Configuration Note

AudioCodes Professional Services - Interoperability Lab

Swisscom SIP Trunk "Enterprise SIP" using AudioCodes Mediant™ BRI/PRI Gateway

Version 7.2

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Document Revision Record

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1 Introduction

This Configuration Note describes how to set up the AudioCodes Gateway for interworking between Swisscom's SIP Trunk environments.

1.1 Intended Audience

The document is intended for engineers, or AudioCodes and Swisscom Partners who are responsible for installing and configuring Swisscom's SIP Trunk for enabling VoIP calls using AudioCodes Gateway.

Note: All references to **Swisscom SIP Trunk** in this document refer to **Swisscom SIP Trunk "Enterprise SIP Standard"**.

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2 Component Information

2.1 AudioCodes Gateway Version

Table 2-1: AudioCodes Gateway Version

Gateway Vendor	AudioCodes
Models	▪ Mediant 500L Gateway ▪ Mediant 500 Gateway ▪ Mediant 800 Gateway ▪ Mediant 1000B Gateway
Software Version	SIP_7.20A.104.001
Protocol	▪ SIP/UDP (to the Swisscom SIP Trunk) ▪ Euro-ISDN over BRI/PRI (to the PSTN PBX)
Additional Notes	None

2.2 Swisscom SIP Trunking Version

Table 2-2: Swisscom Version

Vendor/Service Provider	Swisscom (Switzerland) Ltd.
SSW Model/Service	Swisscom SIP Trunk “Enterprise SIP Standard”
Software Version	▪ E-SBC: 15.5(3)M ▪ Core-SBC: SCZ730m2p ▪ A2: 19.01.2
Protocol	SIP
Additional Notes	None

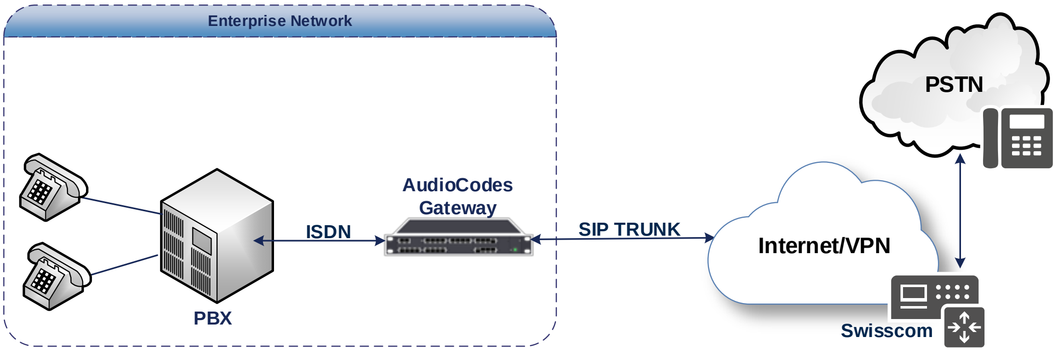
2.3 Interoperability Test Topology

The interoperability testing between AudioCodes Gateway and Swisscom SIP Trunk was done using the following topology setup:

- Enterprise ISDN PBX.
- AudioCodes Gateway is implemented to interconnect between the Enterprise PBX and the SIP Trunk using an AudioCodes Gateway.

The figure below illustrates this test topology:

Figure 2-1: Test Topology between ISDN PBX with Swisscom SIP Trunk



3 Configuring AudioCodes Media Gateway

This chapter provides step-by-step procedures on how to configure the AudioCodes Media Gateway for interworking with the Swisscom SIP Trunk. These configuration procedures are based on the test topology described in Section 2.3 on page 10, and includes the following main areas:

- Gateway IP interface -   Swisscom SIP Trunking environment
- Gateway ISDN interface - PBX environment

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

Notes:

- For implementing Swisscom SIP Trunk based on the configuration described in this section in combination with a SIP PBX, the AudioCodes Media Gateway must be installed with the relevant SBC Software License Keys.
- For information about the License Key, contact your AudioCodes sales representative.
- The scope of this interoperability test and document does **not** cover all security aspects for connecting the SIP Trunk environment. 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.

3.1 Step 1: IP Network Interface Configuration

This step describes how to configure the device's IP network interface.

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

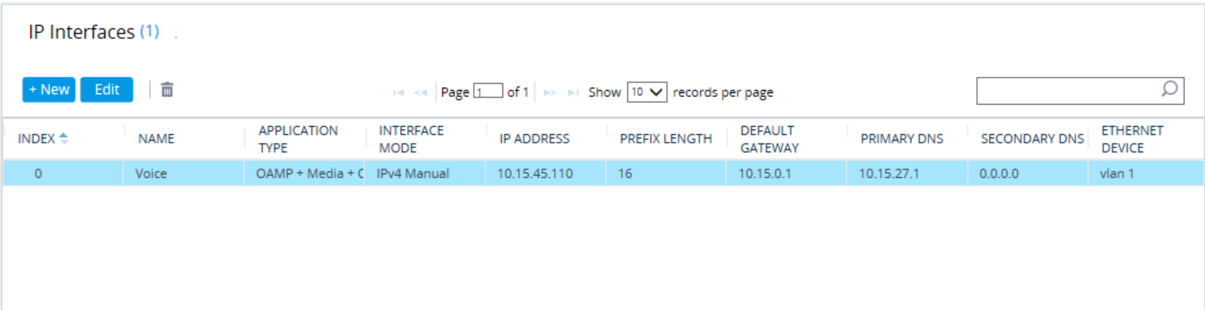
1. Open the IP Interfaces table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **IP Interfaces**).
2. Modify the existing LAN network interface:
   a. Select the 'Index' radio button of the **OAMP + Media + Control** table row, and then click **Edit**.
   b. Configure the interface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

Parameter	Value
Name	**LAN_IF** (arbitrary descriptive name)
Ethernet Device	**vlan 1**
IP Address	**10.15.45.110**
Prefix Length	**16** (subnet mask in bits for 255.255.0.0)
Default Gateway	**10.15.0.1**
Primary DNS	**10.15.27.1**

3. Click **Apply**, and then **Done**.

The configured IP network interface is shown below:

Figure 3-1: Configured Network Interface in IP Interfaces Table



3.2 Step 2: Configure Media Realm

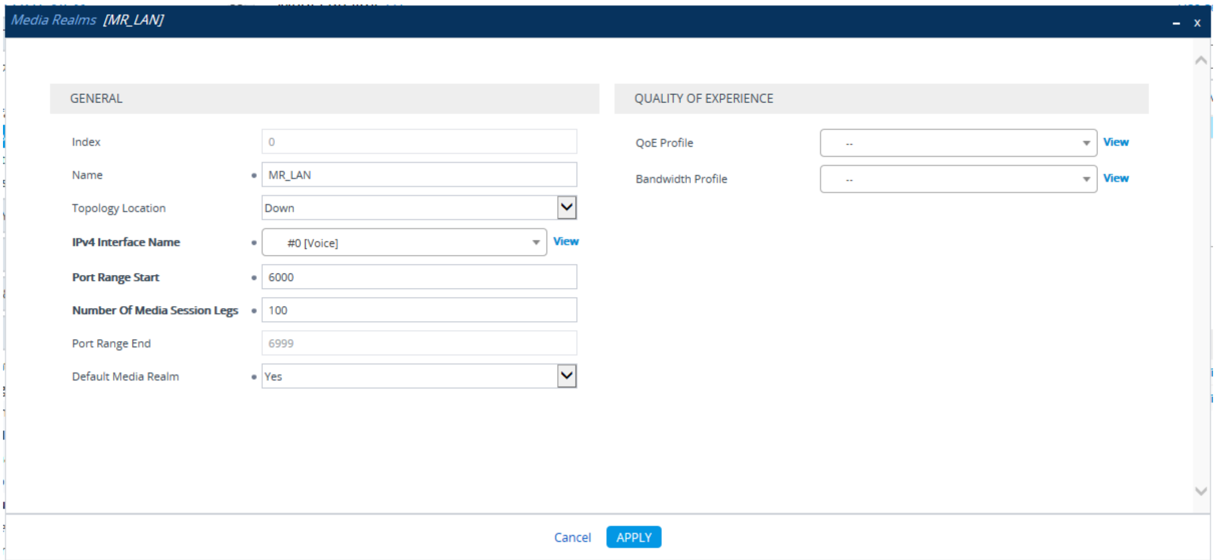
This step describes how to configure Media Realms.

➤ **To configure Media Realms:**

1. Open the Media Realms table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Media Realms**).
2. Add a Media Realm for the LAN interface. You can use the default Media Realm (Index 0), but modify it as shown below:

Parameter	Value
Index	**0**
Name	**MR_LAN** (descriptive name)
IPv4 Interface Name	**Voice**
Port Range Start	**6000** (represents lowest UDP port number used for media on LAN)
Number of Media Session Legs	**100** (media sessions assigned with port range)

Figure 3-2: Configuring Media Realm



3.3 Step 3: Configure SIP Signaling Interface

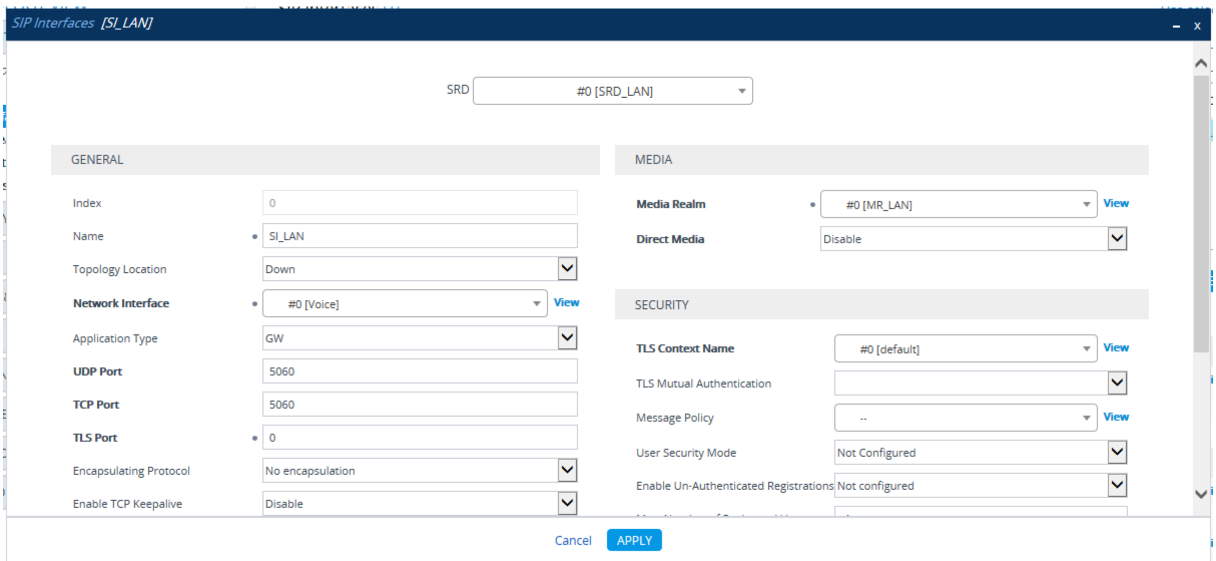
This step describes how to configure SIP Interfaces.

