YX

GoIP User Manual

CONTENT

1 Introduction

1.1 Overview

A VoIP GSM Gateway (GoIP Gateway) is a device which reduces costs when calling from a fixed telephone line to mobile network. It enables direct routing between IP, digital, analog and mobile networks.

GoIP Gateway is now used more and more for telephone carriers to land their IP calls to mobile network. In those areas where fixed line services are unavailable or much more expensive than the mobile cost, GoIP Gateway is an irreplaceable alternative.

The following figure shows a basic topology of GoIP Gateway usage.

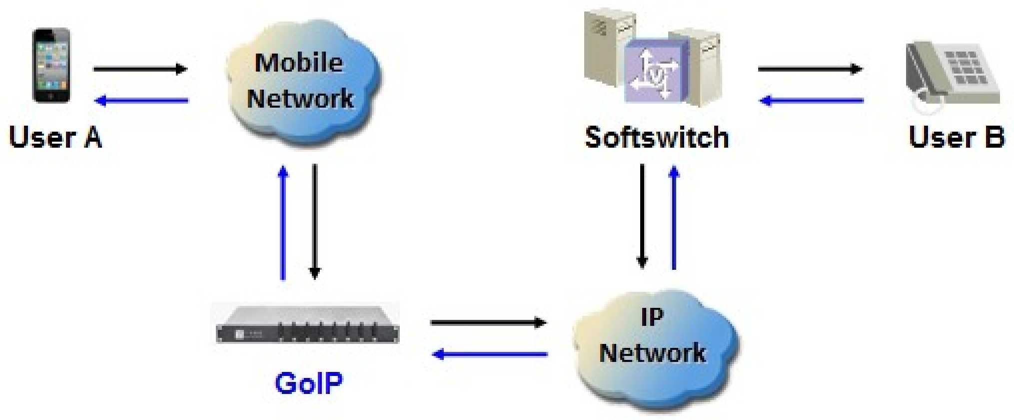


Figure 1.1

1.2 Glossary

VoIP: Voice over Internet Protocol.

SIP: Session Initial Protocol.

DTMF: Dual Tone Multiple Frequency.

IMEI: International Mobile Equipment Identity.

LCR: Least Cost Routing.

USSD: Unstructured Supplementary Service Data.

GSM: Global System Communications.

CDMA: Code Division Multiple Access.

WCDMA: Wideband Code Division Multiple Access.

2 Equipment Information

2.1 Appearance

The following figure shows the front view of YX GoIP Gateway.

Figure 2.1: Front View

The following figure shows the rear view of YX GoIP Gateway.

Figure 2.2: Rear View

2.2 Hardware

The following table shows the hardware information for YX GoIP Gateway.

CPU	KSZ 8695
Memory	

Media DSP	
Power Supply	100-240V AC, 50 ~ 60 Hz
Module:	GSM/CDMA/WCDMA serials
LAN/WAN	
Serial Port	
Antenna	
Card Slot	
Dimensions	● Rack mountable 1U chassis(compatible with 19’’ Rack) ● Width: 482mm ● Height: 44mm ● Depth: 210mm
Weight	2.5 kg
Working Environment	● Temperature: 0 ~ 50 ℃ ● Humidity: 10% ~ 90%

2.3 Software

The following table shows the software information for YX GoIP Gateway.

OS	Embedded Linux OS
Web Server	Built-in Http Server
Firmware	
SIP Client	
DHCP Client	
DHCP Server	
PPPoE	

2.4 Function and Features

This chapter introduces the overall function and features of YX GoIP Gateway.

