GSM Gateway

User Manual

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About this manual

This manual describes the Allo product application and explains how to work and use it major features. It serves as a means to describe the user interface and how to use it to accomplish common tasks. This manual also describes the underlying assumptions and users make the underlying data model.

Document Conventions

In this manual, certain words are represented in different fonts, typefaces, sizes, and weights. This highlighting is systematic; different words are represented in the same style to indicate their inclusion in a specific category. Additionally, this document has different strategies to draw User attention to certain pieces of information. In order of how critical the information is to your system, these items are marked as a note, tip, important, caution, or warning.

Icon	Purpose
	**Note**
	**Tip/Best Practice**
	**Important**
	**Caution**
	**Warning**

- **Bold** indicates the name of the menu items, options, dialog boxes, windows and functions.
- The color blue with underline is used to indicate cross-references and hyperlinks.
- Numbered Paragraphs - Numbered paragraphs are used to indicate tasks that need to be carried out. Text in paragraphs without numbering represents ordinary information.
- The Courier font indicates a command sequence, file type, URL, Folder/File name e.g. www.allo.com

Support Information

Every effort has been made to ensure the accuracy of the document. If you have comments, questions, or ideas regarding the document contact online support: http://support.allo.com

Table of Contents

1. Product Introduction

1.1 Overview

This 4 Port GSM Gateway is a standalone and fan less, easy to install and sturdy construction. This 4 module GSM gateway will bridge IP calls to GSM and vice versa. Use the web interface for easy and quick configuration.

It is a new type of VoIP gateway that allows call terminations from a VoIP network to a GSM network and vice versa. Call connections between IP networks and GSM networks are now bridged seamlessly to extend the voice communication coverage significantly. As the traditional PSTN lines are starting to disappear in developed countries and are not going to be built extensively in under-developed countries, GSM phones are getting more and more popular all over the world with lower service charges. The emergence of Allo bridges is gap between the traditional telephone networks and VoIP networks. As a result, local and worldwide voice communications are more convenience, lower cost, and broader coverage. Mainly includes the following kind of model: **aGG04** - A typical network diagram as below.

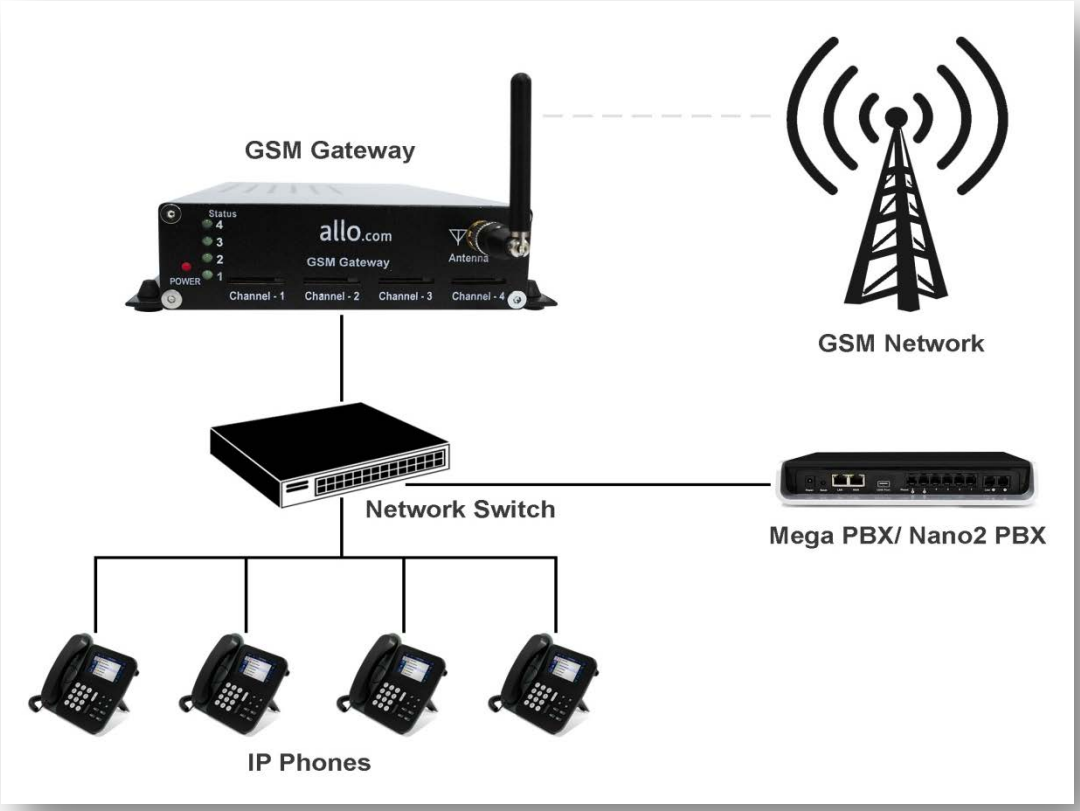


Figure 1: GSM Gateway Overview

1.2 Equipment Structure

1.2.1 Rear View

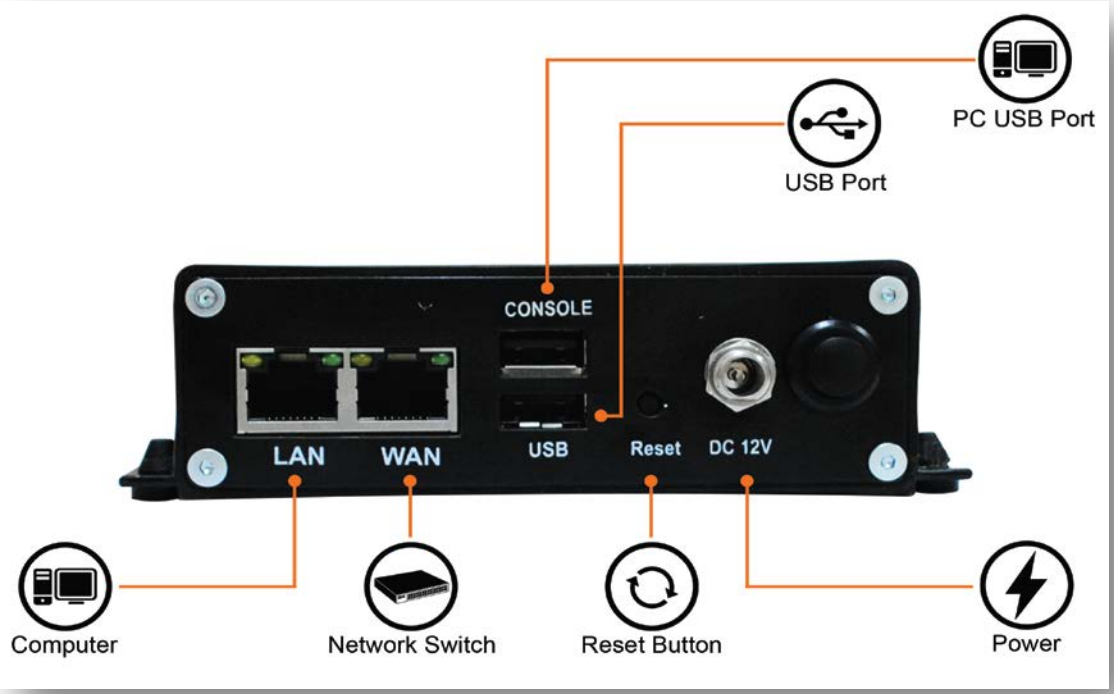


Figure 2: Rear View

1.2.2 Front View

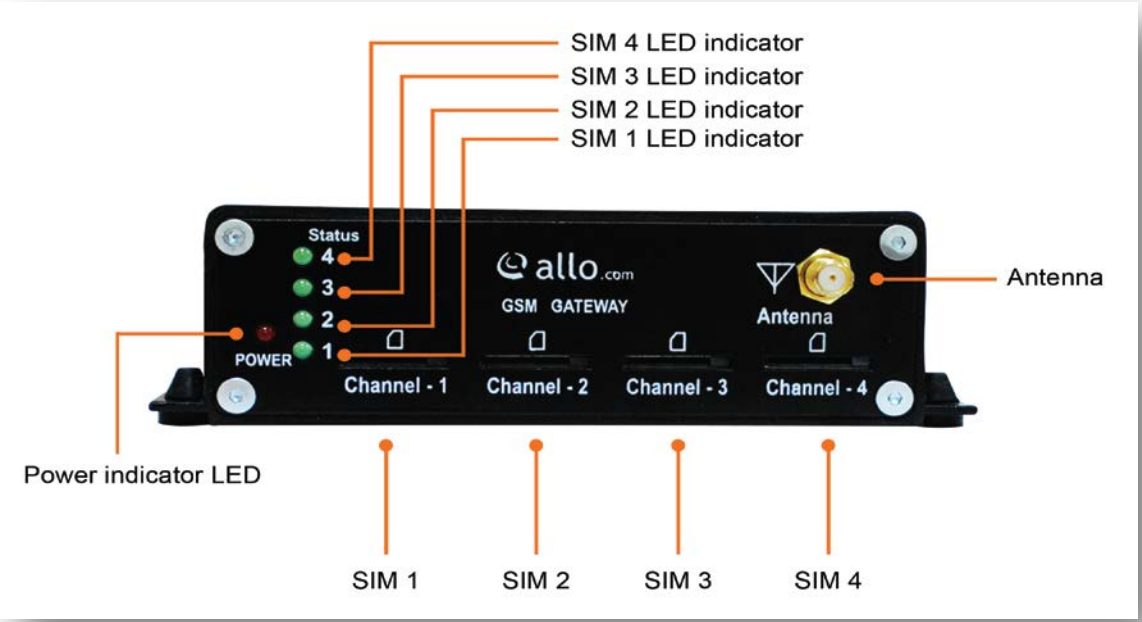


Figure 3: Front View

1.3 Functions and Features

- 4 ports for communication over a SIP-to-cellular connection
- SMS Sending and Receiving Support USSD service
- Send Bulk SMS
- SMS to E-mail
- SIM Balance Alarm
- Caller ID Black List
- Call Duration Limitation for particular SIM
- Call Back
- Call Forwarding (unconditional, no reply, busy, not reachable)
- Open API interface (AMI)
- Gateway control via SMS.(Reboot, Reload, Fetching IP)
- Recharge of SIM balance via GUI
- Flexible call routing
- Configuration and management using a Web browser

1.4 General Hardware Specifications

1.4.1 GSM

- 4 GSM Modules with single SIM card per GSM module.
- Quad-Band GSM 850/900/1800/1900 MHz
- External Antenna
- Echo Cancellation support inbuilt in GSM Module

1.4.2 SIP

- SIP (RFC3261) compliance
- NAT support
- Voice codecs: G729, G.711 A/U law

1.4.3 Network

- Two 10/100 Mbps Ethernet interfaces
- Protocols: DHCP, TCP/UDP, Telnet, HTTP, TFTP

1.5 Warranty

Hardware Warranty: 1 year

If the GSM Gateway was purchased from a Distributor/reseller, please contact the company where the device was purchased for replacement, repair or refund. If the device was purchased directly from Allo.com, contact our Technical Support Team for a RMA (Return Materials Authorization) number before the product is returned. Allo.com reserves the right to remedy warranty policy without prior notification.

Use the power adapter provided with the ALLO PRI Gateway. Do not use a different power adapter as this may damage the device. This type of damage is not covered under warranty.

2. Getting Started with GSM gateway

2.1 Hardware Installation

1. Unpack the items from the box

2. Plug one end of the RJ45 Ethernet cable to your PC & other end into the Management Port (LAN) of the GSM Gateway.

3. Plug one end of the RJ45 Ethernet cable into your Network Switch & other end into the WAN port of the GSM Gateway.

Use Straight-through Ethernet cable to connect between the unit & Router/Switch/PC

4. Insert the GSM SIM in to the GSM Channels.

5. Plug the power cable to the GSM Gateway; wait until the Gateway boot up completely.

Configure your GSM Gateway according to the instructions below.

2.2 Accessing the Web GUI

GSM Gateway WEB GUI can be accessed either through WAN or Management Interface (LAN)

1) Make the setup as described in Hardware setup section, Lets access the WebGUI through Management Interface (LAN)

3) Launch the web browser and enter the URL http://192.168.113.1 which is the default IP address of the Management interface (LAN) of the GSM Gateway.

Figure 4: Login Page

4) Login using the default username & password (Default: Username: admin; Password: admin). It takes you to change the password and login again with the new password. Observe the WAN IP address on the dashboard, this will be used to access the GUI from the WAN interface.

5) If you know the SIM No, you can directly manage the GSM Gateway using below comment by sending the SMS.

ALLOGSM "gateway password" INFO (Command to fetch the IP address of the gateway and basic status of GSM spans)

For an Example: ALLOGSM admin INFO

6) If your network is not enabled with DHCP server, configure the WAN port IP address manually in the **SETTINGS > Network Settings** section as per your requirement.

Recommended Web Browser to access GUI is Mozilla Firefox.

3. Setting up Features

3.1 Dash Board

ALLO GSM Gateway Dash Board summarizes the gateway status with a graphical display. Detailed status of an individual entity is available under the Status Tab or it can be directly accessed by clicking on more.

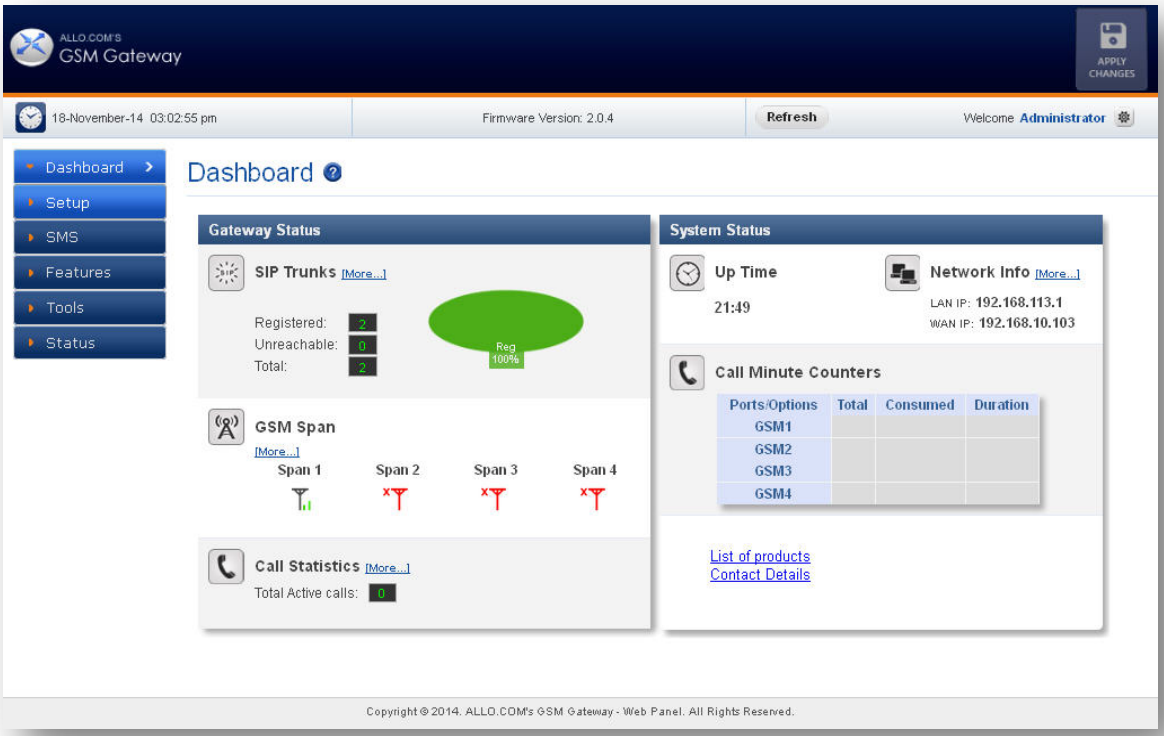


Figure 5: Dashboard

4. SETUP

4.1 Basic

4.1.1 Network

Navigate through **Setup** > **Basic>Network Settings**

WAN Configuration:

DHCP	When enabled and a DHCP server is available, the GSM Gateway will auto configure itself. If DHCP server is not available, select "Static", and fill in the Network Configuration.
IP Address	The static IP address corresponding to your WAN configuration.
Net mask	The Net mask corresponding to your WAN configuration.
Gateway	The IP address corresponding to your network Gateway (Router).
DNS	The IP address corresponding to a DNS server.

LAN Configuration:

LAN Port is a management port. GSM Gateway can be connected back-to-back to a PC or to a LAN network for configuration. It is always recommended to connect back-to-back to a PC. In case, connected to LAN network & if IP series clash is found, IP series can be changed here.