➢ **To configure SIP Interfaces:**

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. Add a SIP Interface for the LAN interface. You can use the default SIP Interface (Index 0), but modify it as shown below:

Parameter	Value
Index	**0**
Name	**SI_LAN** (see note at the end of this section)
Network Interface	**Voice**
Application Type	**GW**
UDP Port	**5060**
TCP Port	**5060**
TLS Port	**0**
Media Realm	**MR_LAN**

Figure 3-3: Configuring SIP-Interface



Note: Current software releases use the string **names** of the configuration entities (e.g., SIP Interface, Proxy Sets, and IP Groups). Therefore, it is recommended to configure each configuration entity with meaningful names for easy identification.

3.4 Step 4: Configure Proxy Set

This step describes how to configure the Proxy Set. The Proxy Set defines the destination address (IP address or FQDN) of the IP entity server.

For the test topology, the Proxy Set needs to be configured for the Swisscom SIP Trunk

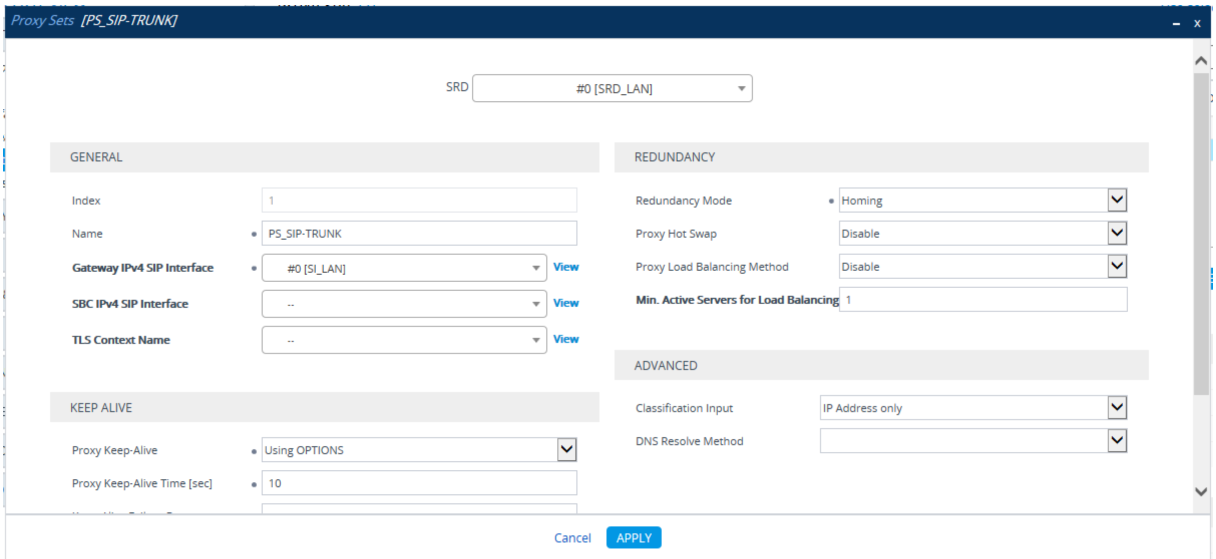
The Proxy Sets will be later applying to the VoIP network by assigning them to IP Groups.

➢ **To configure the Proxy Set:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder >**Proxy Sets**).
2. Add a Proxy Set for the Swisscom SIP Trunk as shown below:

Parameter	Value
Index	**1**
Name	**PS_SIP-TRUNK**
Gateway IPv4 SIP Interface	**SI_LAN**
Proxy Keep-Alive	**Using Options**
Proxy Keep-Alive Time [sec]	**10**
Redundancy Mode	**Homing**

Figure 3-4: Configuring Proxy Set for Swisscom SIP Trunk



3. 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.
4. Click **New**.

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

Parameter	Value
Index	**0**
Proxy Address	**10.254.150.52:5060**
Transport Type	**TCP**
Parameter	**Value**
Index	**1**
Proxy Address	**10.254.150.52:5060**
Transport Type	**UDP**

c. Click **Apply**.

3.5 Step 5: Configure Coders

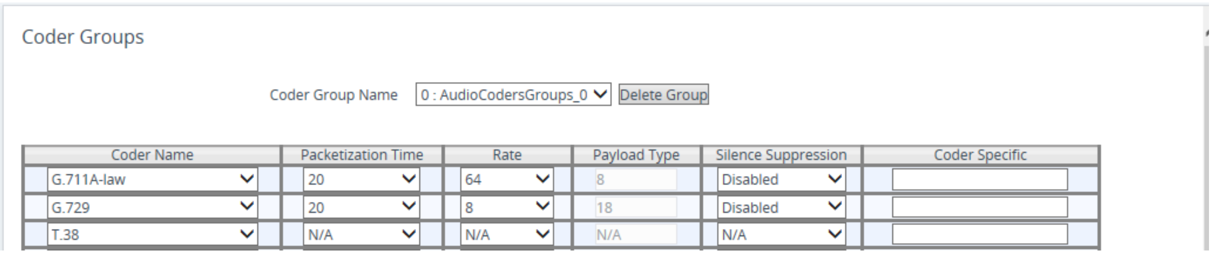
The procedure below describes how to configure coders to ensure that Voice and FAX are negotiated with the Swisscom SIP Trunk while use the coders in specific order.

➢ **To set coders for the Swisscom SIP Trunk:**

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

Coder Name	Payload Type
G.711A-law	**8**
G.729	**18**
T.38	**N/A**

Figure 3-5: Configuring Coders for Swisscom SIP Trunk



3.6 Step 6: Configure IP Profile

This step describes how to configure the IP Profile. The IP Profile defines a set of call capabilities relating to signaling and media.

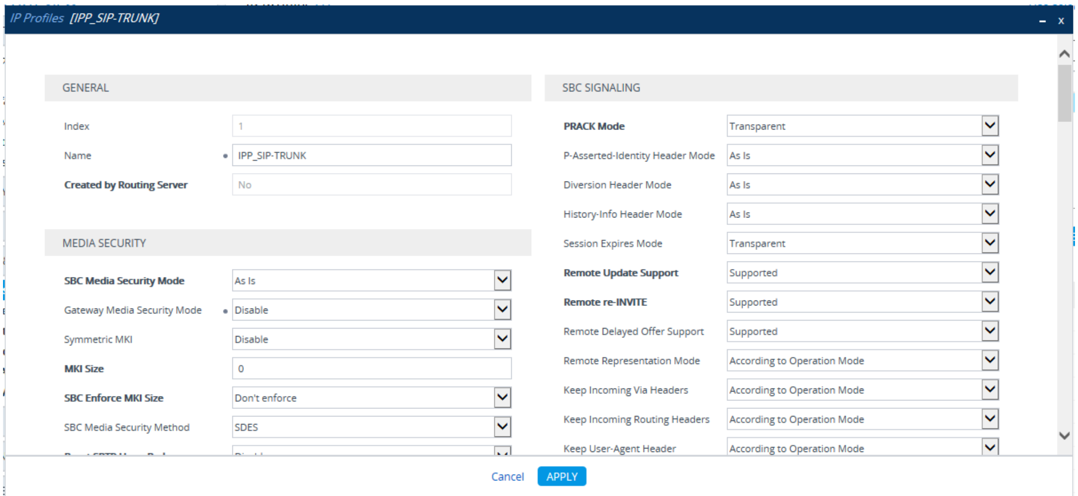
In this interoperability test topology, the IP Profile needs to be configured for the Swisscom SIP trunk IP entity:

➢ **To configure the IP Profile for the Swisscom 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	**IPP_SIP-TRUNK**
Media Security	
Gateway Media Security Mode	**Disable**
GATEWAY	
Early Media	**Enable**
Early 183	**Enable**
Coders Group	**AudioCodersGroups_0**
GATEWAY DTMF	
Is DTMF Used	**Enable**
GATEWAY FAX AND MODEM	
Fax Signaling Method	**No Fax**

Figure 3-6: Configuring IP Profile for Skype for Business Server 2015



3.7 Step 7: Configure IP Group

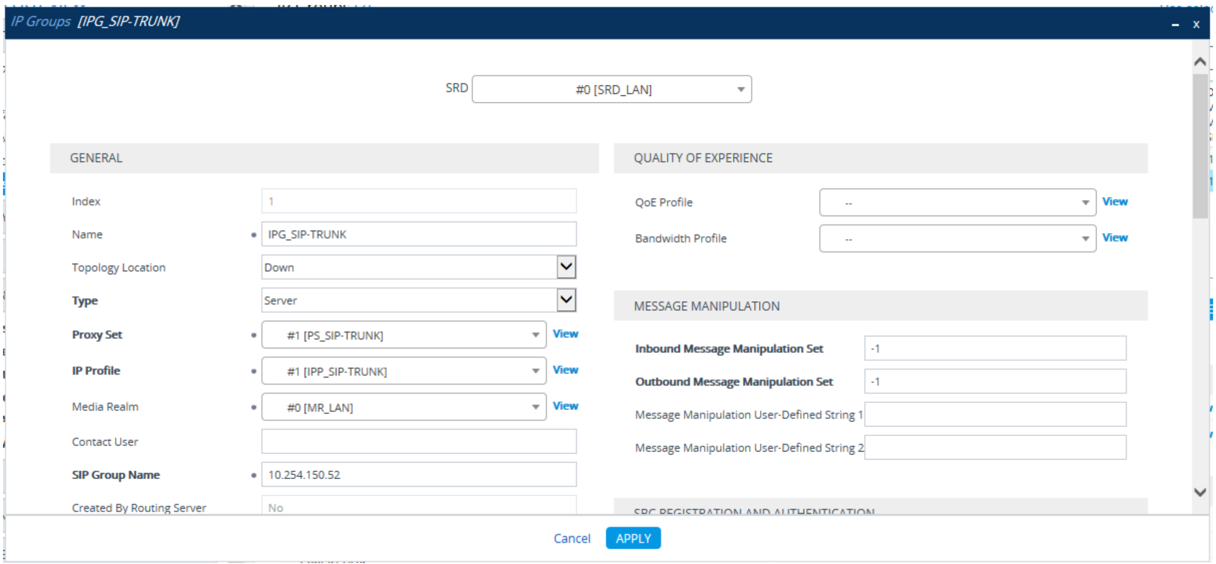
This step describes how to configure the IP Group. The IP Group represents an IP entity on the network with which the Gateway communicates. This can be a server (e.g., IP PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. In this test topology, IP Group is configured for the Swisscom SIP Trunk.

➢ **To configure the IP Group:**

1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
2. Add an IP Group for the Swisscom SIP Trunk as shown below:

Parameter	Value
Index	**1**
Type	**Server**
Description	**IPG_SIP-TRUNK** (arbitrary descriptive name)
Proxy Set ID	**PS_SIP-TRUNK**
IP Profile	**IPP_SIP-TRUNK**
Media Realm Name	**MR_LAN**
SIP Group Name	**10.254.150.52**

Figure 3-7: Configuring IP Group for Swisscom



3.8 Step 8: Configure PSTN Trunk Settings

This step describes how to configure PSTN trunk settings for BRI and PRI PSTN interfaces.

