Configuration Note

AudioCodes Professional Services – Interoperability Lab

Genesys Cloud Contact Center and Generic SIP Trunk using AudioCodes Mediant™ SBC

Version 7.4

Table of Contents

This page is intentionally left blank.

Notice

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, AudioCodes cannot guarantee accuracy of printed material after the Date Published nor can it accept responsibility for errors or omissions. Updates to this document can be downloaded from https://www.audiocodes.com/library/technical-documents.

This document is subject to change without notice.

Date Published: November-15-2021

WEEE EU Directive

Pursuant to the WEEE EU Directive, electronic and electrical waste must not be disposed of with unsorted waste. Please contact your local recycling authority for disposal of this product.

Customer Support

Customer technical support and services are provided by AudioCodes or by an authorized AudioCodes Service Partner. For more information on how to buy technical support for AudioCodes products and for contact information, please visit our website at https://www.audiocodes.com/services-support/maintenance-and-support.

Stay in the Loop with AudioCodes

Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used.

Document Revision Record

LTRT	Description
39478	Initial document release.

Documentation Feedback

AudioCodes continually strives to produce high quality documentation. If you have any comments (suggestions or errors) regarding this document, please fill out the Documentation Feedback form on our Web site at https://online.audiocodes.com/documentation-feedback.

This page is intentionally left blank.

1 Introduction

This Configuration Note describes how to set up the AudioCodes Enterprise Session Border Controller (hereafter, referred to as *SBC*) for interworking between Generic SIP Trunk and the Genesys Cloud Contact Center environment.

You can also use AudioCodes' SBC Wizard tool to automatically configure the SBC based on this interoperability setup. However, it is recommended to read through this document to better understand the various configuration options. For more information on AudioCodes' SBC Wizard including the download option, visit AudioCodes Web site at https://www.audiocodes.com/partners/sbc-interoperability-list.

1.1 Intended Audience

This document is intended for engineers, or AudioCodes and Generic partners who are responsible for installing and configuring Generic SIP Trunk and Genesys Cloud Contact Center for enabling VoIP calls using AudioCodes SBC.

1.2 About Genesys Cloud Contact Center

Genesys Cloud Contact Center Solutions allow companies to manage customer requirements effectively by routing customers to appropriate resources and agents through IVR and consolidated cross-channel management of a customer's interactions. Sophisticated profiling, outbound voice and performance management enables companies to provide very personalized customer care and delivery.

1.3 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. The SBC can be offered as a Virtualized SBC, supporting the following platforms: Hyper-V, AWS, AZURE, AWP, KVM and VMWare.

This page is intentionally left blank.

2 Component Information

2.1 AudioCodes SBC Version

Table 2-1: AudioCodes SBC Version

SBC Vendor	AudioCodes
Models	▪ Mediant 500/L Gateway & E-SBC ▪ Mediant 800B/C Gateway & E-SBC ▪ Mediant 1000B Gateway & E-SBC ▪ Mediant 2600 E-SBC ▪ Mediant 4000/B SBC ▪ Mediant 9000/9030/9080 SBC ▪ Mediant Software SBC (VE/SE/CE)
Software Version	7.40A.100.238 or later
Protocol	SBC can be configured to work SIP/UDP, SIP/TCP or SIP/TLS towards Generic SIP Trunk and / or Genesys Cloud Contact Center, depending on customer requirements.
Additional Notes	None

2.2 Generic SIP Trunking Version

Table 2-2: Generic Version

Vendor/Service Provider	Generic
SSW Model/Service	
Software Version	
Protocol	SIP
Additional Notes	None

2.3 Genesys Cloud Contact Center Version

Table 2-3: Genesys Cloud Contact Center Version

Vendor	Genesys
Model	Contact Center
Software Version	November 3, 2021
Protocol	SIP
Additional Notes	None

2.4 Interoperability Test Topology

The Genesys Cloud Contact Center is connected to the Generic SIP Trunk via an SBC.

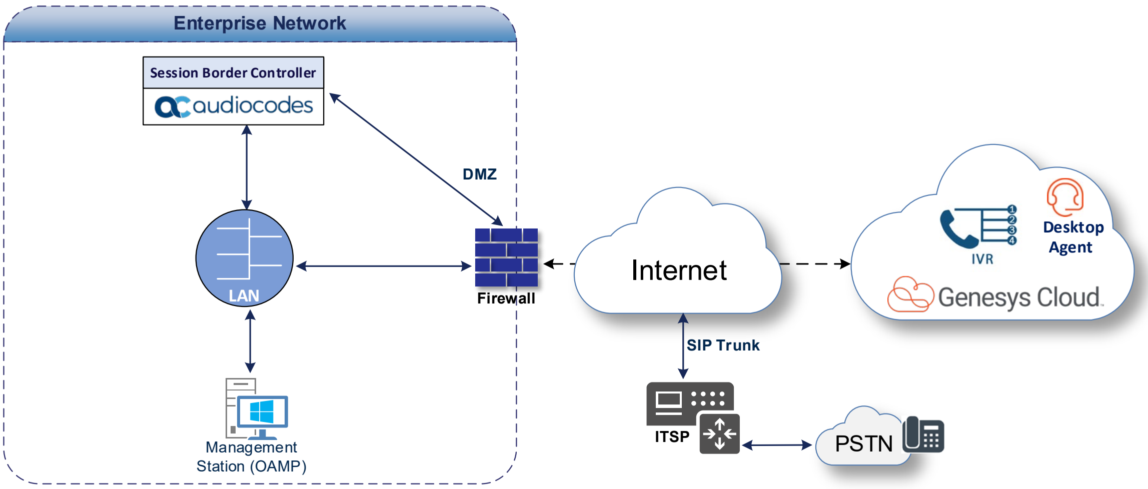
Note: Contact your Genesys Contact Center support channel for more information about topological scenarios.

The interoperability testing between AudioCodes SBC and Generic SIP Trunk with Genesys Cloud Contact Center was done using the following topology setup:

- The Enterprise was deployed with a Genesys Cloud Contact Center as a service using robust Contact Center functionality and interactive voice response (IVR) to efficiently connect customers with the right agents and information at the right time.
- The Enterprise SBC connected the Genesys Cloud Contact Center with the Public PSTN via the Generic SIP Trunk, as an External Trunk over the public network.
- AudioCodes SBC is implemented to interconnect between the SIP Trunk and Genesys Cloud Contact Center
    - **Session:** Real-time voice session using the IP-based Session Initiation Protocol (SIP).
    - **Border:** IP-to-IP network border – both Generic SIP Trunk and the Genesys Cloud Contact Center are located in the public network.

The figure below illustrates this interoperability test topology:

Figure 2-1: Interoperability Test Topology between SBC and Genesys Cloud Contact Center with Generic SIP Trunk



2.4.1 Environment Setup

The interoperability test topology includes the following environment setup:

Table 2-4: Environment Setup

Area	Setup
Network	Both, Generic SIP Trunk and Genesys Cloud Contact Center are located on the Enterprise's (or Service Provider's) WAN
Signaling Transcoding	▪ Genesys Cloud Contact Center operates with SIP-over-UDP, TCP or SIP-over-TLS transport type ▪ Generic SIP Trunk transport type depends on customer requirements ▪ The interoperability test environment used SIP-over-UDP
Codecs Transcoding	▪ Genesys Cloud Contact Center supports OPUS, G.711U-law and G.711A-law coders ▪ Generic SIP Trunk supports G.711U-law, G.711A-law, and G.729 coders
Media Transcoding	Connection to both, Generic SIP Trunk and Genesys Cloud Contact Center can operate with RTP or SRTP media types, depending on the signaling transport type.

Note: The configuration data used in this document, such as IP addresses and FQDNs are used for example purposes only. This data should be configured according to the site specifications.

2.4.2 Known Limitations

There were no limitations observed in the interoperability tests done for the AudioCodes SBC interworking between Genesys Cloud Contact Center and Generic SIP Trunk.

This page is intentionally left blank.

3 Configuring Genesys Cloud

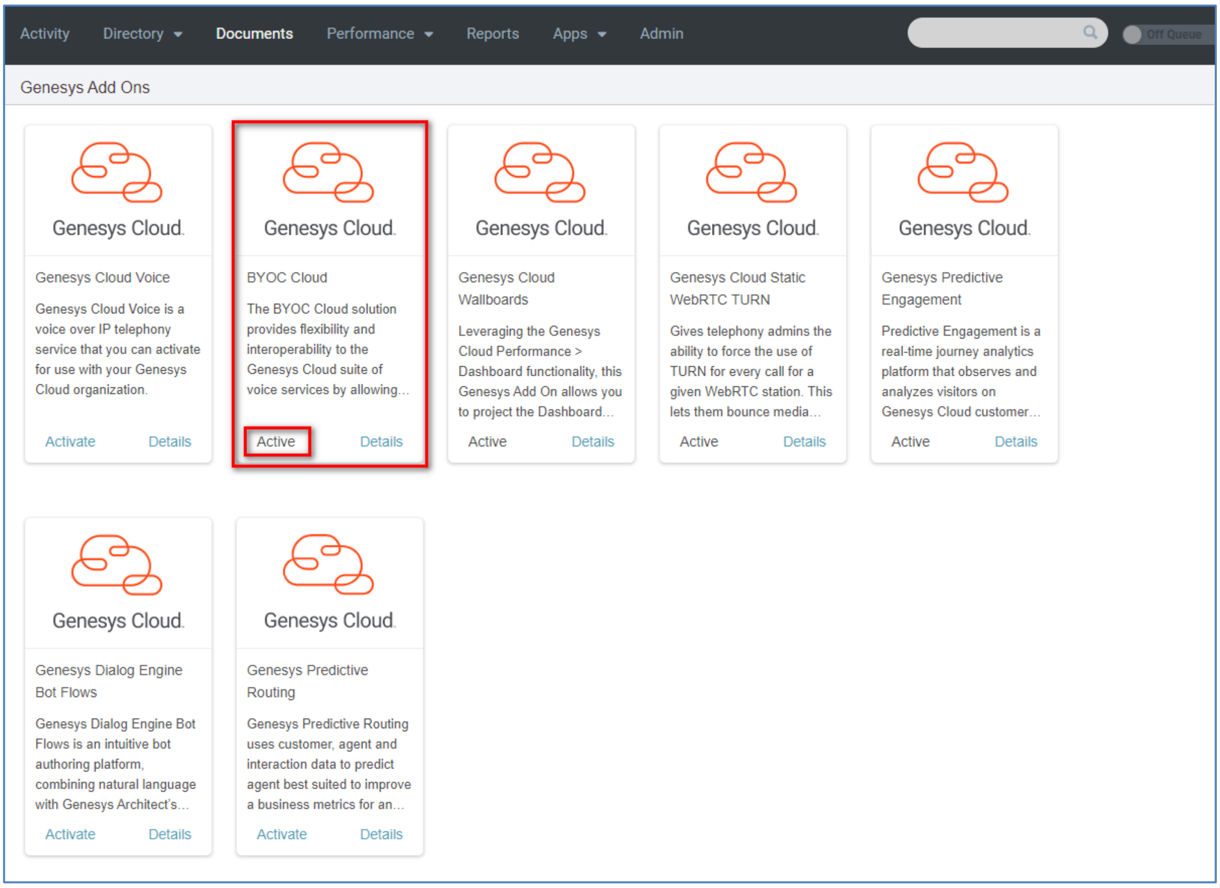
This section describes how to configure new SIP Trunk (BYOC Cloud) on Genesys Cloud Contact Center to operate with Generic SIP Trunk through AudioCodes SBC.

3.1 Prerequisites

Before you begin configuration, make sure that the BYOC Cloud add-on is available and active in your Genesys Cloud organization:

1. From Genesys Desktop, in Admin, select **Genesys Add-Ons** in **Account Settings**.
2. If the BYOC Cloud option is available in your org, you should see a **BYOC Cloud** tile. Otherwise, please reach out to your Genesys contact to ask for it.
3. Make sure to **Activate** this add-on (it will then appear as **Active**).