WAN port IP and LAN port IP address should not be in the same network segment.

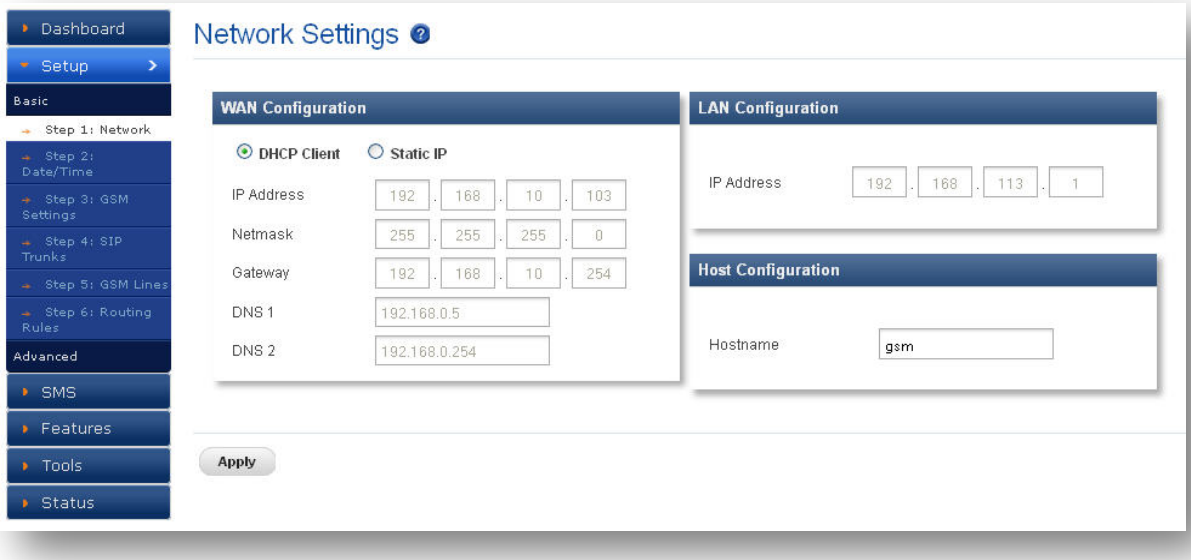


Figure 6: Network Settings

4.1.2 Date/Time

Navigate through **Setup** > **Basic**> **Date/Time**

Configuration Type	Date and Time of the GSM Gateway can be either set manually (uses RTC) or automatically (through NTP). Default: **NTP**
NTP Configuration	**Time Zone:** Select the correct time zone for the location where the GSM Gateway is installed using the Time Zone dropdown box. Default: **Asia/Kolkata** **NTP Server:** URI or IP address of the NTP (Network Time Protocol) server, which will be used to synchronize the date and time. E.g.: 3.in.pool.ntp.org

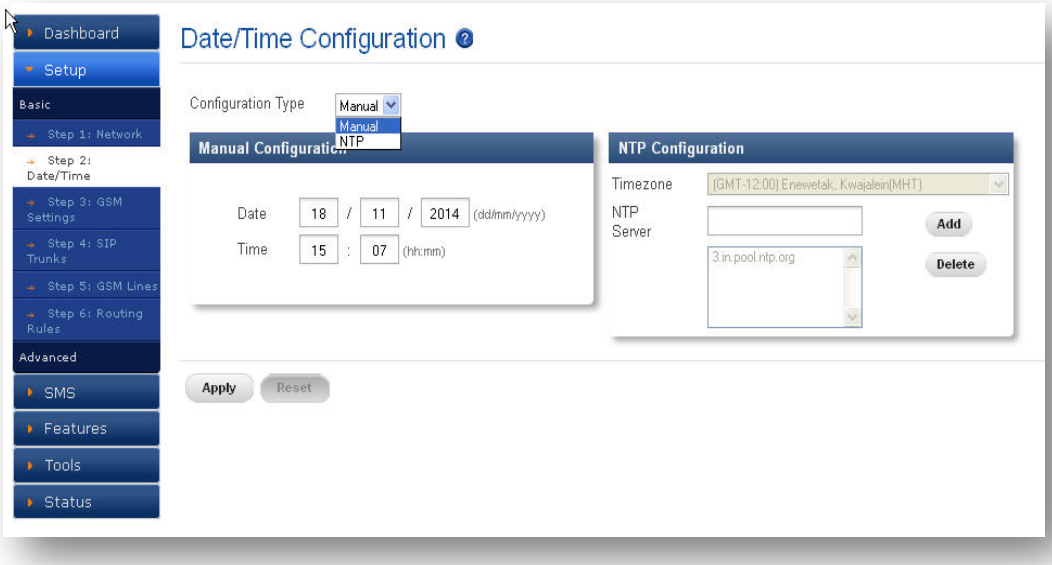


Figure 7: Date/Time Configuration

4.1.3 GSM Settings

Navigate through **Setup** > **Basic**> **GSM Settings**

This section provides the ability to modify the GSM settings depending on the carrier with respect to the service provider

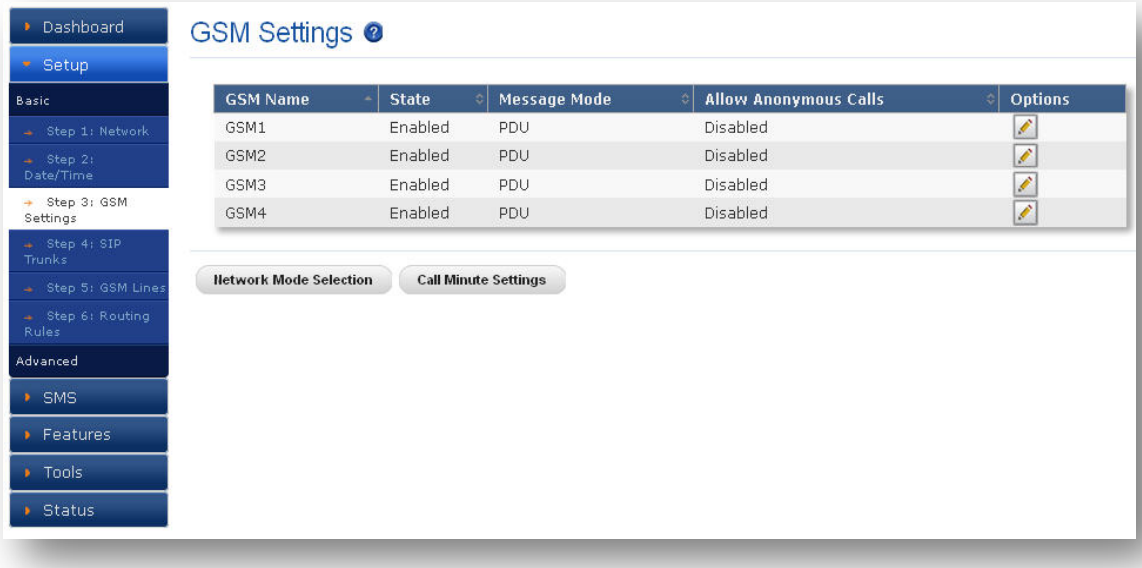


Figure 8: GSM Settings

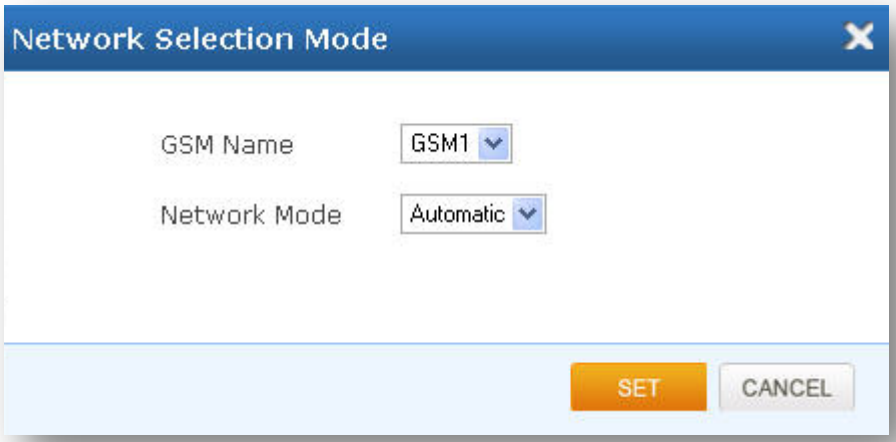


Figure 9: Network Selection Mode

Network Selection Mode

GSM Name- User can select GSM span for which network mode is to be selected.

Network mode- User can select either automatic or manual network mode from the drop down list.

For Manual mode, we have to fetch available networks and user can choose preferable network.

Call Minute Settings

Call minutes can be limited for each span individually. Following are the limits.

1. **Total minutes:**
   a) For every span we can give a limit, i.e. number of minutes. This is the maximum number of minutes a user can make a call on selected span.
   b) User can set alarm over email for the alarm limit.

2. **Call Duration:** Moreover, call duration limiting timer can be selected for the outgoing call. If the call duration feature is enabled, user can't make a single call beyond that limit.

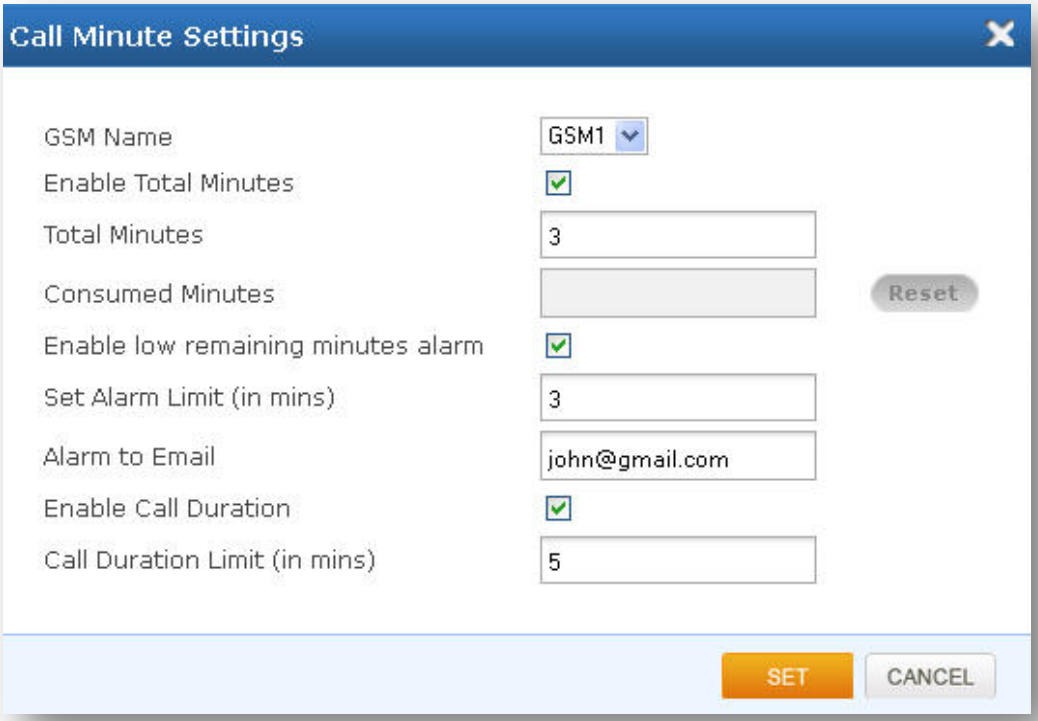


Figure 10: Call Minute Settings

Click GSM Settings Edit button, you will be navigated to Edit GSM Settings screen page. In this page you can edit the GSM name.

Figure 11: Edit GSM Settings

GSM name	GSM name is the name for your in identification
Port State	Enable/ Disable option allow user to change the state as per requirement
SIM PIN	SIM PIN will be given by the provider
SMS Center	A short message service center (SMSC) is a network element in the mobile telephone network
Forward SMS to E-mail	Configure email address where the SMS need to be forwarded
Band Selection	Band selection can be done in Auto, Mono Band mode (850,900E, 1800, 1900 MHz) and Dual band mode (900E/1800 and 900E/1900). Default is **AUTO**
Echo Cancellation	Enable /Disable option allows user to configure according to the requirement

DTMF Detection	DTMF detection in the device controlled be DSP and GSM engine
DTMF Out dialing	DTMF out dialing can be configured in INBAND or USING AT commands
DTMF Duration (msec)	The duration in millisecond of DTMF transmission. Default is : 80
Allow Anonymous Calls	IF this option enabled ,the calls with out caller ID will be allowed

Please configure these settings after contacting support as incorrect updates may lead the system being faulty.

4.1.4 SIP Trunks

Navigate through **Setup** > **Basic** > **SIP Trunks**

SIP Trunks provides the interface to any SIP companion such as VoIP service provider, any SIP server or SIP clients. Add different types of interfaces, and configure the signaling & media settings for each trunk. SIP trunk registration is limited to 10 SIP trunks.

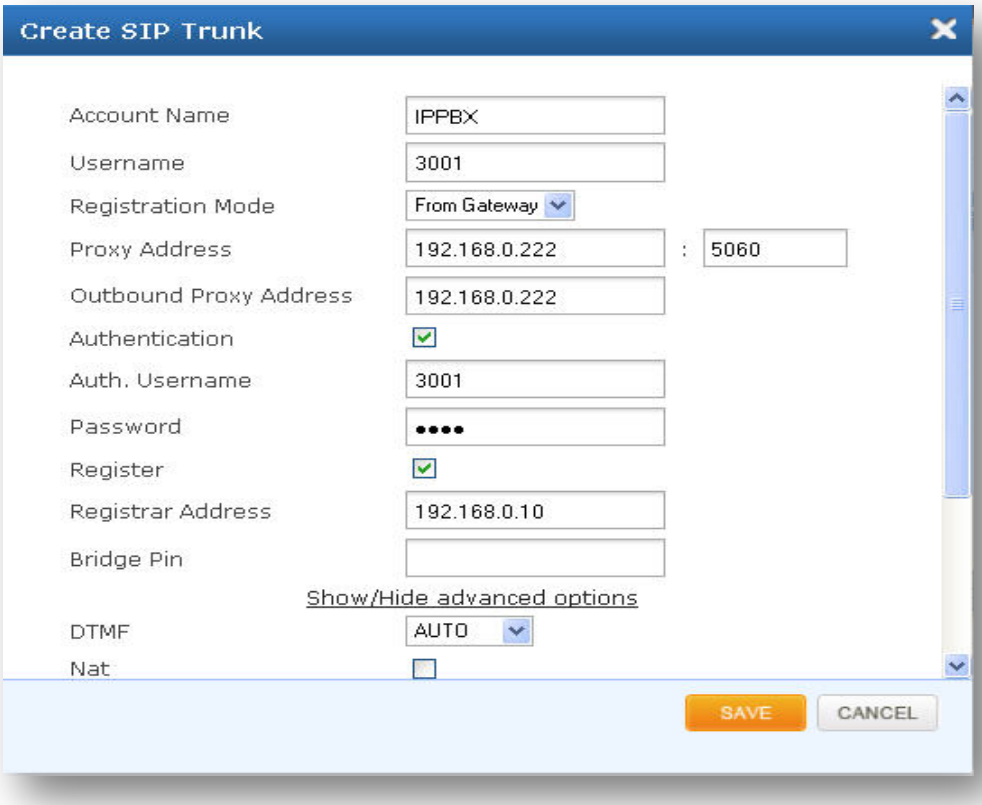


Figure 12: Create SIP Trunk

Account Name	Descriptive name for the SIP Trunk for user's reference.
Username	Username of endpoint (e.g.: IPPBX) will use to authenticate with the gateway.
Registration Mode	Whether the endpoint will register to this gateway or this gateway will register to the endpoint. a. None: No Registration, IP Address peering of the Gateway & Endpoint b. To Gateway: Endpoint (IPPBX or VoIP Service Provider) register with the Gateway c. c) From Gateway: Gateway register to the End point (IPPBX or the VoIP Service Provider)
Authentication	Enable, if Authentication is required by the End point (VOIP Service Provider or IPPBX)
Auth. Username	A username to use only for registration.
Password	Password to authenticate inbound or outbound registrations or calls.
Proxy Address	IP address or hostname with port of the endpoint (VOIP Service Provider or IPPBX) where the calls will be diverted. Default port no.: **5060**
Outbound Proxy Address	IP address or hostname with port of the outbound proxy server. This ensures that all the SIP packets are sent via specified proxy. Specifying the port is not mandatory. Default port no.: **5060** e.g.: 192.168.0.222:5062 OR 192.168.0.222
Register	Enable, if Registration to the End point (VOIP Service Provider or IPPBX) is required.
Registrar Address	IP address or hostname with port of the Registrar server where gateway must register to. Specifying the port is not mandatory. Default port no.: **5060** e.g.: 192.168.0.222:5062 OR 192.168.0.222
Bridge PIN	You can set a PIN for outgoing calls on SIP trunk, thus you will set one more level of security. Leave it blank for unsecured

	mode.

