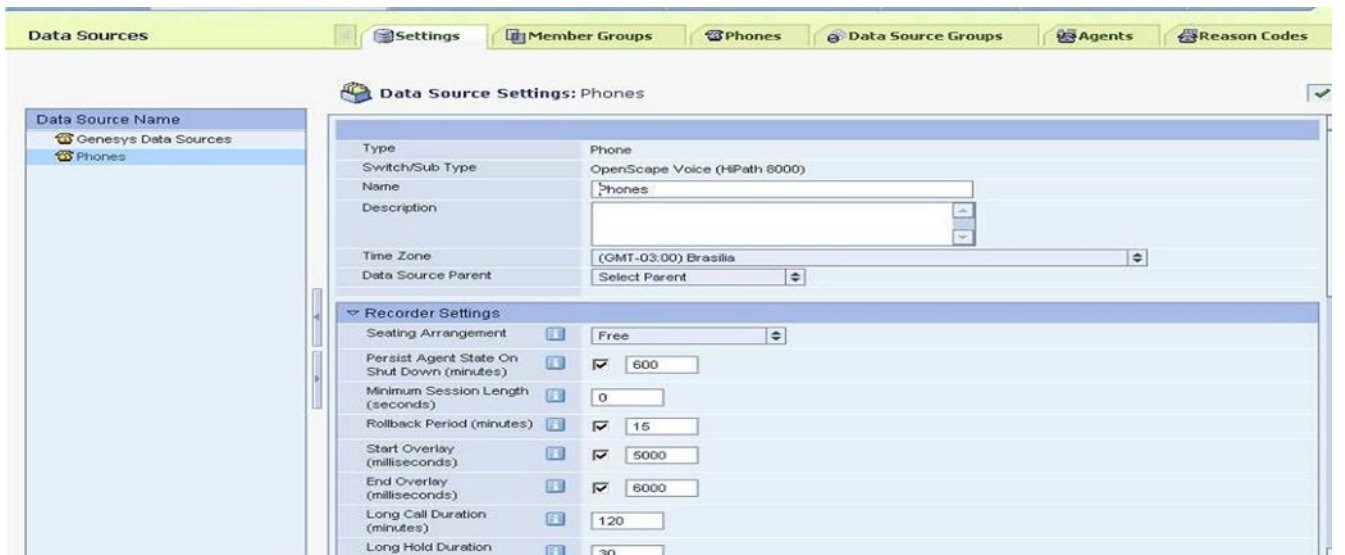


Figure 2-6: Genesys Data Source

The screenshot shows the 'Data Source Settings: Genesys Data Sources' configuration page. The left sidebar lists 'Genesys Data Sources' and 'Phones'. The main area contains the following fields:

- Type: Phone
- Switch/Sub Type: Genesys
- Name: Genesys Data Sources
- Description: (empty text area)
- Time Zone: (GMT-03:00) Brasilia
- Data Source Parent: Select Parent

The 'Recorder Settings' section includes:

- Seating Arrangement: Free
- Persist Agent State On Shut Down (minutes): 600
- Minimum Session Length (seconds): 0
- Rollback Period (minutes): 15
- Start Overlay (milliseconds): 5000
- End Overlay (milliseconds): 6000
- Long Call Duration (minutes): 120
- Long Hold Duration (minutes): 30

Buttons at the bottom include: Import, Export, Reports, Create Data Source, Delete Data Source, Save, and Revert.

Figure 2-7: Member Group Genesys

The screenshot shows the 'Member Groups: Genesys Data Sources' configuration page. The left sidebar lists 'Genesys Data Sources' and 'Phones'. The main area displays a table with the following data:

Name	Type	Recorders
Member Group Genesys	IP Extension Pool	cpspaavrec01,cpspaavrec02

Figure 2-8: Extensions

Extensions Primary/Secondary	Recording Mode	Member Groups	LAN (Screen) Data Source	Workstation Name
1100	Record	Member Group Genesys		
1101	Record	Member Group Genesys		
1102	Record	Member Group Genesys		
1103	Record	Member Group Genesys		
1104	Record	Member Group Genesys		
1105	Record	Member Group Genesys		
1106	Record	Member Group Genesys		
1107	Record	Member Group Genesys		
1108	Record	Member Group Genesys		
1109	Record	Member Group Genesys		
1110	Record	Member Group Genesys		
1111	Record	Member Group Genesys		
1112	Record	Member Group Genesys		
1113	Record	Member Group Genesys		
1114	Record	Member Group Genesys		
1115	Record	Member Group Genesys		
1122	Record	Member Group Genesys		
1123	Record	Member Group Genesys		
1124	Record	Member Group Genesys		
1125	Record	Member Group Genesys		

Figure 2-9: Recorder Integration Service REC01 – Generic SIPREC Adapter

Adapter: Generic SIPREC

Adapter Name	Status	Target
Generic SIPREC	Started	Start
Genesys T-lib	Started	Start
SIP Proxy	Started	Start

Settings

- Adapter Name: Generic SIPREC
- Description: Generic SIPREC Adapter
- Adapter Type: SipRecAdapter
- Startup Type: Automatic
- DataSource: Genesys Data Sources
- SIPREC Device Type: AudioCodes
- Redundancy Type: Recording

General Settings

- SIP Protocol: SIP over UDP
- Listen at IP Address: 10.250.4.133
- Port: 5060

SIP Statistics

- Enable SIP Statistics Gathering:

Buttons: Start, Stop, Restart, Create, Save, Delete

Figure 2-10: Recorder Integration Service REC01 – Genesys T-lib Adapter

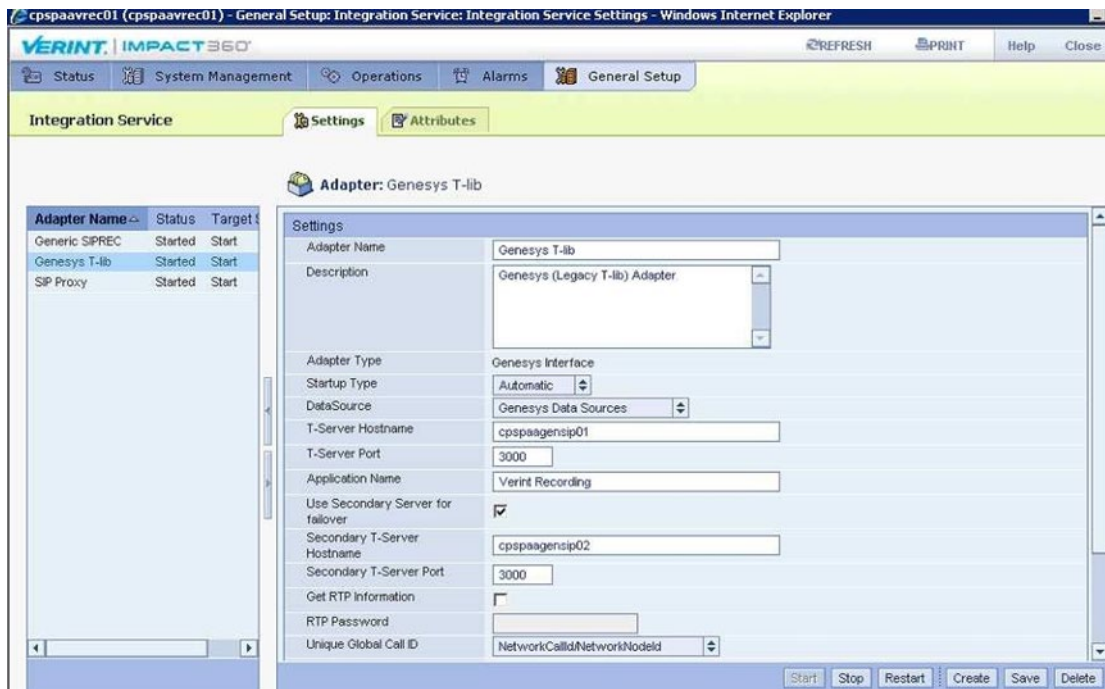


Figure 2-11: Recorder Integration Service REC01 – SIP Proxy Adapter

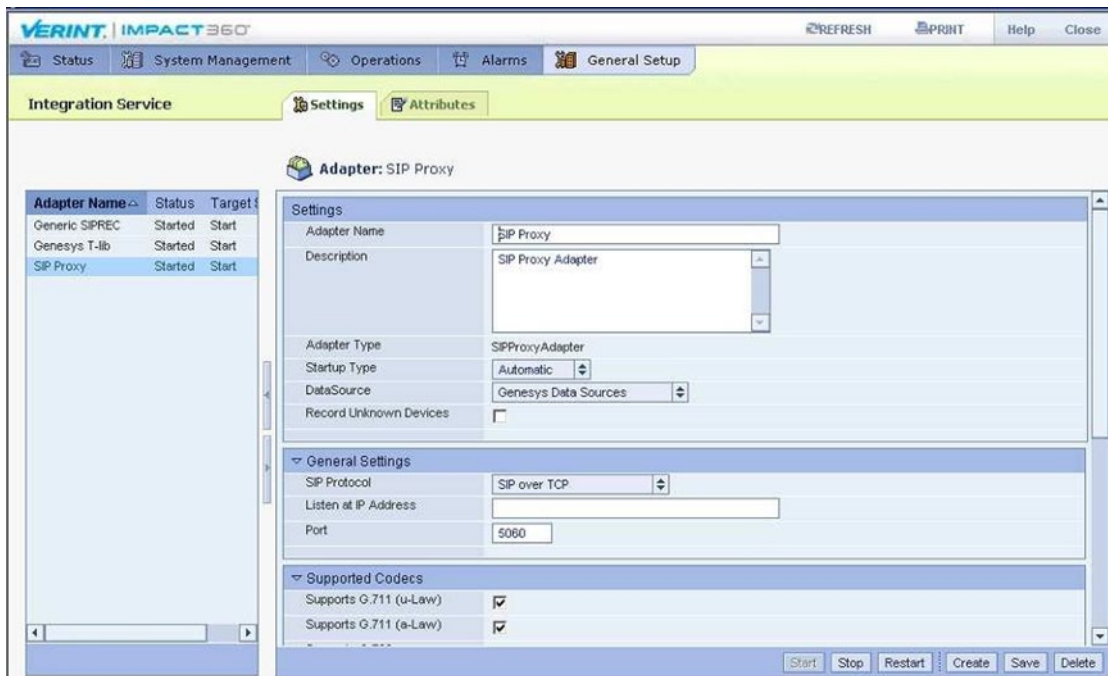


Figure 2-12: Recorder Integration Service REC02 – Generic SIPREC Adapter

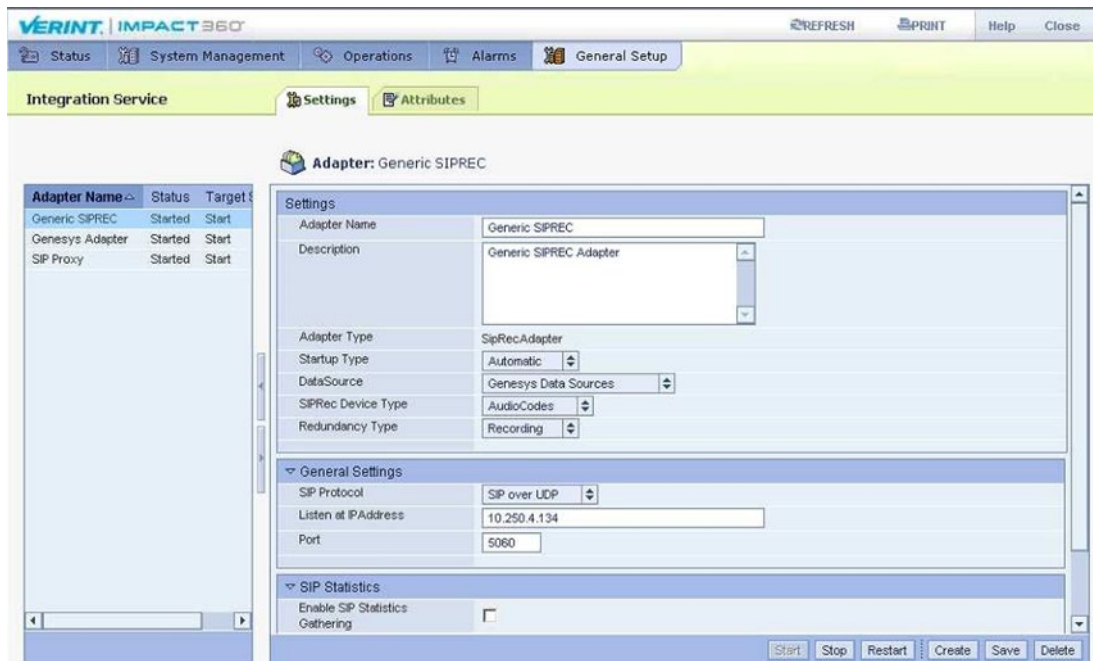


Figure 2-13: Recorder Integration Service REC02 – Genesys Adapter

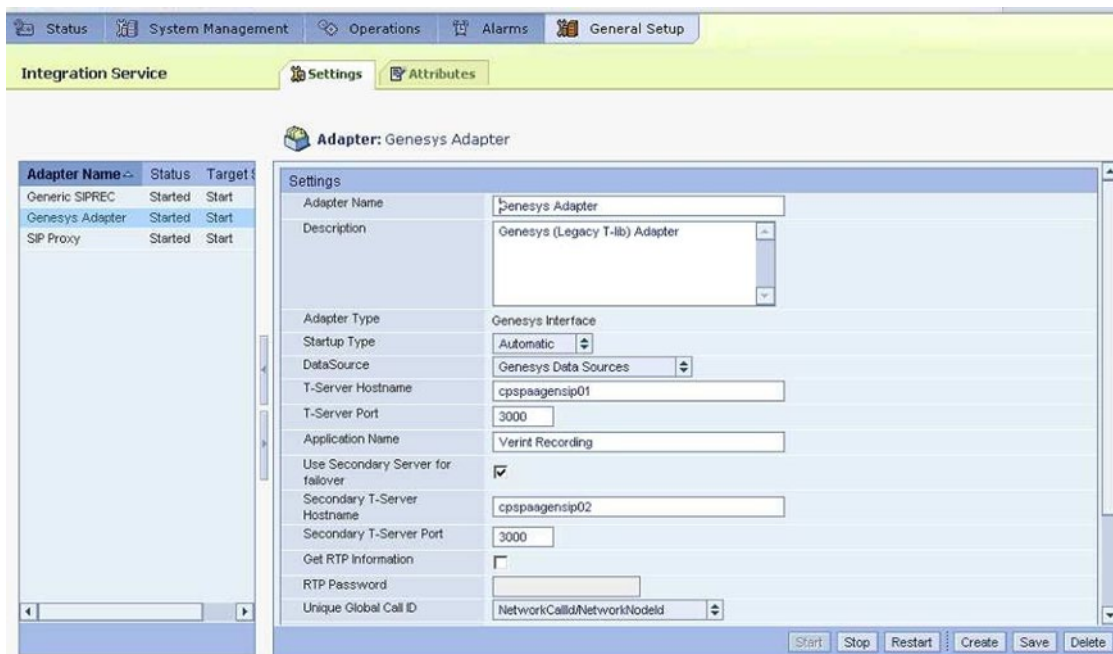


Figure 2-14: Recorder Integration Service REC02 – SIP Proxy Adapter

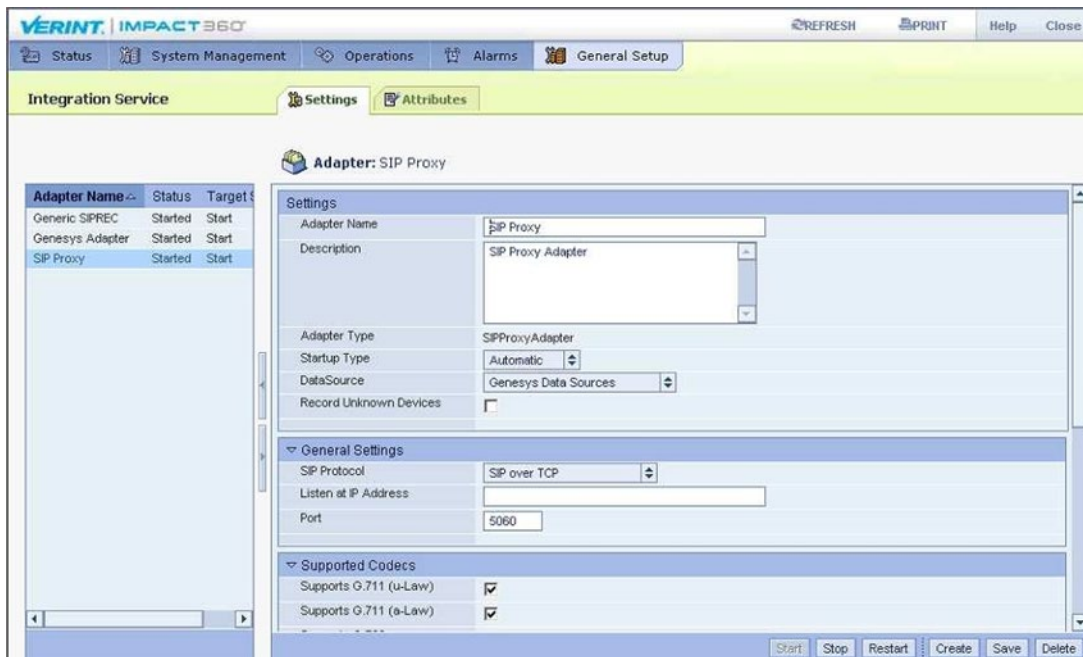


Figure 2-15: Recorder Capture Settings – REC02

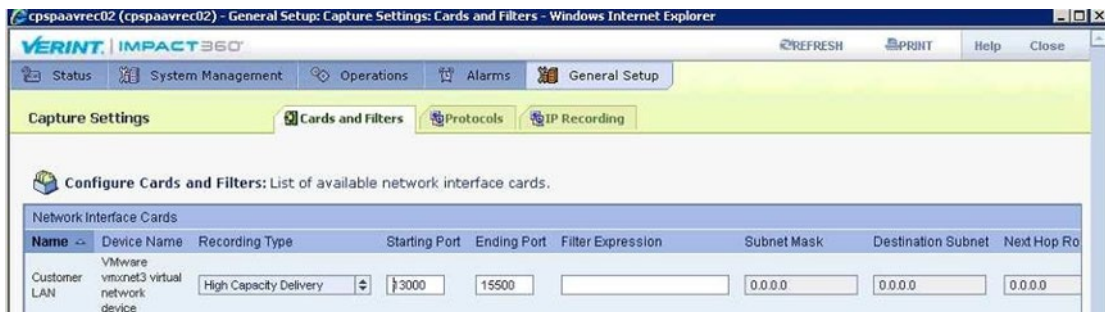


Figure 2-16: Recorder Protocols – REC02

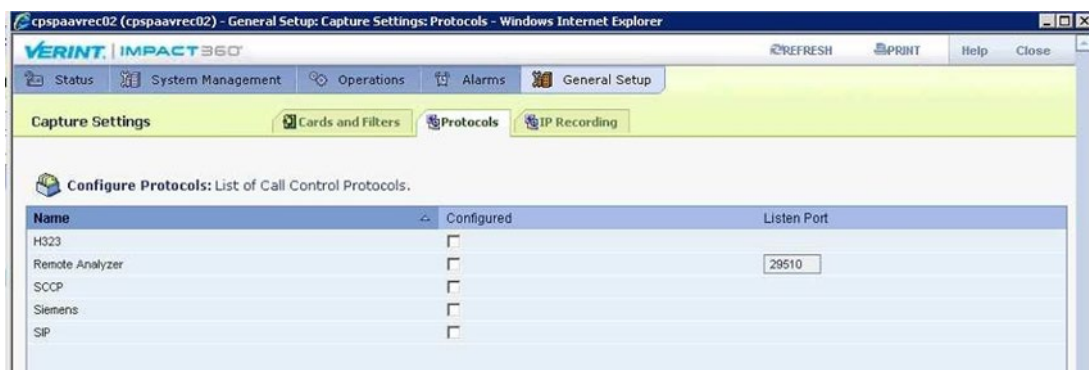


Figure 2-17: IP Recording – REC02

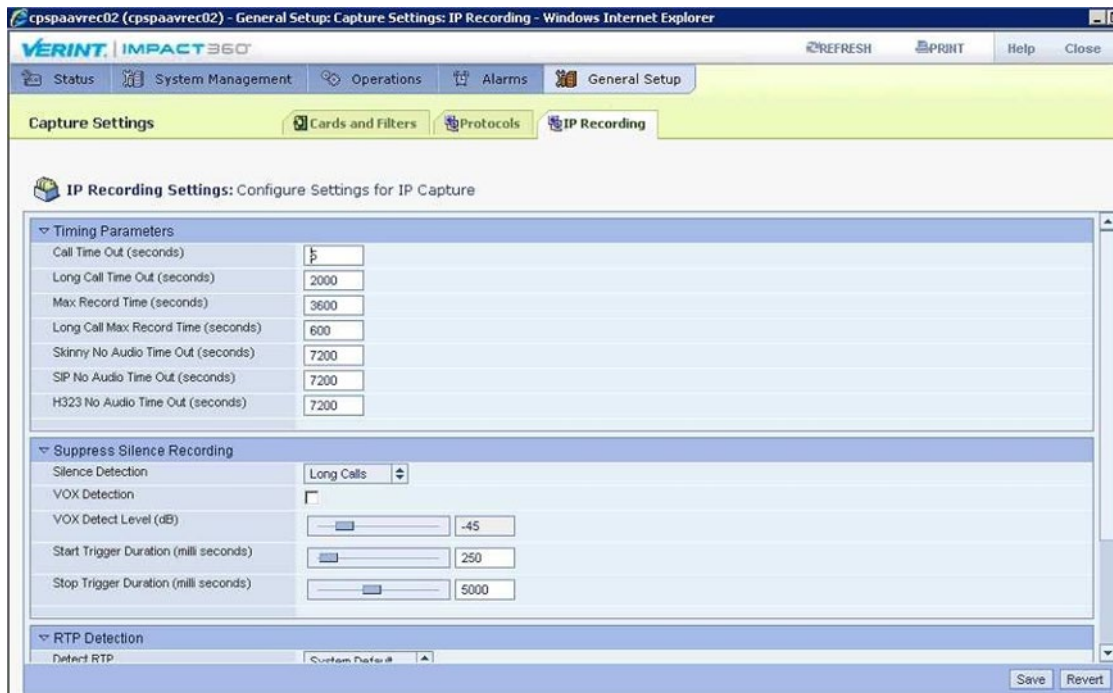


Figure 2-18: Recorder Capture Settings – REC01

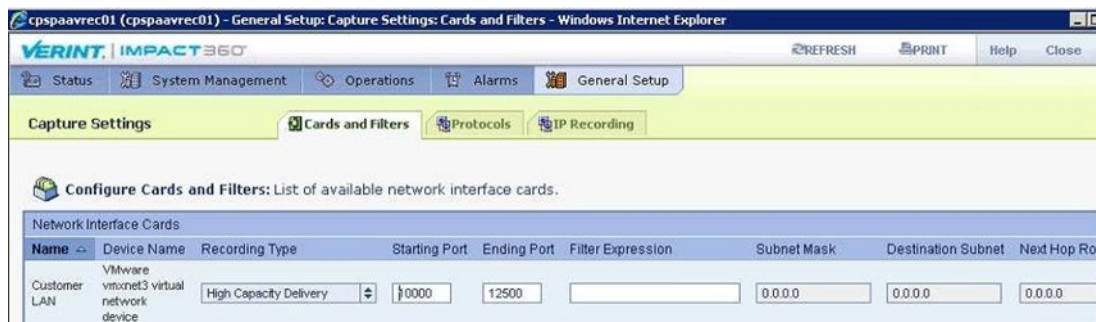


Figure 2-19: Recorder Protocols – REC01

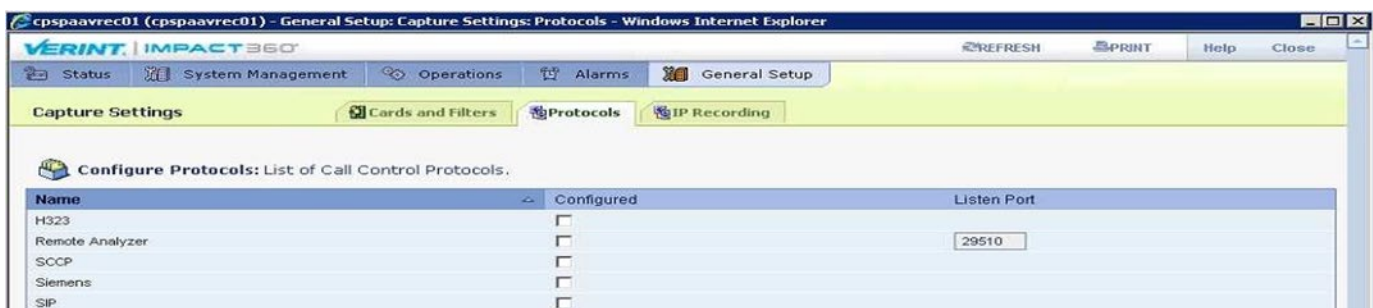


Figure 2-20: IP Recording – REC01

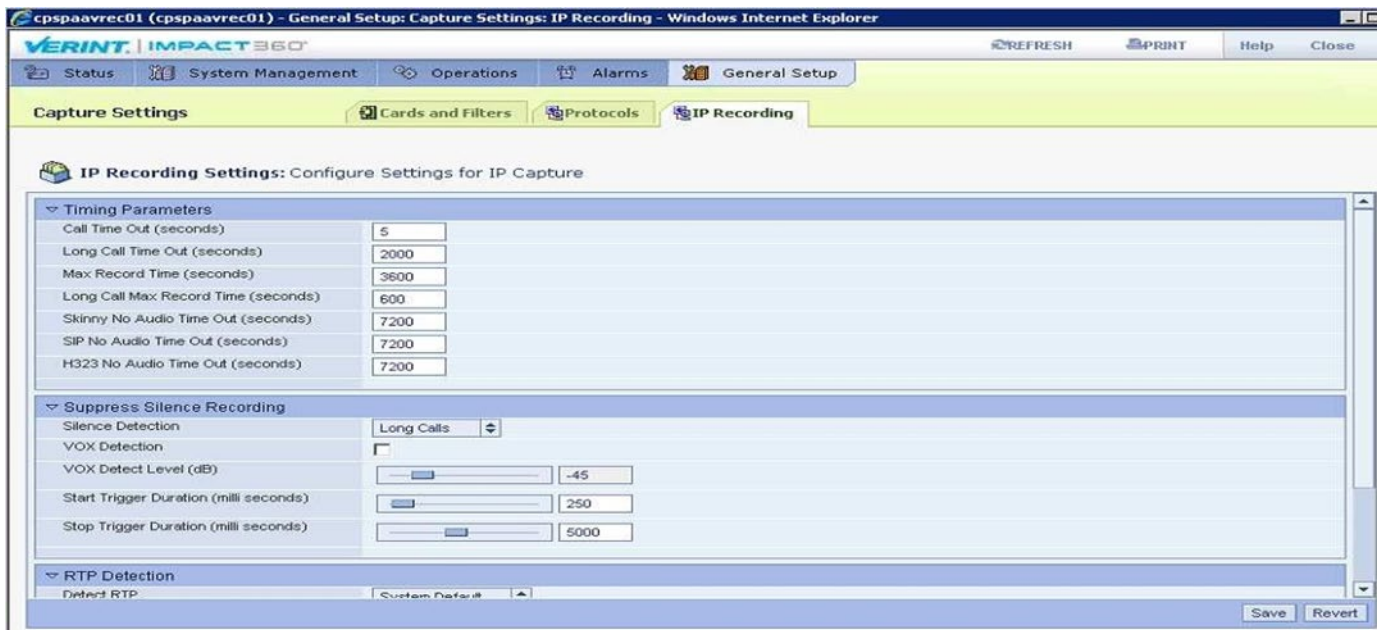


Figure 2-21: User ID Registration of Users in Genesys Data Source

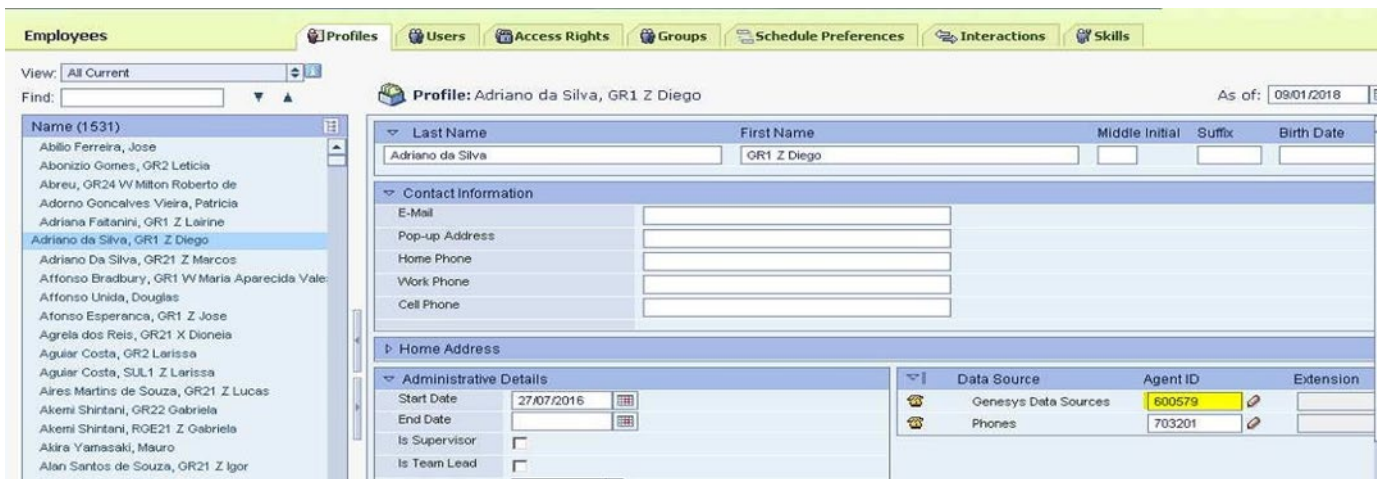


Figure 2-22: System Management \ Custom Data

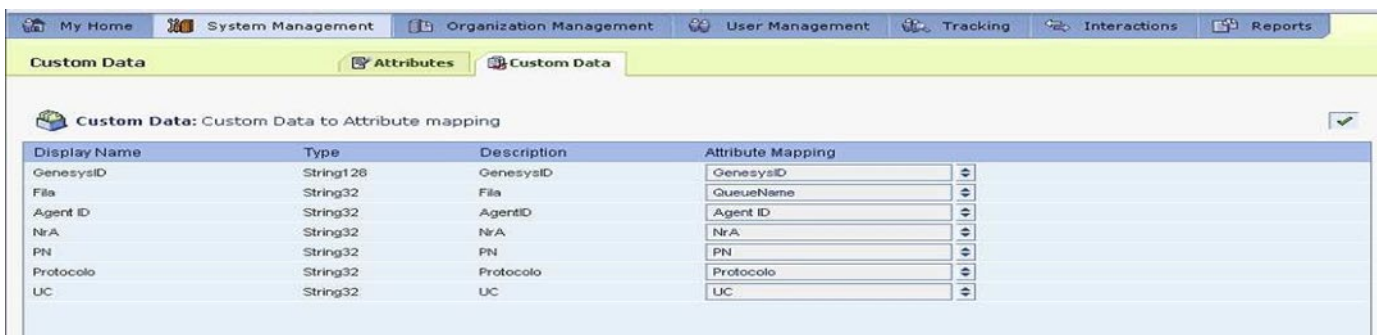


