



SIP Trunk Setup Guide for Metaswitch

### Overview

The following is a guide for configuring Metaswitch SIP Trunks to work with an iPECS LIK with phase 5.5 software. Metaswitch delivers SIP Trunk service as Provisioned (Configured SIP binding) SIP Trunks with Authentication. These trunks have a fixed SIP User to IP address binding; all calls to the SIP User ID are sent to a fixed IP address.

This guide only covers SIP Trunk configuration, other characteristics of the system such as Voice Mail, Station configurations, etc. may be required to meet the customer needs.

Mandatory settings are detailed and comments are provided for several common option settings. Should you need additional assistance contact your distributor.

# Metaswitch SIP Trunk Service

Before you begin to configure the SIP Trunks on the iPECS LIK, you must receive the following information from the Metaswitch service provider.

- SIP ID
- Domain or IPv4 address of the SIP Trunk host
- Authentication User ID:
- Authentication Password:
- Assigned number(s): 1 (NPA) Nxx-xxxx for DID a range should be provided
- Primary DNS

NOTE - You will need to provide a fixed IP for binding.

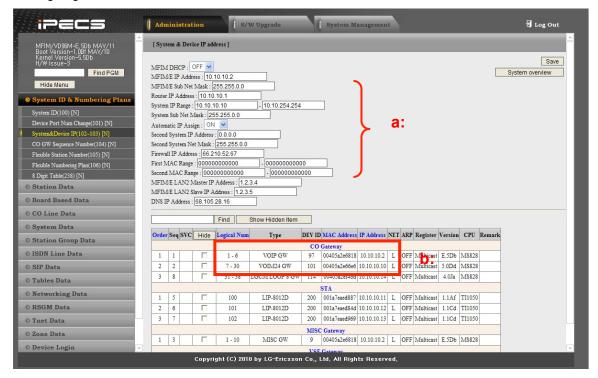
# **Configuration Guide**

All configuration options for SIP Trunks are done through the iPECS LIK Web Admin. Once the Web Admin is opened, use the steps below as a guide to your configuration. Remember this is a guide and the data displayed for the various attributes must be entered as provided from Metaswitch and not the values shown in this guide.



# **Configuration Steps**

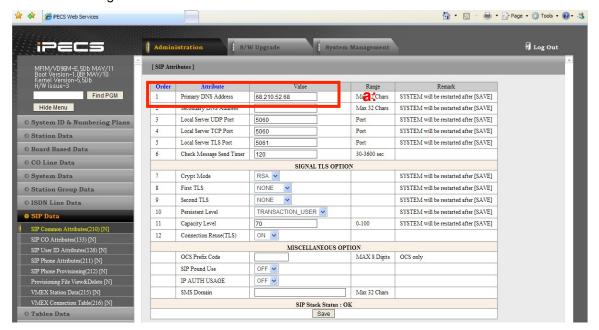
- 1. System & Device IP Web page,
  - a. Enter and Save the System and Device IP addressing parameters.
  - b. Determine the logical number of the VoIP channels to use for the SIP Trunks, you will need this when assigning SIP CO Attributes.





# 2. SIP Common Attributes Web page

- a. Enter the IPv4 address for the Primary DNS Address
- b. Enter other optional attributes including the Secondary DNS.
- c. Save the configuration

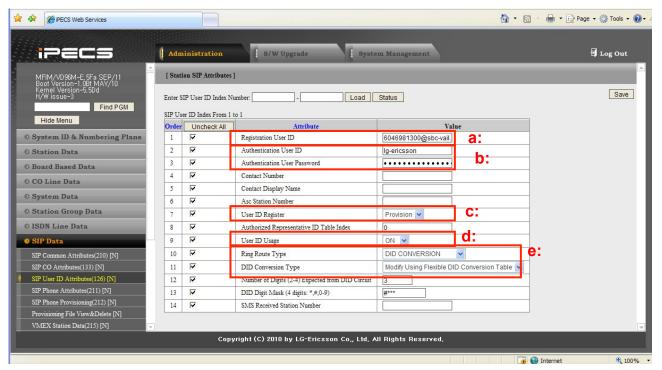




## 3. SIP User ID Attributes Web page

Enter an index range for the SIP User Attributes Table (1 – 300) and click Load

- a. Enter the Registration User ID, which is the User ID from Metaswitch (generally the main DN), with the LIK Firewall address as ID@IP address.,
- b. Enter the Authentication ID and password provided from Metaswitch,
- c. Set "User ID Register" to 'Provision'; Metaswitch SIP Trunks do not require registration.
- d. Set User ID Usage to 'On', and
- e. Select the Ring Route Type to define the routing of incoming calls
  - Extension SIP User Attributes Table Index SIP User Attribute Table index must be set in item 56 of at least one station.
  - Ring Assignment CO/IP Line Type must be Normal (PGM 140) and Ring assignment for CO/IP Lines must be set in PGM 144
  - DID Conversion CO/IP Line Type must be DID (PGM 140), set DID Service Attributes in PGM 145 and, if needed set destinations in Flex table (PGM 231)
- f. Enter other optional attributes
- g. Save the configuration



