



Mida Platform SBC Configuration Guide

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1. Introduction

1.1 Legal Statements

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Mida Platform

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Mida C³ - Cloud Contact Center

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1.2 Preface

This document is part of the official documentation of Mida Solutions products and details functionalities, user interface, option and working modes in detail. The system allows the user to configure all system functions using a simple and intuitive WEB interface. Please refer to the reference table for a complete list of documents relevant for system configuration.

1.3 Audience

The present document addresses both end users and system administrators of the products.

1.4 Notations



This document highlights, where possible, the main parameters and operations through **bold** or *italics* text and all parts that might be critical during system configuration or use. Critical parts are also marked with Warning symbol reported here on the left.

1.5 References

This manual includes references to the following list of documents:

- [1] Mida_Unified_Portal-Administration_&_User_Manual
- [2] Mida_Appliance-Administration_Manual
- [3] MidaRec Gateway-Administration_Manual
- [4] www.midasolutions.com/browsercompatibility

2.SBC configurations

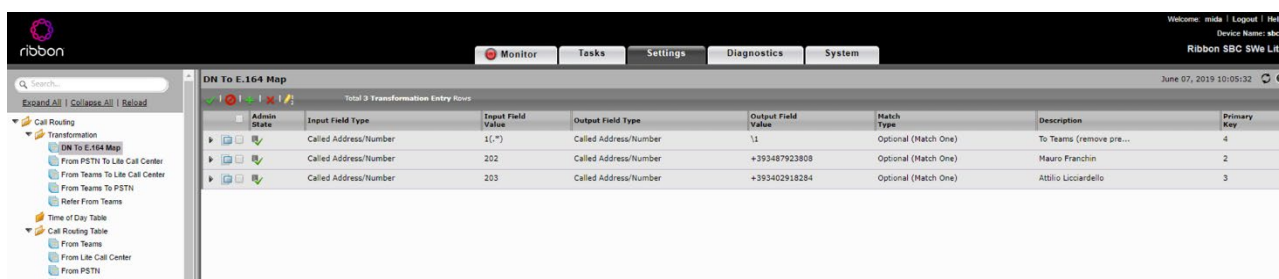


SBC to Mida Platform configurations require to have the SBC Number Transformation Table already set. Do that follow [Ribbon configuration guide](#) or see **Appendix A of this guide for a brief step-by-step guide.**

2.1 Mida C³ - Cloud Contact Center to SBC connection

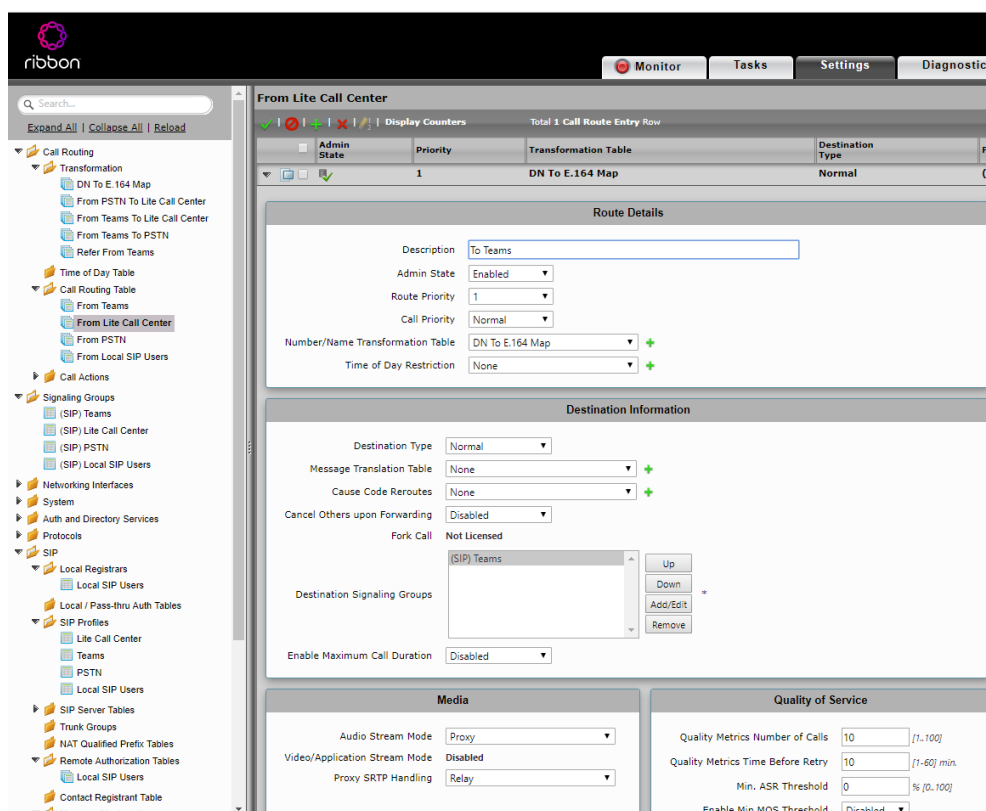
To set up the connection between Mida C³ - Cloud Contact Center and the SBC, follow the next steps and insert values as stated in the screenshots, if no other values are specified.