Figure 3-1: Activate Genesys BYOC Cloud Add-Ons



A Genesys Cloud is deployed in a single AWS region. See [AWS regions for Genesys Cloud deployment] for a list of supported regions, or to determine in which region your Genesys Cloud organization is deployed.

If you have a mechanism to control the IP Addresses, allow it to access to your device (SIP Access Control List):

For SIP, you can find the list of the BYOC Cloud public SIP IP addresses, for each AWS/Genesys Cloud region, here.

If you also need to define a list for the media traffic (Genesys Cloud Media), see IP addresses for the firewall allowlist.

3.2 Adding a New BYOC Cloud Trunk

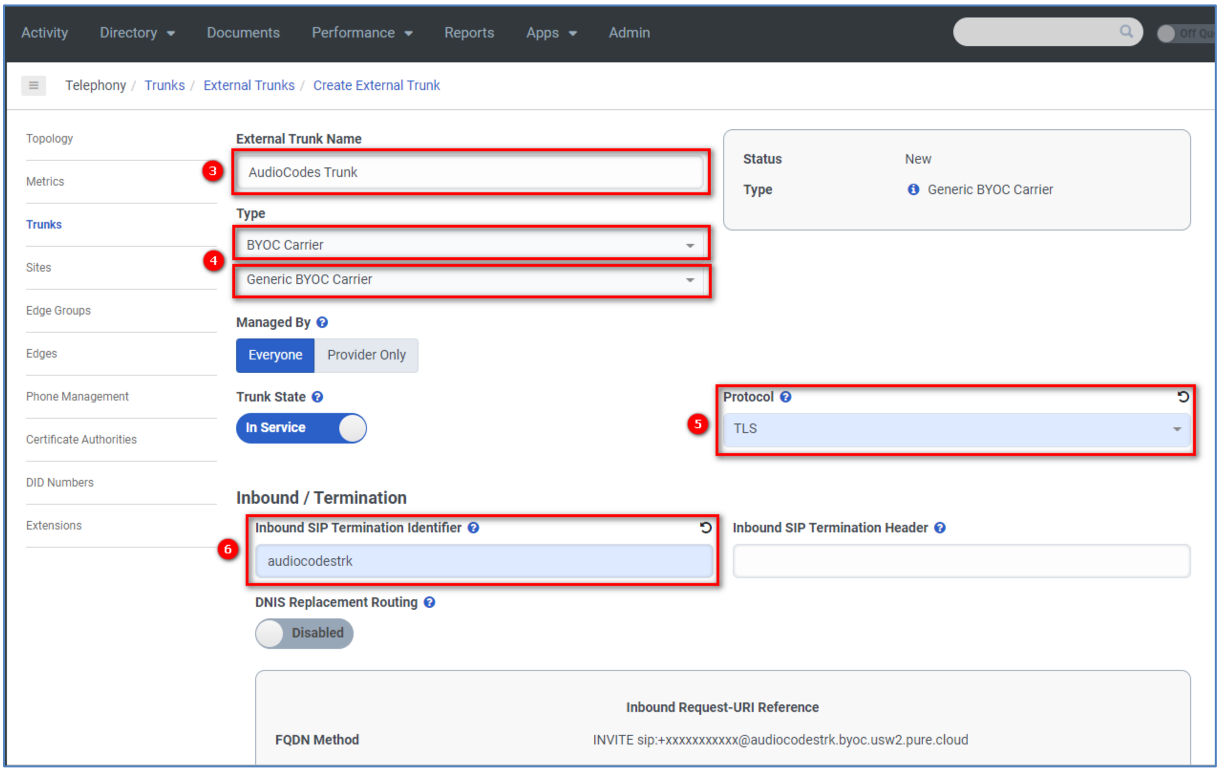
The procedure below describes how add external trunk as Genesys Cloud Bring Your Own Carrier (BYOC) Cloud Trunk.

➢ **To add a new BYOC Cloud Trunk:**

1. From the Genesys Desktop, in Admin, select **Trunks** in **Telephony**.
2. Click on the **Create New** button (**External Trunks** tab).
3. In the 'External Trunk Name' field, enter a name for your External Trunk (e.g., *AudioCodes Trunk*).
4. From the 'Type' drop-down list, select **BYOC Carrier**.
5. From the 'sub-type' drop-down list, select **Generic BYOC Carrier**.
6. From the 'Protocol' drop-down list, select the Transport Protocol used for SIP: UDP/TCP/TLS (e.g., TLS).
7. Under the **Inbound / Termination** group, in the 'Inbound SIP Termination Identifier' field, enter a value (e.g., *audiocodestrk*).

This is the unique identifier of your trunk. It is used for inbound calls to Genesys Cloud, to uniquely identify a customer (Genesys Cloud org.) and a trunk, in a specific region (e.g., *sip:+xxxxxxxxxxx@audiocodestrk.byoc.usw2.pure.cloud*).

Figure 3-2: Adding the new BYOC Cloud Trunk

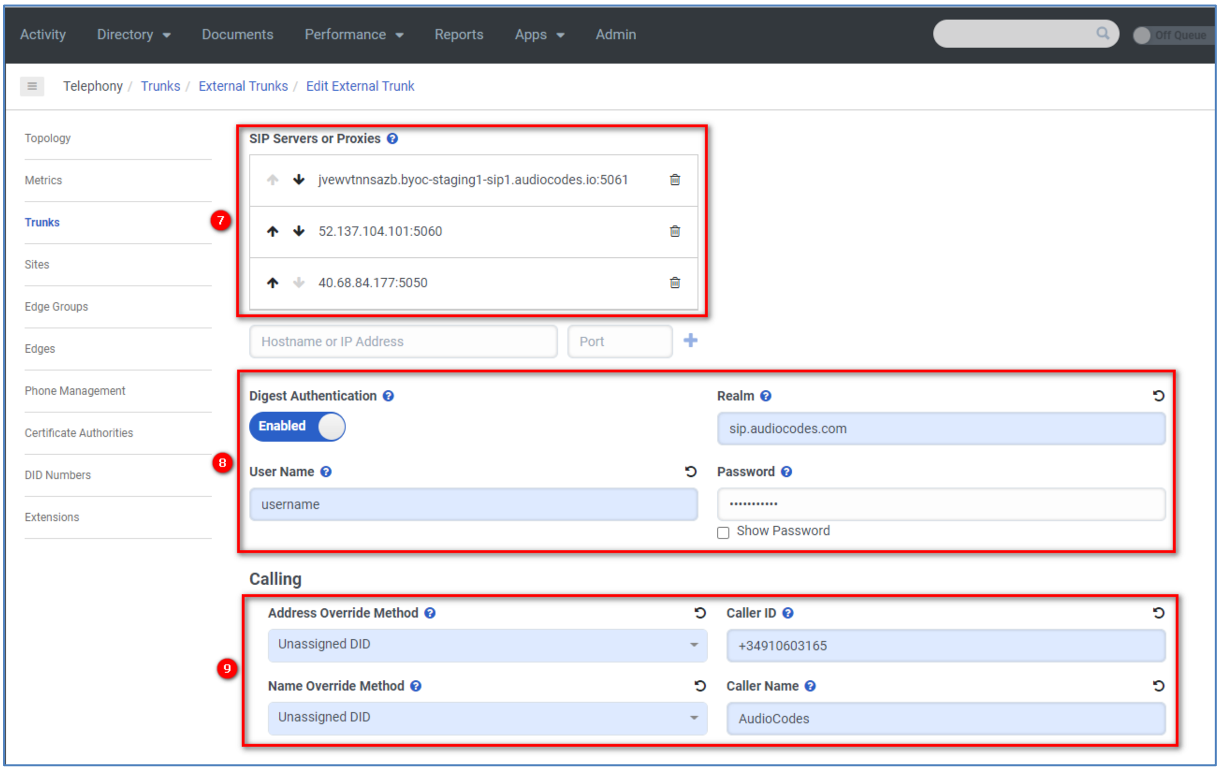


8. Under the **SIP Servers or Proxies group,** under **Outbound**, set the IP Address(es) or FDQN(s) and Port for SIP (to reach AudioCodes SBC).
9. If you have decided to leverage Digest Authentication, under **Outbound**, enable **Digest Authentication** and set the values for **Realm**, **User Name** and **Password**.
10. Under **Outbound**, in **Calling**, from the 'Address Override Method' drop-down list, select **Unassigned DID**.

This is to preserve the CLI of the caller when the call is transferred from Genesys Cloud to your device (from the header).

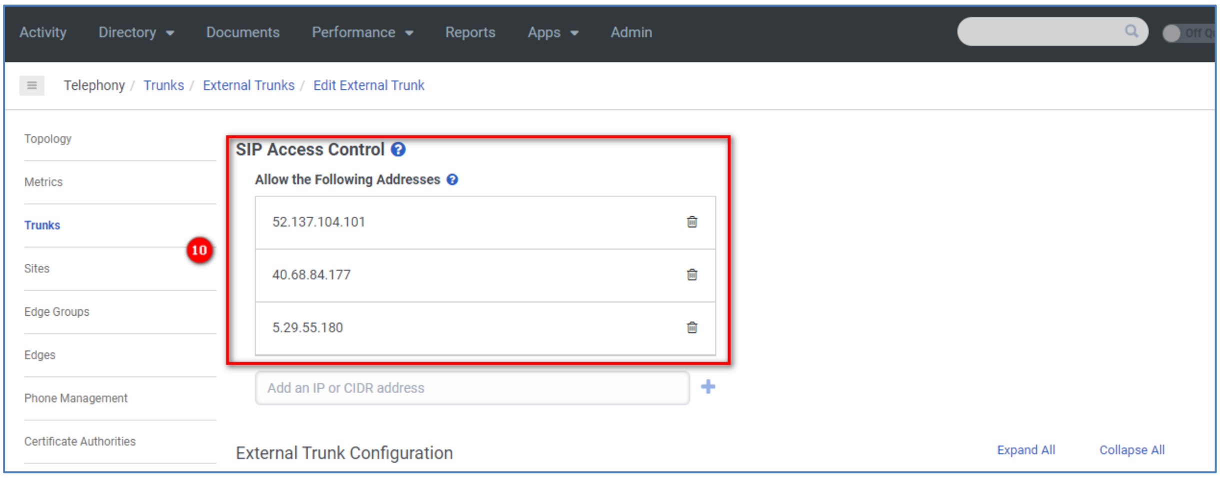
You will also need to define a **Caller ID** value (e.g., an E.164 number +34910603165) which is used as a default value if the CLI of the caller can't be propagated to your system (e.g., a Genesys Cloud Contact Center Agent, with no assigned number, makes a call to your trunk).

Figure 3-3: Definitions of the New BYOC Cloud Trunk



11. Under the **SIP Access Control** group, define the list of IP addresses that your device uses for SIP (e.g., 52.137.104.101). This list is to control access on the Genesys Cloud side.

Figure 3-4: SIP Access Control List



12. Under the **Identity** group, for **Calling** and **Called**, disable **Address Omit + Prefix**. If disabled, when a call comes to the Genesys Cloud and is transferred to your trunk (via the configured BYOC Cloud trunk), the + prefix will be preserved in From and To header. If enabled, the + prefix will be removed (if it is present in the From/To numbers).