3.8.1 Step 8a: Configure the BRI PSTN Interface

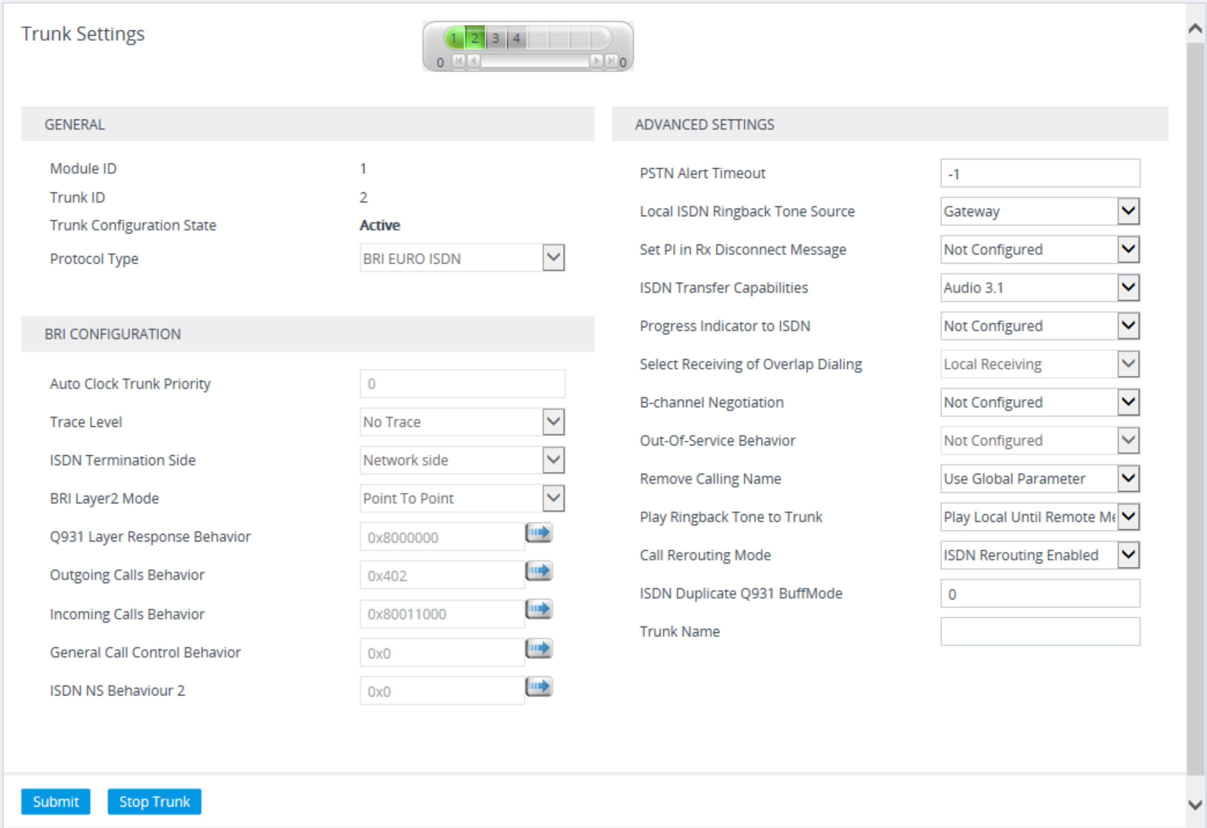
This step describes how to configure the BRI PSTN Interface. Skip to the next step if you have a PRI interface.

To configure the BRI PSTN interface:

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

Parameter	Value
Protocol Type	**BRI EURO ISDN**
ISDN Termination Side	**Network side (for BRI PBX connection)**
BRI Layer2 Mode	**Point To Point**
Q931 Layer Response Behavior	**0x8000000**
Outgoing Calls Behavior	**0x402**
Incoming Calls Behavior	**0x80011000**
Local ISDN Ringback Tone Source	**Gateway**
ISDN Transfer Capabilities	**Audio 3.1**
Select Receiving of Overlap Dialing	**Local Receiving**
Play Ringback Tone to Trunk	**Play Local Until Remote Media Arrive**
Call Rerouting Mode	**ISDN Rerouting Enabled**

Figure 3-8: Configuring BRI PSTN Interface



3. Repeat for all BRI ports available on the device.

3.8.2 Step 8b: Configure the PRI PSTN Interface

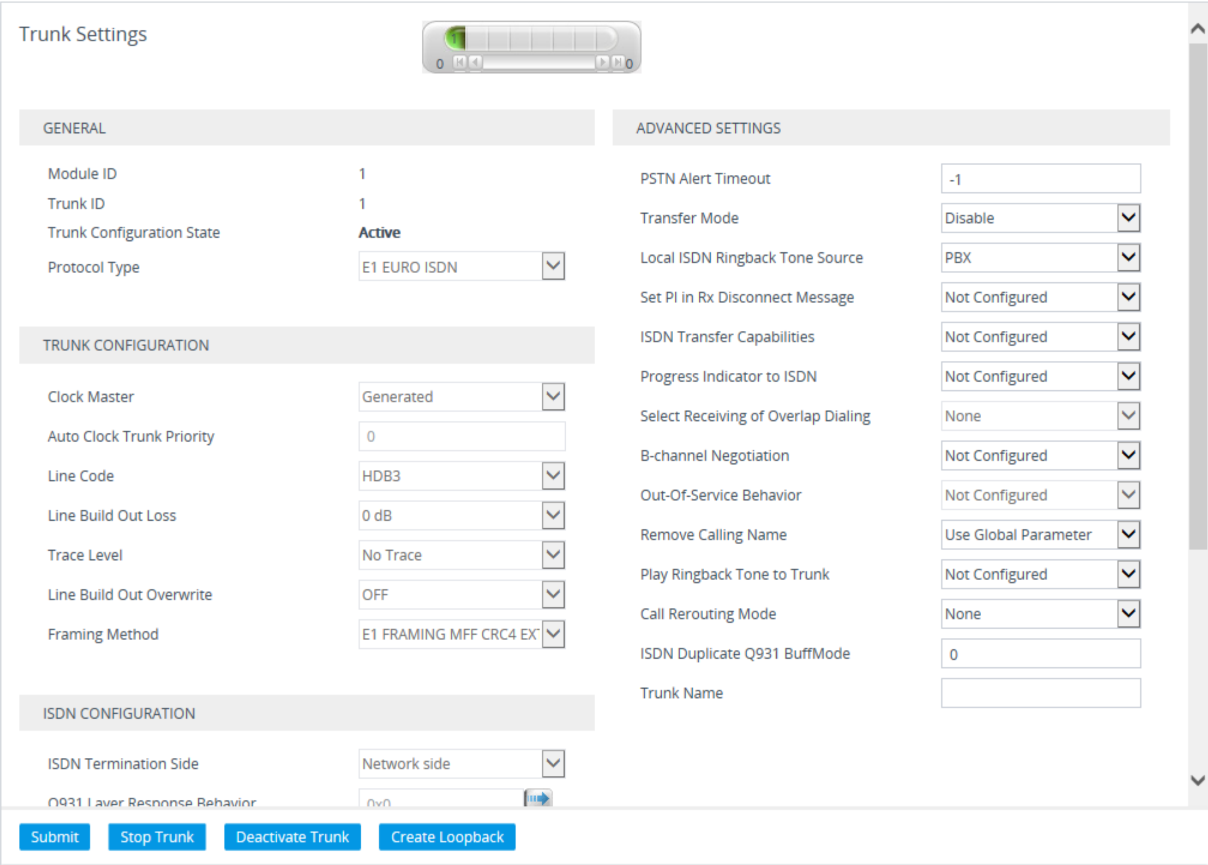
This step describes how to configure the PRI PSTN Interface.

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:

Parameter	Value
Protocol Type	**E1 EURO ISDN**
Clock Master	**Generated**
Line Code	**HDB3**
Framing Method	**E1 Framing MFF CRC4 Ext**
ISDN Termination Side	**Network side**
Outgoing Calls Behavior	**0x402**
Incoming Calls Behavior	**0x80011000**
Transfer Mode	**ECT**
Local ISDN Ringback Tone Source	**Gateway**
ISDN Transfer Capabilities	**Audio 3.1**
Select Receiving of Overlap Dialing	**Local Receiving**
Play Ringback Tone to Trunk	**Play Local Until Remote Media Arrive**
B-channel Negotiation	**Preferred**
Call Rerouting Mode	**ISDN Rerouting Enabled**

Figure 3-9: Configuring PRI PSTN Interface



3. Repeat for all PRI ports available on the device.

3.8.3 Step 8c: Configure the TDM Bus

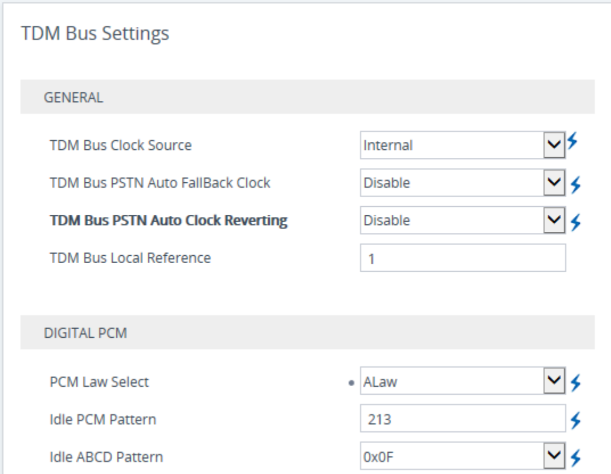
This step describes how to configure the Gateway's TDM bus.

➢ **To configure the TDM bus:**

1. Open the TDM Bus Settings page (**Setup** menu **> Signaling & Media** tab **> Gateway** folder **> TDM Bus Settings**).
2. Configure the TDM bus parameters per your deployment requirements. Below is an example:

Parameter	Value
TDM Bus Clock Source	**Internal**
PCM Law Select	**ALaw**

Figure 3-10: TDM Bus Settings Page



3.9 Step 9: Configure Trunk Group Parameters

This step 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.

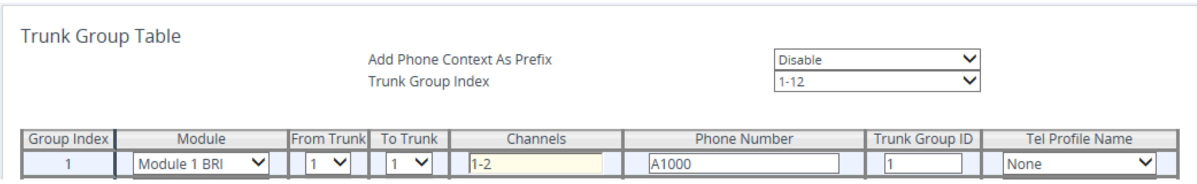
3.9.1 Step 9a: Configure the BRI Trunk Group

This step describes how to configure the BRI Trunk Group. Skip to the next step if you have a PRI interface.

➢ **To configure the BRI Trunk Group Table:**

1. Open the Trunk Group Table page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunk Groups**).
2. Configure each Trunk Group as required. If more than one BRI port is available, on line 1 of the table above, set “To Trunk” to the last BRI port to be used for incoming / outgoing calls between Swisscom and the PBX.