Advanced Options	
DTMF	Set default DTMF mode for sending DTMF digits. Options: ▪ INBAND – sent along with audio (requires 64 kbit codec - alaw, ulaw) ▪ INFO – sent as SIP INFO messages ▪ RFC2833 – sent as RTP packets ▪ AUTO – System automatic selects the mode. Uses RFC2833 if offered, inband otherwise. Default: **AUTO**
NAT	Enable it, to address NAT-related issue for SIP & media sessions for this trunk. *Configuring NAT settings under SIP Global Settings is required) Default:* ***disabled***
Codec Configuration	Choose the available Codecs and set GSMority in the order in which gateway should prefer to send and receive audio. Supported codecs are alaw, ulaw, G.729, G.722

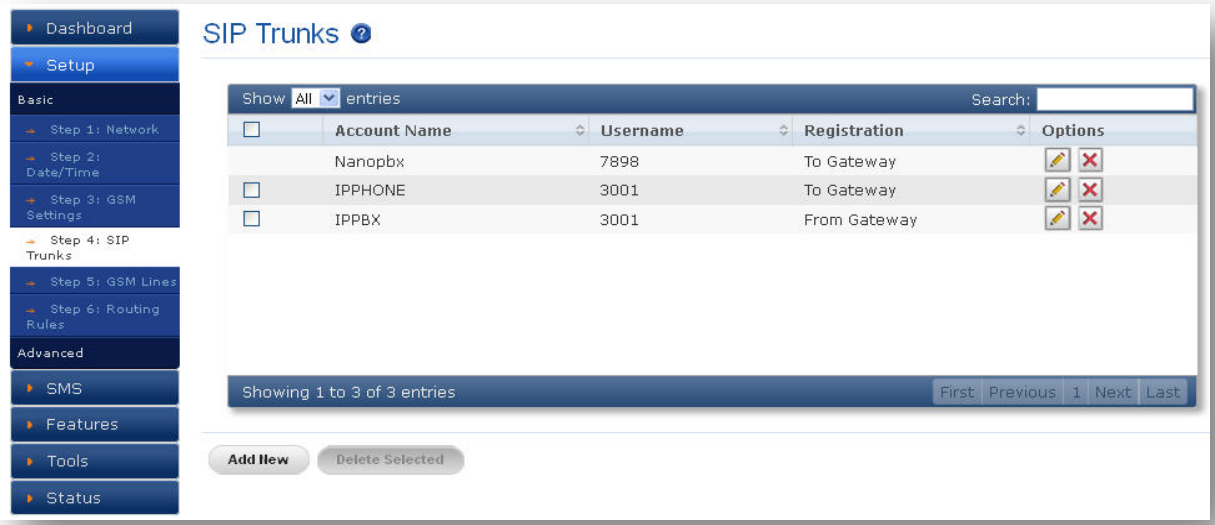


Figure 13: SIP Trunks

4.1.5 GSM Lines

Navigate through **Setup** > **Basic** > **GSM Lines**

GSM Trunks provides the interface to any GSM companion such as GSM service provider. Create an interface for each span.

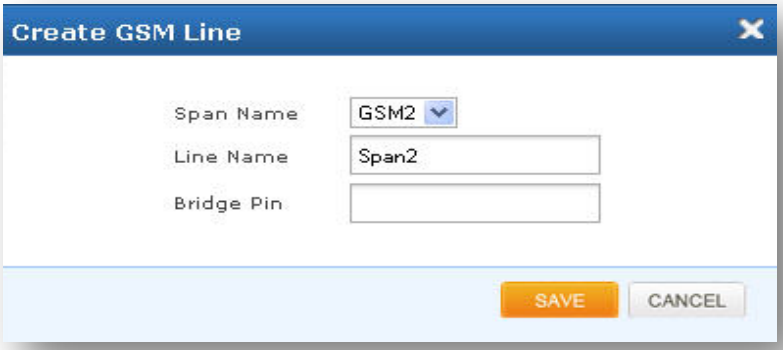


Figure 14: Create GSM Line

Span Name	Select the appropriate GSM Spans.
Trunk Name	Descriptive name for the GSM Trunk for user's reference.
Bridge PIN	You can set a PIN for outgoing call on GSM trunks, thus you can set one level of security. Leave it blank for unsecured mode.

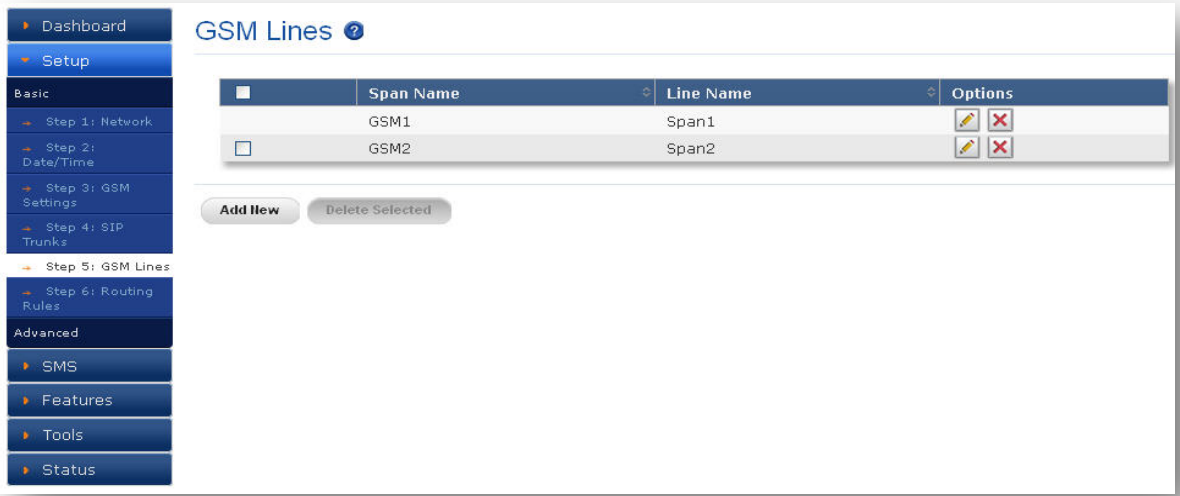


Figure 15: GSM Lines

If you delete GSM lines and already route is present which is using particular GSM lines, you will get the below error message.

Line name is dependency with Calling Groups, Call Routing Rules, & Time Route Groups.

4.1.6 Routing Rules

Navigate through **Setup** > **Basic** > **Routing Rules**

A routing rule determines how a call is handled based upon certain characteristics such as dialed number, inbound trunk, DID, etc.

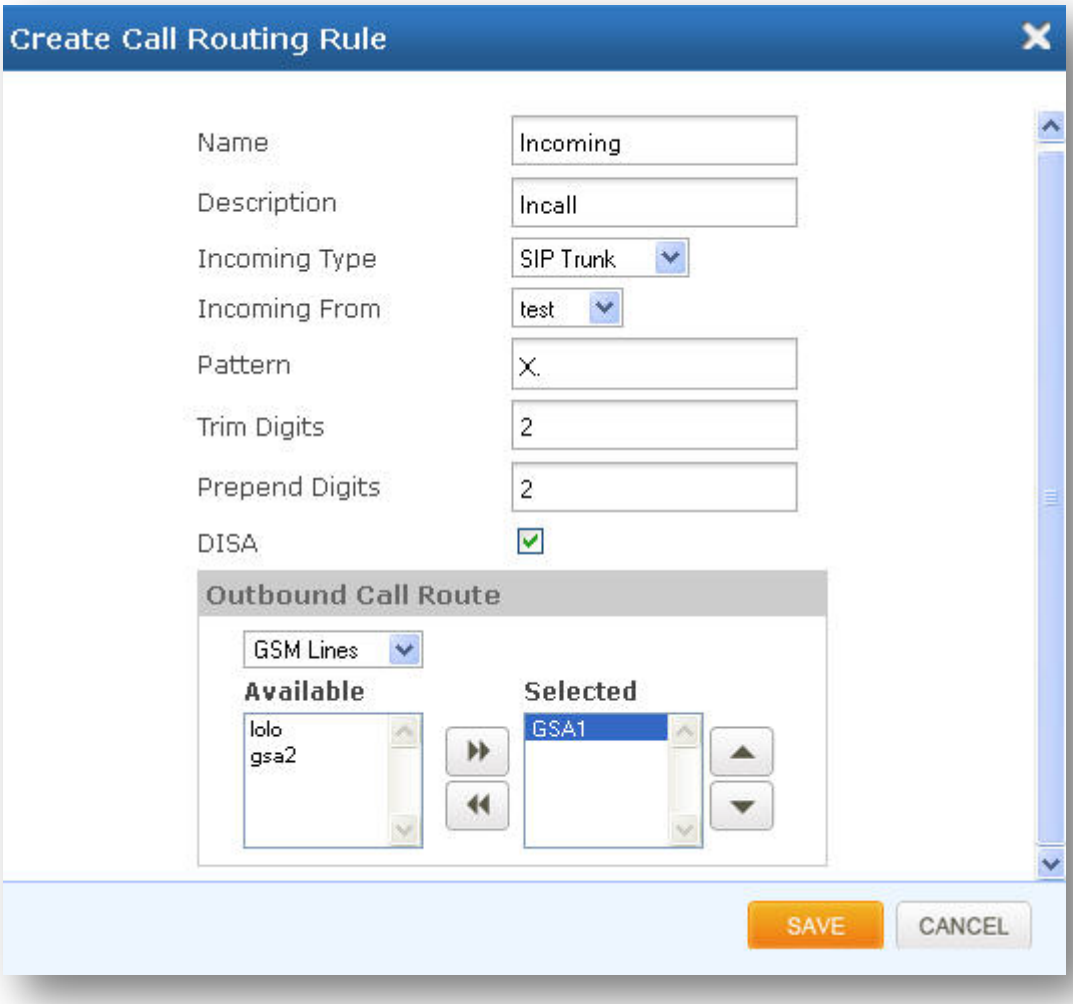


Figure 16: Create Call Routing Rule

Name	Descriptive name for the Call routing rule for user's reference.
Description	Provide the description for the Call routing rule. (Optional)
Incoming Type	Specify how the gateway should match incoming calls. The gateway can match on GSM Trunk, SIP Trunk or Calling Group.
Incoming From	Specify the available trunk or calling group for the selected

	Incoming Type.
Pattern	Specify the pattern to match the dialed string of the incoming call. Pattern: X: Any Digit from 0-9. Z: Any Digit from 1-9. [12345-9]: Any digit from 1 to 9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately. e.g.: X. – match at least one digit 988XXXX – match 988 followed by 4 digits
Trim Digits	Allows you to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk. e.g.: If you configure the pattern as 9X. and you want to strip 9 then you should mention Trim Digits field as 1
Prepend Digits	Specify the digits to be prepended before the call is placed via the selected trunk. Those digits will be prepended after the dialing number is stripped. e.g.: If dialed number is 8789763010 and if you want to prepend 44 as a country code then mention in Prepend digits field as 44
DISA	Available for GSM Incoming routing rule which allow us to receive Dialout number in GSM Gateway. The received number is dialed afterwards.
Outbound Call Routes	Select the preferred trunks or time route groups where calls are to be routed for this Routing rule. Ordering of the trunks in the “Selected” column indicates the order in which call flows on failure.

Figure 17: Call Routing Rules

4.2 Advanced

4.2.1 Calling Groups

Navigate through **Setup** > **Advanced** > **Calling Groups**

A calling group allows a group of trunks to be addressed when creating a call routing rule. Multiple groups can be created with different combination of trunks, but of the same type (SIP or GSM).

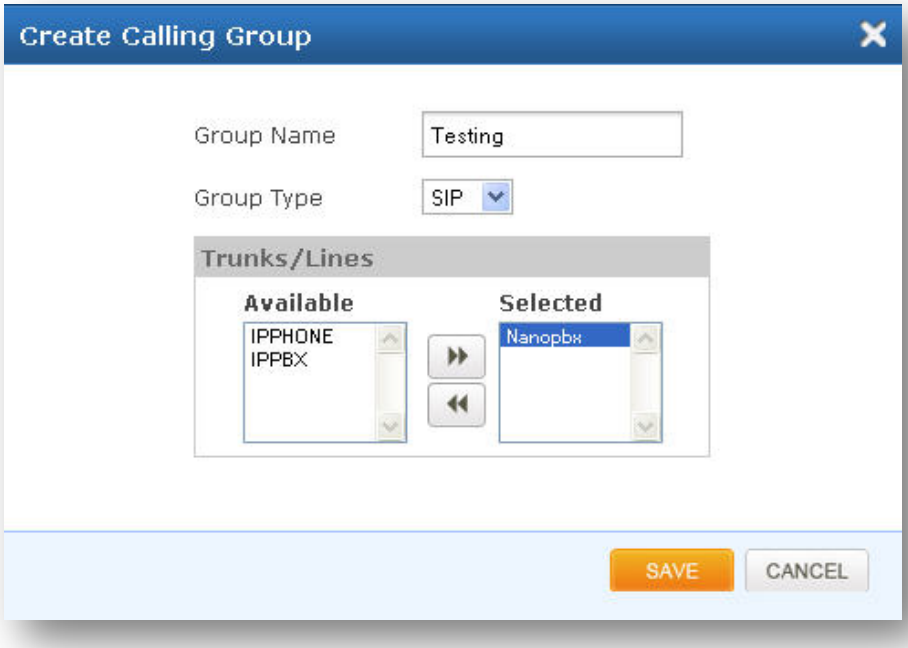


Figure 18: Create Calling Group

Group Name	Descriptive name for the Calling Groups for user's reference.
Group Type	You can select group type as SIP or GSM Trunk. For e.g.: If you select Group Type as SIP, SIP trunks will be listed in available Table.
Trunks	Depending upon the Group Type you have selected, available trunks can be moved to “Selected” column, to be listed under this Calling Group.

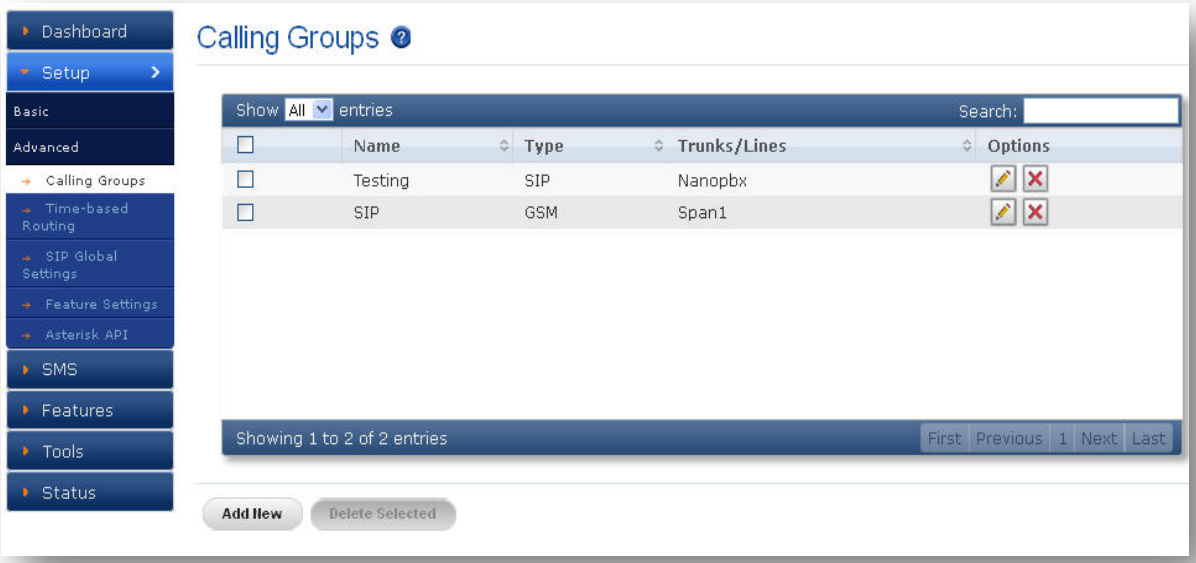


Figure 19: Calling groups

4.2.2 Time-based Routing

Navigate through **Setup** > **Advanced** > **Time-based Routing**

Time routing routes calls to different locations based on the time of day and day of week, when a call is made.