Figure 2-23: Organization Management \ Custom Data

Show disabled fields Import... Print...

Enabled	Display Name	Field Name	Type	Indexed	Data Source	Description	Edit	Up	Down
✓	Fila		String32	False	CTI	Fila			
✓	Agent ID		String32	False	CTI	AgentID			
✓	GenesysID		String128	True	CTI	GenesysID			
✓	NrA		String32	False	CTI	NrA			
✓	PN		String32	False	CTI	PN			
✓	UC		String32	False	CTI	UC			
✓	Protocolo		String32	False	CTI	Protocolo			
	Custom Data 2		String64	True					
	Custom Data 3		String64	True					

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3 Configuring AudioCodes SBC

This chapter provides step-by-step procedures on how to configure AudioCodes SBC for interworking between Verint SIPREC recording system and the Genesys Contact Center. These configuration procedures are based on the homologation test topology described in Section 1.3 on page 9.

This configuration is done using the SBC's embedded Web server (hereafter, referred to as *Web interface*).



Notes:

- For implementing Verint SIPREC and Genesys Contact Center based on the configuration described in this section, AudioCodes SBC must be installed with a License Key that includes the following software features:

- ✓ **SBC**

- ✓ **Security**

- ✓ **SIPREC**

For more information about the License Key, contact your AudioCodes sales representative.