Figure 3-11: Configuring BRI Trunk Group Table



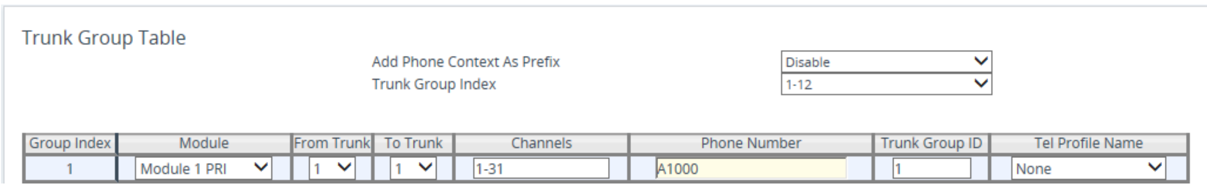
3.9.2 Step 9b: Configure the PRI Trunk Group

This section shows how to configure the PRI Trunk Group.

To configure the PRI Trunk Group Table:

1. Open the Trunk Group Table page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunk Groups**).
2. Configure each Trunk Group as required. If more than one PRI port is available, on line 1 of the table above, set “To Trunk” to the last PRI port to be used for incoming / outgoing calls between Swisscom and the PBX.

Figure 3-12: Configuring PRI Trunk Group Table



3.9.3 Step 9c: Configure Trunk Group Settings

The Trunk Group Settings page allows you to configure the following per trunk group:

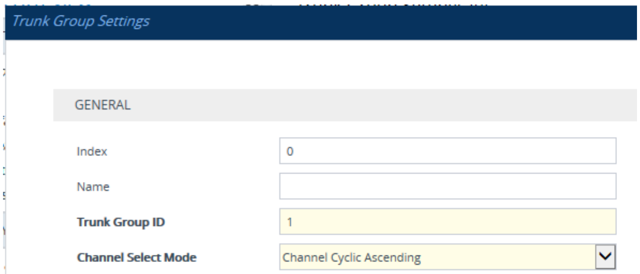
- Channel Select Mode by which IP-to-Tel calls are assigned to the Trunk Group's channels

➤ **To configure the Trunk Group Settings:**

1. Open the Trunk Group Table page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Trunks & Groups** > **Trunk Group Settings**).
2. Click **New**.
3. Configure the following parameters:

Parameter	Value
Trunk Group ID	**1**
Channel Select Mode	**Channel Cyclic Ascending**

Figure 3-13: Configuring Trunk Group Settings



3.10 Step 10: Configure Routing Rules

This step describes how to configure IP-to-Tel and Tel-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to the Trunk Group and vice versa.

3.10.1 Step 10a: Configure Tel-to-IP Routing

This step describes how to configure the Mediant BRI/PRI Gateway Tel-to-IP Routing, whereby all calls from the Trunk Group 1 (i.e., PSTN) are routed to the Swisscom SIP Trunk.

➢ **To configure Tel-to-IP or Outbound IP Routing Rules:**

1. Open the Outbound IP Routing Table page (**Setup** menu **> Signaling & Media** tab **> Gateway** folder **> Routing > Tel -> IP Routing**).
2. Click **New.**
3. Configure a rule for all incoming IP calls. Route them to 'Destination IP Group' **IPG_SIP-TRUNK** (connected to the Swisscom).
4. Click **Apply**.

Figure 3-14: Configured Tel-to-IP Routing Rules

Tel-to-IP Routing (1)

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

INDEX	NAME	SOURCE TRUNK GROUP ID	SOURCE PHONE PREFIX	DESTINATION PHONE PREFIX	DESTINATION IP GROUP	SIP INTERFACE	DESTINATION IP ADDRESS	FORKING GROUP	CONNECTIVITY STATUS
0	PBX-to-SIP-Trunk	-1	*	*	IPG_SIP-TRUNK	--		-1	Not Available

3.10.2 Step 10b: Configure IP-to-Tel Routing

This step describes how to configure Mediant BRI/PRI Gateway IP-to-Tel Routing, whereby all calls from the Swisscom SIP Trunk are routed to Trunk Group 1.

➢ **To configure IP-to-Tel or Inbound IP Routing Rules:**

1. Open the Inbound IP Routing Table page (**Setup** menu **> Signaling & Media** tab **> Gateway** folder **> Routing > IP -> Tel Routing**).
2. Click **New**.
3. Configure a rule for all incoming IP calls, with any destination prefix assigned and route them to 'Trunk Group ID' **1** (connected to the PBX).
4. Click **Apply**.

Figure 3-15: Configured IP-to-Tel Routing Rules

IP-to-Tel Routing (1)

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

INDEX	NAME	SOURCE IP GROUP	SOURCE SIP INTERFACE	SOURCE IP ADDRESS	SOURCE PHONE PREFIX	DESTINATION PHONE PREFIX	TRUNK GROUP ID
0	SIP-Trunk-to-PBX	--	Any			*	1

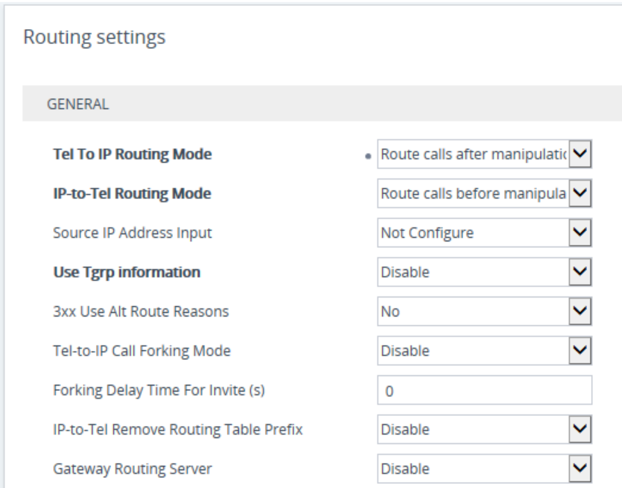
3.10.3 Step 10c: Configure Routing settings

This section identifies the device configuration needed in the Routing settings

➢ **To configure Routing settings:**

1. Open the Routing settings page (**Setup** menu **> Signaling & Media** tab **> Gateway** folder **> Routing > Routing Settings**).
2. From the 'Tel to IP Routing Mode' drop-down list, select **Route calls after manipulation**.
3. Click **Apply**.

Figure 3-16: Routing settings Page



3.11 Step 11: Configure Normalization Rules for E.164 Format for PBX/PSTN Connectivity

Swisscom implements the standard E.164 format, while the PBX or PSTN implements other number formats for dialing. If the Gateway is connected to a PBX or directly to the PSTN, it may need to perform number manipulations for the called and/or calling number to match the PBX or PSTN dialing rules or to match Swisscom E.164 format.

The Gateway entity must therefore be configured with manipulation rules to translate (i.e., normalize) numbers dialed in standard E.164 format to various formats, and vice versa. Manipulation must be performed for outbound calls and inbound calls.

Number manipulation rules are configured in the following Manipulation Tables:

For Tel-to-IP calls:

- Destination Phone Number Manipulation Table for Tel-to-IP Calls
- Source Phone Number Manipulation Table for Tel-to-IP Calls

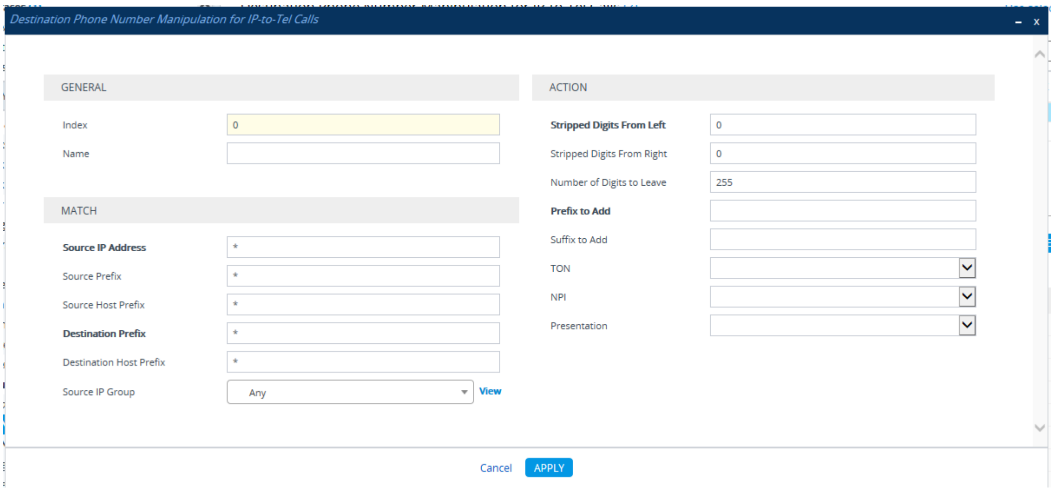
For IP-to-Tel calls:

- Destination Phone Number Manipulation Table for IP-to-Tel Calls
- Source Phone Number Manipulation Table for IP-to-Tel Calls

To configure number manipulation rules:

1. Open the required Number Manipulation page (**Setup** menu **> Signaling & Media** tab **> Gateway** folder **> Manipulations** > **Dest Number IP->Tel** or **Dest Number Tel->IP** or **Source Number IP->Tel** or **Source Number Tel->IP**); the relevant Manipulation table page is displayed.
2. Click the **New** button; the following screen is displayed:

Figure 3-17: Example Dest Number IP->Tel Number Manipulation Rule



3. Configure the matching characteristics.
4. Configure the manipulation actions.
5. Click **Apply** to submit your changes.

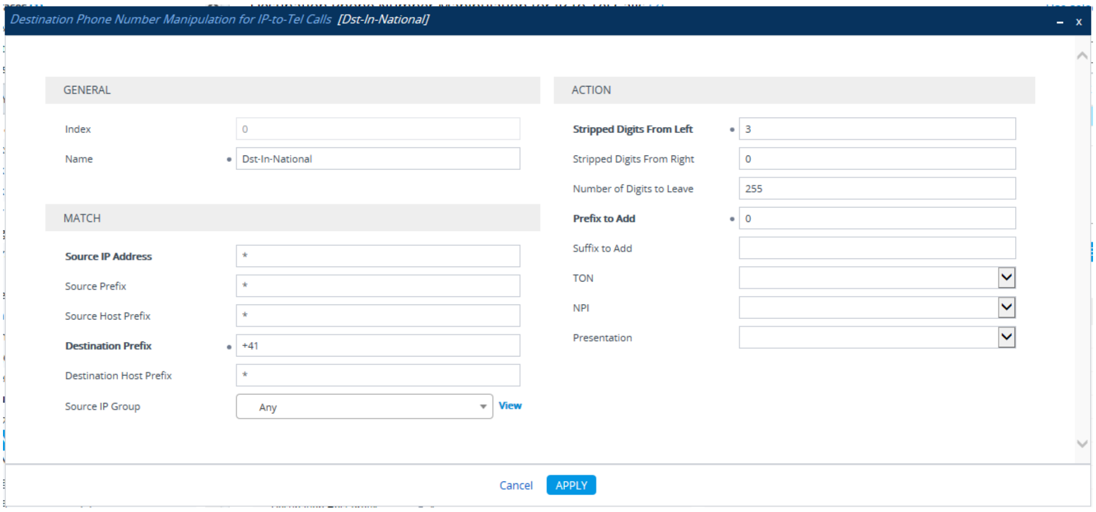
3.11.1 Number Manipulation Examples

Two examples are provided below for number manipulation.

3.11.1.1 Number Manipulation IP to Tel Example

The example below shows a manipulation rule that removes "+41" from the destination number when the destination number prefix "+41".