Group Name	Descriptive name for the Time Route Group for user's reference.
Description	Provide the proper description for the Time based routing rule. (Optional)
Route To Trunk	Select the destination trunk where the call is routed to on matching the time. The destination can be any trunks – GSM Trunks/SIP Trunks.
Duration	Specify the time range for which this routing rule will apply. Format: hh:mm
Days	Select the day/days during which this routing rule will apply.

*Make sure that the current date and time are configured currently under **System**> **Date/Time** Configuration.*

4.2.3 SIP Global Settings

Navigate through **Setup**>**Advanced** > **SIP Global Settings**

SIP Global settings apply to all VoIP traffic.

Port Settings

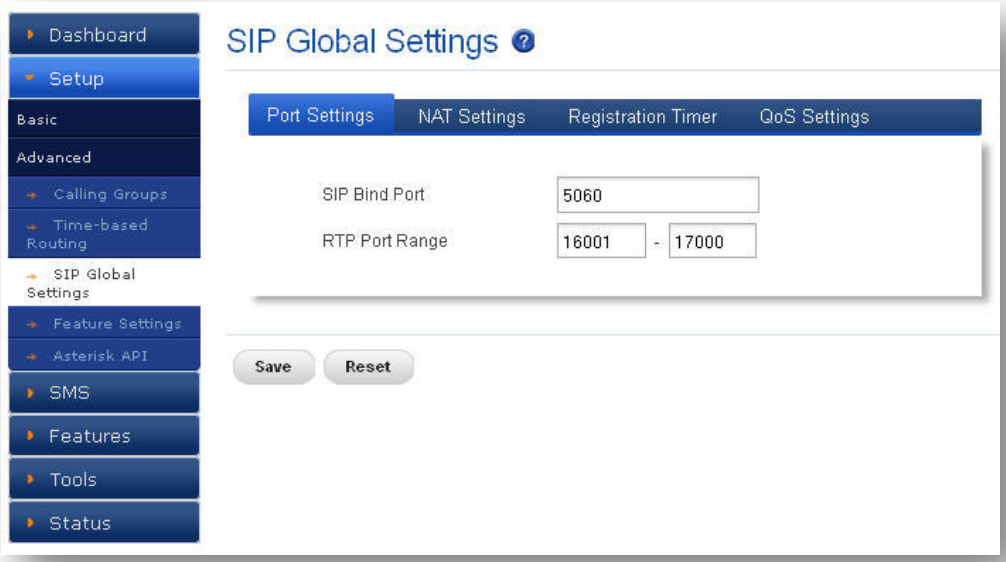


Figure 20: Port Settings

Port Settings	
SIP Bind Port	Choose a port on which to listen for SIP UDP traffic. Default: **5060**
RTP Port Range	Range of port numbers to be used for RTP traffic. Default: **16001- 17000** Make sure you configure this dynamic range of ports on your NAT Router. When the GSM Gateway is behind a NAT and the NAT is configured to do port forwarding with above mentioned port range for UDP ports.

NAT Settings

Navigate through **Setup**> **Advanced**> **SIP Global Settings**

In this NAT Settings, user can select Stun Server IP or External IP function. These functions can help your VoIP device working properly behind NAT.

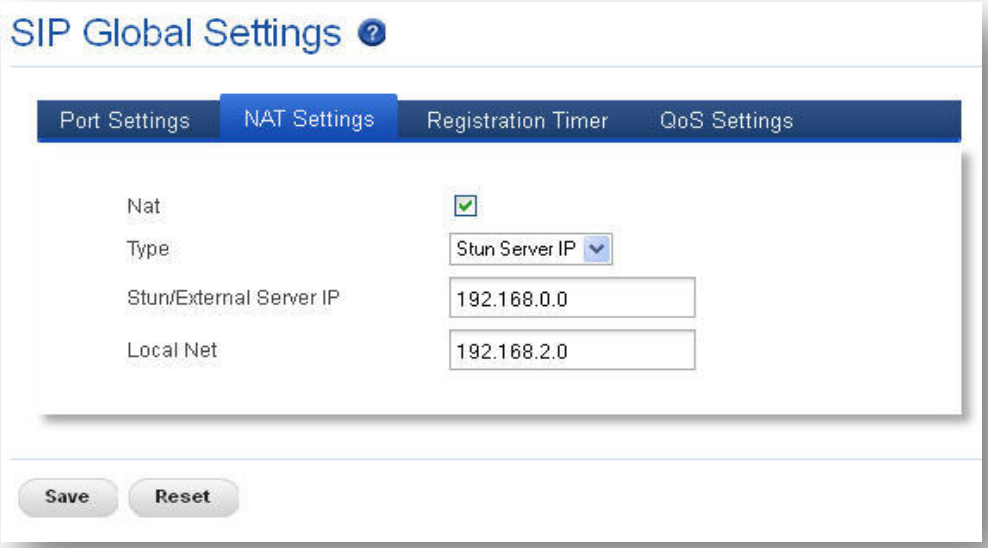


Figure 21: NAT Settings

NAT Settings	
NAT	NAT option is checked, when the GSM Gateway is behind the Router/Firewall. Select either Stun Server IP or External IP. Default: **disabled**
Stun Server IP	If the GSM Gateway is behind a non-symmetric NAT router, it may be necessary to use STUN to allow GSM Gateway to reliably communicate via IP through the router. Enter a STUN server IP address or domain name in the STUN Server field. For a list of public STUN servers, please Refer to: http://www.voip-info.org/wiki/view/STUN
External IP	Enter the NAT Traversal IP address i.e. Public IP Address of your internet, to communicate with Public Network when GSM Gateway is behind the NAT. This IP address will substitute in all outgoing SIP messages instead of Local IP address.
Local Net mask	Entering the Net mask of the local network of the GSM gateway allows it to identify the hosts falling within the same network. E.g.: 192.168.2.0/255.255.255.0

Registration Timer

Navigate through **Setup**> **Advanced**> **SIP Global Settings> Registration Timer**

When the registration timeout is 0, user cannot register the gateway and it will be connected to the server directly. You can reference the setting parameters of the single server mode to register the gateway.

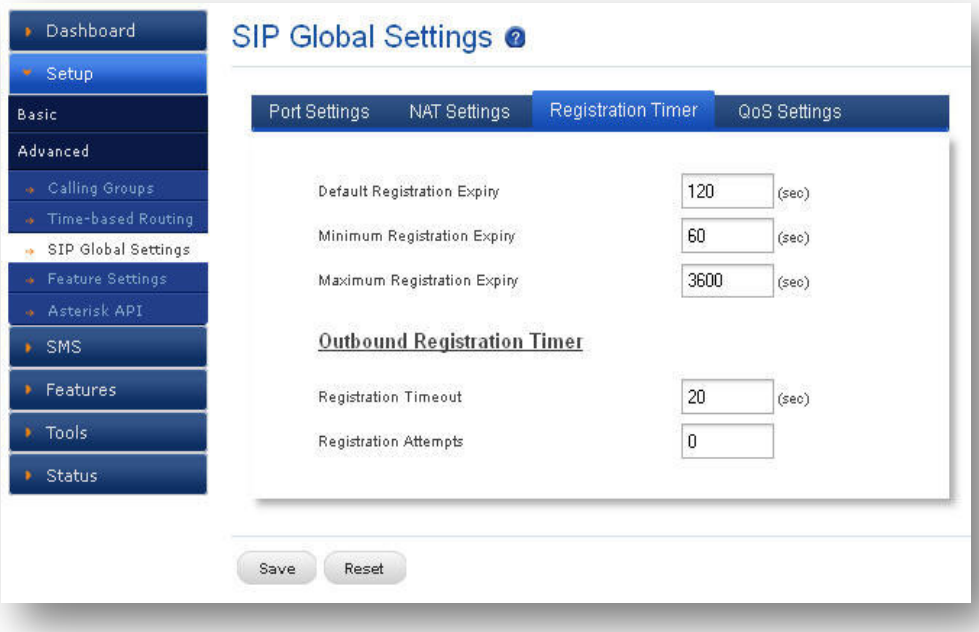


Figure 22: SIP Global Settings

Registration Timer	
Default Registration Expiry	Default duration (in seconds) of incoming/outgoing registrations. Default: **120 sec**
Minimum Registration Expiry	Minimum duration (in seconds) of registrations allowed by the Gateway. Default: **60 sec**
Maximum Registration Expiry	Maximum duration (in seconds) of incoming registrations allowed by the Gateway. Default: **3600 sec**
Registration Timeout	Registration attempt will be retried till this duration (in seconds), if no response from the Registrar. (Outbound Registrations only). Default: **20 sec**
Registration Attempts	Number of registration attempts before giving up with

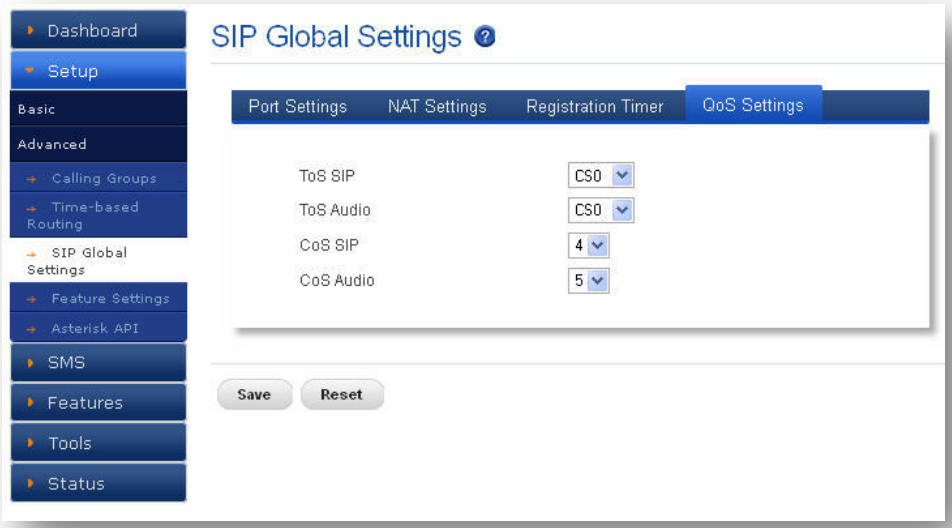
	registrar (Outbound Registrations only). Default: **0** (never give up)

QOS Settings

Navigate through **Setup**> **Advanced**> **SIP Global Settings> QoS Settings**

This QoS feature requires your QoS support of your network to improve voice data traffics. Please consult your network administrator for proper setting.

Signaling QoS improves the performance of SIP signaling. If local network device supports QoS, select this field accordingly.



QOS Settings	
ToS SIP	Sets TOS for SIP packets
ToS Audio	Sets TOS for RTP audio packets.
CoS SIP	Sets 802.1p priority for SIP packets.
CoS Audio	Sets 802.1p priority for RTP audio packets.

4.2.4 Feature Settings

Navigate through **Setup** > **Advanced**> **Feature Settings**

It allows user to configure the call back settings like Max Retries, Retry time and wait time.

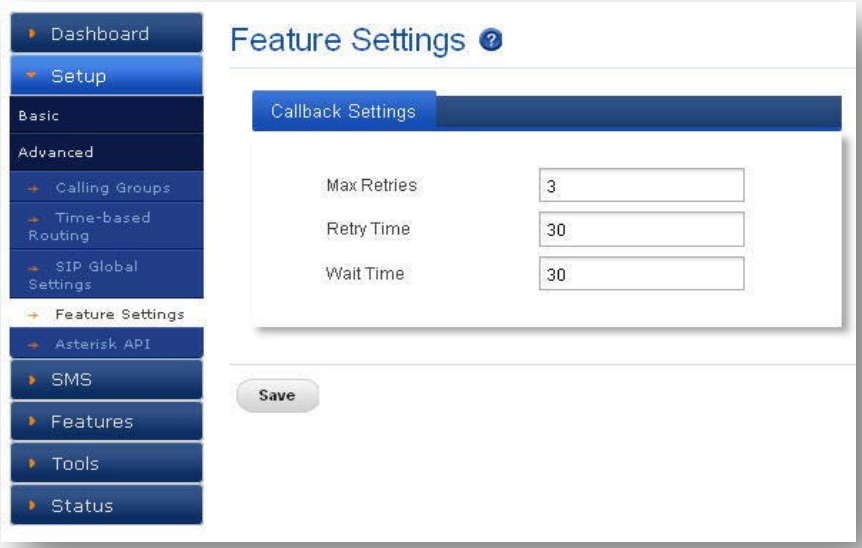


Figure 23: Feature Settings

Call back settings

Max Retries	Number of retries before failing (not including the initial attempt, e.g.0=total of 1 attempt to make the call).
Retry time	It specifies the seconds between retries, don't hammer an unavailable phone.
Wait time	It specifies the seconds to wait for an answer.

4.2.5 Asterisk API

The Asterisk Manager Interface (AMI) allows a client program to connect to an Asterisk instance and issue commands or read events over a TCP/IP stream. GSM Gateway can be configured for AMI in Setup > Advanced > Asterisk API.

General

When you click "Enabled" option, this page is available. It allows users to access the gateway API. And the port number is default: 5038 (fixed).

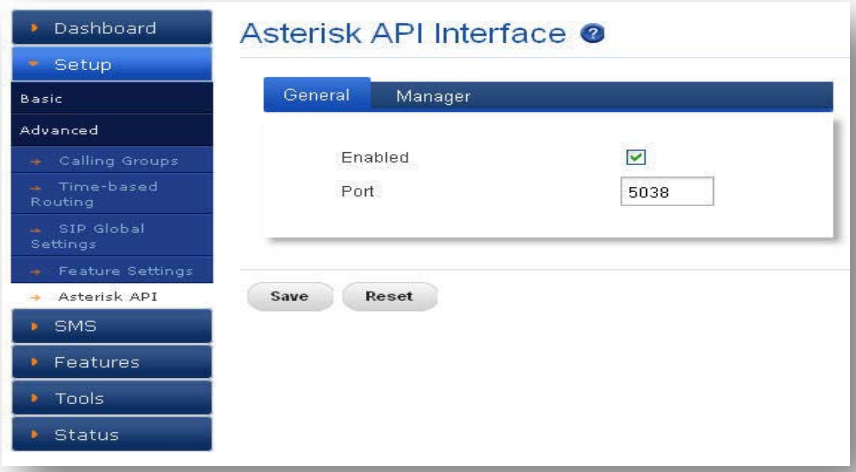


Figure 24: Asterisk API Interface

Asterisk API Interface- Manager

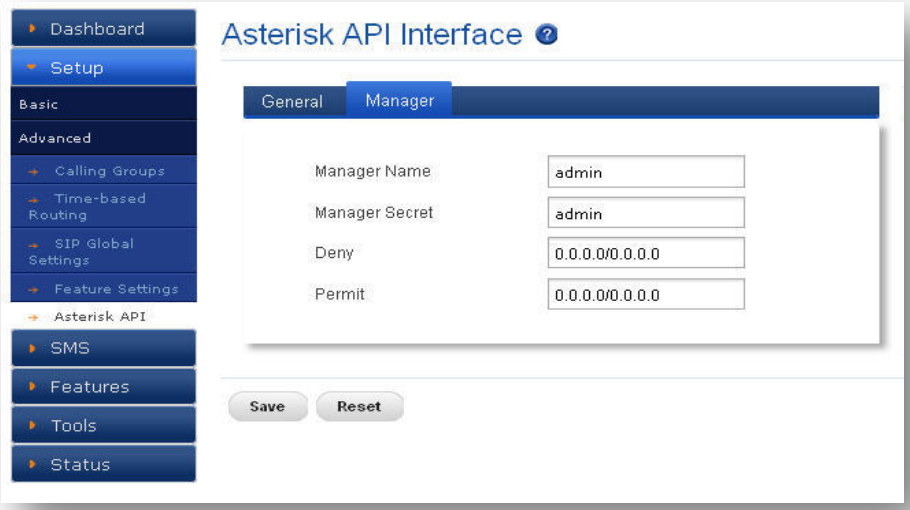


Figure 25: Asterisk API Interface-Manager

Manager Name	Enter the name of the manager for Asterisk API Interface manager. E.g.: Admin
Manager Secret	Enter the password for the manager. Default Password: Admin
Deny	If you want to deny many hosts or networks, use char & as separator. E.g.: 0.0.0.0/0.0.0.0 or 192.168.1.0/255.255.255.0&10.0.0.0/255.0.
Permit	The user wants to permit many hosts or network, use char & as separator. E.g.: 0.0.0.0/0.0.0.0 or 192.168.1.0/255.255.255.0

5. SMS

5.1 Inbox

Navigate through SMS > Inbox

SMS inbox feature allows you to check the inbox of each SIM cards inbox messages

SMS inbox will provide message details such us date and time, message from field information, message body and span information." Options" tab in the SMS inbox help you to delete / View.