1. Go to **Call Routing > Transformation** and create a new **Transformation table** (in the example below we called it "DN to E.164 Map").
This transformation will change the call destination with the proper Teams number.



Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key
<input checked="" type="checkbox"/>	Called Address/Number	1(*)	Called Address/Number	1	Optional (Match One)	To Teams (remove pre...	4
<input checked="" type="checkbox"/>	Called Address/Number	202	Called Address/Number	+393487923809	Optional (Match One)	Mauro Franchin	2
<input checked="" type="checkbox"/>	Called Address/Number	203	Called Address/Number	+393402918284	Optional (Match One)	Attilio Liccardello	3

2. Go to **Call Routing > Call Routing Table** and create a new call route entry (in the example below, "From Lite Call Center")



From Lite Call Center

Total 1 Call Route Entry Row

Admin State	Priority	Transformation Table	Destination Type
<input checked="" type="checkbox"/>	1	DN To E.164 Map	Normal

Route Details

Description: To Teams

Admin State: Enabled

Route Priority: 1

Call Priority: Normal

Number/Name Transformation Table: DN To E.164 Map

Time of Day Restriction: None

Destination Information

Destination Type: Normal

Message Translation Table: None

Cause Code Reroutes: None

Cancel Others upon Forwarding: Disabled

Fork Call: Not Licensed

Destination Signaling Groups: (SIP) Teams

Enable Maximum Call Duration: Disabled

Media

Audio Stream Mode: Proxy

Video/Application Stream Mode: Disabled

Proxy SRTP Handling: Relay

Quality of Service

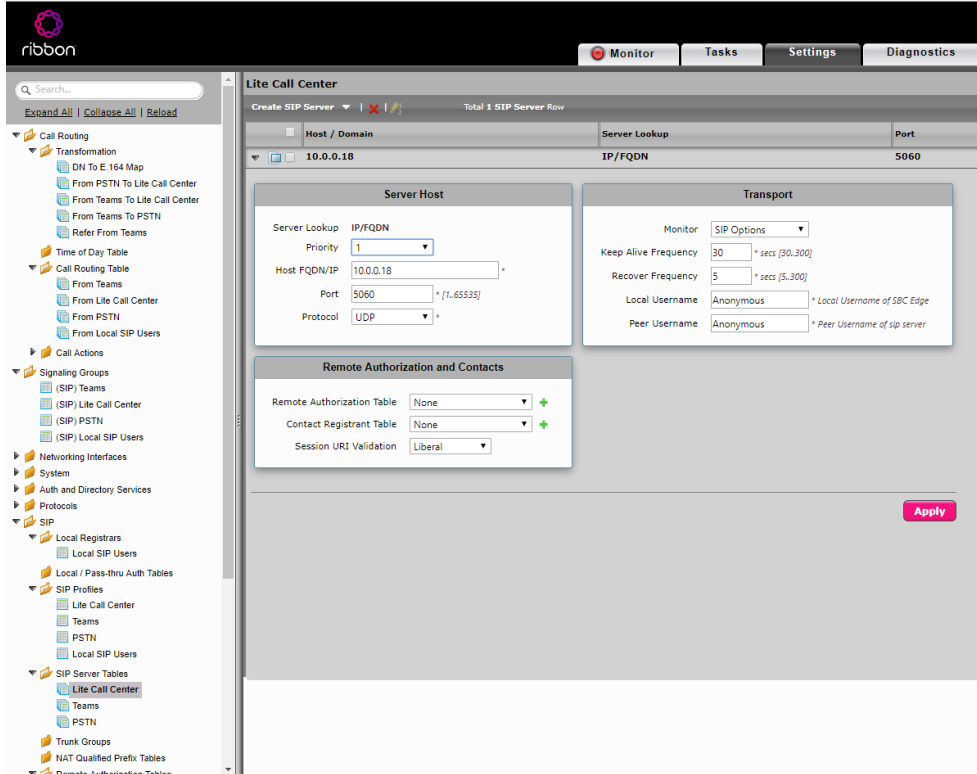
Quality Metrics Number of Calls: 10 [1..100]

Quality Metrics Time Before Retry: 10 [1..60] min.

