SIP Settings

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Restcomm SMSC comes with default JSLEE SIP RA. You can modify the SIP settings using the CLI or GUI.

Using CLI

You can modify the SIP settings by issuing the command `smsc sip modify` with appropriate parameters as described below. You can view the settings by issuing the command `smsc sip show` which will display all the values.

```
Name
    smsc sip modify

SYNOPSIS
    smsc sip modify name cluster-name <clusterName> host <ip> port <port>
    routing-ton <routing address ton> routing-npi <routing address npi>
    routing-range <routing address range> counters-enabled <true | false>
    charging-enabled <true | false> networkid <networkId>

DESCRIPTION
    This command is used to modify SIP settings.

PARAMETERS

    Standard Parameters

    Name              - The name of the SIP Stack that is being modified. Since
                  the Gateway does not allow creating more than one SIP
                  Stack presently, the name is hardcored to "SIP".

    Cluster-name          - The name of the Cluster to which this SIP Stack belongs
                  to. This parameter is not used presently and is meant
                  for future use when Cluster of SIP is enabled.

    Host              - IP address of the remote node to which all SIP messages
                  must be forwarded.

    Port              - Port of the remote node to which all SIP messages
                  must be forwarded.

    routing-ton       - The DefaultSmsRoutingRule will try to match the
                  'dest_addr_ton' of outgoing SMS with the value
                  configured here. If this configured value is null(-1)
                  or not null and matches, the SMSC will compare the
                  'dest_addr_npi' and 'destination_addr' as explained
                  below. If it doesn't match, the SMSC will select the
                  next SIP in the list for matching routing rule.

    routing-npi       - The DefaultSmsRoutingRule will try to match the
                  'dest_addr_npi' of outgoing SMS with the value
```

```
                  configured here. If this configured value is null(-1)
                  or not null and matches, the SMSC will compare the
                  'destination_addr' as below. If it doesn't match, the
                  SMSC will select the next SIP in the list for matching
                  routing rule.

    routing-range       - The DefaultSmsRoutingRule will try to match the
                  'destination_addr' of outgoing SMS with the value
                  configured here. This is a regular java expression and
                  default value is null. If it matches, the SMSC will
                  send the SMS out over this SIP connection. If it
                  doesn't match, the SMSC will select the next ESME in
                  the list for matching routing rule.

    counters-enabled    - Flag to enable or disable counters. Not used presently.

    charging-enabled    - Flag to enable or disable charging for every SMS
                  arriving from SIP.

    networkId     - means to which virtual subnetwork belongs the SIP
                  connector (this is for multi-tenancy support). Default value
                  is 0. If you do not use multi-tenancy support - set this value
                  to 0.
```

Using GUI

Procedure: Managing the SIP Connection using the GUI

1. In the GUI Management Console for SMSC Gateway, click on 'SIPs' in the left panel.
2. The main panel will display the existing SIP settings. You can view and modify the SIP settings in this panel.