Figure 3-5: Omit + Prefix in Calling Number

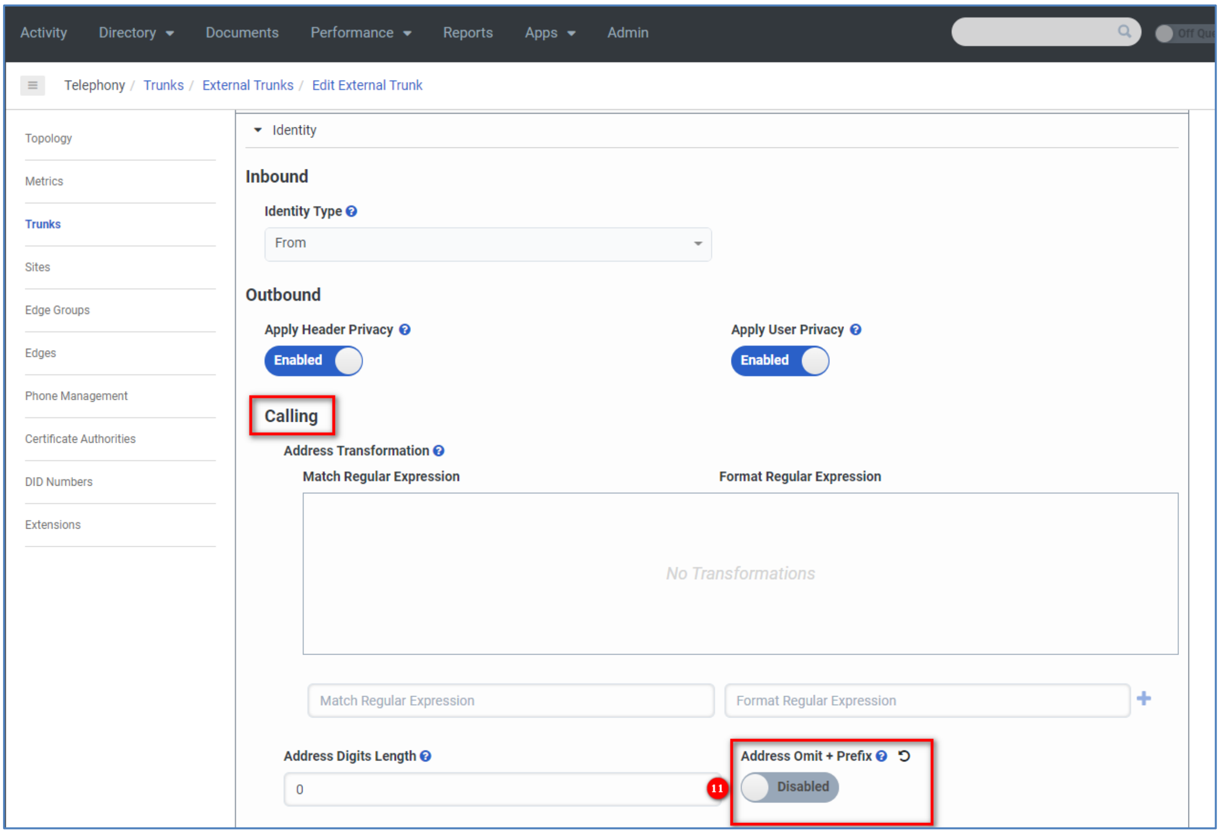
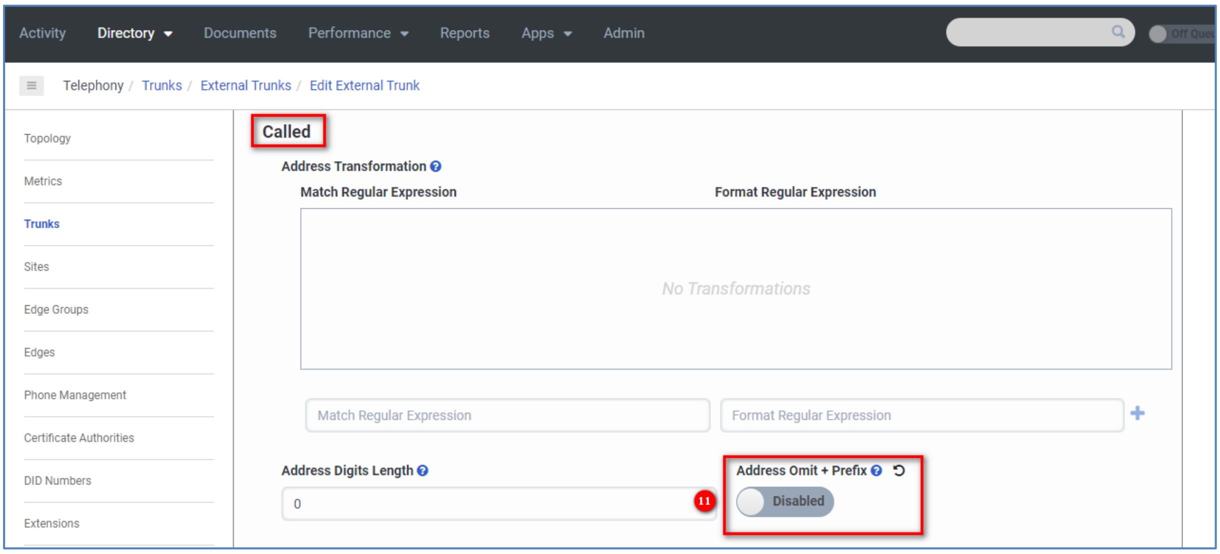
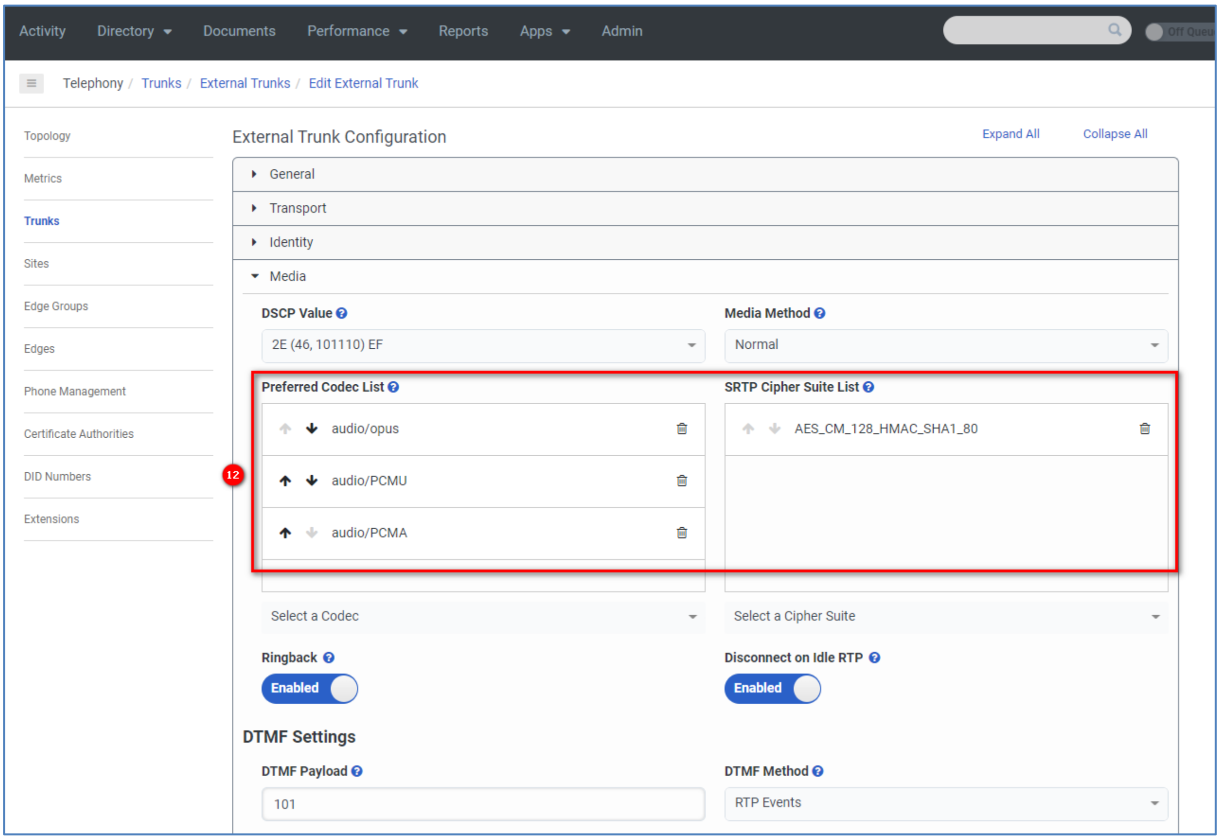


Figure 3-6: Omit + Prefix in Called Number



13. Under the **Media** group, define your **Preferred Codec List**, based on the audio codecs your trunk supports (e.g., audio/opus, audio/PCMU, audio/PCMA). If SRTP is used, you can define the list of specific ciphers.

Figure 3-7: Codec List

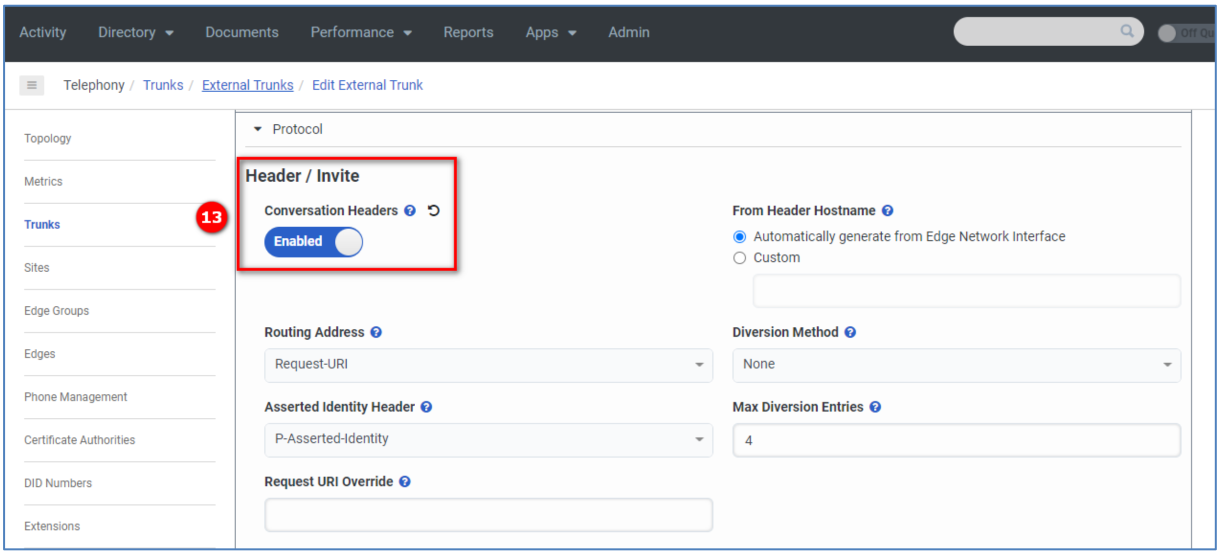


14. Under the **Protocol** group, enable the **Conversation Headers** toggle. By enabling this feature, Genesys Cloud automatically attaches the ConversationId, as a custom SIP header, in the outgoing SIP messages:

   a. For outbound calls (Genesys Cloud to External SIP Trunk), Genesys Cloud adds a **x-inin-cnv** header containing the ConversationId on SIP INVITE.