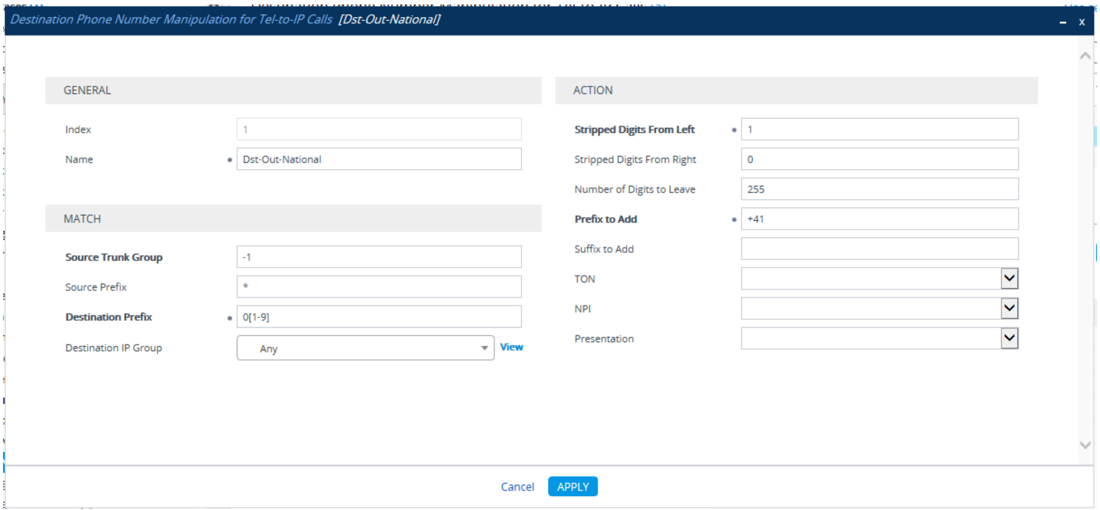
Figure 3-18: Destination Number Manipulation Rule for IP→Tel Calls



3.11.1.2 Number Manipulation Tel to IP Example

The example below shows a National manipulation rule that removes the "0" prefix and adds "+41" to the destination number, when the destination number prefix is "0[1-9]".

Figure 3-19: Destination Number Manipulation Rule for Tel→IP Calls



Note: Adapt the Manipulation Table according to your environment's dial plan.

3.12 Step 12: Configure Message Manipulation Rules

This step 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).

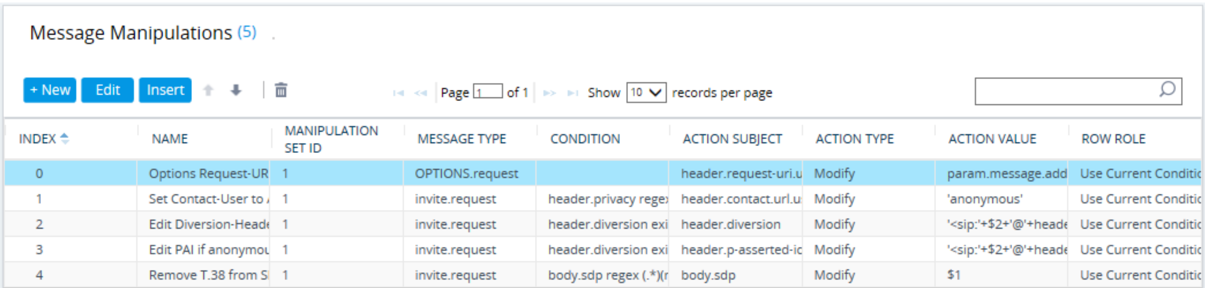
Once you have configured the SIP message manipulation rules, you need to assign them to the GW and determine whether they are applied to inbound or outbound messages.

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

1. Open the Message Manipulations page (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulations**).
2. Configure new manipulation rules (Manipulation Set 1) for Swisscom SIP Trunk according to the table below:

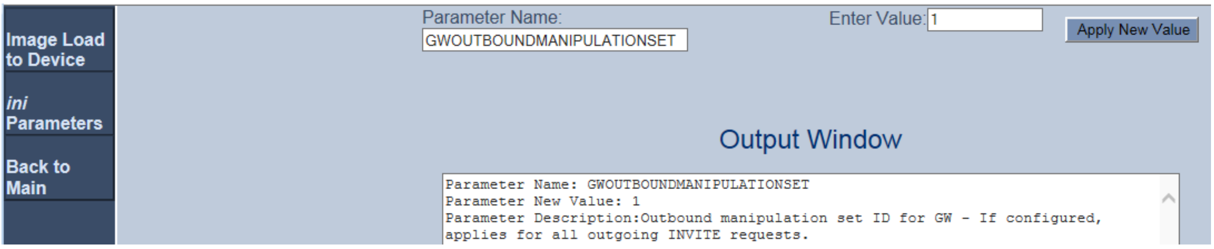
Index	MMS Rule Name	Specific Configuration					
		Set ID	Message Type	Condition	Action Subject	Action Type	Action Value
0	Options Request-URI	1	options.request		header.request-uri.url.host	Modify	param.message.address.dst.address
1	Set Contact-User to Anonymous	1	invite.request	header.privacy regex (id)	header.contact.url.user	Modify	'anonymous'
2	Edit Diversion-Header	1	invite.request	header.diversion exists AND header.diversion regex (<tel:)(.*)(>.*)	header.diversion	Modify	'<sip:'+$2+'@'+header.contact.url.host+$3
3	Edit PAI if anonymous	1	invite.request	header.diversion exists AND header.diversion regex (<sip:)(.*)(@.*) AND header.privacy=='id'	header.p-asserted-identity	Modify	'<sip:'+$2+'@'+header.contact.url.host+'>'
4	Remove T.38 from SDP	1	invite.request	body.sdp regex (.*)(m=image)(.*)	body.sdp	Modify	$1

Figure 3-20: Configured SIP Message Manipulation Rules



3. Assign Manipulation Set ID 1 to the GW outbound messages:
   a. Open the Admin page.
   b. Append the case-sensitive suffix ‘AdminPage’ to the device's IP address in your Web browser's URL field (e.g., http://10.15.45.110/AdminPage).
   c. In the left pane of the page that opens, click ***ini* Parameters**.
   d. In the 'Parameter Name' Enter **GWOUTBOUNDMANIPULATIONSET** and in the 'Enter Value' enter **1**.
   e. Click **Apply New Value**.

Figure 3-21: Assigning Manipulation Set 1 to the GWOUTBOUNDMANIPULATIONSET



3.13 Step 13: Configure Miscellaneous Settings

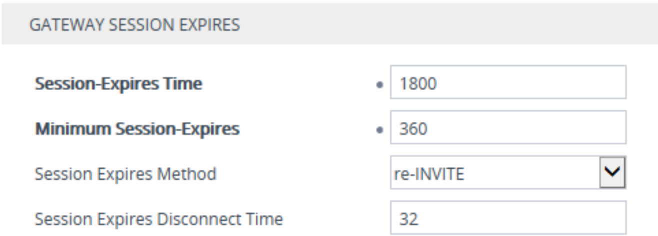
This step describes miscellaneous Gateway configuration

3.13.1 Step 13a: Configure Session-Expires

This step describes how to configure Gateway Session-Expires times.

1. Open the Gateway General Settings page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **SIP Definitions General Settings**).
2. From the 'Session-Expires' set it to **1800**.
3. From the 'Minimum Session-Expires' set it to **360**.
4. Click **Apply**.

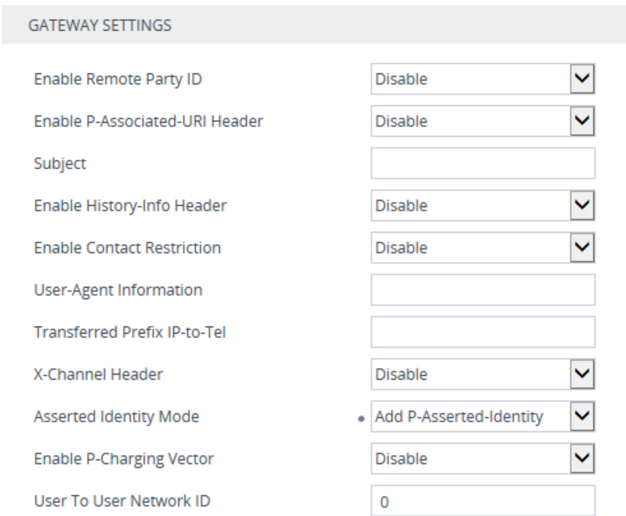
Figure 3-22: General Settings Page



3.13.2 Step 13b: Configure Asserted Identity Mode

1. Open the Message Structure page (**Setup** menu > **Signaling & Media** tab > **SIP Definitions** folder > **Message Structure**).
2. From the 'Asserted Identity Mode' dropdown, select **Add P-Asserted-Identity.**
3. Click **Apply**.

Figure 3-23: Message Structure Page



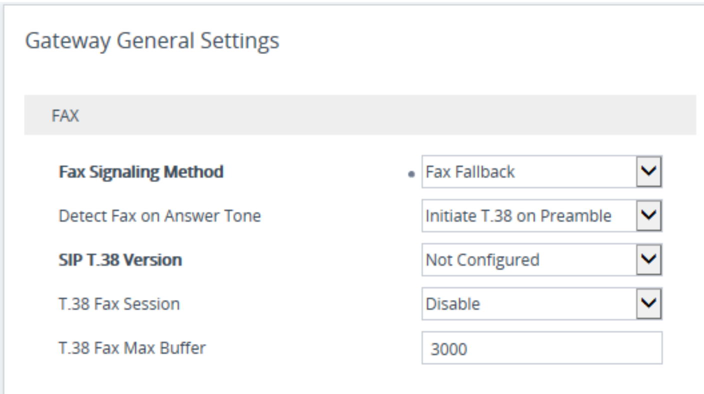
3.13.3 Step 13c: Configure Gateway General Settings

This step identifies the device configuration needed in the Gateway General Settings configuration.

To configure the Gateway General Settings:

1. Open the SIP Proxy & Registration Parameters page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **Gateway General Settings**).
2. From the ‘Fax Signaling Method' drop-down list, select **Fax Fallback**.
3. Click **Apply**.

Figure 3-24: Gateway General Settings Page



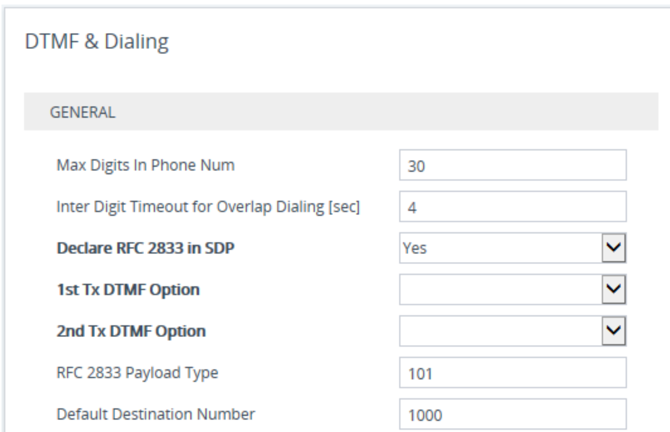
3.13.4 Step 13d: Configure DTMF & Dialing

This step identifies the device configuration needed in the DTMF & Dialing configuration.