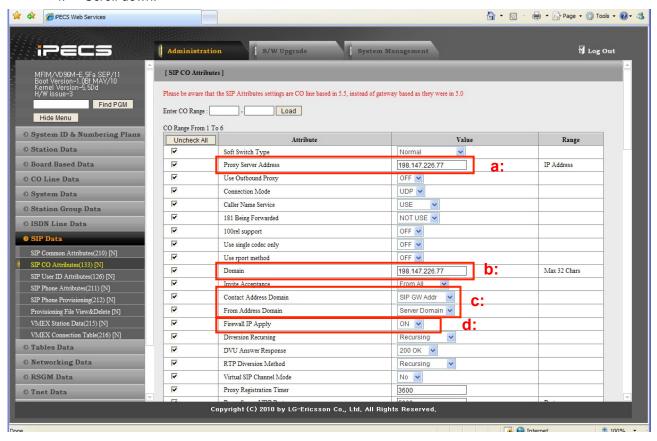


### 4. SIP CO Attribute settings are discussed over the next several pages

## 4-1. SIP CO Attributes web page

Enter the SIP CO Line number range, see System & Device IP, and click Load

- a. Enter the Proxy Server IPv4 Address and set "Use Outbound Proxy" to 'On'. If a Proxy is not used, set "Use Outbound Proxy" to OFF.
- b. Enter the SIP Trunk host domain or IP address,
- c. Set the Contact Address Domain and From Address Domain to SIP G/W Address.
- d. If the system is behind a firewall, turn on "Firewall IP Apply"
- e. Enter other optional SIP CO attributes on the displayed page,
- f. Scroll down.



### Note

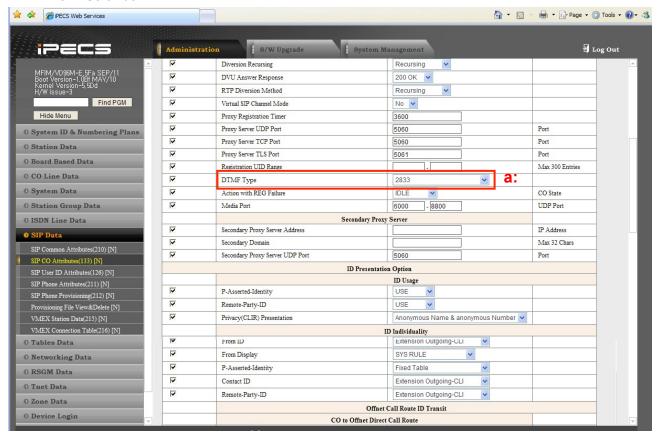
The Contact ID defines a direct route to the user that other elements in the SIP signaling path may use for future request. It also provides bindings information during registration.



### 4-2. SIP CO Attributes web page

Scroll down to display "DTMF Type"

- a. Set "DTMF Type" to 2833
- b. Enter other optional SIP CO attributes on the displayed page,
- c. Scroll dowr



### Notes:

Action with REG Failure (previously FailOver Usage):

If set to Wait Idle and SIP Registration fails or the ID is not registered, the SIP CO is not available for use and returns reorder tone. Otherwise, the SIP Call Setup FailOver Options below define the call routing.

#### Secondary Proxy Server

These Admin fields are not presently supported.

### 4-3. SIP CO Attributes web page

Scroll down to display "ID Individuality" section. This section configures SIP identities the system must send to the host for a normal call placed on the SIP trunk. These must be set properly so the system can send well-constructed SIP messages.

a. Set ID Individuality Ids as follows:

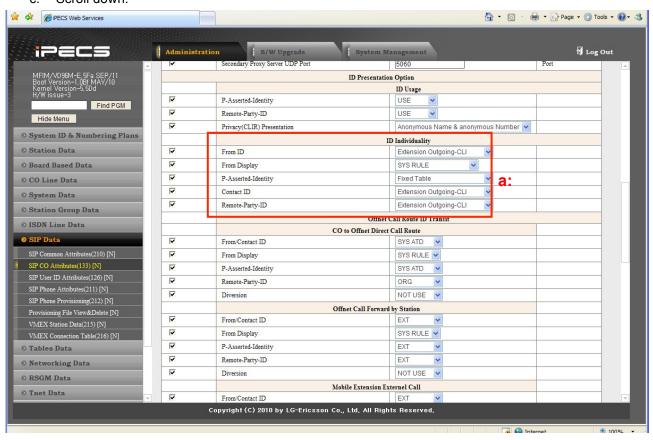
From ID: to provide unique IDs for outgoing calls, set to Extension Outgoing-CLI. The stations DID number should be entered as the Station CLI in the Station ISDN Attributes PGM 114 for use as the From ID for calls from the station. For the same id for all calls, set the From ID to Fixed ID and set the Fixed Table Assignment below to the proper index in the SIP User Attributes Table.