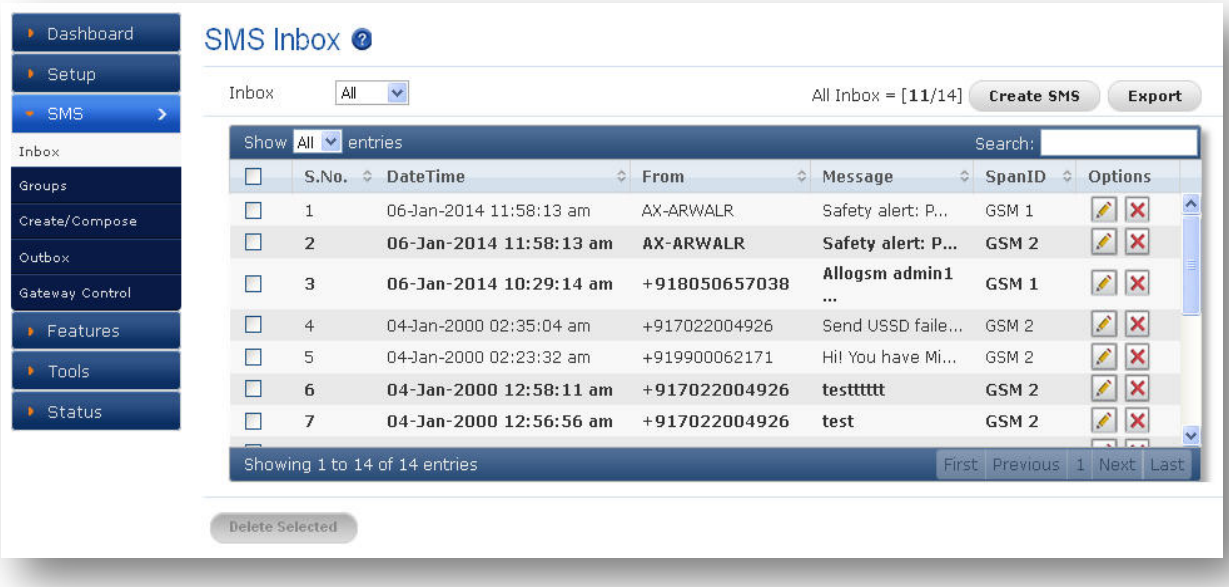


Figure 26: SMS Inbox

Date & Time	Date and time of the INBOX message
From	The source of the INBOX Message
Message	It shows the body of message
Span ID	On which span the message received
Options	Two options are given (View and Delete)
Export	This will allow to export the message in .csv format

Click Edit button, you will be navigated to SMS inbox. In this page you can edit the Message.

5.1.1 Create SMS

Navigate through **SMS** > **Create/Compose.**

Users can create and send the SMS in this page. And you can import numbers from a .CSV file here.

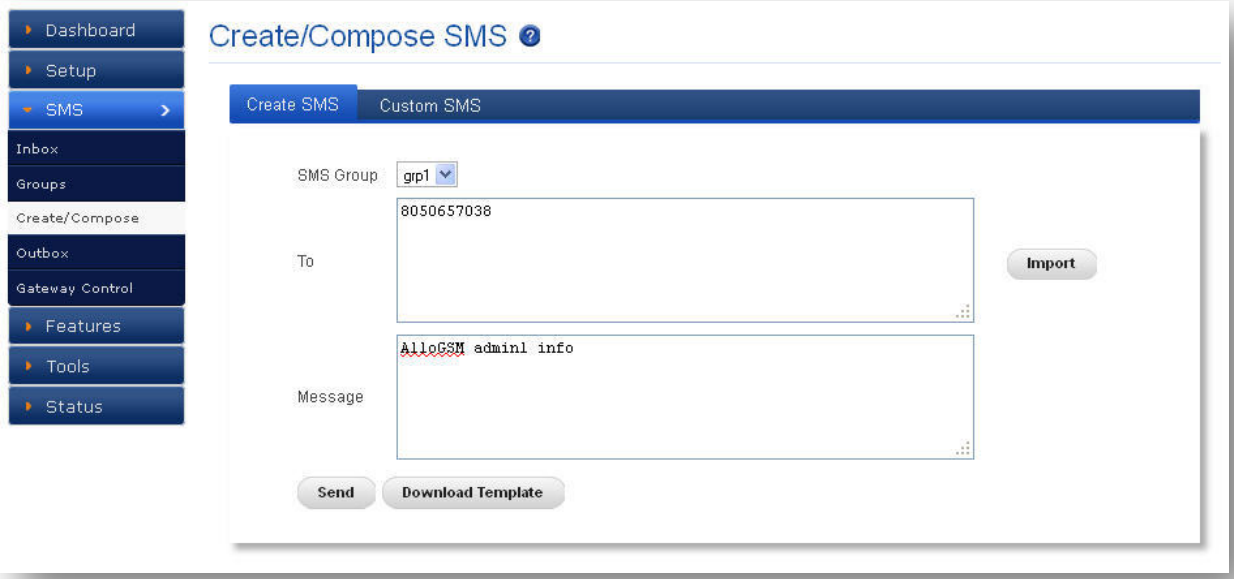


Figure 27: Create SMS

Click Download Template button, to download the GSM gateway Custom SMS.

5.1.2 Custom SMS

Navigate through **SMS** > **Create/Compose.**

This Custom SMS feature allows you to assign *custom* text on a group basis. User can choose the SMS group name and .CSV file for sending SMS. E.g.: GSM_Gateway_Custom SMS (file name).

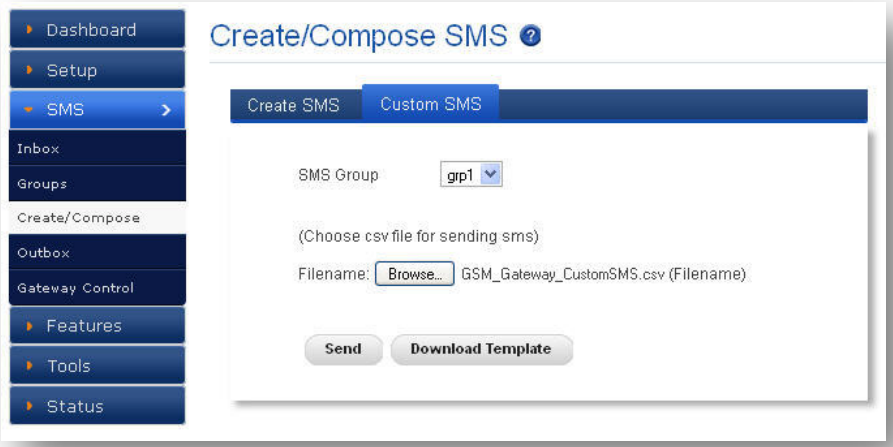


Figure 28: Custom SMS

Click Download Template button, to download the GSM gateway Custom SMS.

5.2 Groups

Navigate through **SMS** > **Groups.**

For sending sms, User has to make group of spans from which we can send SMS. A group can have minimum 1 and maximum 4 spans (for 4 port gateway). This Gateway will automatically schedule sms sending on free spans.

SMS can only be sent on Groups.

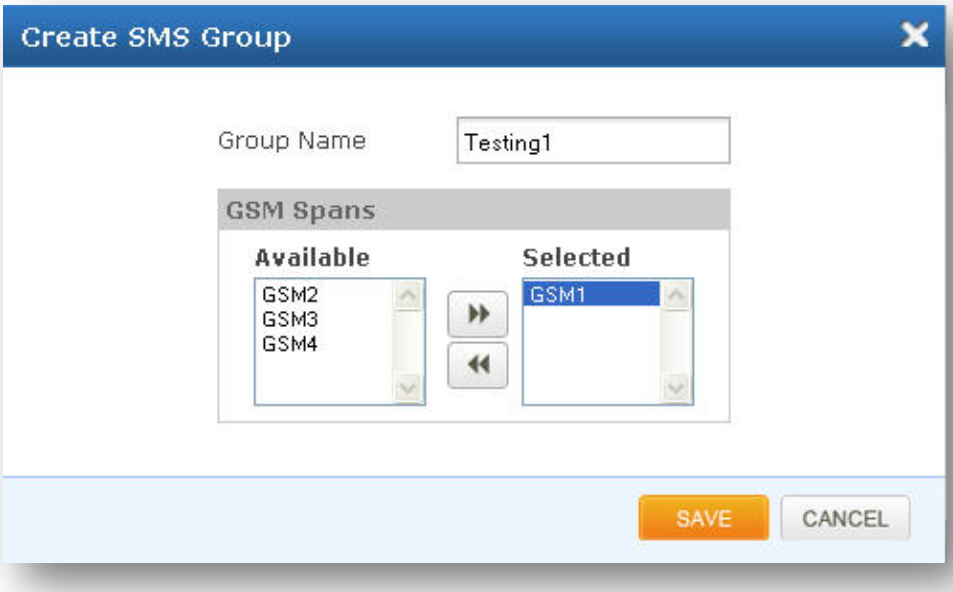


Figure 29: Create SMS Group

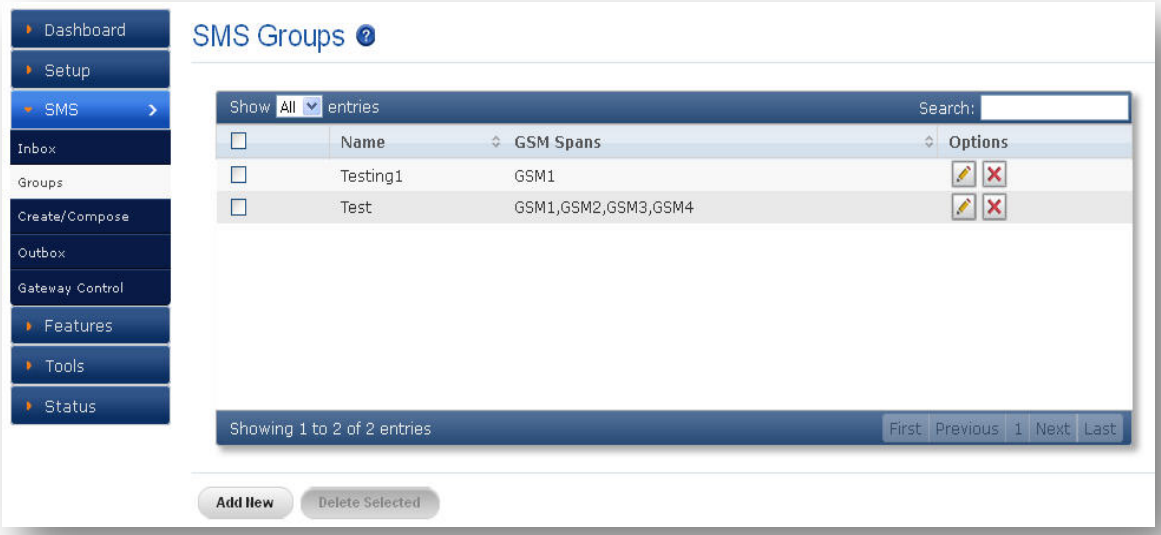


Figure 30: SMS Groups

5.3 Create/Compose

Navigate through **SMS** >**Create SMS**

Create SMS features allow customers to send one to one message and also bulk sms. This feature also allow customer to upload the lists of message recipients *and custom message in .csv format*

GSM	For selecting the GSM channel
To	Destination number separated by semicolons or custom destination number list can be imported using IMPORT button in .csv format
Message	Enter the body of your message.
Custom SMS	Custom messages can be uploaded which is in .csv format. Template is available on the same page.

5.4 Outbox

Navigate through **SMS** >**Outbox**

It shows the status for outgoing SMS and the user can received current message sending status. And also it shows the number of messages sends and failed.

If sending message is failed, user can download the list of filled messages in CSV format.

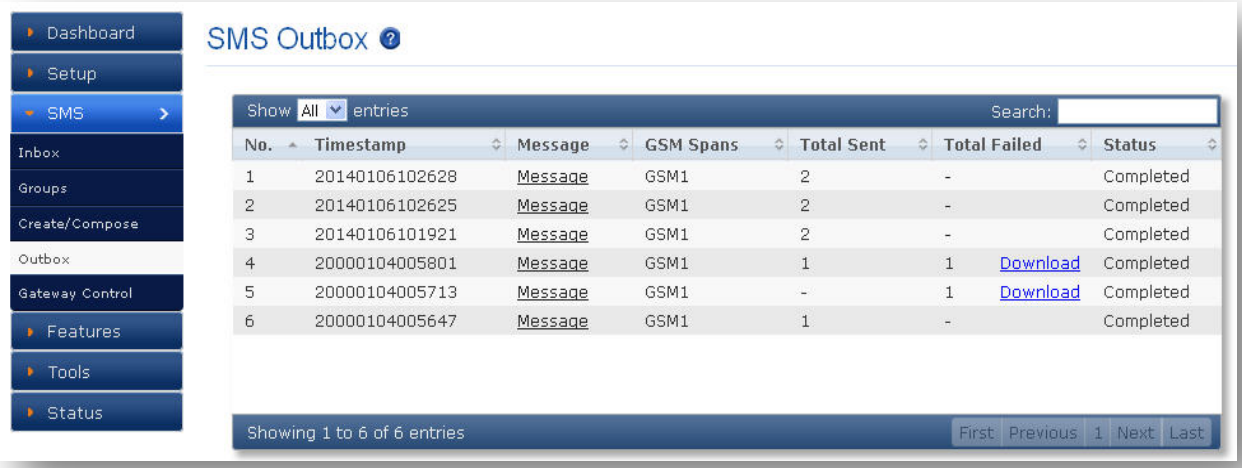


Figure 31: SMS Outbox

5.5 Gateway Control

Navigate through **SMS** >**Outbox**

The GSM gateway can be controlled via SMS for the following commands.

1. **ALLOGSM "*gateway-password*" REBOOT** (Command to reboot the gateway)
2. **ALLOGSM "*gateway-password*" RESTART ASTERISK** (Command to restart asterisk)
3. **ALLOGSM "*gateway-password*" RESTORE CONFIG** (Command to restore configuration to factory defaults)
4. **ALLOGSM "*gateway-password*" INFO** (Command to fetch IP and basic status of GSM spans)

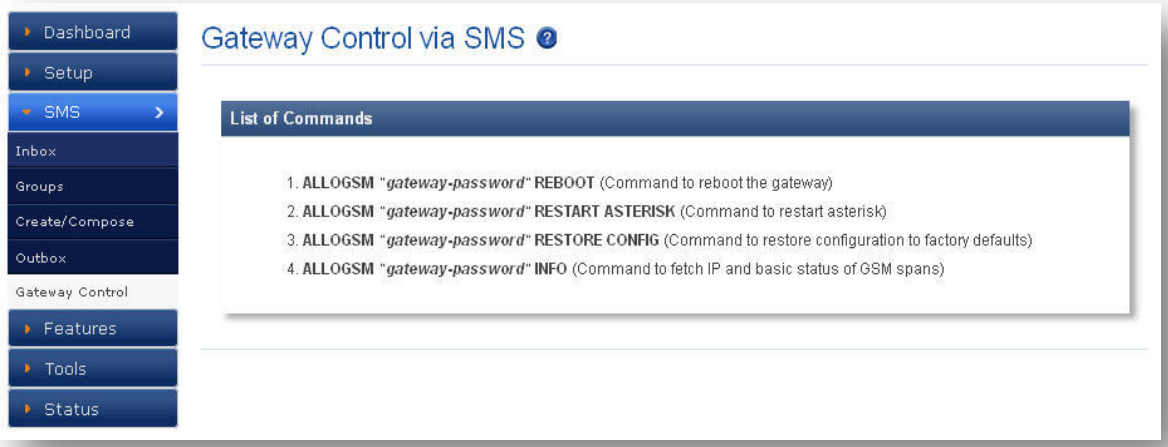


Figure 32: Gateway Control via SMS

6. Features

6.1 Caller List

Navigate through **Features** >**Caller list**

It allows user to store numbers in the Caller list to the Direct Dial memory.