To configure the DTMF & Dialing parameters:

1. Open the SIP DTMF & Dialing page (**Setup** menu > **Signaling & Media** tab > **Gateway** folder > **DTMF and Supplementary > DTMF & Dialing**).
2. From the 'RFC 2833 Payload Type' set it to **101**.
3. Click **Apply**.

Figure 3-25: DTMF & Dialing Page



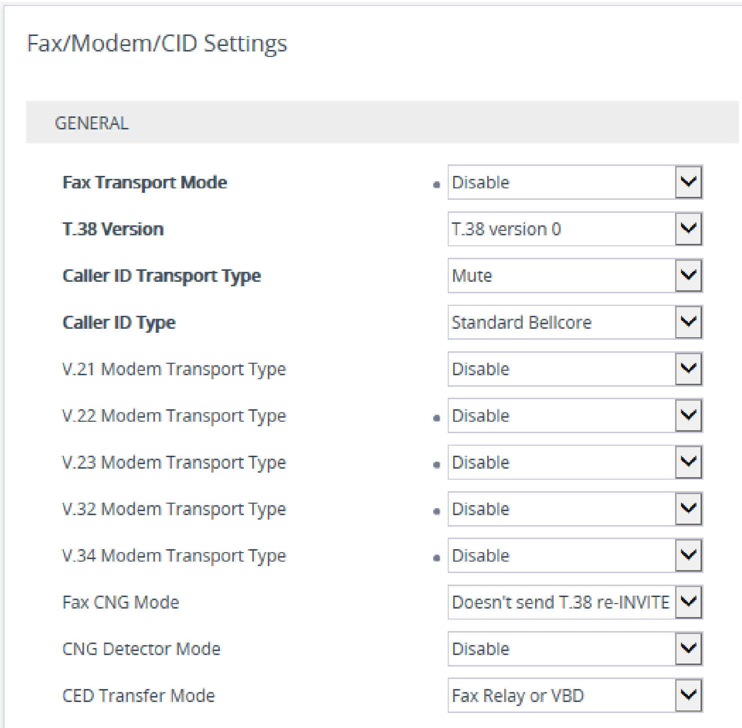
3.13.5 Step 13e: Configure Fax Parameters

This step identifies the device configuration needed in the Fax configuration.

➢ **To configure the Fax parameters:**

1. Open the Fax/Modem/CID Settings page (**Setup** menu > **Signaling & Media** tab > **Media** folder > **Fax/Modem/CID Settings**).
2. From the 'Fax Transport Mode' drop-down list, select **Disable**.
3. From the 'V.22 Modem Transport Type' drop-down list, select **Disable**.
4. From the 'V.23 Modem Transport Type' drop-down list, select **Disable**.
5. From the 'V.32 Modem Transport Type' drop-down list, select **Disable**.
6. From the 'V.34 Modem Transport Type' drop-down list, select **Disable**.
7. Click **Apply**.

Figure 3-26: Fax Settings Page



3.13.6 Step 13f: Configure Parameters using the AdminPage

This step describes how to configure additional Gateway parameters needed via the AdminPage.

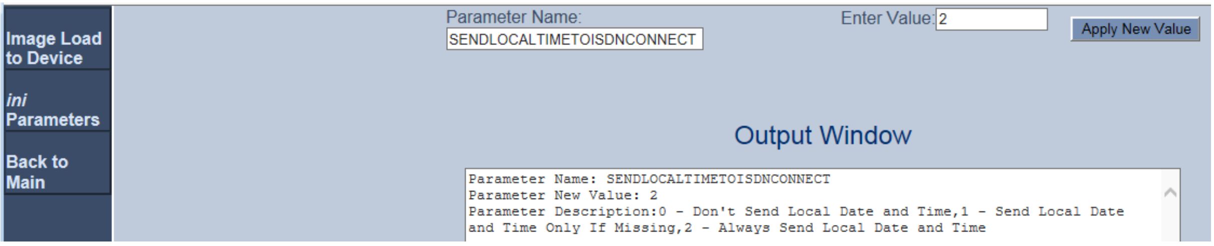
➢ **To configure Parameters using the AdminPage:**

1. Open the Admin page.
2. Append the case-sensitive suffix 'AdminPage' to the device's IP address in your Web browser's URL field (e.g., http://10.15.45.110/AdminPage).
3. In the left pane of the page that opens, click ***ini* Parameters**.
4. Enter these values in the 'Parameter Name' and 'Enter Value' fields:

Parameter	Value	Parameter description
SendLocalTimeToISDNConnect	**2**	The device always sends its local date and time (obtained from its internal clock) in Connect messages.
TransparentCoderOnDataCall	**1**	If the transfer capability of a call from ISDN is "data", open with the transparent coder.

5. Click the **Apply New Value** button for each parameter.

Figure 3-27: Configuring a Parameter in AdminPage



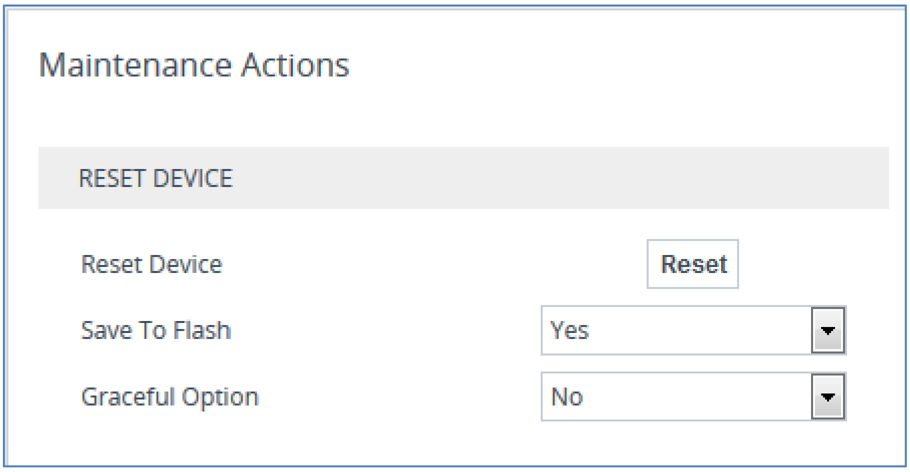
3.14 Step 14: Reset the Gateway

After you have completed the configuration of the Gateway described in this step, save ("burn") the configuration to the Gateway's flash memory with a reset for the settings to take effect.

➢ **To save the configuration to flash memory:**

1. Open the Maintenance Actions page (**Setup** menu > **Administration** tab > **Maintenance** folder > **Maintenance Actions**).

Figure 3-28: Resetting the Gateway



2. Ensure that the 'Burn to FLASH' field is set to **Yes** (default).
3. Click the **Reset** button.

A AudioCodes INI File

The *ini* configuration file of the Gateway with BRI, corresponding to the Web-based configuration as described in Section 3 on page 11, 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 **
;**************

;Board: M800
;HW Board Type: 69  FK Board Type: 72
;Serial Number: 3161551
;Slot Number: 1
;Software Version: 7.20A.104.001
;DSP Software Version: 5014AE3_R => 721.09
;Board IP Address: 10.15.45.110
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 10.15.0.1
;Ram size: 512M   Flash size: 64M   Core speed: 300Mhz
;Num of DSP Cores: 1  Num DSP Channels: 30
;Num of physical LAN ports: 12
;Profile: NONE
;;;Key features:;Board Type: M800 ;E1Trunks=2 ;T1Trunks=2 ;BRITrunks=8
;Security: IPSEC MediaEncryption StrongEncryption EncryptControlProtocol
;Channel Type: RTP DspCh=30 IPMediaDspCh=30 ;DATA features: ;QOE
features: VoiceQualityMonitoring MediaEnhancement ;DSP Voice features:
IpmDetector RTCP-XR ;IP Media: VXML ;Coders: G723 G729 G728 NETCODER GSM-
FR GSM-EFR AMR EVRC-QCELP G727 ILBC EVRC-B AMR-WB G722 EG711 MS_RTA_NB
MS_RTA_WB SILK_NB SILK_WB SPEEX_NB SPEEX_WB OPUS_NB OPUS_WB ;Control
Protocols: MGCP SIP TPNCP CLI FEU=10 TestCall=10 EMS ;Default
features:;Coders: G711 G726;

;------  HW components------
;
; Slot # : Module type : # of ports
;----------------------------------------------
;      1 : BRI         : 4
;      2 : Empty
;      3 : Empty
;----------------------------------------------
;---------  HW components ---------
;
; Slot # : Module type : # of ports : # of DSPs
;----------------------------------------------
;      1 : FALC56      :          1 :         2
;      2 : BRI         :          4 :         2
;      3 : Empty
;      4 : Empty
;      5 : Empty
;      6 : Empty
;----------------------------------------------
```

```
[SYSTEM Params]

SyslogServerIP = 10.254.100.51
EnableSyslog = 1
;IniFileLastUpdateTime is hidden but has non-default value
;IniFileTemplateLastUpdateTime is hidden but has non-default value
;VpFileLastUpdateTime is hidden but has non-default value
TR069ACSPASSWORD = '$1$gQ=='
TR069CONNECTIONREQUESTPASSWORD = '$1$gQ=='
NTPServerIP = '0.0.0.0'
;LastConfigChangeTime is hidden but has non-default value
;BarrierFilename is hidden but has non-default value

[BSP Params]

PCMLawSelect = 1
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
ExitCpuOverloadPercent = 95

[Analog Params]

PolarityReversalType = 1
MinFlashHookTime = 100

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]

EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 1
EP_Num_3 = 0
EP_Num_4 = 0

[PSTN Params]


ProtocolType_1 = 50
ClockMaster = 1
TerminationSide = 1
FramingMethod_1 = 0
LineCode_1 = 0
ISDNIBehavior_1 = 134217728
ISDNInCallsBehavior = 2147553280
ISDNOutCallsBehavior = 1026

[SS7 Params]
```

```
[Voice Engine Params]

FaxTransportMode = 0
V22ModemTransportType = 0
V23ModemTransportType = 0
V32ModemTransportType = 0
V34ModemTransportType = 0
RFC2833TxPayloadType = 101
CallProgressTonesFilename = 'switzerland.dat'

[WEB Params]

LogoWidth = '145'
HTTPSCipherString = 'RC4:EXP'
;HTTPSPkeyFileName is hidden but has non-default value
;HTTPSCertFileName is hidden but has non-default value

[SIP Params]

ROUTEMODETEL2IP = 1
GWDEBUGLEVEL = 5
ISDNRXOVERLAP = 1
SIPSESSIONEXPIRES = 1800
ASSERTEDIDMODE = 1
TRANSPARENTCODERONDATACALL = 1
MINSE = 360
ISFAXUSED = 3
LOCALISDNRBSOURCE = 1
ISDNTRANSFERCAPABILITY = 0
PLAYRBTONE2TRUNK = 3
MSLDAPPRIMARYKEY = 'telephoneNumber'
CALLREROUTINGMODE = 1
SENDLOCALTIMETOISDNCONNECT = 2
GWOUTBOUNDMANIPULATIONSET = 1
ENERGYDETECTORCMD = 587202560
ANSWERDETECTORCMD = 10485760
;GWAPPCONFIGURATIONVERSION is hidden but has non-default value

[SCTP Params]


[VXML Params]


[IPsec Params]