- The scope of this document does **not** cover all security aspects for configuring this topology. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended Security Guidelines* document, which can be found at AudioCodes web site

3.1 IP Network Interface Configuration

This section describes how to configure the SBC's IP network interface. There are several ways to deploy the SBC; however, this test topology employs the following deployment method:

- SBC interfaces with the following IP entities:
 - Verint recording system, located on the LAN
 - Genesys Contact Center, located on the LAN
- The gateway application connects to the PSTN through a E1-Euro-ISDN interface

3.1.1 Configure Network Interface

This section describes how to configure the SBC IP network interface.

➤ **To configure the IP network interfaces:**

1. Open the IP Interfaces table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **IP Interfaces**).
2. Modify the existing network interface:
 - a. Select the 'Index' radio button of the **OAMP + Media + Control** table row, and then click **Edit**.
 - b. Configure the interface as follows:

Parameter	Value
Name	LAN_IF (arbitrary descriptive name)
Ethernet Device	vlan 1
IP Address	192.168.230.49 (LAN IP address of SBC)
Prefix Length	24 (subnet mask in bits for 255.255.255.0)
Default Gateway	192.168.230.1
Primary DNS	8.8.8.8

3. Click **Apply**.

3.2 Configure SIP Signaling Interfaces

This section describes how to configure SIP Interfaces. For the homologation test topology, two SIP Interfaces must be configured – one for the Gateway application and another for the SBC application.

➤ **To configure SIP Interfaces:**

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. Check that default SIP Interface (Index 0) is configured with GW Application Type.
3. Configure a SIP Interface for the SBC application:

Parameter	Value
Index	1
Name	SIPInterface_1
Network Interface	LAN_IF
Application Type	SBC
UDP Port	5070
TCP Port	5070
TLS Port	5071
Media Realm	DefaultRealm

The configured SIP Interfaces are shown in the figure below:

Figure 3-1: Configured SIP Interfaces in SIP Interface Table

SIP Interfaces (2)

+ New Edit | Page 1 of 1 Show 10 records per page

INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATING PROTOCOL	MEDIA REALM
0	SIPInterface_0	DefaultSRD (#)	LAN_IF	GW	5060	5060	5061	No encapsulation	DefaultRealm
1	SIPInterface_1	DefaultSRD (#)	LAN_IF	SBC	5070	5070	5071	No encapsulation	DefaultRealm

3.3 Configure Proxy Sets

This section describes how to configure Proxy Sets. The Proxy Set defines the destination address (IP address or FQDN) of the IP entity server. Proxy Sets can also be used to configure load balancing between multiple servers.

For the homologation test topology, two Proxy Sets need to be configured for the following IP entities:

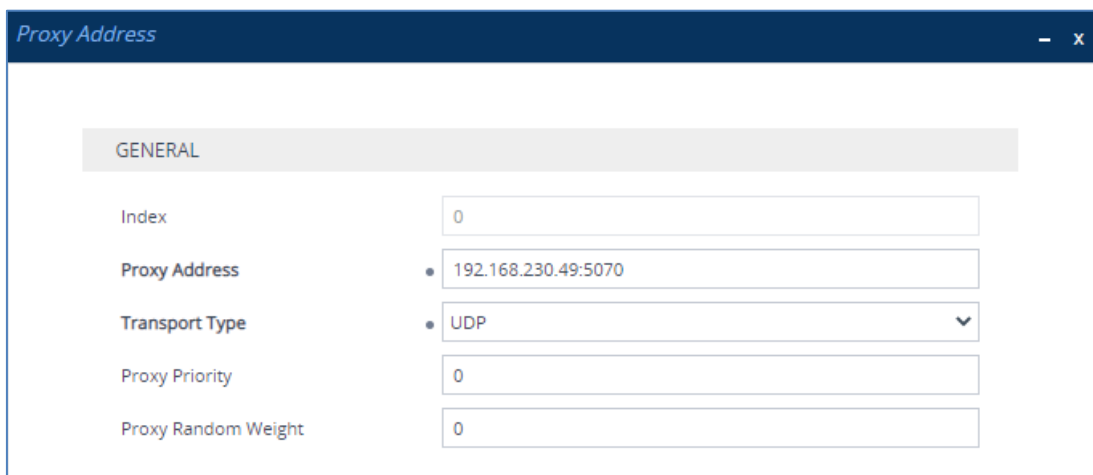
- SBC application
- Verint recording system
- Genesys Contact Center

The Proxy Sets will be later applied to the VoIP network by assigning them to IP Groups.

➤ **To configure Proxy Sets:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**).
2. Modify the default ProxySet_0:
 - a. Select the 'Index' radio button of the ProxySet_0, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
 - b. Click **New**; the following dialog box appears:

Figure 3-2: Configuring Proxy Address for SBC Application



- c. Configure the address of the Proxy Set according to the parameters described in the table below.

Parameter	Value
Index	0
Proxy Address	192.168.230.49:5070 (local Mediant IP address and port for SBC application)
Transport Type	UDP

- d. Click **Apply**.

3. Configure a Proxy Set for the Genesys Contact Center:

Parameter	Value
Index	1
Name	Genesys
SBC IPv4 SIP Interface	SIPInterface_1

Figure 3-3: Configuring Proxy Set for Genesys Contact Center

- a. Select the index row of the Proxy Set that you added, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
- b. Click **New**; the following dialog box appears:

Figure 3-4: Configuring Proxy Address for Genesys Contact Center

- c. Configure the address of the Proxy Set according to the parameters described in the table below.

Parameter	Value
Index	0
Proxy Address	192.168.230.52 (IP address)
Transport Type	UDP

- d. Click **Apply**.

- 4. Add a Proxy Set for the Verint recording server as shown below:

Parameter	Value
Index	2
Name	Verint (arbitrary name)
SBC IPv4 SIP Interface	SIPInterface_1

Figure 3-5: Configuring Proxy Set for Verint Recording Server

- a. Select the index row of the Proxy Set that you added, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
- b. Click **New**; the following dialog box appears:

Figure 3-6: Configuring Proxy Address for Verint Recording Server

- c. Configure the address of the Proxy Set according to the parameters described in the table below.

Parameter	Value
Index	0
Proxy Address	192.168.230.57:5080 (IP address / FQDN and destination port)
Transport Type	UDP

- d. Click **Apply**.

The configured Proxy Sets are shown in the figure below:

Figure 3-7: Configured Proxy Sets in Proxy Sets Table

INDEX	NAME	SRD	GATEWAY IPV4 SIP INTERFACE	SBC IPV4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP
0	ProxySet_0	DefaultSRD (#C	SIPInterface_0	--	60		Disable
1	Genesys	DefaultSRD (#C	--	SIPInterface_1	60		Disable
2	Verint	DefaultSRD (#C	--	SIPInterface_1	60		Disable

3.4 Configure Coders

This section describes how to configure coders (termed *Coder Group*).

➤ **To configure coders:**

1. Open the Coder Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Coder Groups**).
2. Modify default Coder Group:

Parameter	Value
Coder Group Name	AudioCodersGroups_0
Coder Name	<ul style="list-style-type: none"> ▪ G.729 ▪ G.711 A-law ▪ G.711 U-law

Figure 3-8: Configuring Coder Group

Coder Groups

Coder Group Name: 0 : AudioCodersGroups_0 Delete Group

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression	Coder Specific
G.729	20	8	18	Disabled	
G.711A-law	20	64	8	Disabled	
G.711U-law	20	64	0	Disabled	

3.5 Configure IP Groups

This section describes how to configure IP Groups. The IP Group represents an IP entity on the network with which the SBC communicates. This can be a server (e.g., IP PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. Once IP Groups are configured, they are used to configure IP-to-IP routing rules for denoting source and destination of the call.

In this homologation test topology, IP Groups must be configured for the following IP entities:

- Genesys Contact Center
- Verint recording system

➤ **To configure IP Groups:**

1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
2. Configure an IP Group for the Genesys Contact Center:

Parameter	Value
Index	1
Name	Genesys
Type	Server
Proxy Set	Genesys
Media Realm	DefaultRealm

3. Configure an IP Group for the Verint recording system:

Parameter	Value
Index	2
Name	Verint
Type	Server
Proxy Set	Verint
Media Realm	DefaultRealm

The configured IP Groups are shown in the figure below:

Figure 3-9: Configured IP Groups in IP Group Table

INDEX	NAME	SRD	TYPE	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULATION SET	OUTBOUND MESSAGE MANIPULATION SET
0	Default_IPG	DefaultSRD	Server	Not Configured	ProxySet_0	--	DefaultRealm		Enable	-1	-1
1	Genesys	DefaultSRD	Server	Not Configured	Genesys	--	DefaultRealm		Enable	1	1
2	Verint	DefaultSRD	Server	Not Configured	Verint	--	DefaultRealm		Enable	1	1

3.6 Configure IP-to-IP Call Routing Rules

This section describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected.

For the homologation test topology, the following IP-to-IP routing rules need to be configured to route calls between the PSTN network and Genesys Contact Center.

➤ **To configure IP-to-IP routing rules:**

1. Open the IP-to-IP Routing table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing** > **IP-to-IP Routing**).
2. Configure rules as follows:

Index	Source IP Group	Destination Username Pattern	Destination Type	Destination IP Group	Destination Address	Destination Port
0	Any	[0,00]4009xxxx	Dest Address		192.168.230.49	5060 (SBC application listening port)
1	Any		IP Group	Genesys		

The configured routing rules are shown in the figure below:

Figure 3-10: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table

INDEX	NAME	ROUTING POLICY	ALTERNATIVE ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PATTERN	DESTINATION USERNAME PATTERN	DESTINATION TYPE	DESTINATION IP GROUP	DESTINATION SIP INTERFACE	DESTINATION ADDRESS
0		Default_SBCRout	Route Row	Any	All	*	[0,00]4009xxxx	Dest Address	--	--	192.168.230.49
1		Default_SBCRout	Route Row	Any	All	*	*	IP Group	Genesys	--	



Note: The routing configuration may change according to your specific deployment topology.

3.7 Configuring SIP Recording

This section describes SBC’s SIP Recording configuration for recording all calls from Genesys Contact Center by Verint recording system.

3.7.1 Configuring SIP Recording Rules

This section describes how to configure SIP Recording rules through the Web interface. The SIP Recording Rules table lets you configure up to 30 SIP-based media recording rules. A SIP Recording rule defines call routes that you want to record.

➤ **To configure a SIP Recording Routing rule:**

1. Open the SIP Recording Rules table (**Setup** menu > **Signaling & Media** tab > **SIP Recording** folder > **SIP Recording Rules**).
2. Click **New** and configure a SIP recording rule according to the table below:

Index	Recorded IP Group	Peer IP Group	Caller	Recording Server (SRS) IP Group
0	genesys	Any	Peer Party	Verint

The configured SIP recording rules are shown in the figure below:

Figure 3-11: Configured SIP Recording Rules

SIP Recording Rules (1)

+ New Edit | Page 1 of 1 | Show 10 records per page

INDEX ↕	RECORDED IP GROUP	PEER IP GROUP	PEER TRUNK GROUP ID	CALLER	RECORDING SERVER (SRS) IP GROUP
1	Genesys	Any	-1	Peer Party	Verint

3.8 Configure Message Manipulation Rules

This section describes how to configure SIP message manipulation rules. SIP message manipulation rules can include insertion, removal, and/or modification of SIP headers. Manipulation rules are grouped into Manipulation Sets, enabling you to apply multiple rules to the same SIP message (IP entity).

Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

➤ **To configure SIP message manipulation rule:**

1. Open the Message Manipulations page (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulations**).
2. Configure a new manipulation rule (Manipulation Set 1) for Genesys Contact Center.

Parameter	Value
Index	0
Manipulation Set ID	1
Message Type	Refer.Request
Condition	Header.Refer-To regex (.*)(>)(.*)
Action Subject	Header.Refer-To
Action Type	Modify
Action Value	\$1+'?CONTA='+Header.CONTA.Content+'&AGENCIA='+Header.AGENCIA.Content+'&N_CARTAO='+Header.N_CARTAO.Content+'&CPF='+Header.CPF.Content+'\$2+\$3

Figure 3-12: Configuring SIP Message Manipulation Rule 0

The screenshot shows a configuration window titled "Message Manipulations" with the following sections:

- GENERAL:**
 - Index: 0
 - Name: (empty)
 - Manipulation Set ID: 1
 - Row Role: Use Current Condition
- MATCH:**
 - Message Type: Refer.Request
 - Condition: Header.Refer-To regex (.*)(>)(.*)
- ACTION:**
 - Action Subject: Header.Refer-To
 - Action Type: Modify
 - Action Value: \$1+'?CONTA='+Header.CONTA.Content+'&AGENCIA='+Header.AGENCIA.Content+'&N_CARTAO='+Header.N_CARTAO.Content+'&CPF='+Header.CPF.Content+'\$2+\$3

Buttons for "Cancel" and "APPLY" are located at the bottom of the window.

- Configure another manipulation rule (Manipulation Set 1) for Genesys Contact Center.

Parameter	Value
Index	1
Manipulation Set ID	1
Message Type	Refer.Request
Condition	Header.Refer-To regex (.*)(>)(.*)
Action Subject	Header.Refer-To
Action Type	Modify
Action Value	\$1+'&MCI='+Header.MCI.Content+'&SERVICO='+Header.SERVICO.Content+'&X-Genesys-CallUUID='+Header.X-Genesys-CallUUID.Content+\$2+\$3

Figure 3-13: Configuring SIP Message Manipulation Rule 1

Figure 3-14: Example of Configured SIP Message Manipulation Rules

Message Manipulations (2)

+ New Edit Insert ↑ ↓ | Page 1 of 1 | Show 10 records per page

INDEX	NAME	MANIPULATION SET ID	MESSAGE TYPE	CONDITION	ACTION SUBJECT	ACTION TYPE	ACTION VALUE	ROW ROLE
0		1	Refer.Request	Header.Refer-To	Header.Refer-To	Modify	\$1+'?CONTA='+H	Use Current Cor
1		1	Refer.Request	Header.Refer-To	Header.Refer-To	Modify	\$1+'&MCI='+Hea	Use Current Cor

4. Assign Manipulation Set ID 1 to the Genesys IP Group:
 - a. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
 - b. Select the row of the Genesys IP Group, and then click **Edit**.
 - c. Set the 'Inbound Message Manipulation Set' field to **1**.
 - d. Set the 'Outbound Message Manipulation Set' field to **1**.

Figure 3-15: Assigning Manipulation Set to the Genesys IP Group

The screenshot shows a configuration window titled "IP Groups [Genesys]". At the top, there is an "SRD" dropdown menu set to "#0 [DefaultSRD]". The main area is divided into three sections:

- GENERAL:**
 - Index: 1
 - Name: Genesys
 - Topology Location: Down
 - Type: Server
 - Proxy Set: #1 [Genesys] (with a "View" link)
 - IP Profile: -- (with a "View" link)
 - Media Realm: #0 [DefaultRealm] (with a "View" link)
 - Internal Media Realm: -- (with a "View" link)
 - Contact User: (empty field)
 - SIP Group Name: (empty field)
- QUALITY OF EXPERIENCE:**
 - QoE Profile: -- (with a "View" link)
 - Bandwidth Profile: -- (with a "View" link)
- MESSAGE MANIPULATION:**
 - Inbound Message Manipulation Set: 1
 - Outbound Message Manipulation Set: 1
 - Message Manipulation User-Defined String 1: (empty field)
 - Message Manipulation User-Defined String 2: (empty field)
 - Proxy Keep-Alive using IP Group settings: Disable

At the bottom right, there are "Cancel" and "APPLY" buttons.

- e. Click **Apply**.

5. Assign Manipulation Set ID 1 to the Verint IP Group:
 - a. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
 - b. Select the row of the Verint IP Group, and then click **Edit**.
 - c. Set the 'Inbound Message Manipulation Set' field to **1**.
 - d. Set the 'Outbound Message Manipulation Set' field to **1**.

Figure 3-16: Assigning Manipulation Set 1 to the Verint IP Group

The screenshot shows the configuration interface for the 'Verint' IP Group. At the top, there is a dropdown for 'SRD' set to '#0 [DefaultSRD]'. Below this are two main sections: 'GENERAL' and 'QUALITY OF EXPERIENCE'. The 'GENERAL' section includes fields for Index (2), Name (Verint), Topology Location (Down), Type (Server), Proxy Set (#2 [Verint]), IP Profile (--), Media Realm (#0 [DefaultRealm]), Internal Media Realm (--), Contact User, and SIP Group Name. The 'QUALITY OF EXPERIENCE' section includes QoE Profile and Bandwidth Profile, both set to '--'. Below these is the 'MESSAGE MANIPULATION' section, which is currently active. It contains 'Inbound Message Manipulation Set' and 'Outbound Message Manipulation Set', both set to '1'. There are also two empty fields for 'Message Manipulation User-Defined String' and a 'Proxy Keep-Alive using IP Group settings' dropdown set to 'Disable'. At the bottom right, there are 'Cancel' and 'APPLY' buttons.

- e. Click **Apply**.

3.9 Miscellaneous SBC Configuration

This section describes miscellaneous SBC configuration.

3.9.1 Configure SBC to allow Unclassified Calls

This section describes how to allow unclassified calls.

- **To configure SBC to allow unclassified calls:**
- 1. Open the SBC General Settings page (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **SBC General Settings**).
- 2. From the 'Unclassified Calls' drop-down list, select **Allow**.

Figure 3-17: SBC to allow unclassified calls

The screenshot shows the 'SBC General Settings' configuration page. A 'GENERAL' tab is selected. The settings are as follows:

Direct Media	Disable
Unclassified Calls	• Allow
Forking Handling Mode	Latch On First
No Answer Timeout [sec]	600

- 3. Click **Apply**.

3.10 Configuring PSTN Interface (Gateway Application)

This section describes the configuration of the PSTN Interface for connecting to the public switched telephone network (PSTN).

3.10.1 Configure PRI Trunk Settings

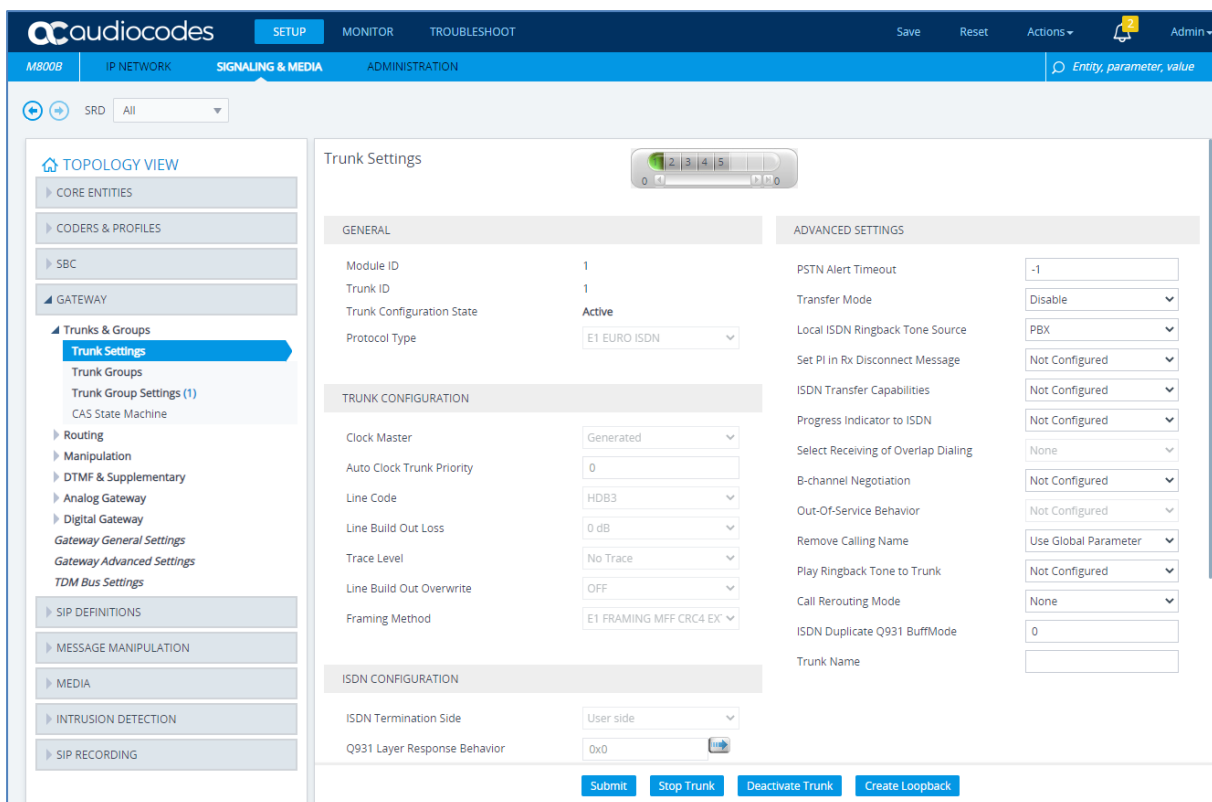
This section describes how to configure the PRI Trunk.

➤ To configure the PRI PSTN interface:

1. Open the Trunk Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunks**).
2. Configure following parameters per PSTN network:

Parameter	Value
Protocol Type	E1 EURO ISDN
Clock Master	<ul style="list-style-type: none"> • Generated (The device is clock master) or • Recovered (The device slaves from the line clock) (per remote side PSTN definitions)
Framing Method	E1 Framing MFF CRC4 Ext (per remote side PSTN definitions)
ISDN Termination Side	User side

Figure 3-18: Configuring the PRI PSTN Interface



3. Click **Submit**.
4. Reset the device with a save-to-flash for your settings to take effect.

3.10.2 Configure Trunk Group Parameters

This section describes how to configure the device's channels, which includes assigning them to Trunk Groups. A Trunk Group is a logical group of physical trunks and channels. A Trunk Group can include multiple trunks and ranges of channels. To enable and activate the device's channels, Trunk Groups must be configured. Channels not configured in this table are disabled. After configuring Trunk Groups, use them to route incoming IP calls to the Tel side, represented by a specific Trunk Group (ID). You can also use Trunk Groups for routing Tel calls to the IP side.