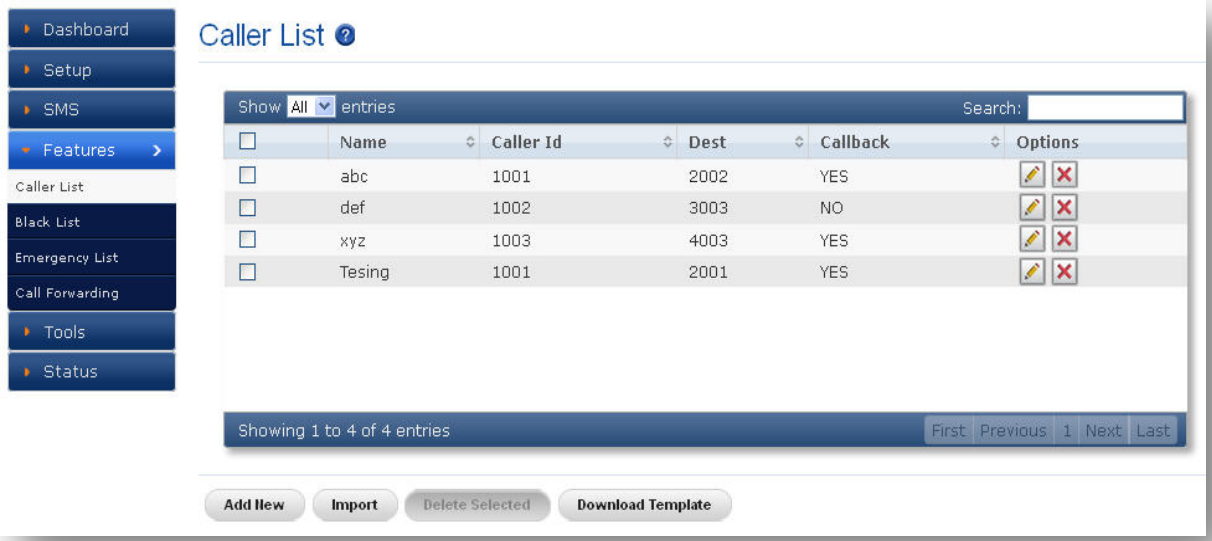


Figure 33: Caller List

Click Download Template button, to download the GSM gateway caller list.

Create Caller list

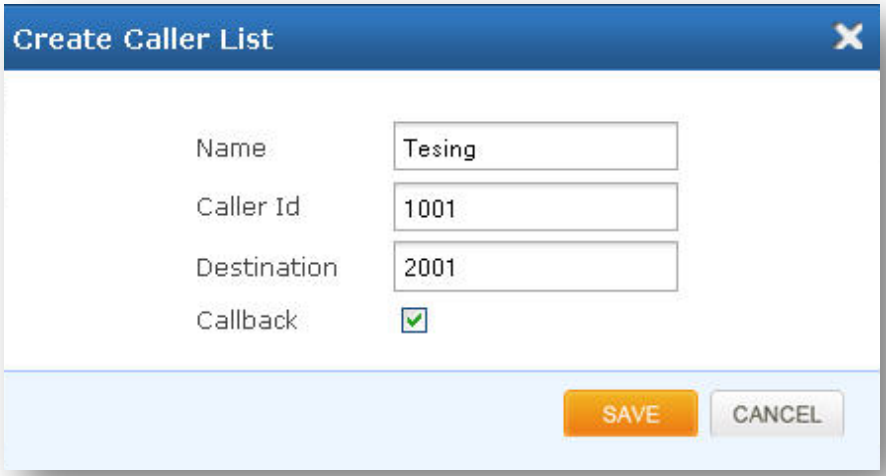


Figure 34: Create Caller list

Name	User can enter the name for creating caller list.: E.g. John

Caller Id	User can make a call to GSM gateway from caller id and a call will be made to destination.
Destination	If the destination does not answer the call, (if call back enabled) a call will be made to Destination.
Callback	If destination answers the call, a call will be made to caller id. Callback settings are available in **Setup**> **Advanced**> **Feature Settings**> **Callback Settings**

Click IMPORT Button, user can import the caller list entries from a .CSV file format.

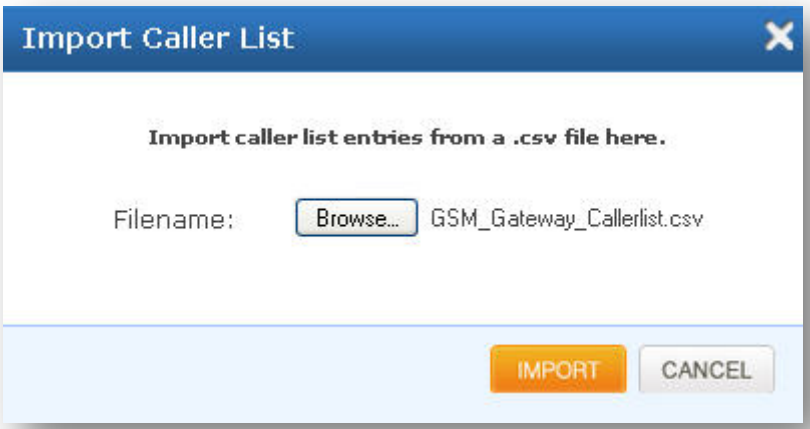


Figure 35: Import Caller List

6.2 Black List

Navigate through **Features** > **Black list**

The Blacklist contains a list of caller numbers that are rejected by the device when an incoming call from the caller is received and authentication is set to Blacklist.

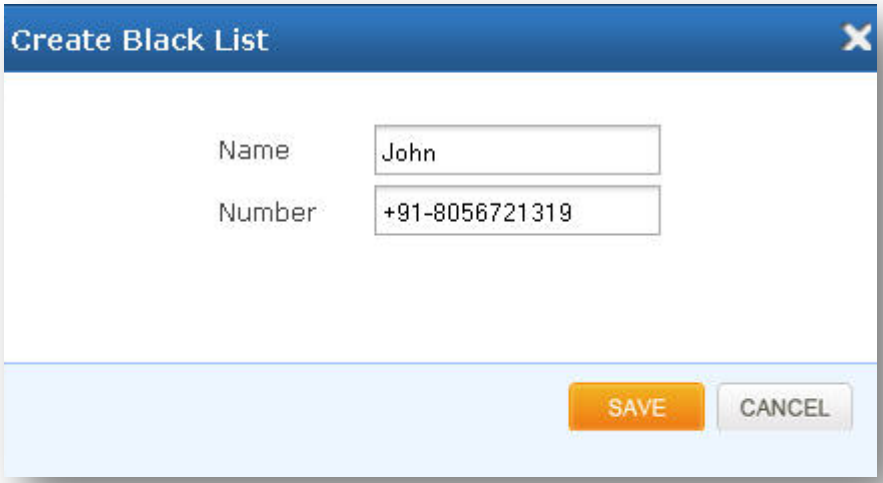


Figure 36: Create Black List

Name	Enter the name for creating Blacklist. E.g. John
Number	Enter the phone number. E.g.: +91 8056721319

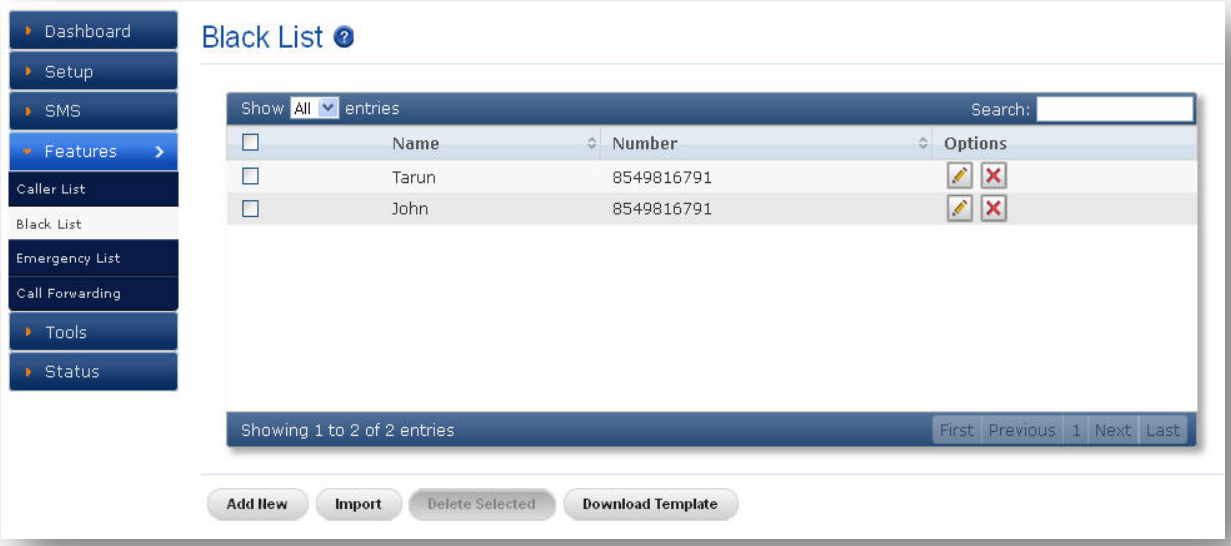


Figure 37: Black List

6.3 Emergency List

Navigate through **Features** > **Emergency list**

It allows user to store emergency numbers in the Caller list to the Direct Dial memory.

In an emergency, call 911 or your local emergency number immediately from ALLO GSM Gateway. An emergency is any situation that requires immediate assistance from the police, fire department or ambulance.

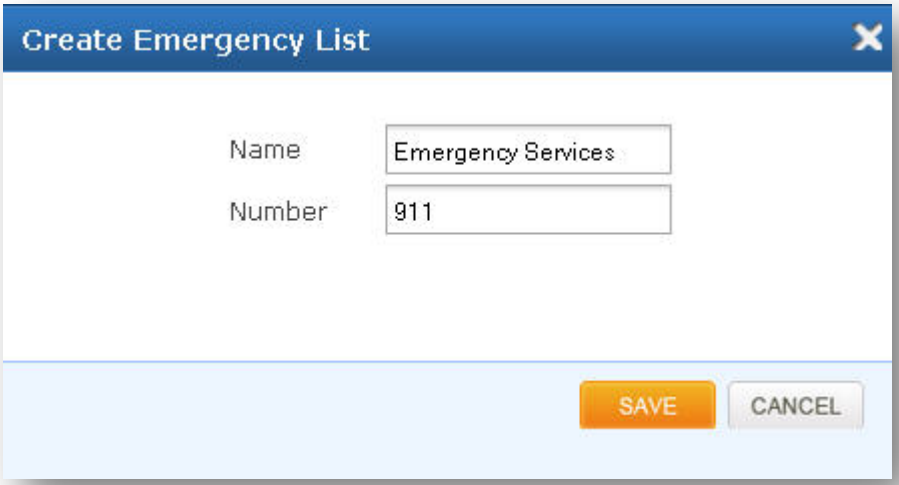


Figure 38: Create Emergency List

Name	Enter the name for creating Emergency list. E.g.: Emergency Services
Number	Enter the phone number. E.g.: 911 (must be in international number format)

6.4 Call Forwarding

Navigate through **Features** > **Call Forwarding**

With this service, you can have calls to your mobile phone forwarded to your office, home phone, or other registered number you registered.

Calls can be forwarded even if you turn off your mobile phone or are already on the line, so you will never miss another call.

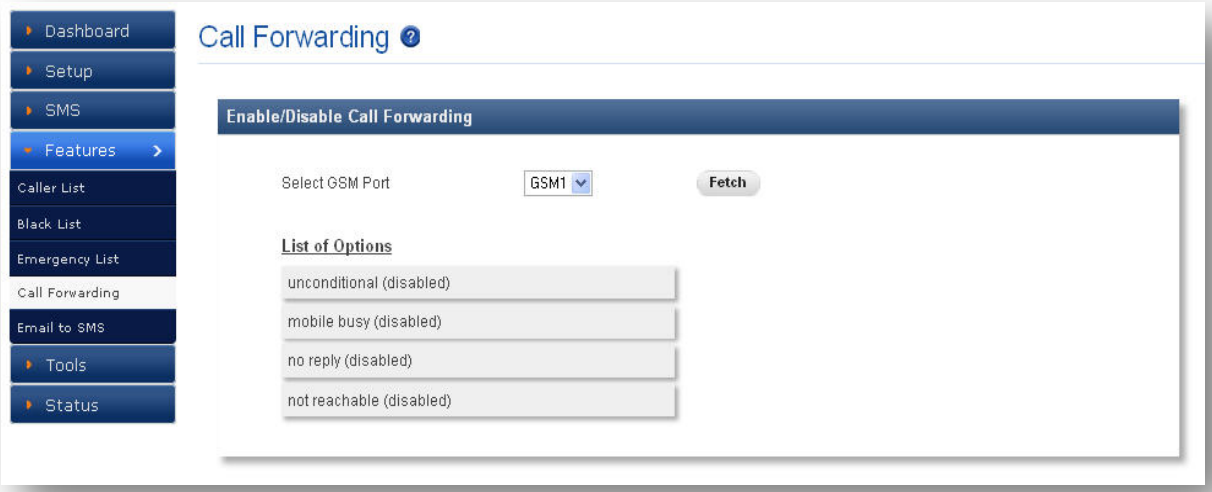


Figure 39: Call Forwarding

Users have to fetch enable /disable call forwarding and choose the list of options.

6.5 Email to SMS

Navigate through **Features** > **Email to SMS**

GSM gateway can receive Email and send content (as mentioned in template) as SMS. To receive Email on gateway, user has to configure an email client. Following are the settings.

Enable	Enabled option.
Mail Server	It specifies Incoming mail Server for POP3 services.
Email ID	Enter the login name of your email account. Note: This option might be different from your email address.
Password	Enter your Password to login your Email.
Username (Optional)	Enter your user name.
Security type	User can choose the security type from the interface.

7. TOOLS

7.1 Diagnostics

Navigate through **Tools** > **Diagnostics**

Analyze the functionality of the GSM Gateway with some of these diagnostic tools provided.

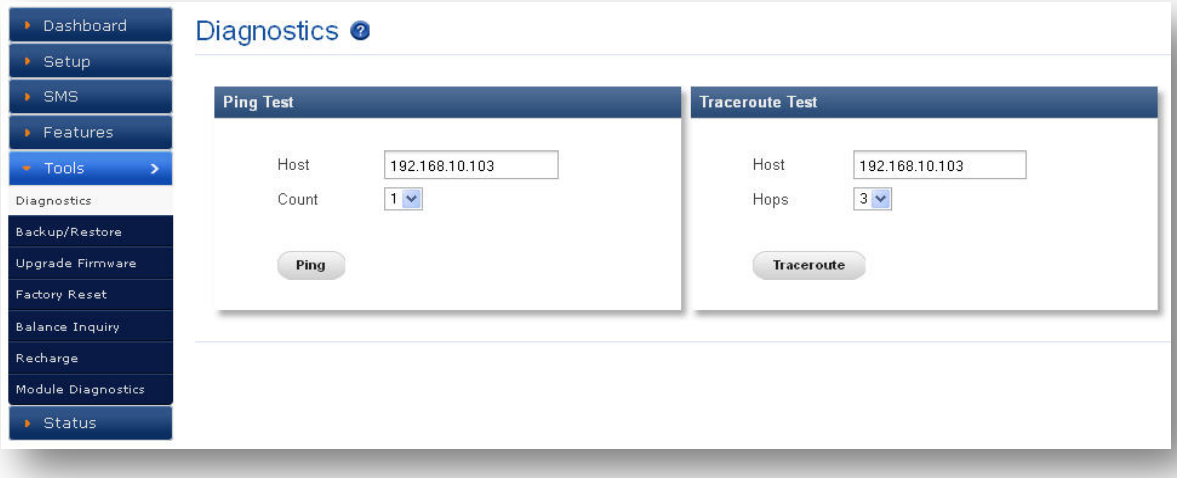


Figure 40: Diagnostics

Diagnostics Ping Result

It is used to check the packet loss and latency time from your SIP end client like IP Phone/ FXS gateways to check the quality of your network connections.

Enter the IP address of the IP Phone in your LAN and enter "PING" Button, it wills displays similar to like this:

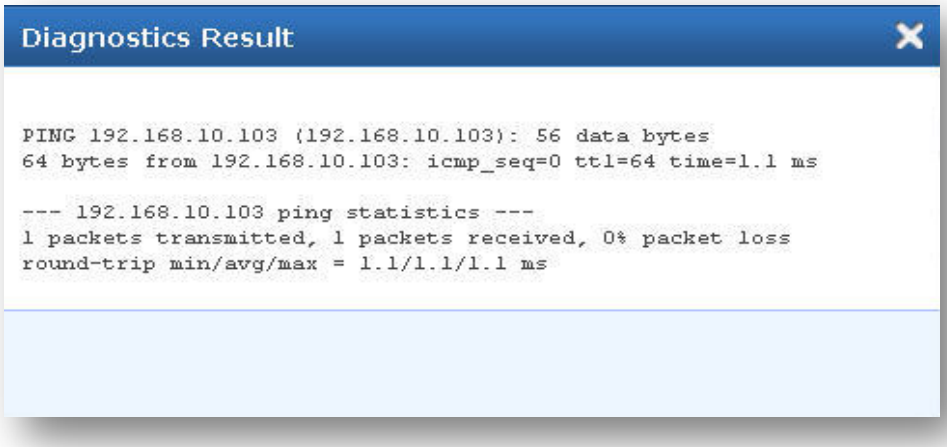


Figure 41: Diagnostics Result

Diagnostics Trace route Result- It is used to determine the route taken by packets across an IP network.