[SNMP Params]


[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion,
TLSContexts_DTLSVersion, TLSContexts_ServerCipherString,
TLSContexts_ClientCipherString, TLSContexts_RequireStrictCert,
TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary,
```

```
TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort,
TLSContexts_OcspDefaultResponse, TLSContexts_DHKeySize;
TLSContexts 0 = "default", 0, 0, "RC4:AES128", "DEFAULT", 0, 0, , , 2560,
0, 1024;

[ \TLSContexts ]


[ AudioCodersGroups ]

FORMAT AudioCodersGroups_Index = AudioCodersGroups_Name;
AudioCodersGroups 0 = "AudioCodersGroups_0";

[ \AudioCodersGroups ]


[ IpProfile ]

FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupName, IpProfile_IsFaxUsed,
IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor,
IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_CNGmode,
IpProfile_VxxTransportType, IpProfile_NSEMode, IpProfile_IsDTMFUsed,
IpProfile_PlayRBTone2IP, IpProfile_EnableEarlyMedia,
IpProfile_ProgressIndicator2IP, IpProfile_EnableEchoCanceller,
IpProfile_CopyDest2RedirectNumber, IpProfile_MediaSecurityBehaviour,
IpProfile_CallLimit, IpProfile_DisconnectOnBrokenConnection,
IpProfile_FirstTxDtmfOption, IpProfile_SecondTxDtmfOption,
IpProfile_RxDTMFOption, IpProfile_EnableHold, IpProfile_InputGain,
IpProfile_VoiceVolume, IpProfile_AddIEInSetup,
IpProfile_SBCExtensionCodersGroupName,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedAudioCodersGroupName,
IpProfile_SBCAllowedVideoCodersGroupName, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel,
IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionMode, IpProfile_SBCHistoryInfoMode,
IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupName,
IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode,
IpProfile_SBCFaxAnswerMode, IpProfile_SbcPrackMode,
IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport,
IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior,
IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport, IpProfile_EnableSymmetricMKI,
IpProfile_MKISize, IpProfile_SBCEnforceMKISize,
IpProfile_SBCRemoteEarlyMediaRTP, IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarly183,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey,
IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource,
IpProfile_GenerateSRTPKeys, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_SBCRemoteReplacesBehavior,
IpProfile_SBCSDPPtimeAnswer, IpProfile_SBCPreferredPTime,
IpProfile_SBCUseSilenceSupp, IpProfile_SBCRTPRedundancyBehavior,
IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,
IpProfile_JitterBufMaxDelay,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
```

```
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCSDPHandleRTCPAttribute,
IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode,
IpProfile_SBCRTCPMux, IpProfile_SBCMediaSecurityMethod,
IpProfile_SBCHandleXDetect, IpProfile_SBCRTCPFeedback,
IpProfile_SBCRemoteRepresentationMode, IpProfile_SBCKeepVIAHeaders,
IpProfile_SBCKeepRoutingHeaders, IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCRemoteMultipleEarlyDialogs,
IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBCDirectMediaTag,
IpProfile_SBCAdaptRFC2833BWToVoiceCoderBW,
IpProfile_CreatedByRoutingServer, IpProfile_SBCFaxReroutingMode,
IpProfile_SBCMaxCallDuration, IpProfile_SBCGenerateRTP,
IpProfile_SBCISUPBodyHandling, IpProfile_SBCISUPVariant,
IpProfile_SBCVoiceQualityEnhancement, IpProfile_SBCMaxOpusBW;
IpProfile 1 = "IPP_SIP-TRUNK", 1, "AudioCodersGroups_0", 0, 10, 10, 46,
24, 0, 0, 0, 0, 0, 1, 0, 1, -1, 1, 0, 2, -1, 1, 4, -1, 1, 1, 0, 0, "",
"", 0, 0, "", "", "", 0, 0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, "", 0, 0,
1, 3, 0, 2, 2, 1, 0, 0, 1, 0, 1, 0, 0, 0, 0, 0, 1, 1, 0, 0, 0, 0, 0, 1,
0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 250, -1, -1, 0, 0, 0, 0, 0, 0, 0, -1,
-1, -1, -1, -1, 0, "", 0, 0, 0, 0, 0, 0, 0, 0, 0;

[ \IpProfile ]


[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart,
CpMediaRealm_MediaSessionLeg, CpMediaRealm_PortRangeEnd,
CpMediaRealm_IsDefault, CpMediaRealm_QoeProfile, CpMediaRealm_BWProfile,
CpMediaRealm_TopologyLocation;
CpMediaRealm 0 = "MR_LAN", "NETIF_LAN", "", 6000, 521, 11209, 1, "", "",
0;

[ \CpMediaRealm ]


[ SBCRoutingPolicy ]

FORMAT SBCRoutingPolicy_Index = SBCRoutingPolicy_Name,
SBCRoutingPolicy_LCREnable, SBCRoutingPolicy_LCRAverageCallLength,
SBCRoutingPolicy_LCRDefaultCost, SBCRoutingPolicy_LdapServerGroupName;
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 1, 0, "";

[ \SBCRoutingPolicy ]


[ SRD ]

FORMAT SRD_Index = SRD_Name, SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers,
SRD_EnableUnAuthenticatedRegistrations, SRD_SharingPolicy,
SRD_UsedByRoutingServer, SRD_SBCOperationMode, SRD_SBCRoutingPolicyName,
SRD_SBCDialPlanName;
SRD 0 = "SRD_LAN", 0, -1, 1, 0, 0, 0, "Default_SBCRoutingPolicy", "";

[ \SRD ]


[ SIPInterface ]
```

```
FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPPort, SIPInterface_TLSPort,
SIPInterface_SRDName, SIPInterface_MessagePolicyName,
SIPInterface_TLSContext, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet, SIPInterface_EncapsulatingProtocol,
SIPInterface_MediaRealm, SIPInterface_SBCDirectMedia,
SIPInterface_BlockUnRegUsers, SIPInterface_MaxNumOfRegUsers,
SIPInterface_EnableUnAuthenticatedRegistrations,
SIPInterface_UsedByRoutingServer, SIPInterface_TopologyLocation;
SIPInterface 0 = "SI_LAN", "NETIF_LAN", 0, 5060, 5060, 0, "SRD_LAN", "",
"default", -1, 0, 500, -1, 0, "MR_LAN", 0, -1, -1, -1, 0, 0;

[ \SIPInterface ]


[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRDName, ProxySet_ClassificationInput, ProxySet_TLSContextName,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName,
ProxySet_SBCIPv4SIPInterfaceName, ProxySet_GWIPv6SIPInterfaceName,
ProxySet_SBCIPv6SIPInterfaceName, ProxySet_MinActiveServersLB,
ProxySet_SuccessDetectionRetries, ProxySet_SuccessDetectionInterval,
ProxySet_FailureDetectionRetransmissions;
ProxySet 0 = "PS_DEFAULT", 0, 60, 0, 0, "SRD_LAN", 0, "", -1, -1, "",
"SI_LAN", "", "", "", 1, 1, 10, -1;
ProxySet 1 = "PS_SIP-TRUNK", 1, 10, 0, 0, "SRD_LAN", 1, "", 1, -1, "",
"SI_LAN", "", "", "", 1, 1, 10, -1;

[ \ProxySet ]


[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_SRDName, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileName,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UUIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2, IPGroup_SIPConnect, IPGroup_SBCPSAPMode,
IPGroup_DTLSContext, IPGroup_CreatedByRoutingServer,
IPGroup_UsedByRoutingServer, IPGroup_SBCOperationMode,
IPGroup_SBCRouteUsingRequestURIPort, IPGroup_SBCKeepOriginalCallID,
IPGroup_TopologyLocation, IPGroup_SBCDialPlanName,
IPGroup_CallSetupRulesSetId;
IPGroup 0 = 0, "IPG_DEFAULT", "PS_DEFAULT", "", "", -1, 0, "SRD_LAN", "",
0, "", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "", "$1$gQ==", 0, "", "", 0,
"", "", 0, 0, "default", 0, 0, -1, 0, 0, 0, "", -1;
IPGroup 1 = 0, "IPG_SIP-TRUNK", "PS_SIP-TRUNK", "10.254.150.52", "", -1,
0, "SRD_LAN", "MR_LAN", 1, "IPP_SIP-TRUNK", -1, -1, -1, 0, 0, "", 0, -1,
-1, "", "", "$1$gQ==", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -1,
0, 0, 0, "", -1;
```

```
[ \IPGroup ]


[ PREFIX ]

FORMAT PREFIX_Index = PREFIX_RouteName, PREFIX_DestinationPrefix,
PREFIX_DestAddress, PREFIX_SourcePrefix, PREFIX_ProfileName,
PREFIX_MeteringCodeName, PREFIX_DestPort, PREFIX_DestIPGroupName,
PREFIX_TransportType, PREFIX_SrcTrunkGroupID,
PREFIX_DestSIPInterfaceName, PREFIX_CostGroup, PREFIX_ForkingGroup,
PREFIX_CallSetupRulesSetId, PREFIX_ConnectivityStatus;
PREFIX 0 = "PBX-to-SIP-Trunk", "*", "", "*", "", "", 0, "IPG_SIP-TRUNK",
-1, -1, "", "", -1, -1, "Not Available";

[ \PREFIX ]


[ TrunkGroup ]

FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum,
TrunkGroup_FirstTrunkId, TrunkGroup_FirstBChannel,
TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber,
TrunkGroup_ProfileName, TrunkGroup_LastTrunkId, TrunkGroup_Module;
TrunkGroup 1 = 2, 0, 1, 2, "B100", "", 1, 2;

[ \TrunkGroup ]


[ NumberMapIp2Tel ]

FORMAT NumberMapIp2Tel_Index = NumberMapIp2Tel_ManipulationName,
NumberMapIp2Tel_DestinationPrefix, NumberMapIp2Tel_SourcePrefix,
NumberMapIp2Tel_SourceAddress, NumberMapIp2Tel_SrcHost,
NumberMapIp2Tel_DestHost, NumberMapIp2Tel_NumberType,
NumberMapIp2Tel_NumberPlan, NumberMapIp2Tel_RemoveFromLeft,
NumberMapIp2Tel_RemoveFromRight, NumberMapIp2Tel_LeaveFromRight,
NumberMapIp2Tel_Prefix2Add, NumberMapIp2Tel_Suffix2Add,
NumberMapIp2Tel_IsPresentationRestricted, NumberMapIp2Tel_SrcIPGroupName;
NumberMapIp2Tel 0 = "Dst-In-National", "+41", "*", "*", "*", "*", 255,
255, 3, 0, 255, "0", "", 255, "Any";
NumberMapIp2Tel 1 = "Dst-In-International", "+", "*", "*", "*", "*", 255,
255, 1, 0, 255, "00", "", 255, "Any";

[ \NumberMapIp2Tel ]


[ NumberMapTel2Ip ]