Min. ASR Threshold: 0 [% [0..100]]

Enable Min MOS Threshold: Disabled

- Go to **SIP > SIP Server Tables** and create a new SIP Server ("Lite Call Center")



The screenshot displays the Mida Solutions SBC Configuration interface. The left sidebar shows a tree view with the following structure:

- Call Routing
 - Transformation
 - DN To E.164 Map
 - From PSTN To Lite Call Center
 - From Teams To Lite Call Center
 - From Teams To PSTN
 - Refer From Teams
 - Time of Day Table
 - Call Routing Table
 - From Teams
 - From Lite Call Center
 - From PSTN
 - From Local SIP Users
 - Call Actions
- Signaling Groups
 - (SIP) Teams
 - (SIP) Lite Call Center
 - (SIP) PSTN
 - (SIP) Local SIP Users
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols
- SIP
 - Local Registrars
 - Local SIP Users
 - Local / Pass-thru Auth Tables
 - SIP Profiles
 - Lite Call Center
 - Teams
 - PSTN
 - Local SIP Users
 - SIP Server Tables
 - Lite Call Center
 - Teams
 - PSTN
 - Trunk Groups
 - NAT Qualified Prefix Tables
 - Demote Authorization Tables

The main panel shows the 'Lite Call Center' configuration page. The 'Create SIP Server' tab is active, and the 'Total 1 SIP Server Row' is displayed. The configuration is as follows:

Host / Domain	Server Lookup	Port
10.0.0.18	IP/FQDN	5060

The 'Server Host' section contains the following fields:

- Server Lookup: IP/FQDN
- Priority: 1
- Host FQDN/IP: 10.0.0.18
- Port: 5060
- Protocol: UDP

The 'Transport' section contains the following fields:

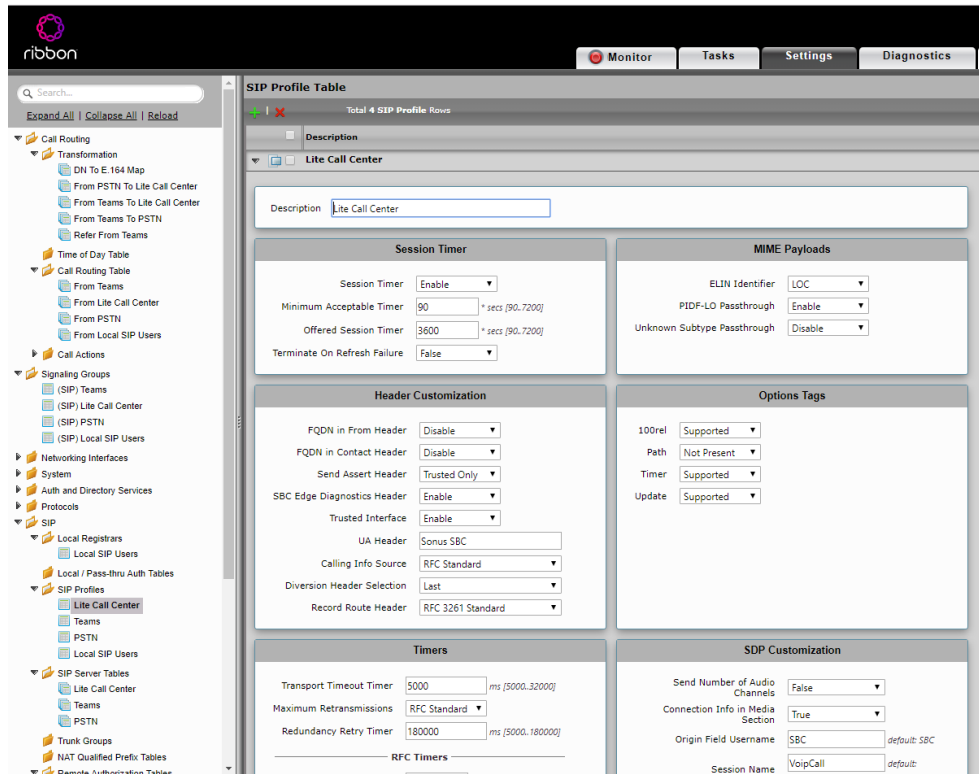
- Monitor: SIP Options
- Keep Alive Frequency: 30
- Recover Frequency: 5
- Local Username: Anonymous
- Peer Username: Anonymous

The 'Remote Authorization and Contacts' section contains the following fields:

- Remote Authorization Table: None
- Contact Registrant Table: None
- Session URI Validation: Liberal

An 'Apply' button is located at the bottom right of the configuration panel.