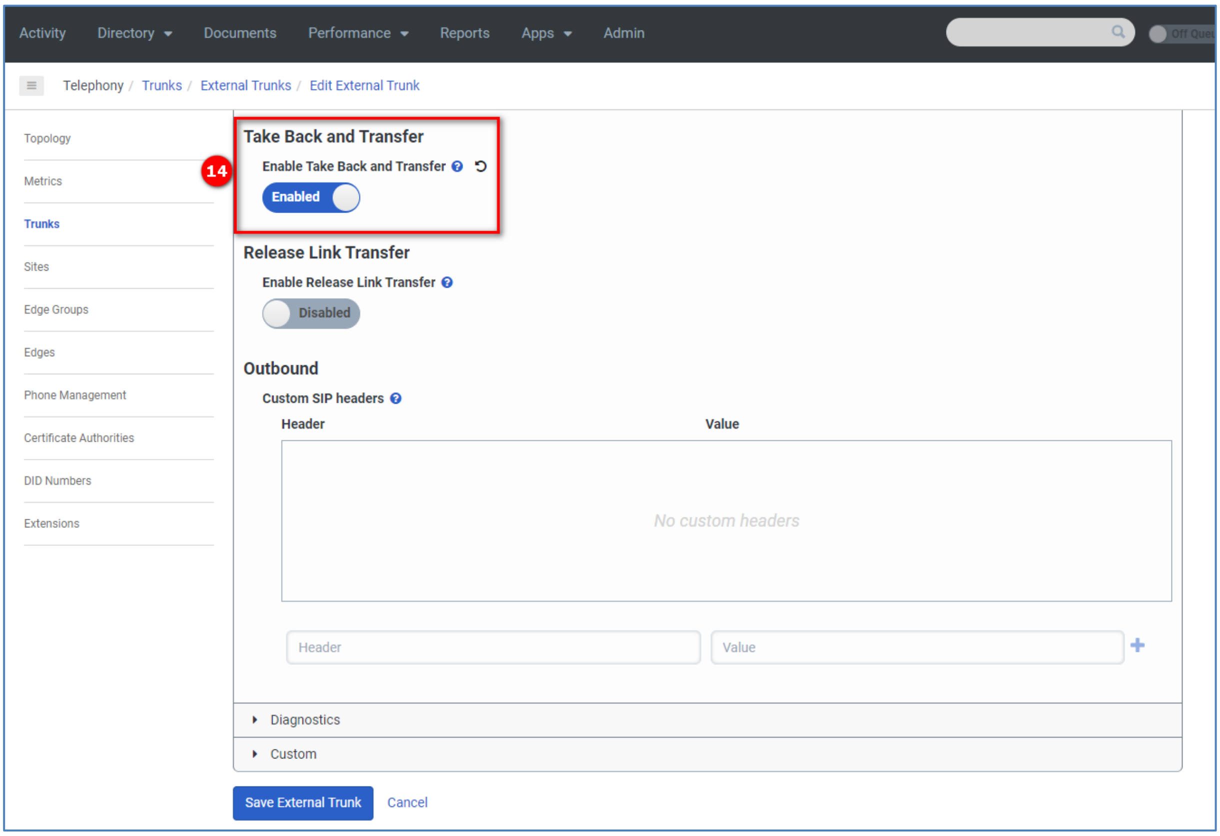
   b. For inbound calls (External SIP Trunk to Genesys Cloud), Genesys Cloud adds a **x-inin-cnv** header containing the ConversationId in the 200 OK (following the received SIP INVITE).

Figure 3-8: Enable Conversation Headers



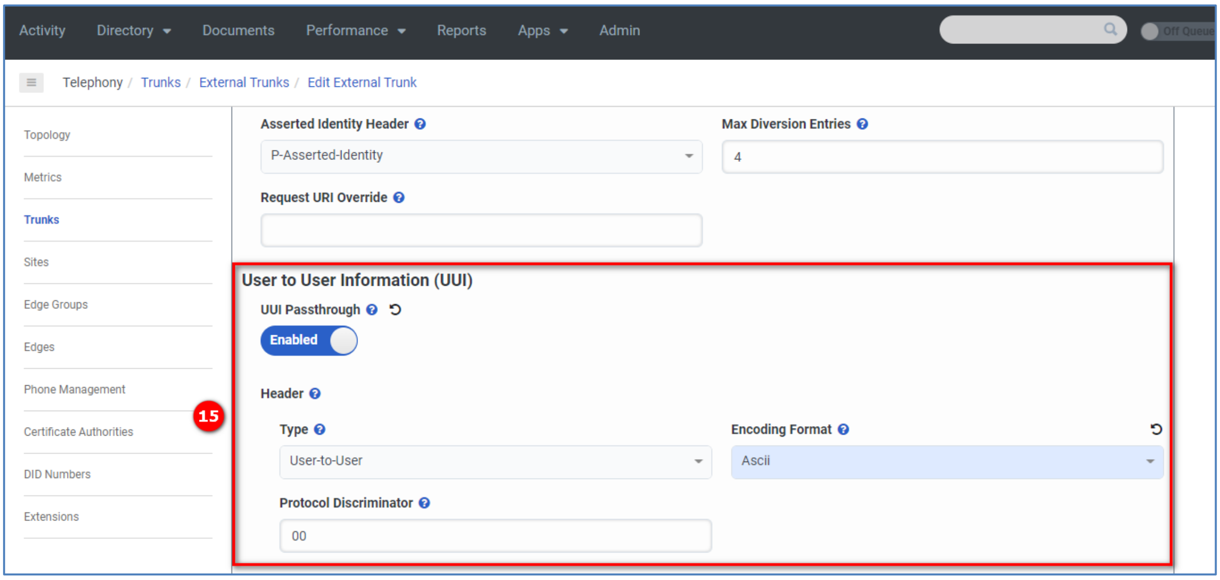
15. Under the **Protocol** group, enable **Take Back and Transfer**. This enables support of REFER on Genesys Cloud.

Figure 3-9: Enable REFER



16. To pass any additional data from Genesys Cloud to the external device through the SIP Trunk, enable User to User Information (UUI) Data on the BYOC Cloud trunk. Under the **Protocol** group, in **User to User Information (UUI)**, enable the **UUI Passthrough** toggle and select the **Header Type** (User-to-User) and **Encoding Format** (hex, ascii) you want to use.

Figure 3-10: Enabling UUI Header



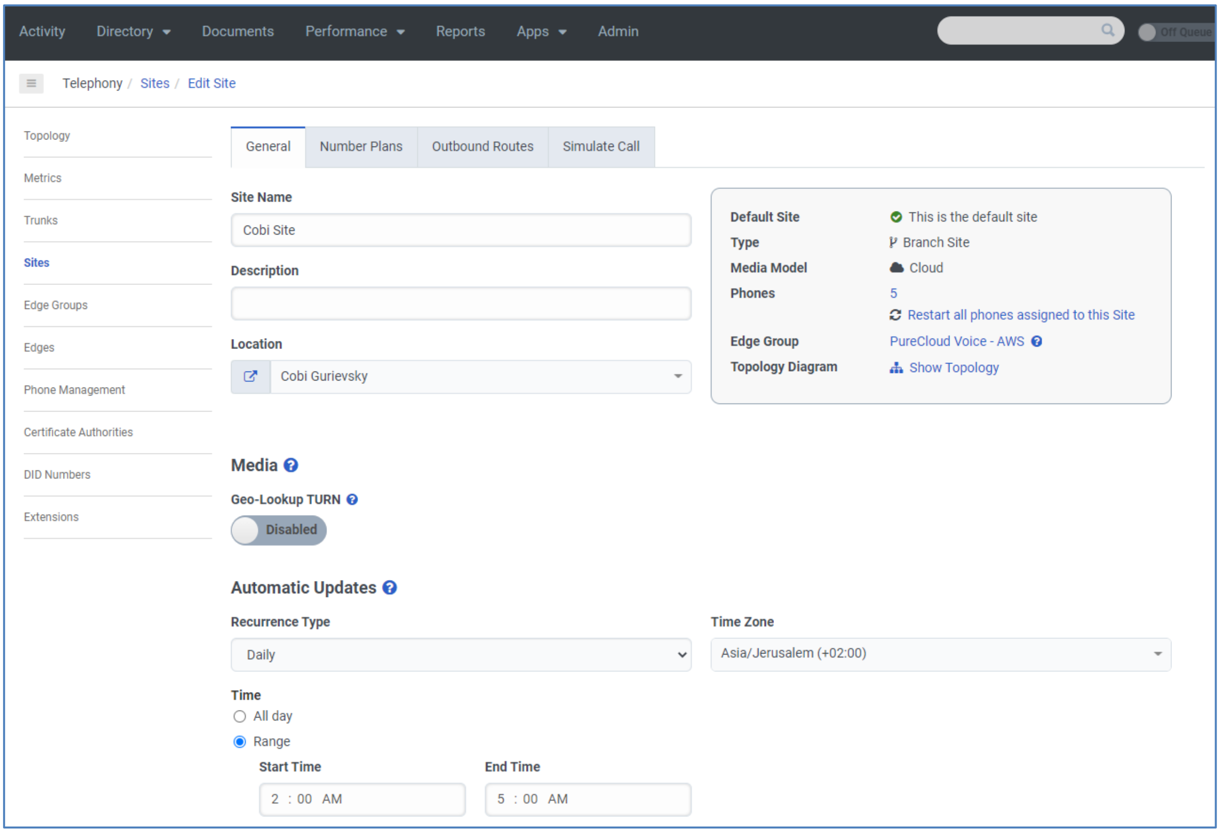
3.3 Adding a New Outbound Route on Site

The procedure below describes how to add an external trunk, created in the previous section, as an outbound route for a specific site. We need to make sure that this trunk will be selected.

➢ **To add a new Outbound Route on Site:**

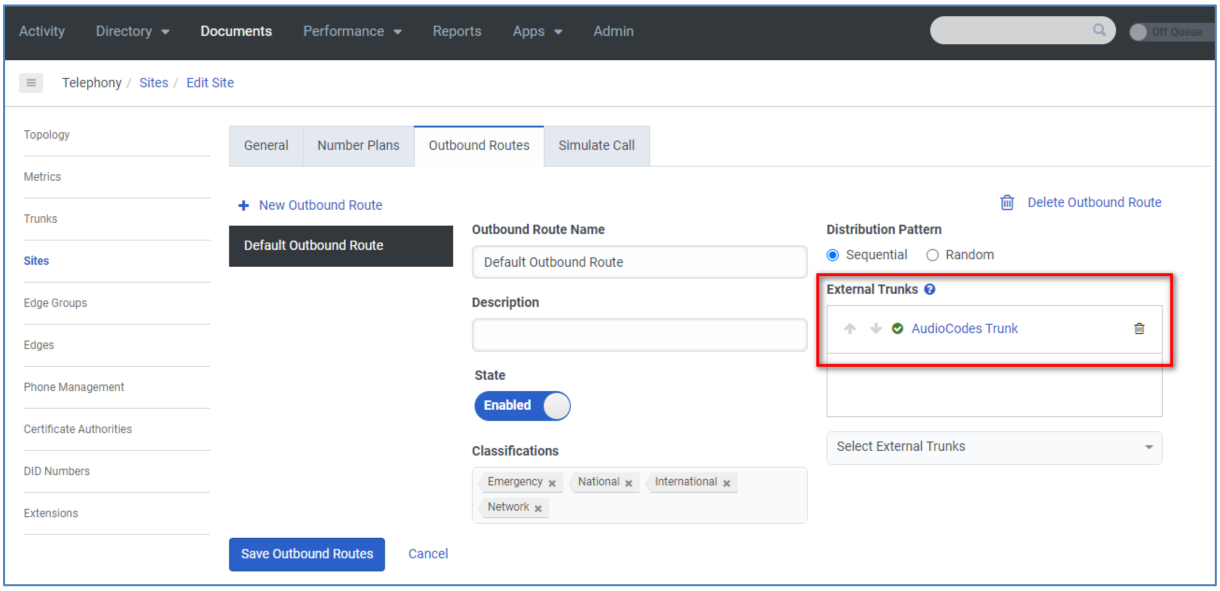
1. From the Genesys Desktop, in Admin, navigate to **Sites** in **Telephony**, and then select the site where you want to add your BYOC Cloud trunk (or create a new one):

Figure 3-11: Site Selection



2. Navigate to the **Outbound Routes** tab, and then add an external trunk, created in the previous section (e.g., *AudioCodes Trunk*) for Default Outbound Route:

Figure 3-12: Outbound Routes



Optionally, you can add a dedicated Number Plan (under ***Number Plans*** tab), to match specific DID numbers.

For a more deeper description, refer to the BYOC Cloud quick start guide.