➤ **To configure the PRI PSTN interface:**

1. Open the Trunk Group table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunk Groups**).

Figure 3-19: Configuring PRI Trunk Group Table

GROUP INDEX	MODULE	FROM TRUNK	TO TRUNK	CHANNELS	PHONE NUMBER	TRUNK GROUP ID	TEL PROFILE NAME
1	Module 1 PRI	1	1	1-31	9990	1	None
2							None
3							None
4							None

2. Configure Trunk Group as required.

3.10.3 Configure Tel to IP Routing

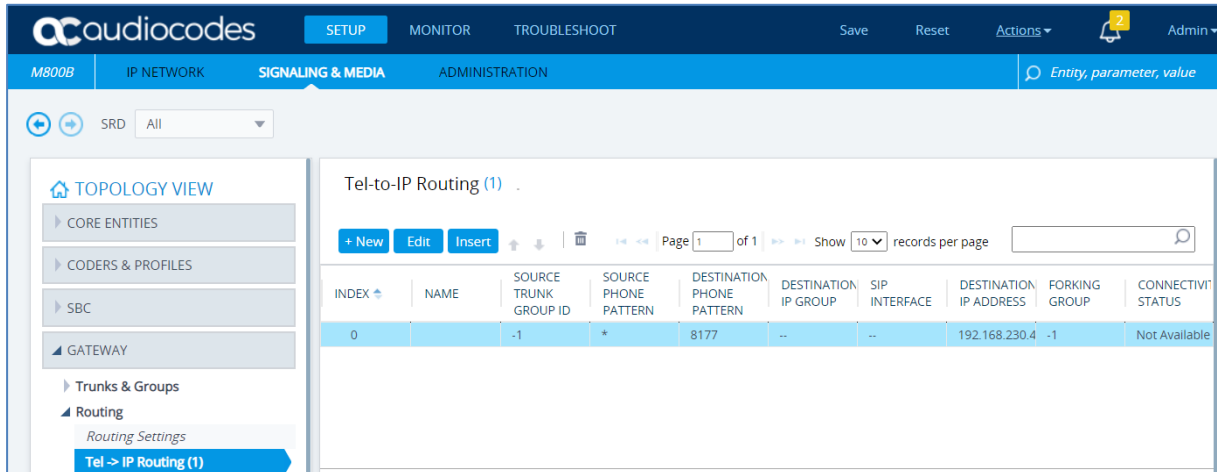
This section describes how to configure Mediant Tel-to-IP routing rules that are used to route calls from the Tel side to an IP destination.

➤ **To configure Tel-to-IP Routing Rules:**

1. Open the IP-to-Tel Routing table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Routing** > **Tel -> IP Routing**).
2. Click **New** and configure following parameters (per homologation test environment):

Parameter	Value
Destination Phone Pattern	8177
Destination IP Address	192.168.230.49 (SBC application address)
Destination Port	5070 (SBC application listening port)

Figure 3-20: Configuring Tel to IP Routing Rules



3. Click **Apply**.

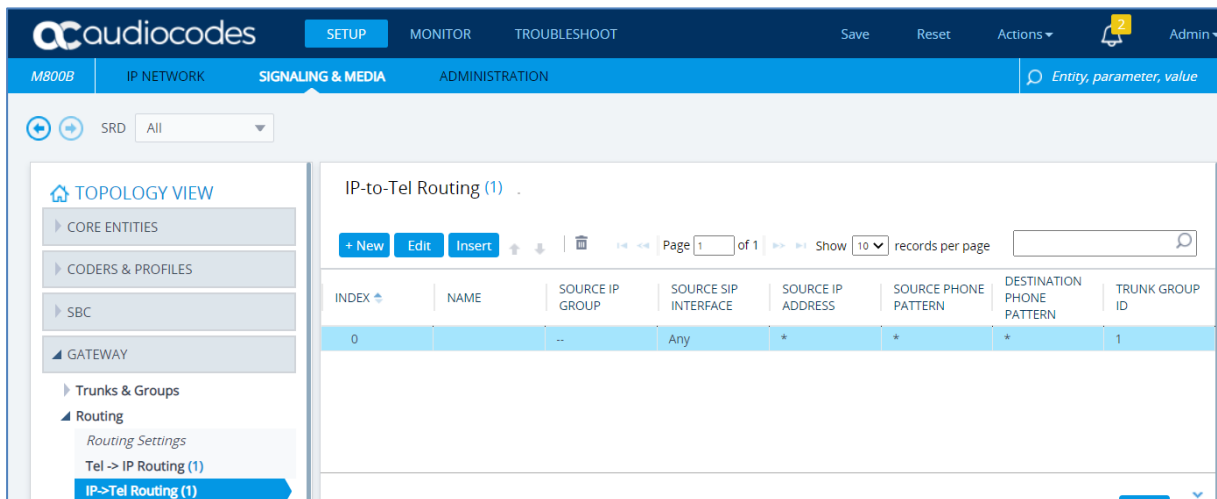
3.10.4 Configure IP to Tel Routing

This section describes how to configure Mediant IP-to-Tel routing rules to route incoming IP calls to Trunk Groups. The specific channel pertaining to the Trunk Group to which the call is routed is determined according to the Trunk Group's channel selection mode. When there is more than one TDM interface, you can choose to route calls based on incoming IP SIP call message to a specific TDM port i.e., Trunk Group.

➤ **To configure IP-to-Tel Routing Rules:**

1. Open the IP-to-Tel Routing table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Routing** > **IP -> Tel Routing**).

Figure 3-21: Configuring IP to Tel Routing Rules



2. Configure a rule for all incoming IP calls, route them to **Trunk Group ID 1** (connected to the PSTN).
3. Click **Apply**.

3.10.5 Configure Number Manipulation Rules

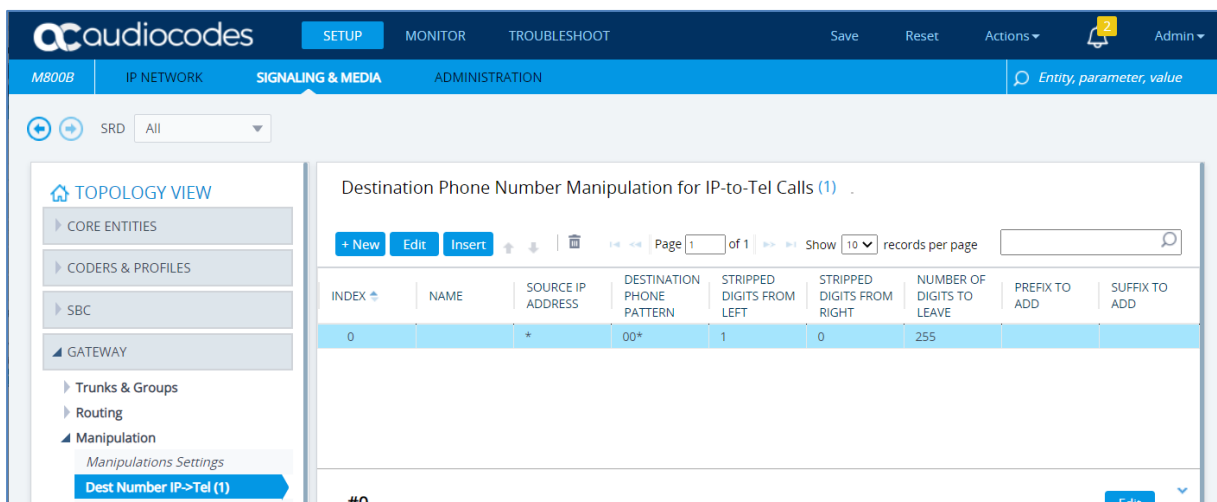
This section describes how to configure Mediant number manipulation rules for manipulating source and destination telephone numbers for IP-to-Tel and Tel-to-IP calls.

➤ **To configure IP-to-Tel destination number manipulation rules:**

1. Open the Dest Number IP -> Tel Number manipulation table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Manipulation** > **Dest Number IP -> Tel**).
2. Click **New** and configure following parameters (per homologation test environment):

Index	Destination Phone Pattern	Stripped Digits From Left	Destination Port
0	00*	1	5070 (SBC application listening port)

Figure 3-22: Configuring IP-to-Tel Destination Number Manipulation Rule



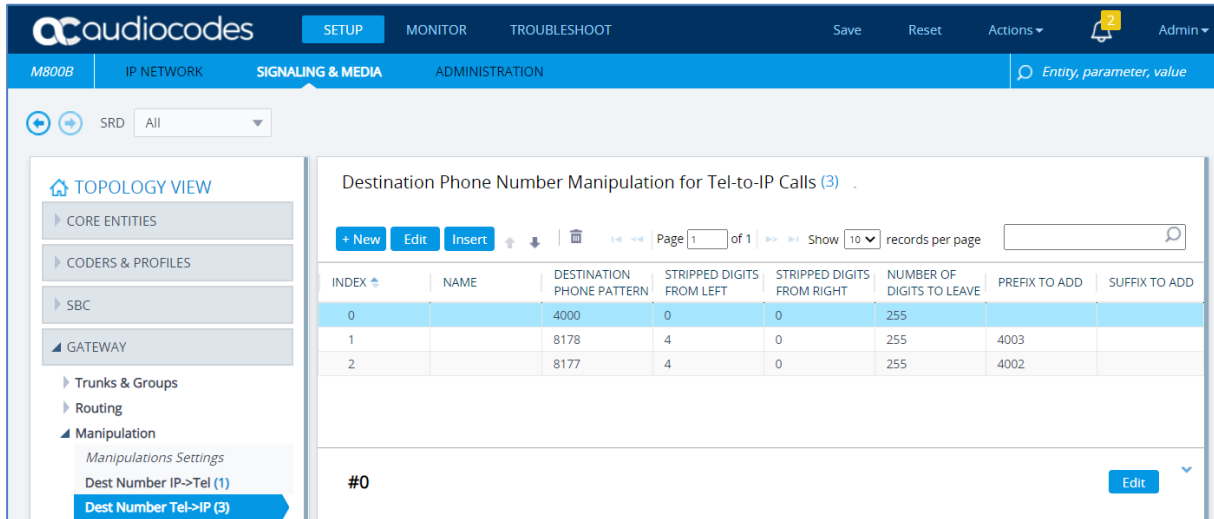
3. Click **Apply**.