- Go to **SIP > SIP Profiles** and create a new entry ("Lite Call Center")



The screenshot displays the Mida Solutions SBC Configuration interface. The left sidebar shows a tree view of configuration categories, with 'SIP Profiles' expanded and 'Lite Call Center' selected. The main panel shows the 'SIP Profile Table' configuration for 'Lite Call Center'.

SIP Profile Table
Total 4 SIP Profile Rows

Description
Lite Call Center

Session Timer

- Session Timer: Enable
- Minimum Acceptable Timer: 90 * secs [90..7200]
- Offered Session Timer: 3600 * secs [90..7200]
- Terminate On Refresh Failure: False

MIME Payloads

- ELIN Identifier: LOC
- PIDF-LO Passthrough: Enable
- Unknown Subtype Passthrough: Disable

Header Customization

- FQDN in From Header: Disable
- FQDN in Contact Header: Disable
- Send Assert Header: Trusted Only
- SBC Edge Diagnostics Header: Enable
- Trusted Interface: Enable
- UA Header: Sonus SBC
- Calling Info Source: RFC Standard
- Diversion Header Selection: Last
- Record Route Header: RFC 3261 Standard

Options Tags

- 100rel: Supported
- Path: Not Present
- Timer: Supported
- Update: Supported

Timers

- Transport Timeout Timer: 5000 ms [5000..32000]
- Maximum Retransmissions: RFC Standard
- Redundancy Retry Timer: 180000 ms [5000..180000]

SDP Customization

- Send Number of Audio Channels: False
- Connection Info in Media Section: True
- Origin Field Username: SBC default: SBC
- Session Name: VoipCall default:

- Go to **Signaling Groups** and create a new entry ("Signaling Group")

ribbon Monitor Tasks Settings Diagnostics System

Search... Expand All Collapse All Reload

- Call Routing
 - Transformation
 - DN To E.164 Map
 - From PSTN To Lite Call Center
 - From Teams To Lite Call Center
 - From Teams To PSTN
 - Refer From Teams
 - Time of Day Table
 - Call Routing Table
 - From Teams
 - From Lite Call Center
 - From PSTN
 - From Local SIP Users
- Call Actions
- Signaling Groups
 - (SIP) Teams
 - (SIP) Lite Call Center
 - (SIP) PSTN
 - (SIP) Local SIP Users
- Networking Interfaces
- System
- Auth and Directory Services
- Protocols
- SIP
 - Local Registrars
 - Local SIP Users
 - Local / Pass-thru Auth Tables
 - SIP Profiles
 - Lite Call Center
 - Teams
 - PSTN
 - Local SIP Users
 - SIP Server Tables
 - Lite Call Center
 - Teams
 - PSTN
 - Trunk Groups
 - NAT Qualified Prefix Tables
 - Remote Authentication Tables

Signaling Group Table

Total 4 Signaling Group Rows

Type	Description	Admin State	Service Status	Display
SIP	Teams		Up	Counters Channels Sessions
SIP	Lite Call Center		Up	Counters Channels Sessions

SIP Channels and Routing

Action Set Table: +

Call Routing Table: +

No. of Channels: * [1..960]

SIP Profile: +

SIP Mode:

Agent Type:

SIP Server Table: +

Load Balancing:

Channel Hunting:

Notify Lync CAC Profile:

Challenge Request:

Media Information

Supported Audio Modes: Add/Edit Remove

Supported Video/Application Modes:

Proxy Local SRTP Crypto Profile ID: +

Allow Refresh SIP:

RTCP Multiplexing:

Mapping Tables

SIP IP Details

Signaling/Media Source IP:

Signaling DSCP: * [0..63]

NAT Traversal

ICE Support:

Static NAT - Outbound

Outbound NAT Traversal:

Static NAT - Inbound

Detection:

Listen Ports

Total 1 SIP Listen Port Row

Port	Protocol	TLS Profile ID
5060	UDP	N/A

Federated IP/FQDN

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
10.0.0.18	255.255.255.255

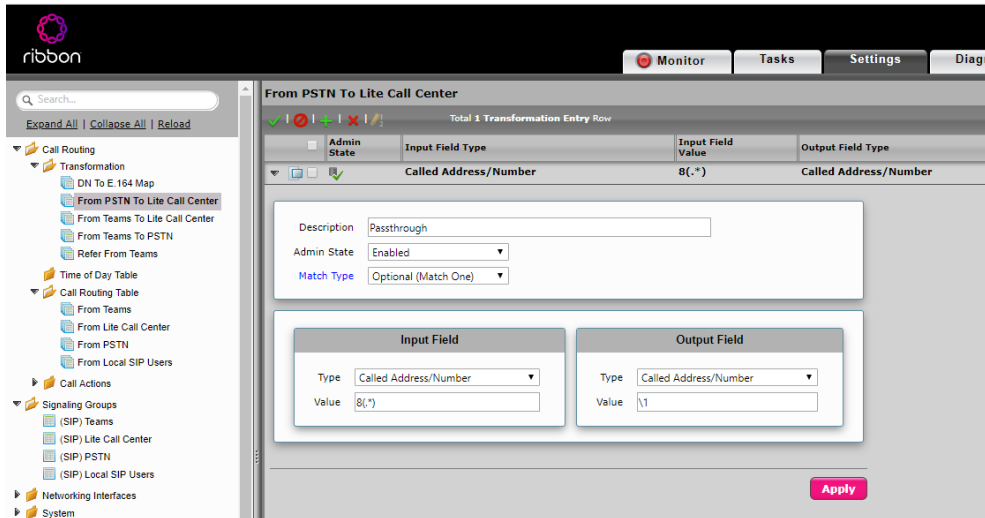
Message Manipulation:

Apply

2.2 PSTN to SBC connection

To set up the connection between PSTN Mida C³ – Cloud Contact Center and the Mida C³ – Cloud Contact Center, follow the next steps and insert values as stated in the screenshots, if no other values are specified.

1. Go to **Call Routing > Transformation** and create a new **Transformation table** (in the example below we called it "From PSTN to Mida C³ – Cloud Contact Center").
This transformation will lead desired calls from the PSTN to the Mida C³ – Cloud Contact Center
In **Value**, it is possible to insert the desired prefix. All calls incoming to the SBC with that prefix will be redirected to Mida C³ – Cloud Contact Center.

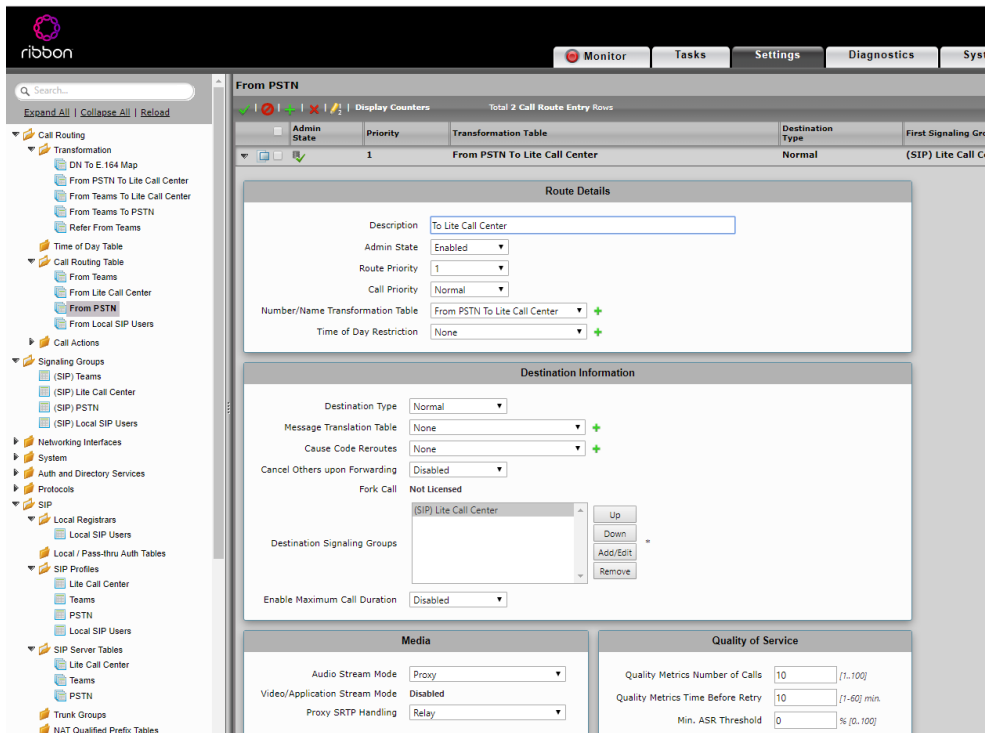


The screenshot shows the 'Transformation' configuration page in the Ribbon SBC interface. The left sidebar shows the navigation tree with 'Call Routing > Transformation' selected. The main panel is titled 'From PSTN To Lite Call Center' and shows a table with one entry. The entry details are as follows:

Admin State	Input Field Type	Input Field Value	Output Field Type
Enabled	Called Address/Number	8(*)	Called Address/Number

Below the table, the 'Match Type' is set to 'Optional (Match One)'. The 'Input Field' section shows 'Type' as 'Called Address/Number' and 'Value' as '8(*)'. The 'Output Field' section shows 'Type' as 'Called Address/Number' and 'Value' as '1'. An 'Apply' button is at the bottom right.

2. Go to **Call Routing > Call Routing Table** and create a new call route entry (in the example below, "From PSTN")

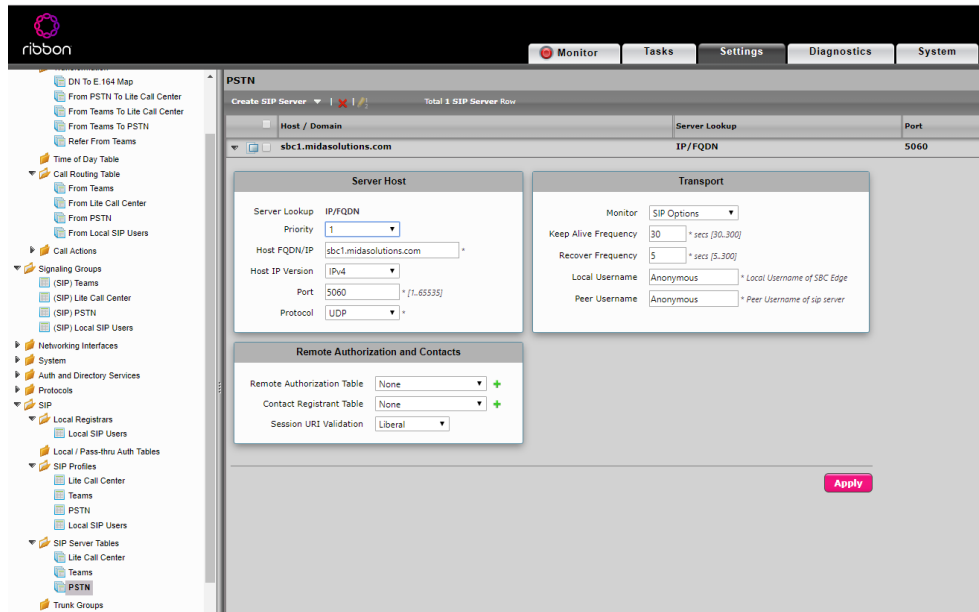


The screenshot shows the 'Call Routing Table' configuration page in the Ribbon SBC interface. The left sidebar shows the navigation tree with 'Call Routing > Call Routing Table' selected. The main panel is titled 'From PSTN' and shows a table with one entry. The entry details are as follows:

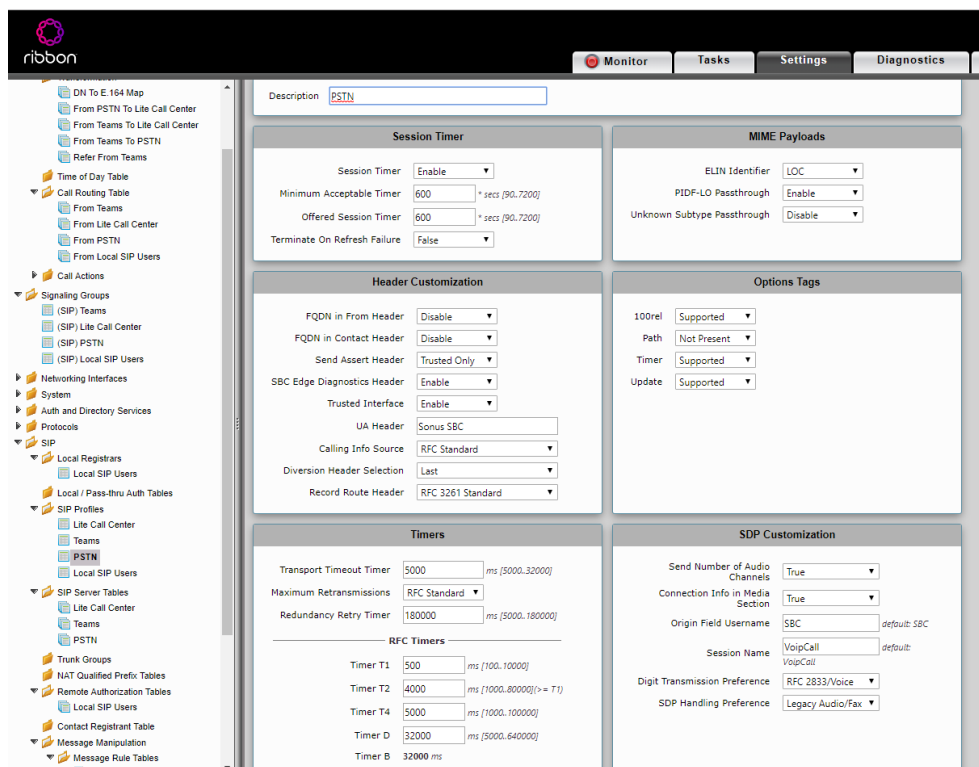
Admin State	Priority	Transformation Table	Destination Type	First Signaling Group
Enabled	1	From PSTN To Lite Call Center	Normal	(SIP) Lite Call Center

Below the table, the 'Route Details' section shows 'Description' as 'To Lite Call Center', 'Admin State' as 'Enabled', 'Route Priority' as '1', 'Call Priority' as 'Normal', 'Number/Name Transformation Table' as 'From PSTN To Lite Call Center', and 'Time of Day Restriction' as 'None'. The 'Destination Information' section shows 'Destination Type' as 'Normal', 'Message Translation Table' as 'None', 'Cause Code Reroutes' as 'None', 'Cancel Others upon Forwarding' as 'Disabled', 'Fork Call' as 'Not Licensed', 'Destination Signaling Groups' as '(SIP) Lite Call Center', and 'Enable Maximum Call Duration' as 'Disabled'. The 'Media' section shows 'Audio Stream Mode' as 'Proxy', 'Video/Application Stream Mode' as 'Disabled', and 'Proxy SRTP Handling' as 'Relay'. The 'Quality of Service' section shows 'Quality Metrics Number of Calls' as '10' (range [1..100]), 'Quality Metrics Time Before Retry' as '10' (range [1..60] min), and 'Min. ASR Threshold' as '0' (range [% (0..100)]).