4 Configuring AudioCodes SBC

This section provides step-by-step procedures on how to configure AudioCodes SBC for interworking between Genesys Cloud Contact Center and the Generic SIP Trunk. These configuration procedures are based on the interoperability test topology described in Section 2.4 on page 10, and includes the following main areas:

- SBC LAN interface – Management Station
- SBC WAN interface - Generic SIP Trunking and Genesys Cloud environment

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

Notes:

- For implementing Genesys Cloud and Generic SIP Trunk based on the configuration described in this section, AudioCodes SBC must be installed with a License Key that includes the following software features:
  - **Number of SBC sessions** (based on requirements)
  - **Transcoding sessions** (only if media transcoding is needed)
  - **Coders** (based on requirements)

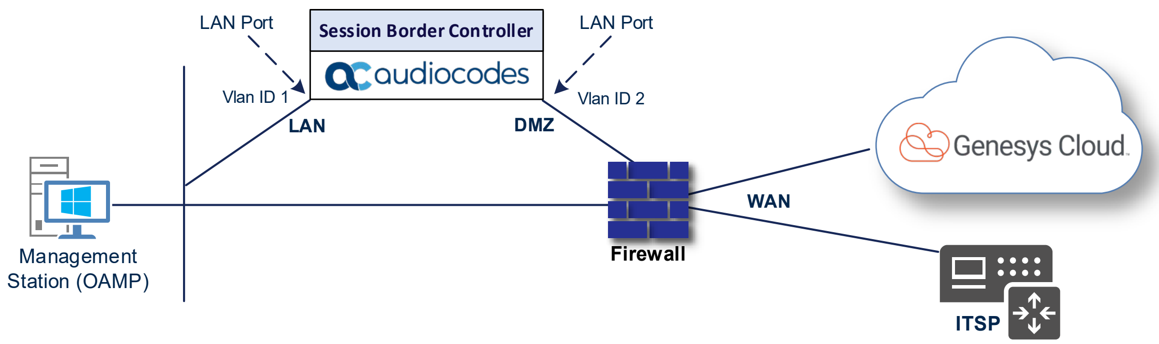
  For more information about the License Key, contact your AudioCodes sales representative.
- If your SBC is deployed in a virtual environment and transcoding is required, your virtual machine must have a minimum of two vCPUs. For more information, please refer to the appropriate *Installation Manual,* which can be found on AudioCodes website.
- 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

4.1 IP Network Interfaces Configuration

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

- SBC interfaces with the following IP entities:
  - Management Servers, located on the LAN
  - Genesys Cloud Contact Center and Generic SIP Trunk, located on the WAN
- SBC connects to the WAN through a DMZ network
- Physical connection: The type of physical connection depends on the method used to connect to the Enterprise's network. In the interoperability test topology, SBC connects to the LAN and DMZ using dedicated ethernet ports (i.e., two ports and two network cables are used).
- SBC also uses two logical network interfaces:
  - LAN (VLAN ID 1)
  - DMZ (VLAN ID 2)

Figure 4-1: Network Interfaces in Interoperability Test Topology



4.1.1 Configure VLANs

This section describes how to configure VLANs for each of the following interfaces:

- LAN (assigned the name "LAN_IF")
- WAN (assigned the name "WAN_IF")

➢ **To configure the VLANs:**

1. Open the Ethernet Device table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **Ethernet Devices**).
2. There will be one existing row for VLAN ID 1 and underlying interface GROUP_1.
3. Add another VLAN ID 2 for the WAN side.

4.1.2 Configure Network Interfaces

This section describes how to configure the IP network interfaces for each of the following interfaces:

- LAN Interface (assigned the name "LAN_IF")
- WAN Interface (assigned the name "WAN_IF")

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

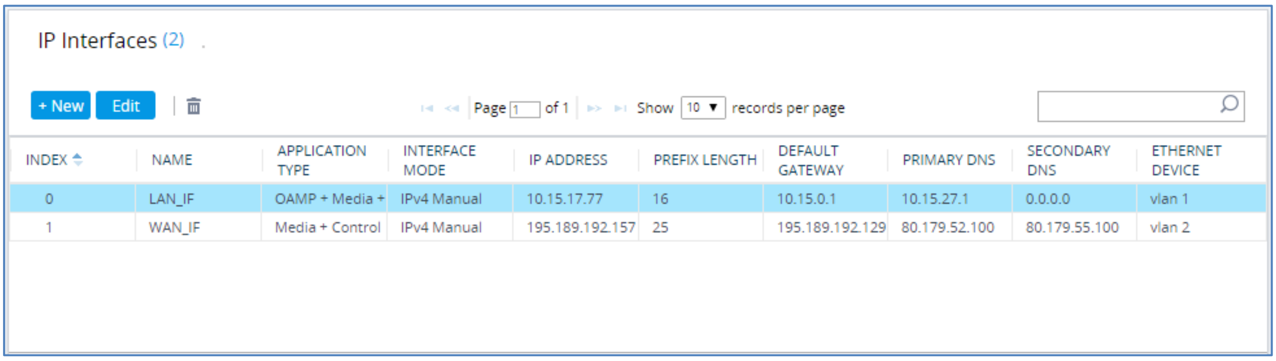
1. Open the IP Interfaces table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **IP Interfaces**).
2. Configure the IP interfaces as follows (your network parameters might be different):

Table 4-1: Configuration Example of the Network Interface Table

Index	Application Types	Interface Mode	IP Address	Prefix Length	Gateway	DNS	I/F Name	Ethernet Device
0	OAMP+ Media + Control	IPv4 Manual	10.15.77.77	16	10.15.0.1	10.15.27.1	LAN_IF	vlan 1
1	Media + Control (as this interface points to the internet, enabling OAMP is not recommended)	IPv4 Manual	195.189.192.157 (DMZ IP address of SBC)	25	195.189.192.129 (router's IP address)	According to your Internet provider's instructions	WAN_IF	vlan 2

The configured IP network interfaces are shown below:

Figure 4-2: Configured Network Interfaces in IP Interfaces Table



4.2 SIP TLS Connection Configuration (Optional)

This section describes how to configure the SBC for using a TLS connection with the Genesys Cloud.

Note: This configuration is essential for a secure SIP TLS connection *only* and that's why if your connection to Genesys Cloud is over TCP or UDP, skip this section.

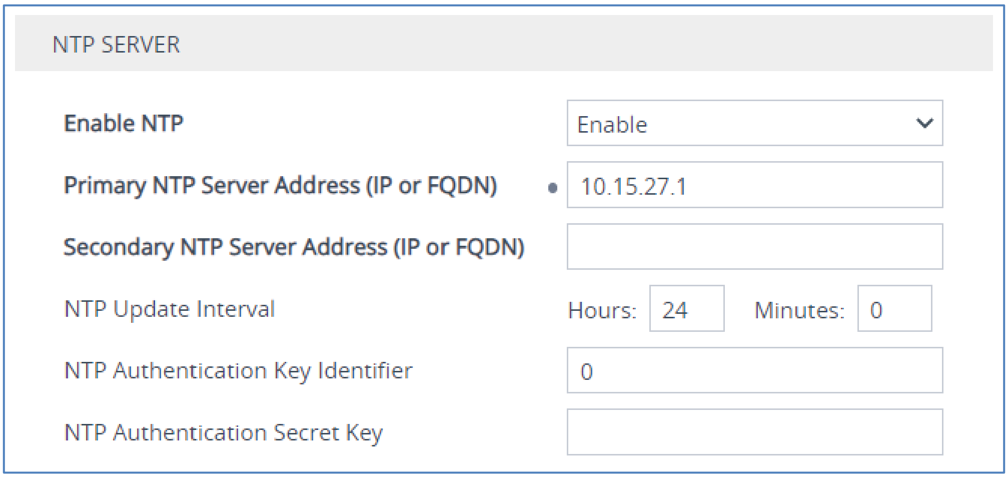
4.2.1 Configure the NTP Server Address

This section describes how to configure the NTP server's IP address. It is recommended to implement an NTP server (Microsoft NTP server or another global server) to ensure that the SBC receives the current date and time. This is necessary for validating certificates of remote parties. It is important, that NTP Server will locate on the OAMP IP Interface (LAN_IF in our case) or will be accessible through it.

➢ **To configure the NTP server address:**

1. Open the Time & Date page (**Setup** menu > **Administration** tab > **Time & Date**).
2. In the 'Primary NTP Server Address' field, enter the IP address of the NTP server (e.g., **10.15.27.1**).

Figure 4-3: Configuring NTP Server Address



3. Click **Apply**.

4.2.2 Create a TLS Context for Genesys Cloud

This section describes how to configure TLS Context in the SBC.

➢ **To configure the TLS version:**

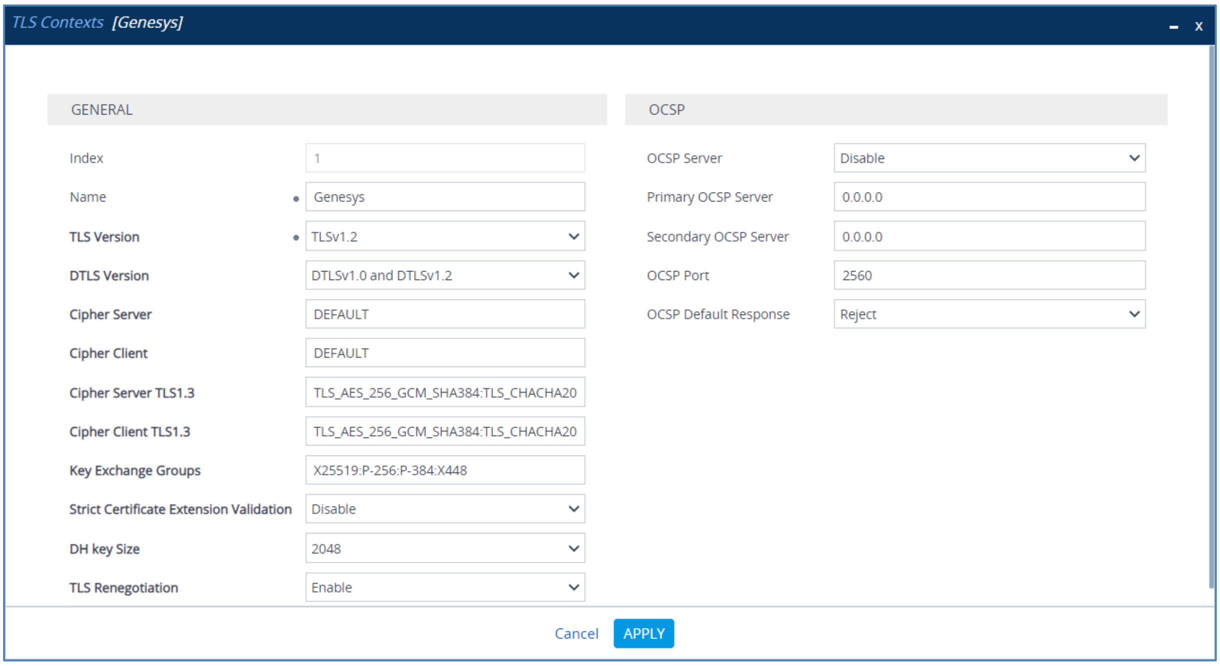
1. Open the TLS Contexts table (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. Create a new TLS Context by clicking **New** at the top of the interface, and then configure the parameters using the table below as reference.

> **Note:** You can use the existing (default) TLS Context, but we recommend creating an additional one, dedicated to our setup.

Table 4-2: New TLS Context

Index	Name	TLS Version
1	Genesys (arbitrary descriptive name)	TLSv1.2
All other parameters can be left unchanged with their default values.		

Figure 4-4: Configuring TLS Context for Genesys Cloud



3. Click **Apply**.

4.2.3 Configure a Certificate

This section describes how to deploy Genesys Trusted Root Certificates on the SBC.

➢ **To load trusted root certificates:**

1. Open the TLS Contexts page (**Setup** menu > **IP Network** tab > **Security** folder > **TLS Contexts**).
2. In the TLS Contexts page, select the required TLS Context index row, and then click the Trusted Root Certificates link, located at the bottom of the TLS Contexts page; the Trusted Certificates page appears.
3. Click the **Import** button, and then select all Root Certificates obtained from Genesys or from your Certification Authority to load.
4. Click **OK**; the certificate is loaded to the device and listed in the Trusted Certificates store.

4.3 Configure Media Realms

This section describes how to configure Media Realms. The simplest configuration is to create two Media Realms - one for the SIP Trunk traffic and one for the Genesys traffic.