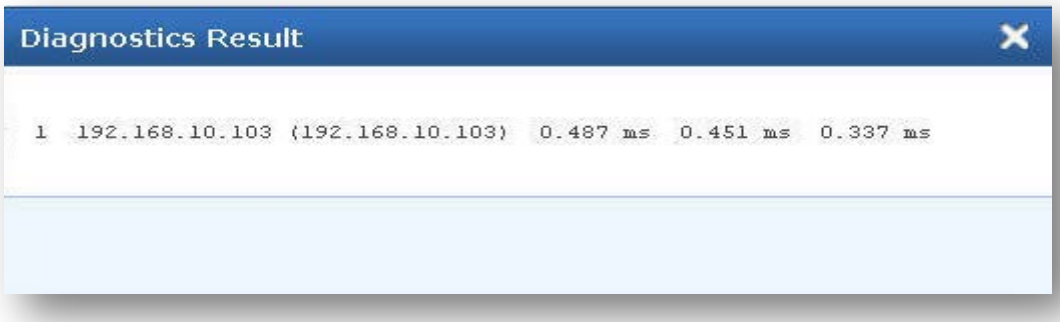


Figure 42: Diagnostics Trace Route Result

7.2 Backup/Restore

Navigate through **Tools** > **Backup/Restore**

Back Up:

Allow you to take the back up of the System configurations & save it to the local PC.

Restore:

Restoring from a new upload or backup file will destroy all current configurations and require a system reboot. All calls will be dropped and all current configurations will be destroyed.

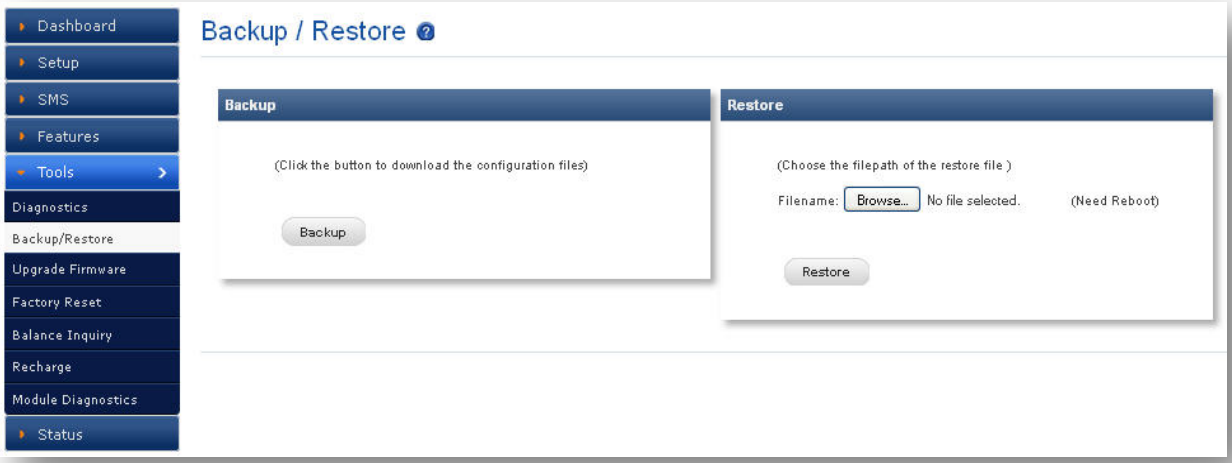


Figure 43: Backup/Restore

Administrator password will not be restored on restoration. So you should still use same credentials as before restoration

7.3 Upgrade Firmware

Navigate through **Tools** > **Upgrade Firmware**

The Firmware Upgrade page allows you to update the GSM Gateway with the latest release available, which can contain key updates, added functionalities and bug fixes. When a new release is available, download it and save to your local PC. Then, browse for the file, and click the Upload button. Now your GSM Gateway will display a Progress Screen and will prompt you when your GSM Gateway is about to reboot. Let your GSM Gateway reboot, and wait for the Blue LED's to come back on.

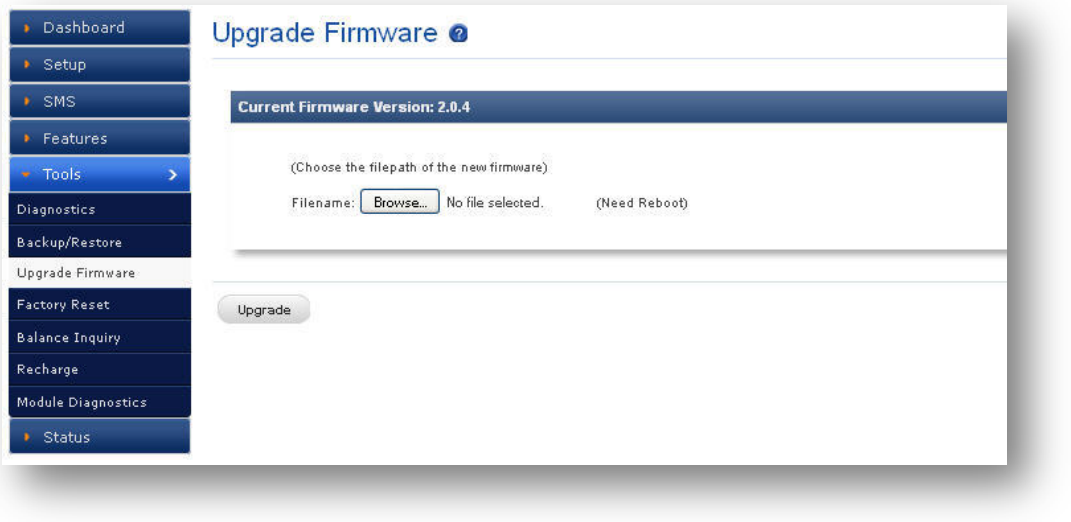


Figure 44: Upgrade Firmware

During firmware upgrade there should not be any power or network disturbances, which may leads to GSM Gateway board faulty. Firmware up-gradation process will take few minutes.

7.4 Factory Reset

Navigate through **Tools** > **Factory Reset**

This feature allows the system admin to erase the user configuration and the device will reset to the Factory Default Settings.

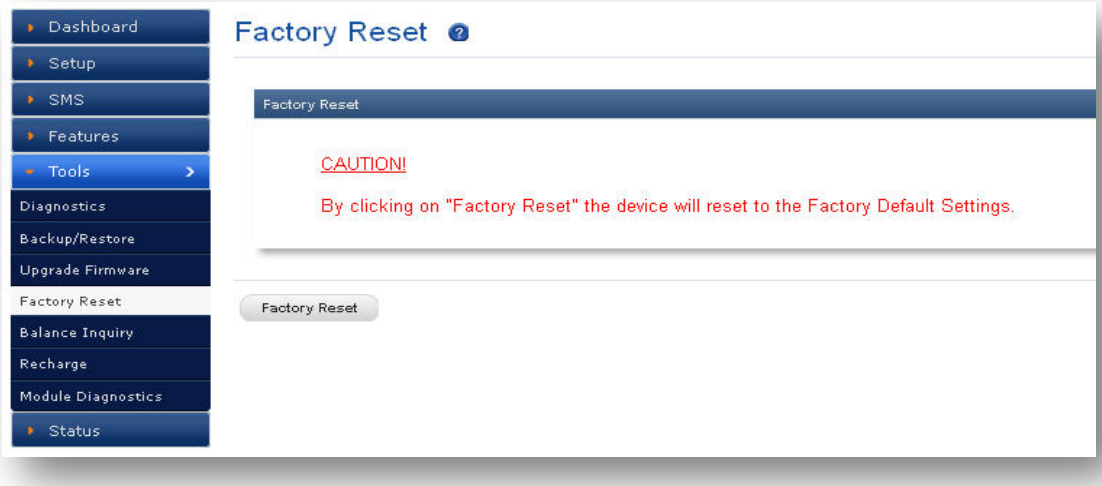


Figure 45: Factory Reset

Hardware Factory Reset will erase the call reports and software Factory Reset shall retain the previous CDR reports.

By clicking on Factory Reset, the device will reset to the Factory Default Settings.

7.5 Balance Inquiry

Navigate through **Tools** > **Balance Inquiry**

This feature allow system admin to get UPTO DATE information regarding the account balance in each SIM ,this information can also send to admin email address. Administrator can monitor the balance history using the Tab “Balance History”

The balance information can be forwarded to a mobile no by selecting (Checking) the option “” Send SMS for every Balance Inquiry “”

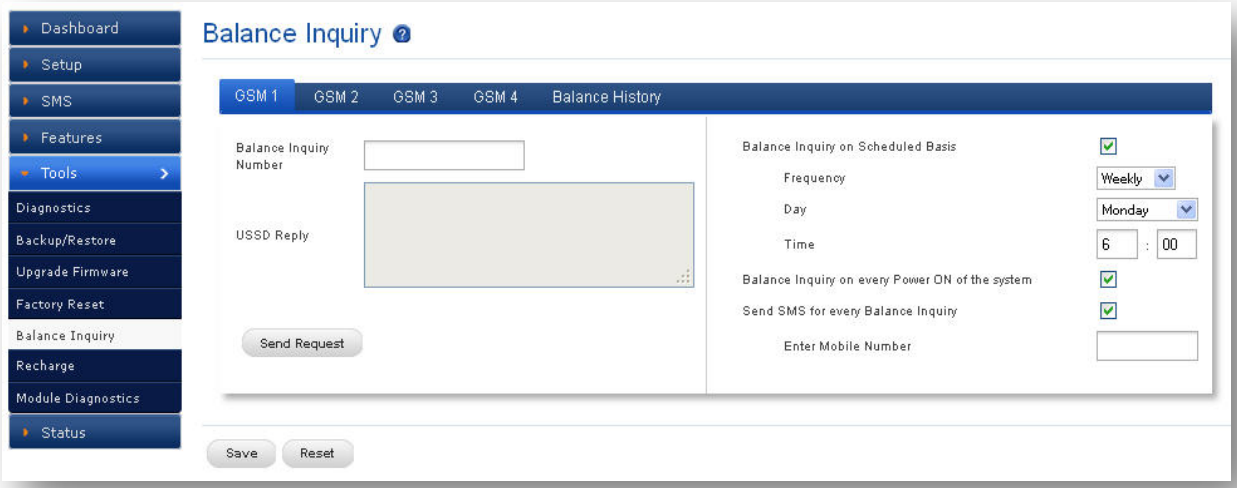


Figure 46: Balance Enquiry

Balance Inquiry Number	This is the number provided by GSM service provider to check balance ex : *123# or *141# …etc
USSD Reply	**Unstructured Supplementary Service Data** (USSD) is a protocol used by GSM cellular telephones to communicate with the service provider's computers. USSD messages are up to 182 alphanumeric characters in length.
Balance Inquiry on Scheduled Basis	Customer can configure the schedule for checking balance by selecting this option
Frequency	schedule for checking balance –Daily ,weekly, monthly
Day	schedule for checking balance in day (Mon- SUN)
Time	schedule for checking balance in time settings *Hours*: *Minutes* (*HH:MM*)
Balance Inquiry on every Power ON of the system	This option will check balance inquiry on every power ON
Send SMS for every Balance Inquiry	This option will send balance information to selected mobile no after each balance check
Enter Mobile Number	The mobile no to be configured to receive for forwarded balance information form the Gateway
Balance history	This will provide the history of balance check

7.6 Recharge

Navigate through **Tools** > **Recharge**

This option allow system admin to recharge the pre paid SIM card using the wed GUI

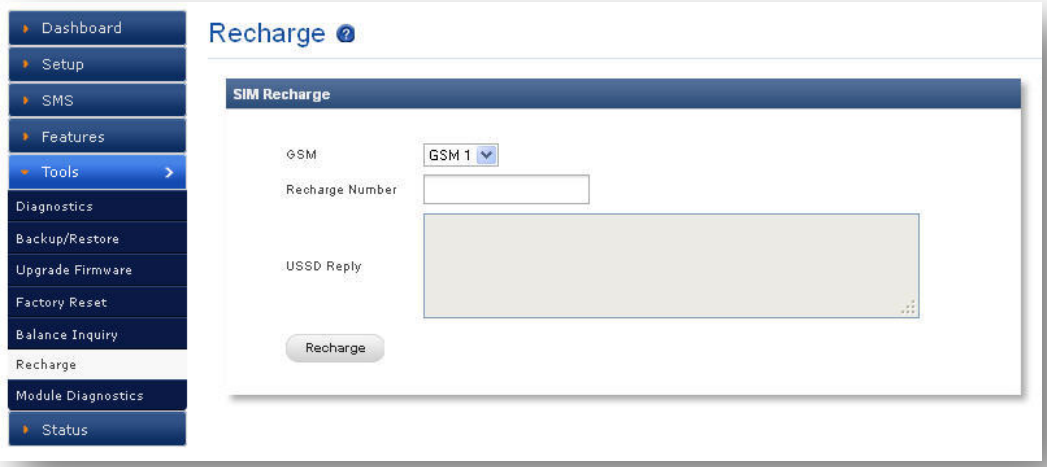


Figure 47: Recharge

7.7 Module Diagnostics

Navigate through **Tools** > **Module Diagnostics**

We have manual module resets here in case of failure.

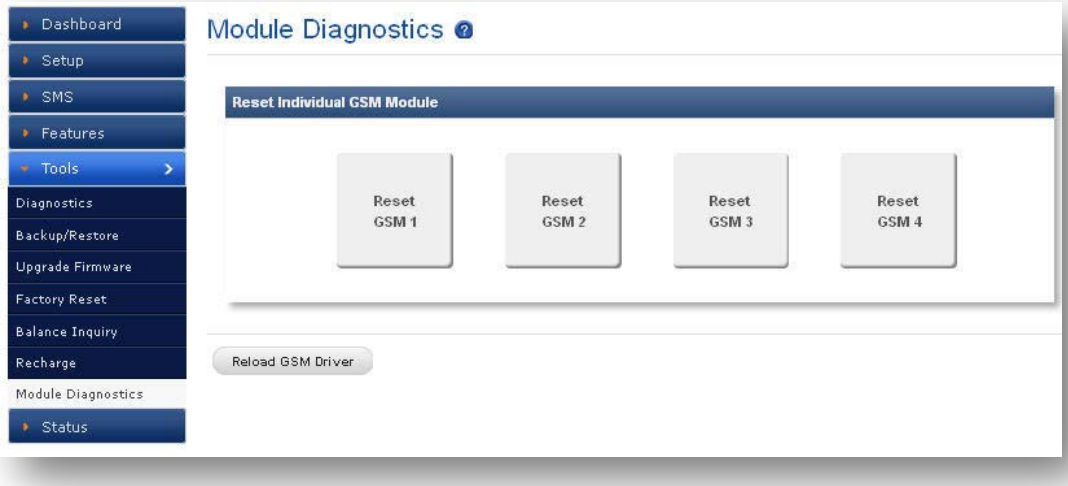


Figure 48: Module Diagnostics

8. Status

8.1 Call Reports

Navigate through **Status** > **Call Reports**

Call Reports displays a detailed list of calls pass through the GSM Gateway. The list can be generated on the bases of date range, CDR count, latest 50 entries or all entries. Generated report can also be exported to local PC as *CSV* file.

To create a new report select the Extension Range or Date range and click the "Generate" Report button. A list with call details will display in the Call Reports section. You can either export to your local PC or Print the Call reports.