➤ **To configure Tel-to-IP destination number manipulation rules:**

1. Open the Dest Number Tel -> IP Number manipulation table (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Manipulation** > **Dest Number Tel -> IP**).
2. Configure following rules (per homologation test environment):

Index	Destination Phone Pattern	Stripped Digits From Left	Prefix to Add	Presentation
0	4000			Allowed
1	8178	4	4003	
2	8177	4	4002	

Figure 3-23: Configuring Tel-to-IP Destination Number Manipulation Rules



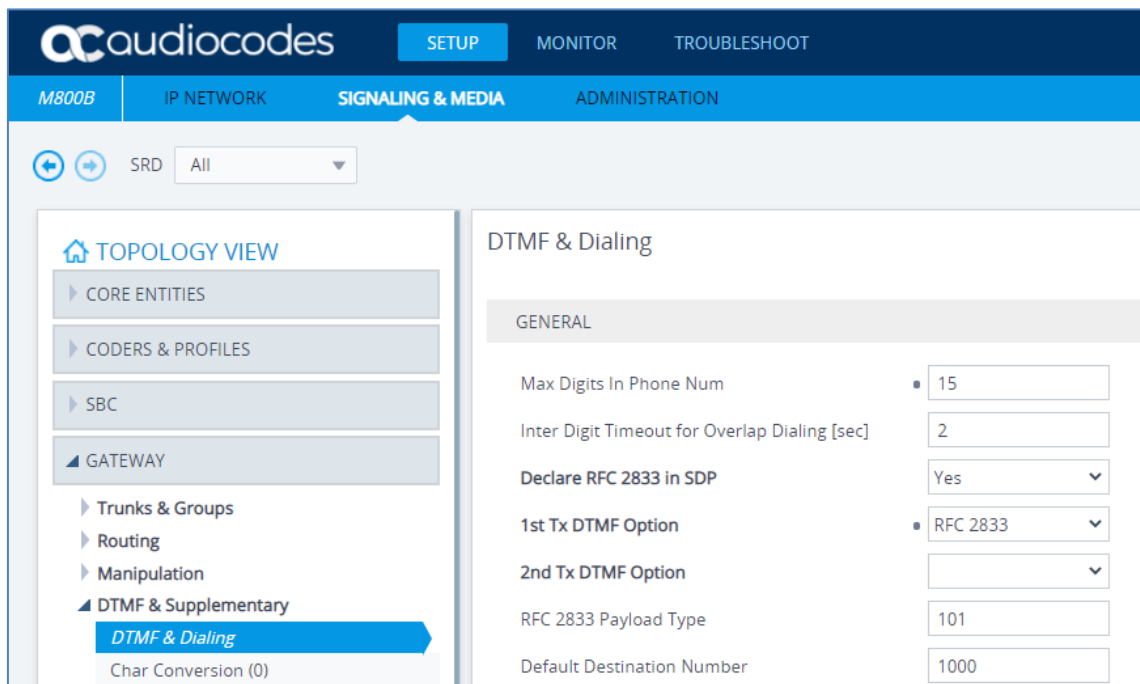
3.10.6 Configure DTMF and Dialing

This section describes how to configure DTMF and dialing parameters.

➤ **To configure DTMF and Dialing:**

1. Open the Gateway Supplementary Services Settings page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **DTMF & Supplementary** > **DTMF & Dialing**).
2. For the 'Max Digits In Phone Num' parameter, configure **15**.
3. From the '1st Tx DTMF Option' drop-down list, select **RFC 2833**.

Figure 3-24: Configuring DTMF and Dialing



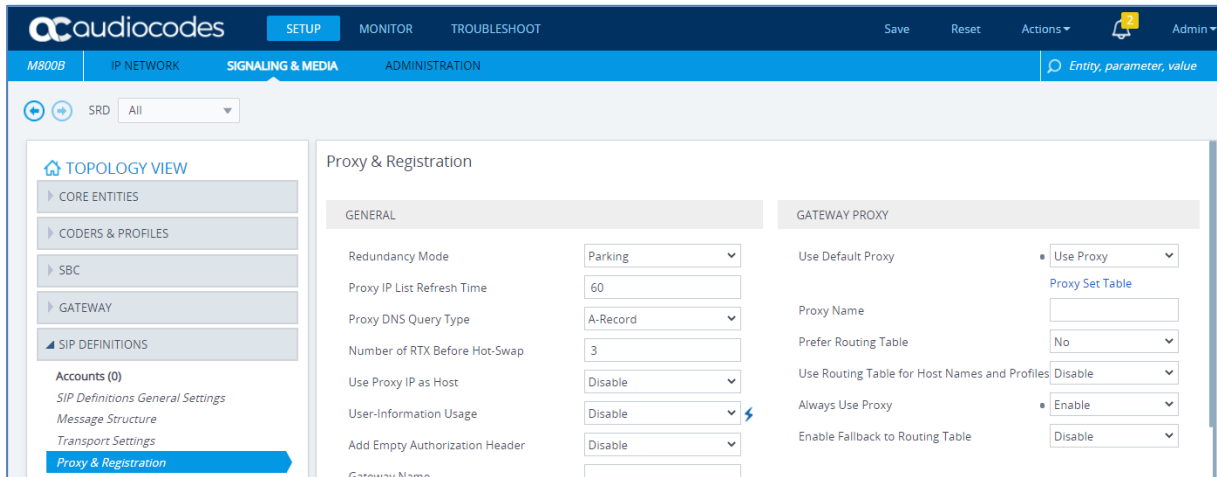
4. Click **Apply**.

3.10.7 Configure Registration Parameters

This section describes how to configure the SIP Proxy and Registration parameters.

- **To configure the SIP Proxy & Registration parameters:**
 1. Open the SIP Proxy & Registration Parameters page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Proxy & Registration**).
 2. From the 'Use Default Proxy' drop-down list, select **Use Proxy**.
 3. From the 'Always Use Proxy' drop-down list, select **Enable**.

Figure 3-25: Configuring Proxy & Registration Parameters



4. Click **Apply**.

A AudioCodes INI File

The *ini* configuration file of the SBC, corresponding to the Web-based configuration as described in Section 3 on page 23, is shown below:



Note: To load or save an *ini* file, use the Configuration File page (**Setup** menu > **Administration** tab > **Maintenance** folder > **Configuration File**).

```
;*****  
;** Ini File **  
;*****  
  
[SYSTEM Params]  
  
SyslogServerIP = 192.168.250.164  
EnableSyslog = 1  
;VpFileLastUpdateTime is hidden but has non-default value  
NTPServerIP = '0.0.0.0'  
  
[BSP Params]  
  
PCMLawSelect = 1  
UdpPortSpacing = 10  
EnterCpuOverloadPercent = 99  
ExitCpuOverloadPercent = 95  
  
[Analog Params]  
  
[ControlProtocols Params]  
  
AdminStateLockControl = 0  
  
[PSTN Params]  
  
ProtocolType = 1  
FramingMethod = c  
LineCode = 2  
  
[Voice Engine Params]  
  
CallProgressTonesFilename = 'br_tons_m600_m1k.dat'  
  
[WEB Params]  
  
HTTPSCipherString = 'RC4:EXP'  
Languages = 'en-US', '', '', '', '', '', '', ''  
  
[SIP Params]  
  
MAXDIGITS = 15  
ISPROXYUSED = 1  
GWDEBUGLEVEL = 5
```

```

ALWAYSSENDDTOPROXY = 1
ALLOWUNCLASSIFIEDCALLS = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
ANSWERDETECTORCMD = 10486144
SYSLOGOPTIMIZATION = 1
FIRSTTXDTMFOPTION = 4

[SNMP Params]

[ DeviceTable ]

FORMAT Index = VlanID, UnderlyingInterface, DeviceName, Tagging, MTU;
DeviceTable 0 = 1, "GROUP_1", "vlan 1", 0, 1500;

[ \DeviceTable ]

[ InterfaceTable ]

FORMAT Index = ApplicationTypes, InterfaceMode, IPAddress, PrefixLength,
Gateway, InterfaceName, PrimaryDNSServerIPAddress,
SecondaryDNSServerIPAddress, UnderlyingDevice;
InterfaceTable 0 = 6, 10, 192.168.230.49, 24, 192.168.230.1, "LAN_IF",
8.8.8.8, , "vlan 1";

[ \InterfaceTable ]

[ AudioCodersGroups ]

FORMAT Index = Name;
AudioCodersGroups 0 = "AudioCodersGroups_0";

[ \AudioCodersGroups ]

[ CpMediaRealm ]

FORMAT Index = MediaRealmName, IPv4IF, IPv6IF, RemoteIPv4IF,
RemoteIPv6IF, PortRangeStart, MediaSessionLeg, PortRangeEnd,
TCPPortRangeStart, TCPPortRangeEnd, IsDefault, QoeProfile, BWProfile,
TopologyLocation;
CpMediaRealm 0 = "DefaultRealm", "LAN_IF", "", "", "", 6000, 256, 8559,
0, 0, 1, "", "", 0;

[ \CpMediaRealm ]

[ SBCRoutingPolicy ]

FORMAT Index = Name, LCREnable, LCRAverageCallLength, LCRDefaultCost,
LdapServerGroupName;
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 0, 1, "";

[ \SBCRoutingPolicy ]
    
```

```

[ SRD ]

FORMAT Index = Name, BlockUnRegUsers, MaxNumOfRegUsers,
EnableUnAuthenticatedRegistrations, SharingPolicy, UsedByRoutingServer,
SBCOperationMode, SBCRoutingPolicyName, SBCDialPlanName,
AdmissionProfile;
SRD 0 = "DefaultSRD", 0, -1, 1, 0, 0, 0, "Default_SBCRoutingPolicy", "",
"";

[ \SRD ]

[ SIPInterface ]

FORMAT Index = InterfaceName, NetworkInterface,
SCTPSecondaryNetworkInterface, ApplicationType, UDPPort, TCPPort,
TLSPort, SCTPPort, AdditionalUDPPorts, AdditionalUDPPortsMode, SRDName,
MessagePolicyName, TLSContext, TLSMutualAuthentication,
TCPKeepaliveEnable, ClassificationFailureResponseType,
PreClassificationManSet, EncapsulatingProtocol, MediaRealm,
SBCDirectMedia, BlockUnRegUsers, MaxNumOfRegUsers,
EnableUnAuthenticatedRegistrations, UsedByRoutingServer,
TopologyLocation, PreParsingManSetName, AdmissionProfile,
CallSetupRulesSetId;
SIPInterface 0 = "SIPInterface_0", "LAN_IF", "", 0, 5060, 5060, 5061, 0,
"", 0, "DefaultSRD", "", "default", -1, 0, 500, -1, 0, "DefaultRealm", 0,
-1, -1, -1, 0, 0, "", "", -1;
SIPInterface 1 = "SIPInterface_1", "LAN_IF", "", 2, 5070, 5070, 5071, 0,
"", 0, "DefaultSRD", "", "default", -1, 0, 500, -1, 0, "DefaultRealm", 0,
-1, -1, -1, 0, 0, "", "", -1;

[ \SIPInterface ]

[ ProxySet ]

FORMAT Index = ProxyName, EnableProxyKeepAlive, ProxyKeepAliveTime,
ProxyLoadBalancingMethod, IsProxyHotSwap, SRDName, ClassificationInput,
TLSContextName, ProxyRedundancyMode, DNSResolveMethod,
KeepAliveFailureResp, GWIPv4SIPInterfaceName, SBCIPv4SIPInterfaceName,
GWIPv6SIPInterfaceName, SBCIPv6SIPInterfaceName, MinActiveServersLB,
SuccessDetectionRetries, SuccessDetectionInterval,
FailureDetectionRetransmissions;
ProxySet 0 = "ProxySet_0", 0, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "",
"SIPInterface_0", "", "", "", 1, 1, 10, -1;
ProxySet 1 = "Genesys", 0, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "", "",
"SIPInterface_1", "", "", 1, 1, 10, -1;
ProxySet 2 = "Verint", 0, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "", "",
"SIPInterface_1", "", "", 1, 1, 10, -1;

[ \ProxySet ]

[ IPGroup ]

FORMAT Index = Type, Name, ProxySetName, VoiceAICConnector, SIPGroupName,
ContactUser, SipReRoutingMode, AlwaysUseRouteTable, SRDName, MediaRealm,
InternalMediaRealm, ClassifyByProxySet, ProfileName, MaxNumOfRegUsers,
InboundManSet, OutboundManSet, RegistrationMode, AuthenticationMode,
MethodList, SBCServerAuthType, OAuthHTTPService, EnableSBCClientForking,
SourceUriInput, DestUriInput, ContactName, Username, Password, UIFormat,
QOEProfile, BWProfile, AlwaysUseSourceAddr, MsgManUserDef1,