➢ **To configure Media Realms:**

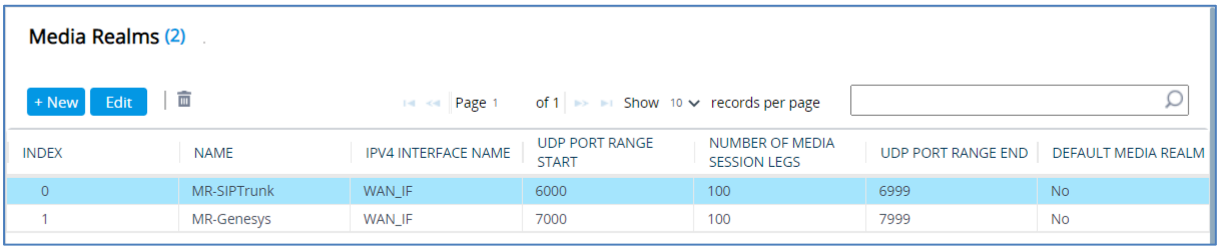
1. Open the Media Realms table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Media Realms**).
2. Configure Media Realms as follows (you can use the default Media Realm (Index 0), but modify it):

Table 4-3: Configuration Example Media Realms in Media Realm Table

Index	Name	Topology Location	IPv4 Interface Name	Port Range Start	Number of Media Session Legs
0	MR-SIPTrunk (arbitrary name)		WAN_IF	6000	100 (media sessions assigned with port range)
1	MR-Genesys (arbitrary name)	Up	WAN_IF	7000	100 (media sessions assigned with port range)

The configured Media Realms are shown in the figure below:

Figure 4-5: Configured Media Realms in Media Realm Table


Media Realms (2)

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

INDEX	NAME	IPV4 INTERFACE NAME	UDP PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	UDP PORT RANGE END	DEFAULT MEDIA REALM
0	MR-SIPTrunk	WAN_IF	6000	100	6999	No
1	MR-Genesys	WAN_IF	7000	100	7999	No

4.4 Configure SIP Signaling Interfaces

This section describes how to configure SIP Interfaces. For the interoperability test topology, towards the SIP Trunk and towards the Genesys Cloud SIP Interfaces must be configured for the SBC.

➢ **To configure SIP Interfaces:**

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. Configure SIP Interfaces. You can use the default SIP Interface (Index 0), but modify it as shown in the table below. The table below shows an example of the configuration. You can change some parameters according to your requirements.

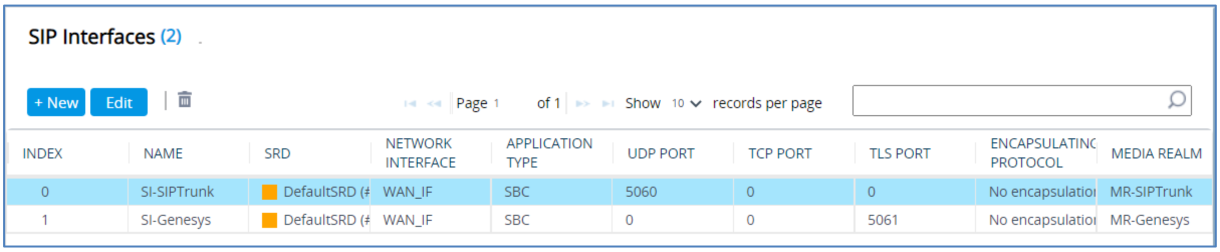
Note: The following example used TLS connectivity between the SBC and Genesys Cloud.

Table 4-4: Configured SIP Interfaces in SIP Interface Table

Index	Name	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Classification Failure Response Type	Media Realm
0	**SI-SIPTrunk** (arbitrary name)	**WAN_IF**	**SBC**	**5060** (according to Service Provider requirement)	**0**	**0**	**0** (Recommended to prevent DoS attacks)	**MR-SIPTrunk**
1	**SI-Genesys** (arbitrary name)	**WAN_IF**	**SBC**	**0**	**0**	**5061** (as configured on Genesys site)	**0** (Recommended to prevent DoS attacks)	**MR-Genesys**

The configured SIP Interfaces are shown in the figure below:

Figure 4-6: 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	ENCAPSULATINC PROTOCOL	MEDIA REALM
0	SI-SIPTrunk	DefaultSRD (#	WAN_IF	SBC	5060	0	0	No encapsulatio	MR-SIPTrunk
1	SI-Genesys	DefaultSRD (#	WAN_IF	SBC	0	0	5061	No encapsulatio	MR-Genesys

4.5 Configure Proxy Sets and Proxy Address

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

- Generic SIP Trunk
- Genesys Cloud

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

➢ **To configure Proxy Sets:**

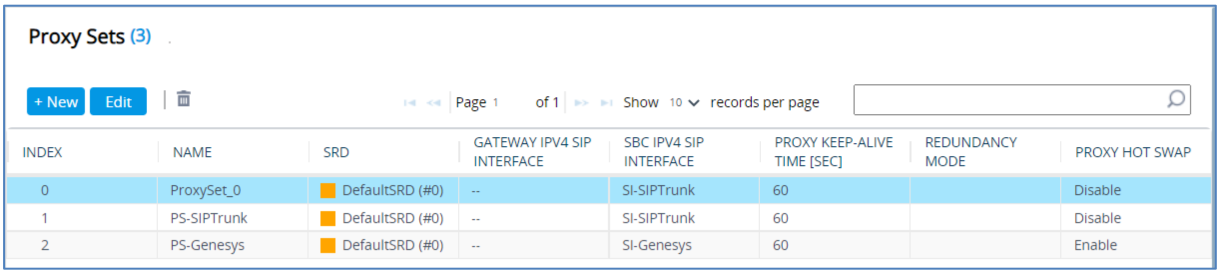
1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**).
2. Configure Proxy Sets as shown in the table below:

Table 4-5: Configuration Example Proxy Sets in Proxy Sets Table

Index	Name	SBC IPv4 SIP Interface	TLS Context Name	Proxy Keep-Alive	Proxy Hot Swap	Proxy Load Balancing Method	DNS Resolve Method
1	PS-SIPTrunk (arbitrary name)	SI-SIPTrunk	Default	Using Options	-	-	-
2	PS-Genesys (arbitrary name)	SI-Genesys	Genesys	Using Options	Enable	Round Robin	SRV

The configured Proxy Sets are shown in the figure below:

Figure 4-7: Configured Proxy Sets in Proxy Sets Table



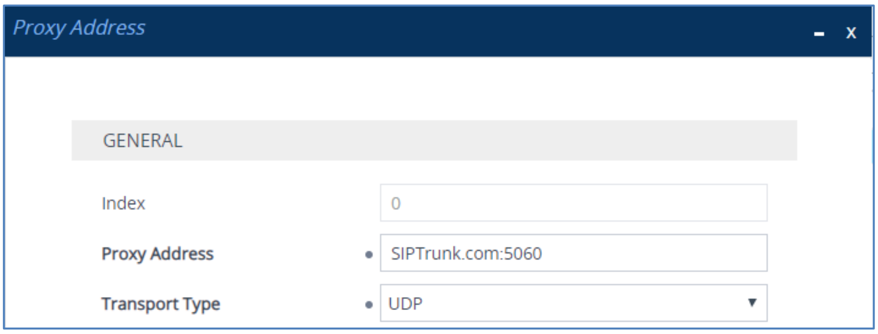
4.5.1 Configure a Proxy Address

This section shows how to configure a Proxy Address.

➢ **To configure a Proxy Address for SIP Trunk:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**), click the Proxy Set **PS-SIPTrunk**, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
2. Click **+New**; the following dialog box appears:

Figure 4-8: Configuring Proxy Address for SIP Trunk



3. Configure the address of the Proxy Set according to the parameters described in the table below:

Table 4-6: Configuration Proxy Address for SIP Trunk

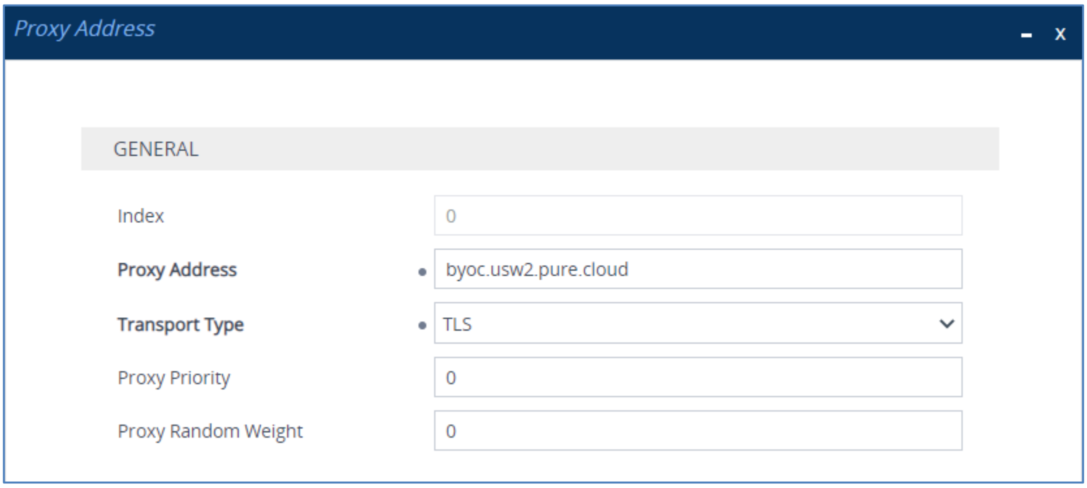
Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	SIPTrunk.com:5060 (SIP Trunk IP / FQDN and port)	UDP	0	0

4. Click **Apply**.

➢ To configure a Proxy Address for Genesys Cloud:

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**) and then click the Proxy Set **PS-Genesys**, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
2. Click **+New**; the following dialog box appears:

Figure 4-9: Configuring Proxy Address for Genesys Cloud



3. Configure the address of the Proxy Set according to the parameters described in the table below:

Table 4-7: Configuration Proxy Address for Genesys Cloud

Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	byoc.usw2.pure.cloud (FQDN of the appropriated Genesys BYOC Cloud region)	TLS	0	0

4. Click **Apply**.

4.6 Configure Coders

This section describes how to configure coders (termed *Coder Group*). As Genesys Cloud supports the OPUS coder while the network connection to Generic SIP Trunk may restrict operation with a dedicated coders list, you need to add a Coder Group with the supported coders for each leg, the Genesys Cloud and the Generic SIP Trunk.

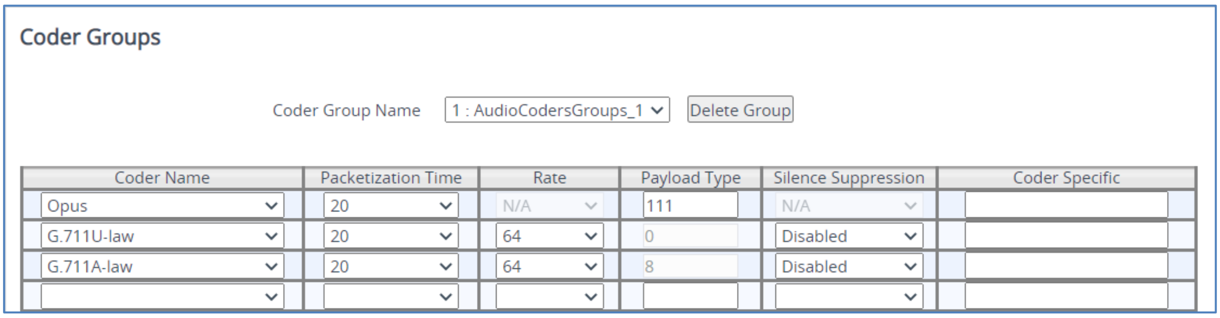
Note that the Coder Group ID for this entity will be assigned to its corresponding IP Profile in the next step.