From Display: to provide Caller Name display on outgoing calls, set to Sys Rule. A Name should be entered for the Station Name Display and, in Station ISDN Attributes PGM 114, set COLP Display, CLIP Display and CLI Name Display to 'On'.

*P-Asserted ID*: the proven trusted identity of the sender also used in LIK to reference the SIP User Attribute Table index with the proper Authentication credentials. Set *P-Asserted ID* to Fixed ID and set the Fixed Table Assignment below to the proper index in the SIP User Attributes Table. This can also be achieved by setting *P-Asserted ID* to Extension SIP-User-ID-Table and entering the proper index in item 56 in Station Attributes.

Contact ID: to provide an accurate direct contact, set the Contact ID to the Extension Outgoing CLI, the domain is the Contact Address Domain and should be VoIP Gateway Address. See section 4.1. Remote Party ID: is a rarely used privacy header that can be set to the Extension Outgoing CLI option.

- Enter other optional SIP CO attributes on the displayed page,
- c. Scroll down.

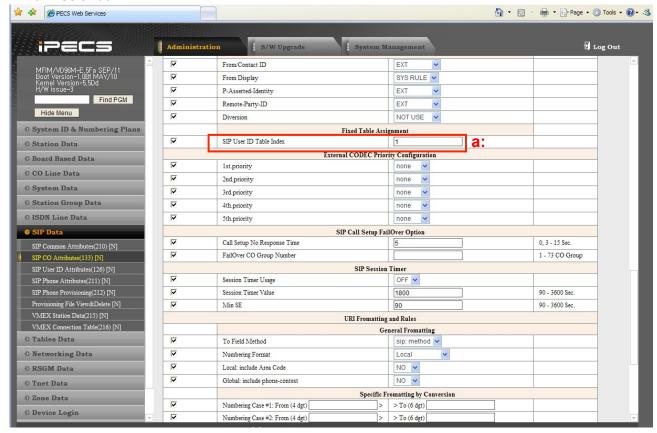




### 4-4. SIP CO Attributes Web page

Scroll down to display Fixed Table Assignment

- a. Enter the SIP User ID Attribute Table index (PGM 126) for the Fixed Table Assignment,
- b. Enter other optional SIP CO attributes on the displayed page,
- c. Scroll down



#### Notes:

#### Codec:

This section defines the priority for the various available Codecs selected from the drop-down menu.

### SIP Call Setup FailOver Options

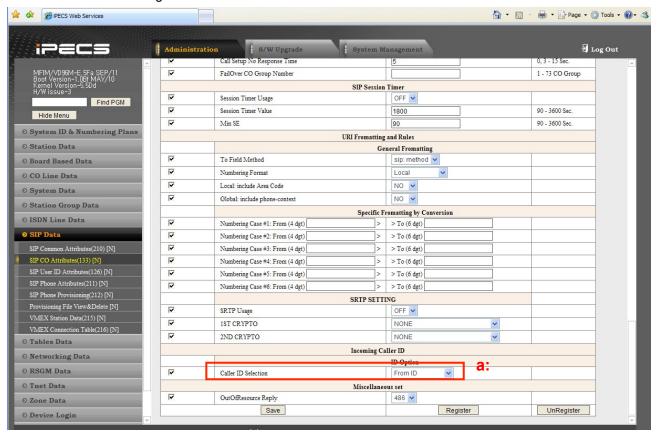
This section establishes the Fail-over routing when enabled (see "Action with REG Failure") and no response is received for a SIP Invite (the call setup fails).



### 4-5. SIP CO Attributes web page

Scroll down to the bottom of page

- a. Set the Caller ID selection to "From ID", generally the received From ID contains the "Display Name" for an incoming call. To display the Name for an incoming call, in Station ISDN Attributes PGM 114, set COLP Display, CLIP Display and CLI Name display to 'On'.
- b. Save the configuration



NOTE - There is no need to Register; Metaswitch SIP trunks employ a fixed binding.

# 5. Other System Admin that may need attention based on the SIP Trunk configuration:

### A. Station Attributes

Item 56 - SIP User ID Table Index, if "Extension SIP User ID Table is select for any ID, a valid index must be set.

### B. Station ISDN Attributes:

Item 1 - COLP Display, set to On

Item 2 - CLIP Display, set to On

Item 6 - Station CLI, set to station DID number

Item 8 - CLI Name Display, set to On

# C. Station Name Display

## D. CO/IP Attributes:

CO Type, set to DID CO/IP Group

# E. CO Ring Assignments

### F. DID Service Attributes

### G. ISDN CO Line Attributes

COLP Table Index, set to Station CLI CLIP Table Index, set to Station CLI Enblock Sending, set to On Station CLI Type, set to Station CLI 1

H. DID Flex Conversion Table





**PARTNER CERTIFIED SOLUTION** 

I hereby certify that

LG Ericsson LIK version LIK5.5Fc SIP protocol with MetaSphere CFS version 7.3

has been demonstrated to have successfully completed the Metaswitch

Networks Interoperability Test



12/5/2011

Chief Technology Officer

Certification subject to any detailed restrictions listed at www.metaswitch.com.