```

```

MsgManUserDef2, SIPConnect, SBCPSAPMode, DTLSContext,
CreatedByRoutingServer, UsedByRoutingServer, SBCOperationMode,
SBCRouteUsingRequestURIPort, SBCKeepOriginalCallID, TopologyLocation,
SBCDialPlanName, CallSetupRulesSetId, Tags, SBCUserStickiness,
UserUDPPortAssignment, AdmissionProfile, ProxyKeepAliveUsingIPG,
SBCAltRouteReasonsSetName, TeamsLocalMediaOptimization,
TeamsLocalMOInitialBehavior, SIPSourceHostName;
IPGroup 0 = 0, "Default_IPG", "ProxySet_0", "", "", "", -1, 0,
"DefaultSRD", "DefaultRealm", "", 1, "", -1, -1, -1, 0, 0, "", -1, "", 0,
-1, -1, "", "", "$1$gQ==", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -
1, 0, 0, 0, "", -1, "", 0, 0, "", 0, "", 0, 0, "";
IPGroup 1 = 0, "Genesys", "Genesys", "", "", "", -1, 0, "DefaultSRD",
"DefaultRealm", "", 1, "", -1, 1, 1, 0, 0, "", -1, "", 0, -1, -1, "", "",
"$1$gQ==", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -1, 0, 0, 0, "",
-1, "", 0, 0, "", 0, "", 0, 0, "";
IPGroup 2 = 0, "Verint", "Verint", "", "", "", -1, 0, "DefaultSRD",
"DefaultRealm", "", 1, "", -1, 1, 1, 0, 0, "", -1, "", 0, -1, -1, "", "",
"$1$gQ==", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -1, 0, 0, 0, "",
-1, "", 0, 0, "", 0, "", 0, 0, "";

[ \IPGroup ]

[ PREFIX ]

FORMAT Index = RouteName, DestinationPrefix, DestAddress, SourcePrefix,
ProfileName, MeteringCodeName, DestPort, DestIPGroupName, TransportType,
SrcTrunkGroupID, DestSIPInterfaceName, CostGroup, ForkingGroup,
CallSetupRulesSetId, DestTags, SrcTags;
PREFIX 0 = "", "8177", "192.168.230.49", "*", "", "", 5070, "", -1, -1,
"", "", -1, -1, "", "";

[ \PREFIX ]

[ TrunkGroup ]

FORMAT Index = TrunkGroupNum, FirstTrunkId, FirstBChannel, LastBChannel,
FirstPhoneNumber, ProfileName, LastTrunkId, Module;
TrunkGroup 0 = 1, 0, 1, 31, "9990", "", 0, 1;

[ \TrunkGroup ]

[ NumberMapIp2Tel ]

FORMAT Index = ManipulationName, DestinationPrefix, SourcePrefix,
SourceAddress, SrcHost, DestHost, NumberType, NumberPlan, RemoveFromLeft,
RemoveFromRight, LeaveFromRight, Prefix2Add, Suffix2Add,
IsPresentationRestricted, SrcIPGroupName;
NumberMapIp2Tel 0 = "", "00*", "*", "*", "*", "*", 255, 255, 1, 0, 255,
"", "", 255, "Any";

[ \NumberMapIp2Tel ]

[ NumberMapTel2Ip ]

FORMAT Index = ManipulationName, DestinationPrefix, SourcePrefix,
NumberType, NumberPlan, RemoveFromLeft, RemoveFromRight, LeaveFromRight,

```



```

Prefix2Add, Suffix2Add, IsPresentationRestricted, SrcTrunkGroupID,
DestIPGroupName;
NumberMapTel2Ip 0 = "", "4000", "*", 255, 255, 0, 0, 255, "", "", 0, -1,
"Any";
NumberMapTel2Ip 1 = "", "8178", "*", 255, 255, 4, 0, 255, "4003", "",
255, -1, "Any";
NumberMapTel2Ip 2 = "", "8177", "*", 255, 255, 4, 0, 255, "4002", "", 0,
-1, "Any";

[ \NumberMapTel2Ip ]

[ PstnPrefix ]

FORMAT Index = RouteName, DestPrefix, TrunkGroupId, SourcePrefix,
SourceAddress, ProfileName, SrcIPGroupName, DestHostPrefix,
SrcHostPrefix, SrcSIPInterfaceName, TrunkId, CallSetupRulesSetId,
DestType, DestTags, SrcTags;
PstnPrefix 0 = "", "*", 1, "*", "*", "", "", "", "", "Any", -1, -1, 0,
"", "";

[ \PstnPrefix ]

[ ProxyIp ]

FORMAT Index = ProxySetId, ProxyIpIndex, IpAddress, TransportType,
Priority, Weight;
ProxyIp 0 = "1", 0, "192.168.230.52", 0, 0, 0;
ProxyIp 1 = "2", 0, "192.168.230.57:5080", 0, 0, 0;
ProxyIp 2 = "0", 0, "192.168.230.49:5070", 0, 0, 0;

[ \ProxyIp ]

[ IP2IPRouting ]

FORMAT Index = RouteName, RoutingPolicyName, SrcIPGroupName,
SrcUsernamePrefix, SrcHost, DestUsernamePrefix, DestHost, RequestType,
MessageConditionName, ReRouteIPGroupName, Trigger, CallSetupRulesSetId,
DestType, DestIPGroupName, DestSIPInterfaceName, DestAddress, DestPort,
DestTransportType, AltRouteOptions, GroupPolicy, CostGroup, DestTags,
ModifiedDestUserName, SrcTags, IPGroupSetName, RoutingTagName,
InternalAction;
IP2IPRouting 0 = "", "Default_SBCRoutingPolicy", "Any", "*", "*",
"[0,00]4009xxxx", "*", 0, "", "Any", 0, -1, 1, "", "", "192.168.230.49",
5060, -1, 0, 0, "", "", "", "", "", "default", "";
IP2IPRouting 1 = "", "Default_SBCRoutingPolicy", "Any", "*", "*", "*",
"*, 0, "", "Any", 0, -1, 0, "Genesys", "", "", 0, -1, 0, 0, "", "", "",
"", "", "default", "";

[ \IP2IPRouting ]

[ MessageManipulations ]

FORMAT Index = ManipulationName, ManSetID, MessageType, Condition,
ActionSubject, ActionType, ActionValue, RowRole;
MessageManipulations 0 = "", 1, "Refer.Request", "Header.Refer-To regex
(.*)(>)(.*)", "Header.Refer-To", 2,

```

```

"$1+'?CONTA='+Header.CONTA.Content+'&AGENCIA='+Header.AGENCIA.Content+'&N
_CARTAO='+Header.N_CARTAO.Content+'&CPF='+Header.CPF.Content+$2+$3", 0;
MessageManipulations 1 = "", 1, "Refer.Request", "Header.Refer-To regex
(.*)(>)(.*)", "Header.Refer-To", 2,
"$1+'&MCI='+Header.MCI.Content+'&SERVICO='+Header.SERVICO.Content+'&X-
Genesys-CallUUID='+Header.X-Genesys-CallUUID.Content+$2+$3", 0;

[ \MessageManipulations ]

[ GwRoutingPolicy ]

FORMAT Index = Name, LCREnable, LCRAverageCallLength, LCRDefaultCost,
LdapServerGroupName;
GwRoutingPolicy 0 = "GwRoutingPolicy", 0, 0, 1, "";

[ \GwRoutingPolicy ]

[ ResourcePriorityNetworkDomains ]

FORMAT Index = Name, Ip2TelInterworking;
ResourcePriorityNetworkDomains 1 = "dsn", 0;
ResourcePriorityNetworkDomains 2 = "dod", 0;
ResourcePriorityNetworkDomains 3 = "drsn", 0;
ResourcePriorityNetworkDomains 5 = "uc", 1;
ResourcePriorityNetworkDomains 7 = "cuc", 0;

[ \ResourcePriorityNetworkDomains ]

[ SIPRecRouting ]

FORMAT Index = RecordedIPGroupName, RecordedSourcePrefix,
RecordedDestinationPrefix, ConditionName, PeerIPGroupName,
PeerTrunkGroupID, Caller, SRSIPGroupName, SRSRedundantIPGroupName;
SIPRecRouting 1 = "Genesys", "*", "*", "", "Any", -1, 2, "Verint", "";

[ \SIPRecRouting ]

[ MaliciousSignatureDB ]

FORMAT Index = Name, Pattern;
MaliciousSignatureDB 0 = "SIPVicious", "Header.User-Agent.content prefix
'friendly-scanner';
MaliciousSignatureDB 1 = "SIPScan", "Header.User-Agent.content prefix
'sip-scan";
MaliciousSignatureDB 2 = "Smapi", "Header.User-Agent.content prefix
'smapi";
MaliciousSignatureDB 3 = "Sipsak", "Header.User-Agent.content prefix
'sipsak";
MaliciousSignatureDB 4 = "Sipcli", "Header.User-Agent.content prefix
'sipcli";
MaliciousSignatureDB 5 = "Sivus", "Header.User-Agent.content prefix
'SIVuS";
MaliciousSignatureDB 6 = "Gulp", "Header.User-Agent.content prefix
'Gulp";
    
```

```
MaliciousSignatureDB 7 = "Sipv", "Header.User-Agent.content prefix
'sipv'";
MaliciousSignatureDB 8 = "Sundayddr Worm", "Header.User-Agent.content
prefix 'sundayddr'";
MaliciousSignatureDB 9 = "VaxIPUserAgent", "Header.User-Agent.content
prefix 'VaxIPUserAgent'";
MaliciousSignatureDB 10 = "VaxSIPUserAgent", "Header.User-Agent.content
prefix 'VaxSIPUserAgent'";
MaliciousSignatureDB 11 = "SipArmyKnife", "Header.User-Agent.content
prefix 'siparmyknife'";

[ \MaliciousSignatureDB ]

[ AudioCoders ]

FORMAT Index = AudioCodersGroupId, AudioCodersIndex, Name, pTime, rate,
PayloadType, Sce, CoderSpecific;
AudioCoders 0 = "AudioCodersGroups_0", 0, 3, 2, 19, -1, 0, "";
AudioCoders 1 = "AudioCodersGroups_0", 1, 1, 2, 90, -1, 0, "";
AudioCoders 2 = "AudioCodersGroups_0", 2, 2, 2, 90, -1, 0, "";

[ \AudioCoders ]
```

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