➢ **To configure coders:**

1. Open the Coder Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Coder Groups**).
2. Configure a Coder Group for Genesys Cloud:

Parameter	Value
Coder Group Name	**AudioCodersGroups_1**
Coder Name	▪ **Opus** ▪ **G.711 U-law** ▪ **G.711 A-law**

Figure 4-10: Configuring Coder Group for Genesys Cloud



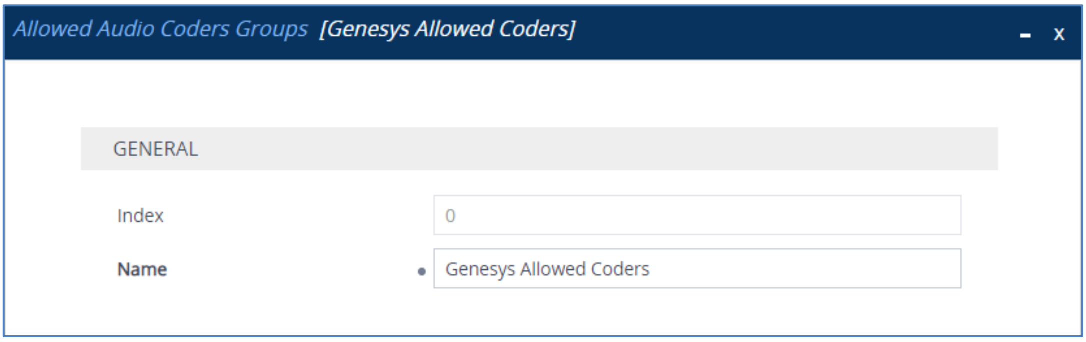
3. Click **Apply,** and then confirm the configuration change in the prompt that pops up.

The procedure below describes how to configure an Allowed Coders Group to ensure that the voice sent to the Genesys Cloud, uses the dedicated coders list whenever possible. Note that this Allowed Coders Group ID will be assigned to the IP Profile belonging to the Genesys Cloud in the next step.

➢ **To set a preferred coder for the Genesys Cloud:**

1. Open the Allowed Audio Coders Groups table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **Allowed Audio Coders Groups**).
2. Click **New**, and then configure a name for the Allowed Audio Coders Group for the Genesys Cloud.

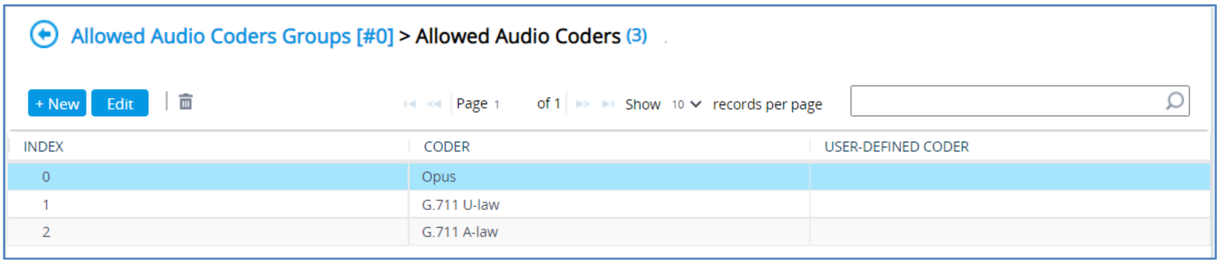
Figure 4-11: Configuring Allowed Coders Group for the Genesys Cloud



3. Click **Apply.**
4. Select the new row that you configured, and then click the **Allowed Audio Coders** link located below the table; the Allowed Audio Coders table opens.
5. Click **New**, and then configure an Allowed Coders as follows:

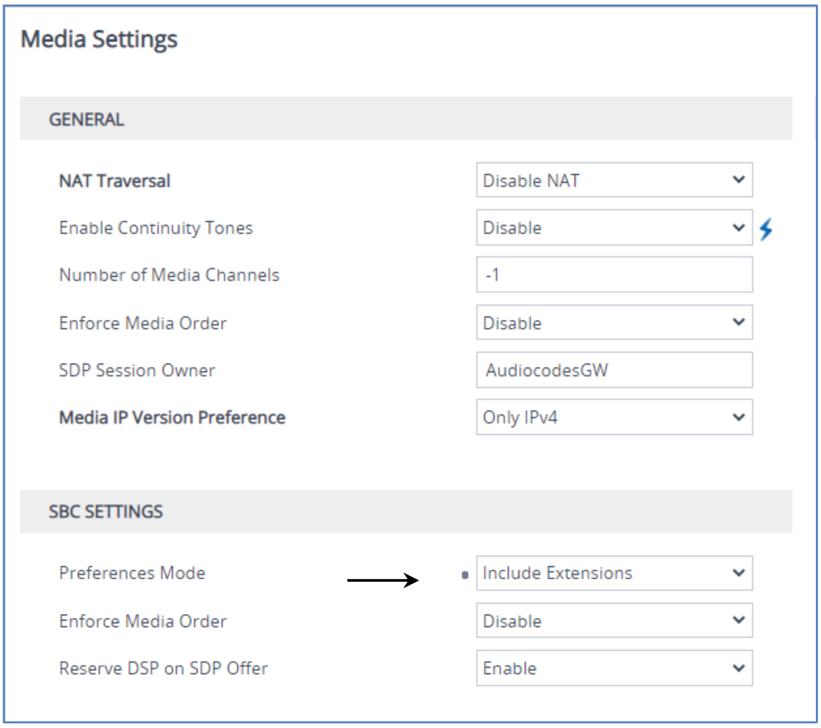
Index	Coder
0	Opus
1	G.711 U-law
2	G.711 A-law

Figure 4-12: Configuring Allowed Coders for the Genesys Cloud



6. Open the Media Settings page (**Setup** menu > **Signaling & Media** tab > **Media** folder > **Media Settings**).

Figure 4-13: SBC Preferences Mode



7. From the '**Preferences Mode**' drop-down list, select **Include Extensions**.
8. Click **Apply**.

4.7 Configure IP Profiles

This section describes how to configure IP Profiles. The IP Profile defines a set of call capabilities relating to signaling (e.g., SIP message terminations such as REFER) and media (e.g., coder and transcoding method).

In this interoperability test topology, IP Profiles need to be configured for the following IP entities:

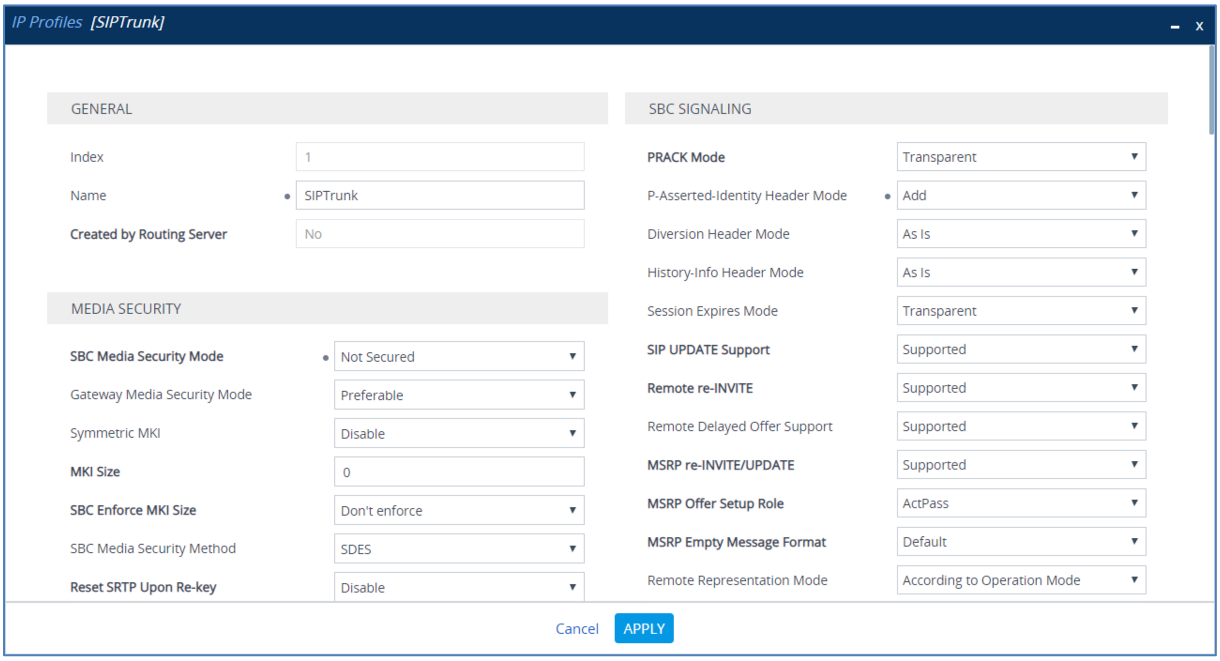
- Generic SIP trunk – to operate in non-secure mode using RTP and SIP over UDP
- Genesys Cloud – to operate in secure mode using SRTP and SIP over TLS

➢ **To configure an IP Profile for the Generic SIP Trunk:**

1. Open the IP Profiles table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Click **New**, and then configure the parameters as follows:

Parameter	Value
General	
Index	**1**
Name	**SIPTrunk**
Media Security	
SBC Media Security Mode	**Not Secured**
SBC Signaling	
P-Asserted-Identity Header Mode	**Add** (required for anonymous calls)

Figure 4-14: Configuring IP Profile for Generic SIP Trunk



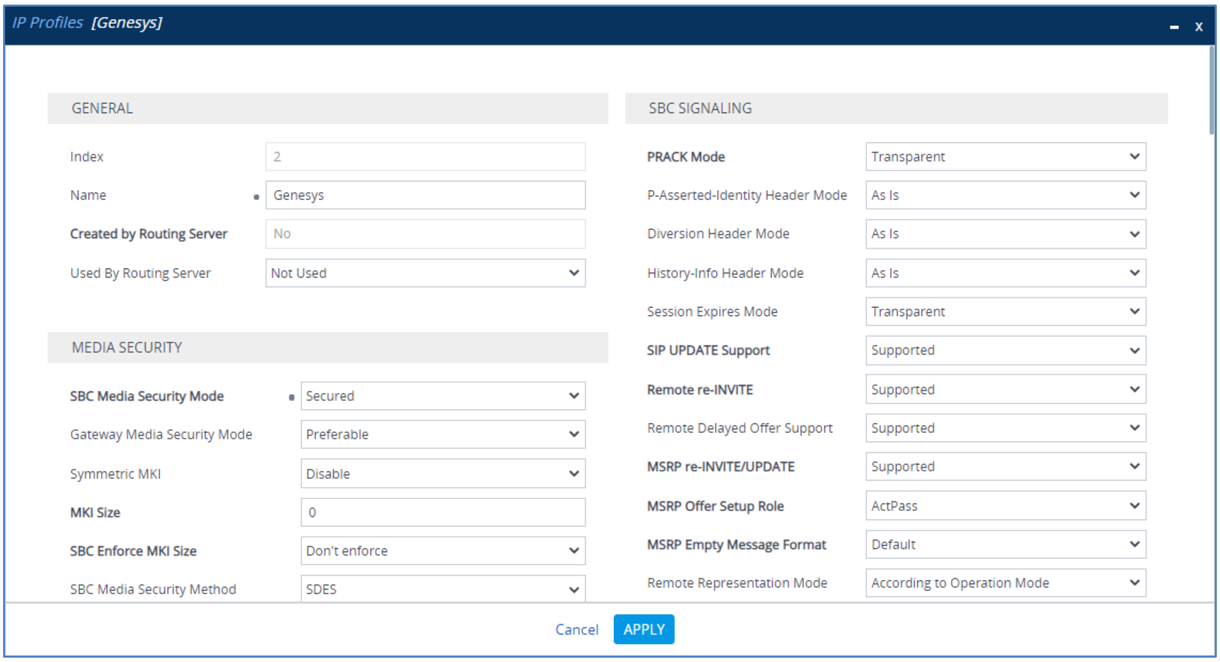
3. Click **Apply**.