FORMAT NumberMapTel2Ip_Index = NumberMapTel2Ip_ManipulationName,
NumberMapTel2Ip_DestinationPrefix, NumberMapTel2Ip_SourcePrefix,
NumberMapTel2Ip_NumberType, NumberMapTel2Ip_NumberPlan,
NumberMapTel2Ip_RemoveFromLeft, NumberMapTel2Ip_RemoveFromRight,
NumberMapTel2Ip_LeaveFromRight, NumberMapTel2Ip_Prefix2Add,
NumberMapTel2Ip_Suffix2Add, NumberMapTel2Ip_IsPresentationRestricted,
NumberMapTel2Ip_SrcTrunkGroupID, NumberMapTel2Ip_DestIPGroupName;
NumberMapTel2Ip 0 = "Dst-Out-International", "00", "*", 255, 255, 2, 0,
255, "+", "", 255, -1, "Any";
NumberMapTel2Ip 1 = "Dst-Out-National", "0[1-9]", "*", 255, 255, 1, 0,
255, "+41", "", 255, -1, "Any";
```

```
NumberMapTel2Ip 2 = "Dst-Out-NoZero", "[1-9]xx.", "*", 255, 255, 0, 0,
255, "+41", "", 255, -1, "Any";

[ \NumberMapTel2Ip ]


[ SourceNumberMapIp2Tel ]

FORMAT SourceNumberMapIp2Tel_Index =
SourceNumberMapIp2Tel_ManipulationName,
SourceNumberMapIp2Tel_DestinationPrefix,
SourceNumberMapIp2Tel_SourcePrefix, SourceNumberMapIp2Tel_SourceAddress,
SourceNumberMapIp2Tel_SrcHost, SourceNumberMapIp2Tel_DestHost,
SourceNumberMapIp2Tel_NumberType, SourceNumberMapIp2Tel_NumberPlan,
SourceNumberMapIp2Tel_RemoveFromLeft,
SourceNumberMapIp2Tel_RemoveFromRight,
SourceNumberMapIp2Tel_LeaveFromRight, SourceNumberMapIp2Tel_Prefix2Add,
SourceNumberMapIp2Tel_Suffix2Add,
SourceNumberMapIp2Tel_IsPresentationRestricted,
SourceNumberMapIp2Tel_SrcIPGroupName;
SourceNumberMapIp2Tel 0 = "Src-In-National", "*", "+41", "*", "*", "*",
255, 255, 3, 0, 255, "0", "", 255, "Any";
SourceNumberMapIp2Tel 1 = "Src-In-International", "*", "+", "*", "*",
"*", 255, 255, 1, 0, 255, "00", "", 255, "Any";
SourceNumberMapIp2Tel 2 = "Src-In-Anonymous", "*", "anonymous", "*", "*",
"*", 255, 255, 0, 0, 0, "", "", 1, "Any";

[ \SourceNumberMapIp2Tel ]


[ SourceNumberMapTel2Ip ]

FORMAT SourceNumberMapTel2Ip_Index =
SourceNumberMapTel2Ip_ManipulationName,
SourceNumberMapTel2Ip_DestinationPrefix,
SourceNumberMapTel2Ip_SourcePrefix, SourceNumberMapTel2Ip_NumberType,
SourceNumberMapTel2Ip_NumberPlan, SourceNumberMapTel2Ip_RemoveFromLeft,
SourceNumberMapTel2Ip_RemoveFromRight,
SourceNumberMapTel2Ip_LeaveFromRight, SourceNumberMapTel2Ip_Prefix2Add,
SourceNumberMapTel2Ip_Suffix2Add,
SourceNumberMapTel2Ip_IsPresentationRestricted,
SourceNumberMapTel2Ip_SrcTrunkGroupID;
SourceNumberMapTel2Ip 0 = "Src-Out-International", "*", "00", 255, 255,
2, 0, 255, "+", "", 255, -1;
SourceNumberMapTel2Ip 1 = "Src-Out-National", "*", "0", 255, 255, 1, 0,
255, "+41", "", 255, -1;
SourceNumberMapTel2Ip 2 = "Src-Out-NoZero", "*", "[1-9]xx.", 255, 255, 0,
0, 255, "+41", "", 255, -1;

[ \SourceNumberMapTel2Ip ]


[ PstnPrefix ]

FORMAT PstnPrefix_Index = PstnPrefix_RouteName, PstnPrefix_DestPrefix,
PstnPrefix_TrunkGroupId, PstnPrefix_SourcePrefix,
PstnPrefix_SourceAddress, PstnPrefix_ProfileName,
PstnPrefix_SrcIPGroupName, PstnPrefix_DestHostPrefix,
PstnPrefix_SrcHostPrefix, PstnPrefix_SrcSIPInterfaceName,
PstnPrefix_TrunkId, PstnPrefix_CallSetupRulesSetId, PstnPrefix_DestType;
PstnPrefix 0 = "SIP-Trunk-to-PBX", "*", 2, "", "", "", "", "", "", "Any",
-1, -1, 0;
```

```
[ \PstnPrefix ]


[ CauseMapIsdn2Sip ]

FORMAT CauseMapIsdn2Sip_Index = CauseMapIsdn2Sip_IsdnReleaseCause,
CauseMapIsdn2Sip_SipResponse;
CauseMapIsdn2Sip 0 = 21, 603;

[ \CauseMapIsdn2Sip ]


[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_ProxySetId, ProxyIp_ProxyIpIndex,
ProxyIp_IpAddress, ProxyIp_TransportType;
ProxyIp 0 = "1", 0, "10.254.150.52:5060", 1;
ProxyIp 1 = "1", 1, "10.254.150.52:5060", 0;

[ \ProxyIp ]


[ RedirectNumberMapIp2Tel ]

FORMAT RedirectNumberMapIp2Tel_Index =
RedirectNumberMapIp2Tel_ManipulationName,
RedirectNumberMapIp2Tel_DestinationPrefix,
RedirectNumberMapIp2Tel_RedirectPrefix,
RedirectNumberMapIp2Tel_SourceAddress, RedirectNumberMapIp2Tel_SrcHost,
RedirectNumberMapIp2Tel_DestHost, RedirectNumberMapIp2Tel_NumberType,
RedirectNumberMapIp2Tel_NumberPlan,
RedirectNumberMapIp2Tel_RemoveFromLeft,
RedirectNumberMapIp2Tel_RemoveFromRight,
RedirectNumberMapIp2Tel_LeaveFromRight,
RedirectNumberMapIp2Tel_Prefix2Add, RedirectNumberMapIp2Tel_Suffix2Add,
RedirectNumberMapIp2Tel_IsPresentationRestricted;
RedirectNumberMapIp2Tel 0 = "Rn-In-National", "*", "+41", "*", "*", "*",
0, 0, 3, 0, 255, "0", "", 255;
RedirectNumberMapIp2Tel 1 = "Rn-In-International", "*", "+", "*", "*",
"*", 0, 0, 1, 0, 255, "00", "", 255;

[ \RedirectNumberMapIp2Tel ]


[ RedirectNumberMapTel2Ip ]

FORMAT RedirectNumberMapTel2Ip_Index =
RedirectNumberMapTel2Ip_ManipulationName,
RedirectNumberMapTel2Ip_DestinationPrefix,
RedirectNumberMapTel2Ip_RedirectPrefix,
RedirectNumberMapTel2Ip_NumberType, RedirectNumberMapTel2Ip_NumberPlan,
RedirectNumberMapTel2Ip_RemoveFromLeft,
RedirectNumberMapTel2Ip_RemoveFromRight,
RedirectNumberMapTel2Ip_LeaveFromRight,
RedirectNumberMapTel2Ip_Prefix2Add, RedirectNumberMapTel2Ip_Suffix2Add,
RedirectNumberMapTel2Ip_IsPresentationRestricted,
RedirectNumberMapTel2Ip_SrcTrunkGroupID;
RedirectNumberMapTel2Ip 0 = "Rn-Out-International", "*", "00", 255, 255,
2, 0, 255, "+", "", 255, -1;
```

```
RedirectNumberMapTel2Ip 1 = "Rn-Out-National", "*", "0", 255, 255, 1, 0,
255, "+41", "", 255, -1;
RedirectNumberMapTel2Ip 2 = "Rn-Out-NoZero", "*", "[1-9]xx.#", 255, 255,
0, 0, 255, "+41", "", 255, -1;

[ \RedirectNumberMapTel2Ip ]


[ CodersGroup0 ]

;
;  *** TABLE CodersGroup0 ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts.
;

[ \CodersGroup0 ]


[ MessageManipulations ]

FORMAT MessageManipulations_Index =
MessageManipulations_ManipulationName, MessageManipulations_ManSetID,
MessageManipulations_MessageType, MessageManipulations_Condition,
MessageManipulations_ActionSubject, MessageManipulations_ActionType,
MessageManipulations_ActionValue, MessageManipulations_RowRole;
MessageManipulations 0 = "Options Request-URI", 1, "OPTIONS.request", "",
"header.request-uri.url.host", 2, "param.message.address.dst.address", 0;
MessageManipulations 1 = "Set Contact-User to Anonymous", 1,
"invite.request", "header.privacy regex (id)", "header.contact.url.user",
2, "'anonymous'", 0;
MessageManipulations 2 = "Edit Diversion-Header", 1, "invite.request",
"header.diversion exists AND header.diversion regex (<tel:)(.*)(>.*)",
"header.diversion", 2, "'<sip:'+$2+'@'+header.contact.url.host+$3", 0;
MessageManipulations 3 = "Edit PAI if anonymous", 1, "invite.request",
"header.diversion exists AND header.diversion regex (<sip:)(.*)(@.*) AND
header.privacy=='id'", "header.p-asserted-identity", 2,
"'<sip:'+$2+'@'+header.contact.url.host+'>'", 0;
MessageManipulations 4 = "Remove T.38 from SDP", 1, "invite.request",
"body.sdp regex (.*)(m=image)(.*)", "body.sdp", 2, "$1", 0;

[ \MessageManipulations ]


[ GwRoutingPolicy ]

FORMAT GwRoutingPolicy_Index = GwRoutingPolicy_Name,
GwRoutingPolicy_LCREnable, GwRoutingPolicy_LCRAverageCallLength,
GwRoutingPolicy_LCRDefaultCost, GwRoutingPolicy_LdapServerGroupName;
GwRoutingPolicy 0 = "GwRoutingPolicy", 0, 1, 0, "";

[ \GwRoutingPolicy ]


[ ResourcePriorityNetworkDomains ]

FORMAT ResourcePriorityNetworkDomains_Index =
ResourcePriorityNetworkDomains_Name,
ResourcePriorityNetworkDomains_Ip2TelInterworking;
ResourcePriorityNetworkDomains 1 = "dsn", 1;
```

```
ResourcePriorityNetworkDomains 2 = "dod", 1;
ResourcePriorityNetworkDomains 3 = "drsn", 1;
ResourcePriorityNetworkDomains 5 = "uc", 1;
ResourcePriorityNetworkDomains 7 = "cuc", 1;

[ \ResourcePriorityNetworkDomains ]


[ AudioCoders ]

FORMAT AudioCoders_Index = AudioCoders_AudioCodersGroupId,
AudioCoders_AudioCodersIndex, AudioCoders_Name, AudioCoders_pTime,
AudioCoders_rate, AudioCoders_PayloadType, AudioCoders_Sce,
AudioCoders_CoderSpecific;
AudioCoders 0 = "AudioCodersGroups_0", 0, 1, 2, 90, -1, 0, "";
AudioCoders 1 = "AudioCodersGroups_0", 1, 3, 2, 19, -1, 0, "";
AudioCoders 2 = "AudioCodersGroups_0", 2, 4, 255, 255, -1, 0, "";

[ \AudioCoders ]
```

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