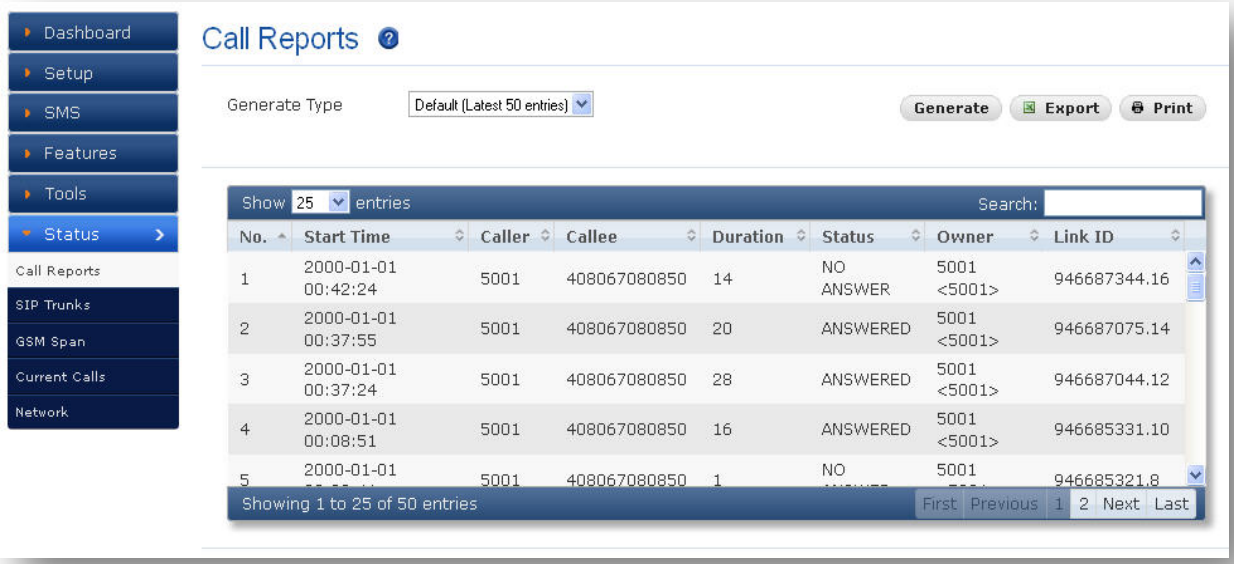


Figure 49: Call Reports

Generate Type	**Default (last 50 entries)**-Show the last 50 calls **Based on date range**- filter the calls according to the FROM and TO date entries **Based on CDR count** – Based on CDR entry serial no **All** – Display all
Show entries	How many entries in the single page
NO	Serial no of the CDR entry

Start Time	Stating time of the call
Caller	Who originates a call
Callee	The person who is called by the caller
Duration	The duration of the call
Status	**NO ANSWER** –If the call landing on the Gateway not answered **Answer** –If the call landing on the gateway answered **Busy** – If the call landing on the gateway rejected by the user
Owner	Caller ID of the caller
Link ID	Unique no generated by the Gateway
Generate	This button used when you sort CRD with generate type
Export	To export CDR in .csv format
Print	To Print CDR

8.2 SIP Trunk Status

Navigate through **Status** > **SIP Trunks**

SIP Trunk Status page displays detailed status of each SIP trunks available on the GSM Gateway.

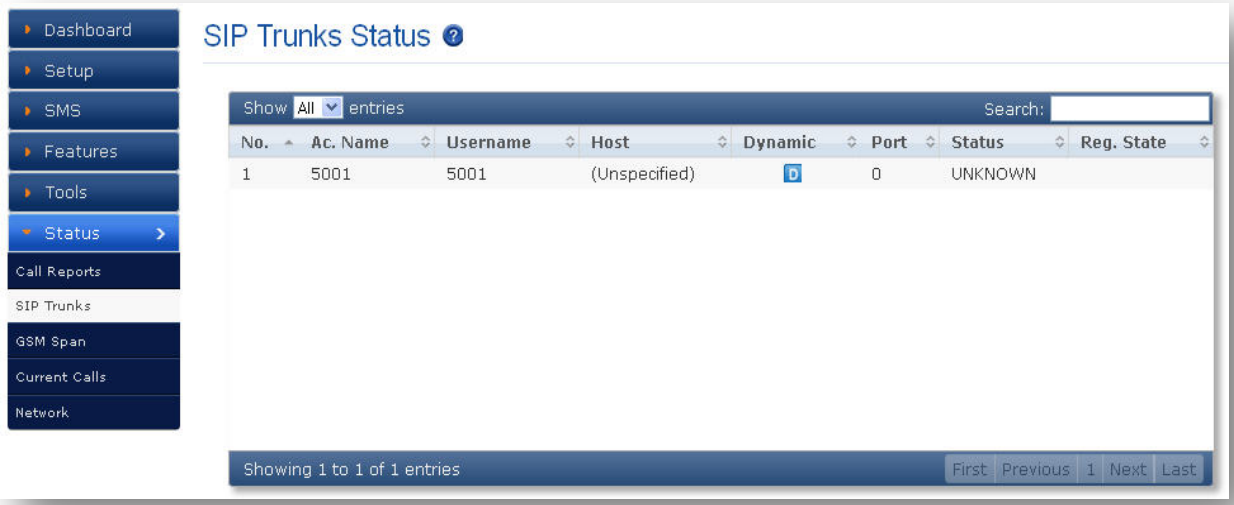


Figure 50: SIP Trunk Status

Status	Reg. State	Description	
OK	Registered	Configured, Registered & reachable	-
OK	-	Configured & Reachable, but no Registration	-
OK	Request Sent	Configured, but Host not responding or unreachable	Check Registrar Address
OK	Rejected	Configured & reachable, but Registration failure	Check Authentication
UNREACHABLE	Registered	Configured, Registered, but not reachable	Check Proxy Address
UNREACHABLE	-	Configured, but not reachable	Check Proxy Address
UNKNOWN	-	Not Registered	Client not registered

Dynamic: Host IP is obtained dynamically on registration.

8.3 GSM Span Status

Navigate through **Status** > **GSM Span**

GSM Span Status page displays detailed status of each span name, Signal Strength, Service Provider info and channel status available on the GSM Gateway.

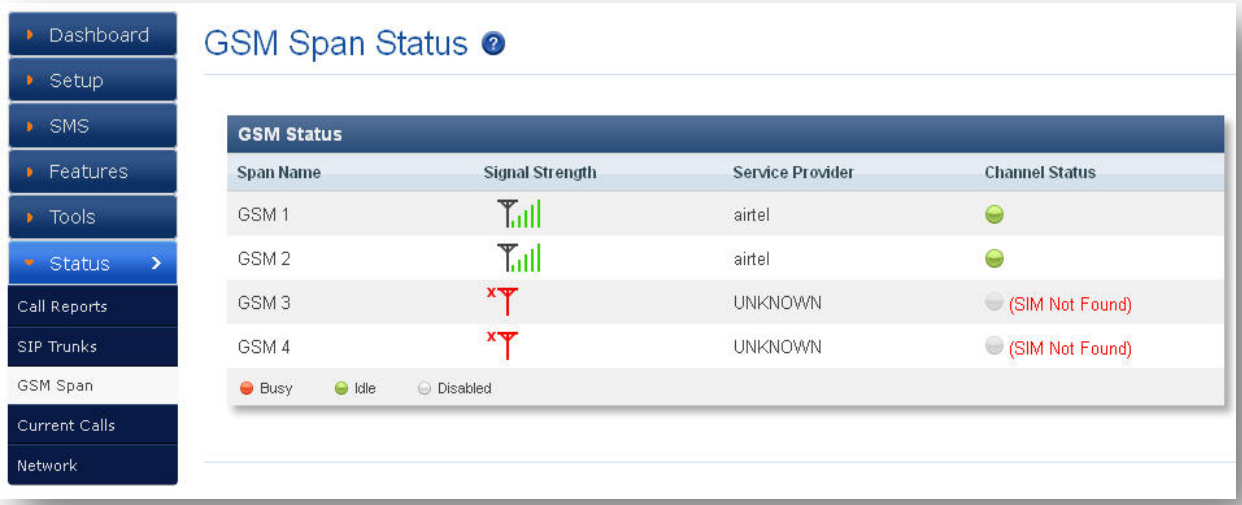


Figure 51: GSM Span Status

Channel Status	Description
	Channel is Busy
	Channel is Idle and ready to receive or make calls
	Channel is not active(SIM not found)

8.4 Current Calls

Navigate through **Status** > **Current Calls**

Current Calls page displays detailed status of the real time calls available on GSM Gateway.

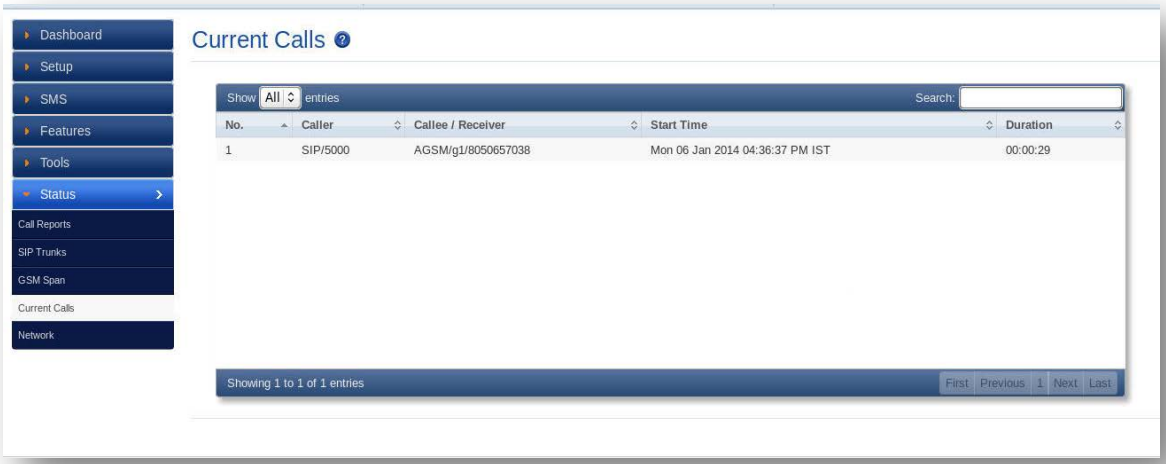


Figure 52: Current Calls

8.5 Network Status

Navigate through **Status** > **Network**

Network Status page displays detailed status of the network configuration on GSM Gateway.

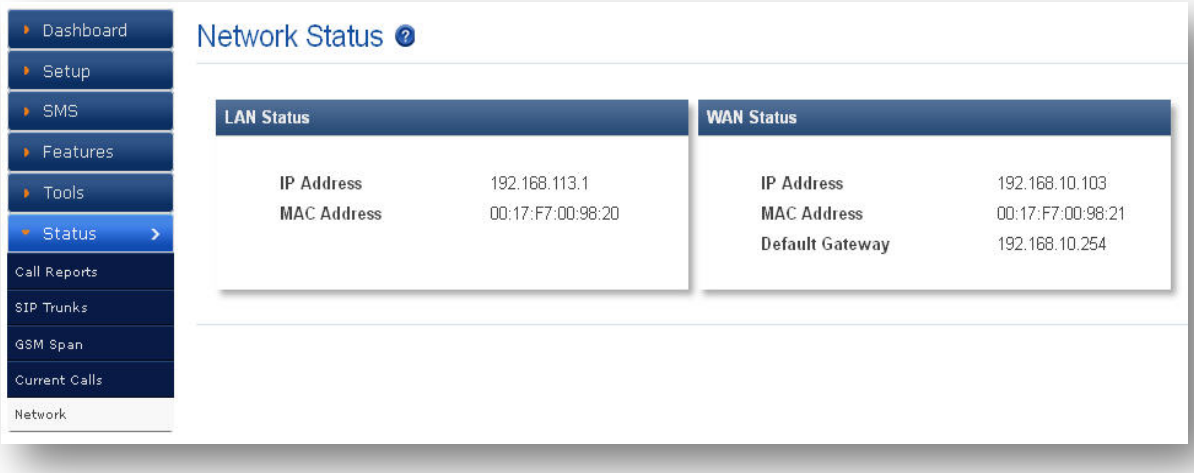


Figure 53: Network Status

9. Administrator

9.1 Reboot

Refer Below screen shot for Reboot, WEB settings, Email Settings and Log out

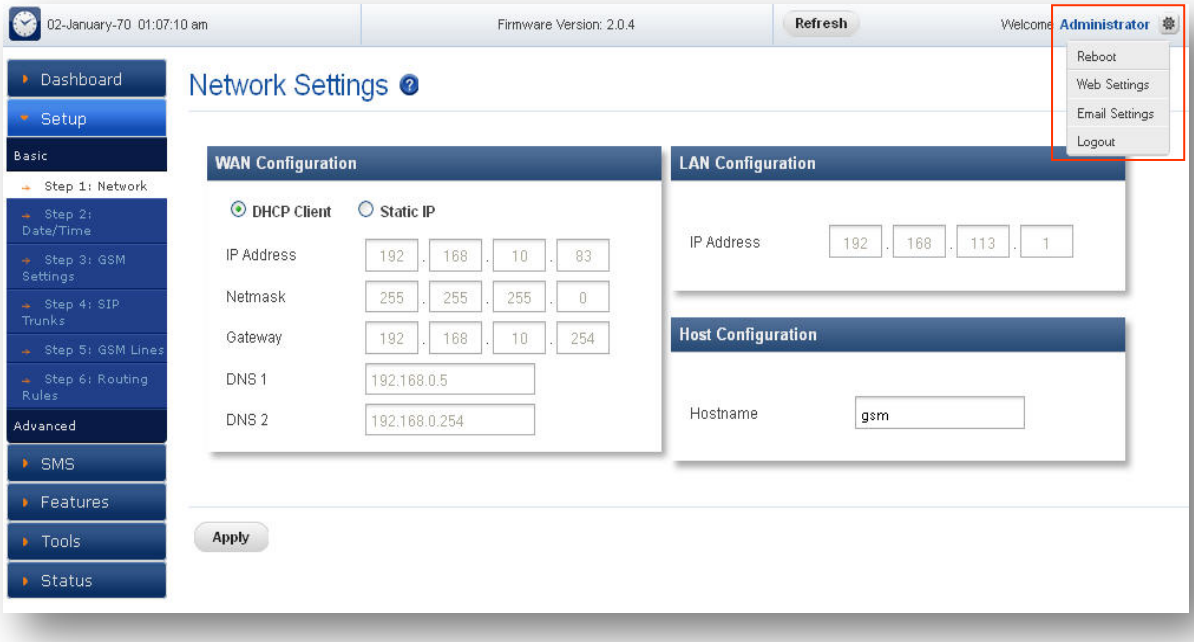


Figure 54: Reboot

Reboot option will help customers / Technicians to reboot the device from GUI

9.2 Web Settings

Navigate through **Administrator** > **Web Settings**

Session Timeout	Duration after which current web login session expires. Default: 3600 sec
Pagination	Number of entries in a table per page to be displayed.
Change Password	Modify Administrator password here.

9.3 Email settings

Navigate through **Administrator** > **Email Settings**

Mail Server	The IP Address or domain name of the SMTP server
Email-ID	Specify the Administrator Email ID.
Username	Authorized username of the Admin Email ID
Password	Authorized password of the Admin Email ID

TLS Support	TLS setting to require mail to be transmitted via a secure connection when users correspond with specific domains and email addresses.

10. Appendix

10.1 SMS Sending and Receiving Options in Allo's GSM PCI cards for Asterisk

1. Sending SMS

There are two modes for sending SMS such as PDU mode and text mode. PDU mode supports a few languages' characters for sending. Text mode currently supports only English characters.

To send SMS, chan_allogsm provides three methods:

i. Sending SMS on the Asterisk console (only supports English characters).

Usage: allogsm send sms <destination number> <sms contents>

E.g. **allogsm send sms 1 135xxxxxxxx "Hello World!"**

ii. Sending SMS over Linux Shell.

Usage: asterisk - rx "allogsm send sms <destination number> <sms contents>"

E.g. **asterisk – rx "allogsm send sms 1 135xxxxxxxx \"Hello World! \""**

iii. Send SMS over dialplan.

Usage: SendSMS (, destination number, sms content)

E.g. **SendSMS (1,135xxxxxxxx, "Hello World!")**

2. Receiving SMS

The sms will be saved in /var/log/asterisk/sms/receive_message when system receives. At this moment, it will trigger corresponding contexts and sms priorities over dialplan.

E.g. **exten =>sms, 1, xxxxxxxxxxx** (must use sms so that trigger sms receiving). It might be different for each span

3. Additional Variable settings in Dialplan.

Sending Side

i. Set (CHAR_CODING=encoding); set up encoding mode of sending sms. Currently supported GSM for text mode and HEX for PDU.

E.g. Set (CHAR_CODING=GSM)

ii. Set (SMSC=sms centre number); set up centre number of sms.

E.g. Set (SMSC=861380755500)

Receiving Side

i. ${SMSSRC}; sms sender number.

ii. ${SMSTXT}; sms contents.

iii. ${SMSPDU}; PDU codes of sms.

Thank you for choosing

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