➢ To configure IP Profile for the Genesys Cloud:

1. Open the IP Profiles table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Click **New**, and then configure the parameters as follows:

Parameter	Value
General	
Index	**2**
Name	**Genesys** (arbitrary descriptive name)
Media Security	
SBC Media Security Mode	**Secured** (if TLS/SRTP implemented)
SBC Media	
Extension Coders Group	**AudioCodersGroups_1**
Allowed Audio Coders	**Genesys Allowed Coders**
Allowed Coders Mode	**Restriction and Preference**
RFC 2833 Mode	**Extend**
RFC 2833 DTMF Payload Type	**101**

Figure 4-15: Configuring IP Profile for Genesys Cloud



3. Click **Apply**.

> **Note:** The configuration may change according to specific SIP Trunk requirements.

4.8 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 interoperability test topology, IP Groups must be configured for the following IP entities:

- Generic SIP Trunk
- Genesys Cloud Contact Center

➢ **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 Generic SIP Trunk:

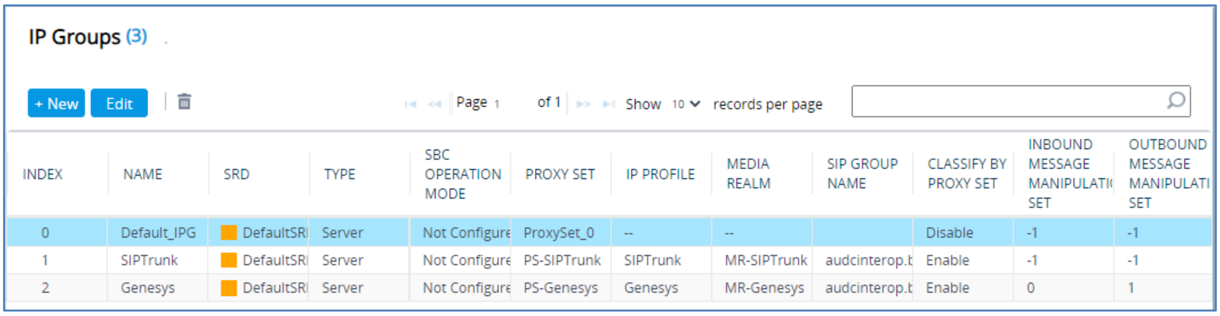
Parameter	Value
Index	**1**
Name	**SIPTrunk**
Type	**Server**
Proxy Set	**PS-SIPTrunk**
IP Profile	**SIPTrunk**
Media Realm	**MR-SIPTrunk**
SIP Group Name	**audiocodestrk.byoc.usw2.pure.cloud** (according to requirement)

3. Configure an IP Group for the Genesys Cloud:

Parameter	Value
Index	**2**
Name	**Genesys**
Topology Location	**Up**
Type	**Server**
Proxy Set	**PS-Genesys**
IP Profile	**Genesys**
Media Realm	**MR-Genesys**
SIP Group Name	**audiocodestrk.byoc.usw2.pure.cloud** (according to requirement)

The configured IP Groups are shown in the figure below:

Figure 4-16: Configured IP Groups in IP Group Table



IP Groups (3)

INDEX	NAME	SRD	TYPE	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULATI SET	OUTBOUND MESSAGE MANIPULATI SET
0	Default_IPG	DefaultSRI	Server	Not Configure	ProxySet_0	--	--		Disable	-1	-1
1	SIPTrunk	DefaultSRI	Server	Not Configure	PS-SIPTrunk	SIPTrunk	MR-SIPTrunk	audcinterop.t	Enable	-1	-1
2	Genesys	DefaultSRI	Server	Not Configure	PS-Genesys	Genesys	MR-Genesys	audcinterop.t	Enable	0	1

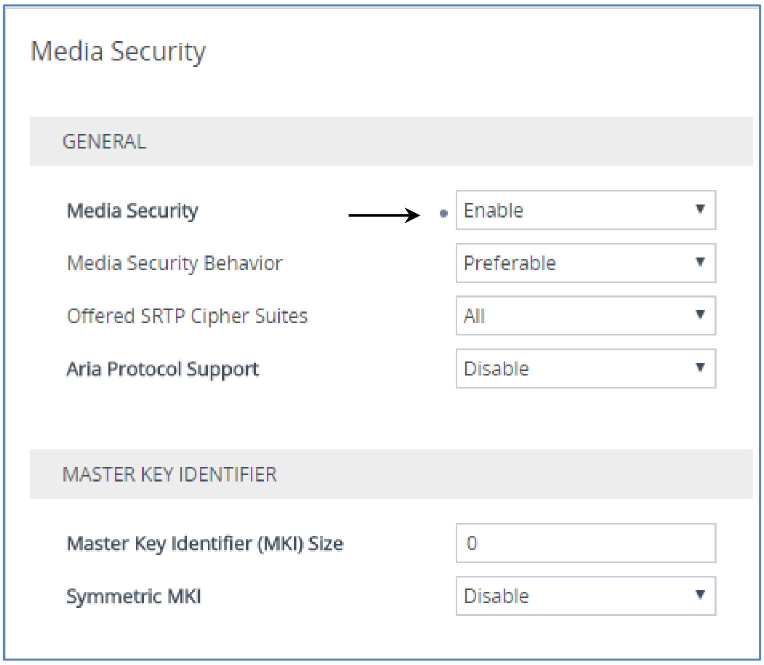
4.9 Configure SRTP

This section describes how to configure media security. If connectivity between AudioCodes SBC and Genesys Cloud was implemented in secure mode (TLS/SRTP), configure the SBC to operate in the same manner.

➢ **To configure media security:**

1. Open the Media Security page (**Setup menu > Signaling & Media tab > Media folder > Media Security**).
2. From the 'Media Security' drop-down list, select **Enable** to enable SRTP.

Figure 4-17: Configuring SRTP



3. Click **Apply**.

4.10 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 interoperability test topology, the following IP-to-IP routing rules need to be configured to route calls between Genesys Cloud Contact Center and Generic SIP Trunk:

- Terminate SIP OPTIONS messages on the SBC that are received from any entity
- Calls from Genesys Cloud to Generic SIP Trunk
- Calls from Generic SIP Trunk to Genesys Cloud

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

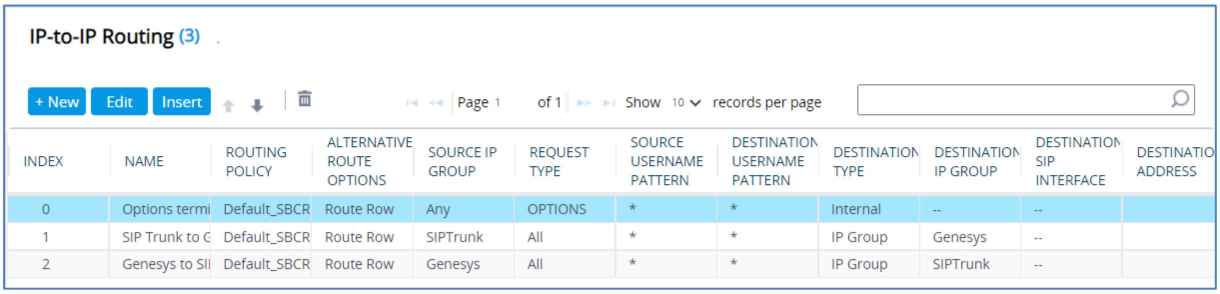
1. Open the IP-to-IP Routing table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing** > **IP-to-IP Routing**).
2. Configure routing rules as shown in the table below:

Table 4-8: Configuration IP-to-IP Routing Rules

Index	Name	Source IP Group	Request Type	Call Triger	ReRoute IP Group	Dest Type	Dest IP Group	Internal Action
0	Terminate OPTIONS	Any	OPTIONS			Internal		Reply (Response ='200')
1	Genesys to SIP Trunk (arbitrary name)	Genesys				IP Group	SIPTrunk	
2	SIP Trunk to Genesys (arbitrary name)	SIPTrunk				IP Group	Genesys	

The configured routing rules are shown in the figure below:

Figure 4-18: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table



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

4.11 Configure Number Manipulation Rules (Optional)

This section describes how to configure IP-to-IP manipulation rules. These rules manipulate the SIP Request-URI user part (source or destination number). The manipulation rules use the configured IP Groups (as configured in Section 4.8 on page 32) to denote the source and destination of the call.

Note: Adapt the manipulation table according to your environment dial plan.

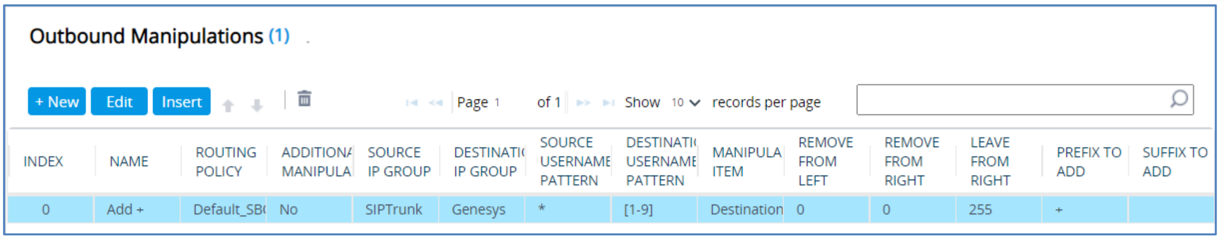
For example, for this interoperability test topology, a manipulation is configured to add the "+" (plus sign) to the destination number (if it not exists) for calls from the Generic SIP Trunk to the Genesys Cloud for any destination username pattern.

➢ **To configure a number manipulation rule:**

1. Open the Outbound Manipulations table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Manipulation** > **Outbound Manipulations**).
2. Configure the rules according to your setup.

The figure below shows an example of configured IP-to-IP outbound manipulation rules for calls between Genesys Cloud IP Group and Generic SIP Trunk IP Group:

Figure 4-19: Example of Configured IP-to-IP Outbound Manipulation Rules



Outbound Manipulations (1)

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

INDEX	NAME	ROUTING POLICY	ADDITIONA MANIPULA	SOURCE IP GROUP	DESTINATI IP GROUP	SOURCE USERNAME PATTERN	DESTINATI USERNAME PATTERN	MANIPULA ITEM	REMOVE FROM LEFT	REMOVE FROM RIGHT	LEAVE FROM RIGHT	PREFIX TO ADD	SUFFIX TO ADD
0	Add +	Default_SB	No	SIPTrunk	Genesys	*	[1-9]	Destination	0	0	255	+	

Rule Index	Description
0	Calls from SIP Trunk IP Group to Genesys Cloud IP Group with the prefix destination. Can be any number with digits 1 to 9. Add "+" to the prefix of the destination number.

4.12 Miscellaneous Configuration

This section describes a miscellaneous SBC configuration.

4.12.1 Optimizing CPU Cores Usage for a Specific Service (Relevant for Mediant 9000 and Software SBC only)

This section describes how to optimize the SBC's CPU cores usage for a specified profile to achieve maximum capacity for that profile. The supported profiles include:

- SIP profile: Improves SIP signaling performance, for example, SIP calls per second (CPS)
- SRTP profile: Improves maximum number of SRTP sessions
- Transcoding profile: Enables all DSP-required features, for example, transcoding and voice in-band detectors

➢ **To optimize core allocation for a profile:**

1. Open the SBC General Settings page (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **SBC General Settings**).
2. From the 'SBC Performance Profile' drop-down list, select the required profile:

3. Click **Apply**, and then reset the device with a burn-to-flash for your settings to take effect.

his page is intentionally left blank.

International Headquarters
1 Hayarden Street,
Airport City
Lod 7019900, Israel
Tel: +972-3-976-4000
Fax: +972-3-976-4040

AudioCodes Inc.
200 Cottontail Lane
Suite A101E
Somerset NJ 08873
Tel: +1-732-469-0880
Fax: +1-732-469-2298

Contact us: https://www.audiocodes.com/corporate/offices-worldwide
website: https://www.audiocodes.com

Document #: LTRT-39478