3. Go to **SIP > SIP Server Tables** and create a new SIP Server ("PSTN").



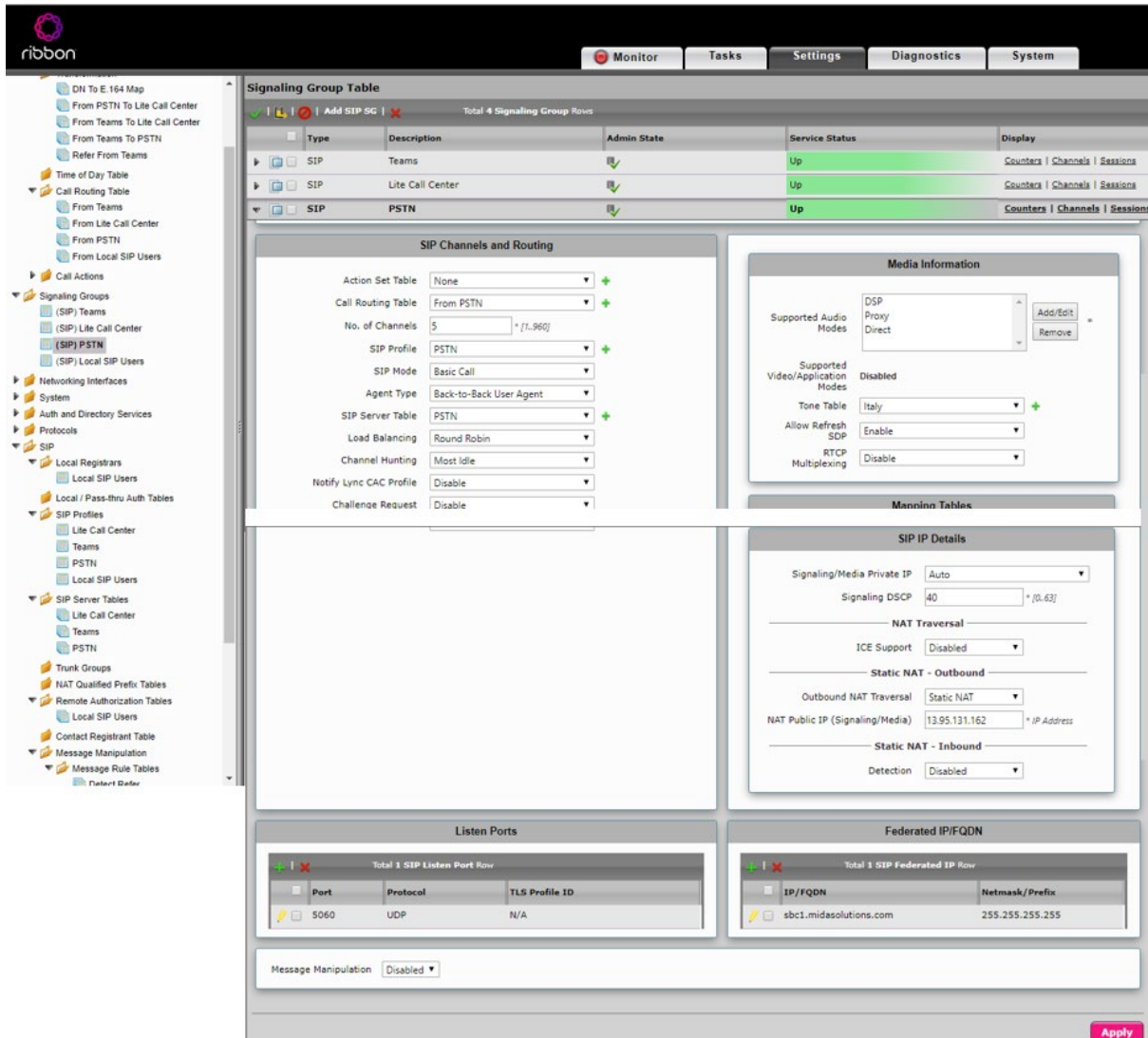
- Go to **SIP > SIP Profiles** and create a new entry ("PSTN"). Leave everything as default.



5. Go to **Signaling Groups** and create a new entry ("PSTN").



NAT configuration may not be necessary.



The screenshot shows the Ribbon SBC configuration interface. The left sidebar contains a tree view of configuration categories including Signaling Groups, SIP, and NAT. The main area displays the 'Signaling Group Table' with a table listing three groups: Teams, Lite Call Center, and PSTN. The PSTN group is selected, and its configuration details are shown in the right-hand panels.

Signaling Group Table

Type	Description	Admin State	Service Status	Display
SIP	Teams	Up	Up	Counters Channels Sessions
SIP	Lite Call Center	Up	Up	Counters Channels Sessions
SIP	PSTN	Up	Up	Counters Channels Sessions

SIP Channels and Routing

- Action Set Table: None
- Call Routing Table: From PSTN
- No. of Channels: 5
- SIP Profile: PSTN
- SIP Mode: Basic Call
- Agent Type: Back-to-Back User Agent
- SIP Server Table: PSTN
- Load Balancing: Round Robin
- Channel Hunting: Most Idle
- Notify Lync CAC Profile: Disable
- Challenge Request: Disable

Media Information

- Supported Audio Modes: DSP Proxy, Direct
- Supported Video/Application Modes: Disabled
- Tone Table: Italy
- Allow Refresh SDP: Enable
- RTCP Multiplexing: Disable

SIP IP Details

- Signaling/Media Private IP: Auto
- Signaling DSCP: 40
- NAT Traversal: Disabled
- Static NAT - Outbound: Static NAT
- NAT Public IP (Signaling/Media): 13.95.131.162
- Static NAT - Inbound: Disabled

Listen Ports

Port	Protocol	TLS Profile ID
5060	UDP	N/A

Federated IP/FQDN

IP/FQDN	Netmask/Prefix
sbcl.midasolutions.com	255.255.255.255

Message Manipulation: Disabled

Apply

2.3 SBC to Teams – Teams Direct Routing

To configure Teams Direct Routing, follow the SBC vendor guide:

- Ribbon: [Best Practice - Configuring SBC Edge for Microsoft Teams Direct Routing](#)