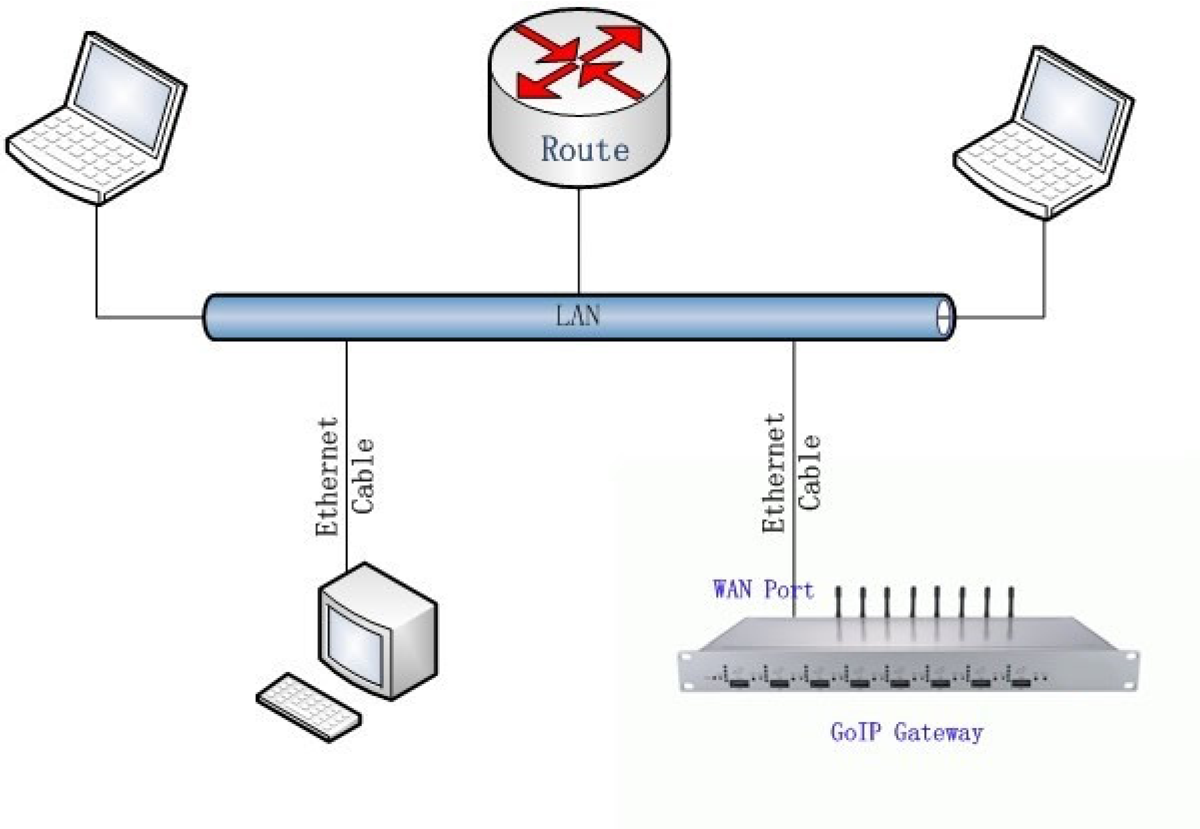
- Hot-line call
- Dial pattern
- Dial prefix manipulation
- Phone book
- CDR
- White list

3 Equipment Installation

This chapter describes how to install a new GoIP Gateway to a physical network environment, how to initialize it and start it in a proper way.

3.1 Network Setup

Network is a prerequisite to install GoIP Gateway. The following figure shows the topology of LAN with a VoIP Gateway connected.



Note: WAN Port will be used to connect GoIP Gateway to the LAN.

3.2 Equipment IP Address

The default IP of GoIP Gateway WAN port is 192.168.1.67, while the default LAN port IP is 192.167.1.1.

3.3 Equipment Connection

Follow the steps below to install the GoIP Gateway to LAN.

1) Fix the antenna to the GoIP Gateway. (Optional)

2) Insert SIM card(s) to slots.
3) Connect an Ethernet Cable to the WAN port of GoIP Gateway. The other end of the Ethernet Cable should be connected to LAN route or switch.
4) Connect an Ethernet Cable to the LAN port of GoIP Gateway. The other end of the Ethernet Cable should be connected to PC or other network device. (Optional)
5) Plug in the GoIP Gateway.

3.4 LED Indicators

There are a set of LED lights in the front of GoIP Gateway. Lights will be on or glittering when the GoIP Gateway is power on and running. The following table describes various meanings of status corresponding to LED lights in different display color.

Power	It indicates whether the system is running or not.
Run x	
SW x	

4 Web Settings

This chapter describes how to set up GoIP Gateway through Web Page. There is a built-in web server which can be accessed at URL: http://GATEWAY_IP/, while GATEWAY_IP is the WAN IP address of the GoIP Gateway, such as 192.168.1.67.

As an example, the following introduction will base on the GoIP Gateway with WAN IP 192.168.1.67.

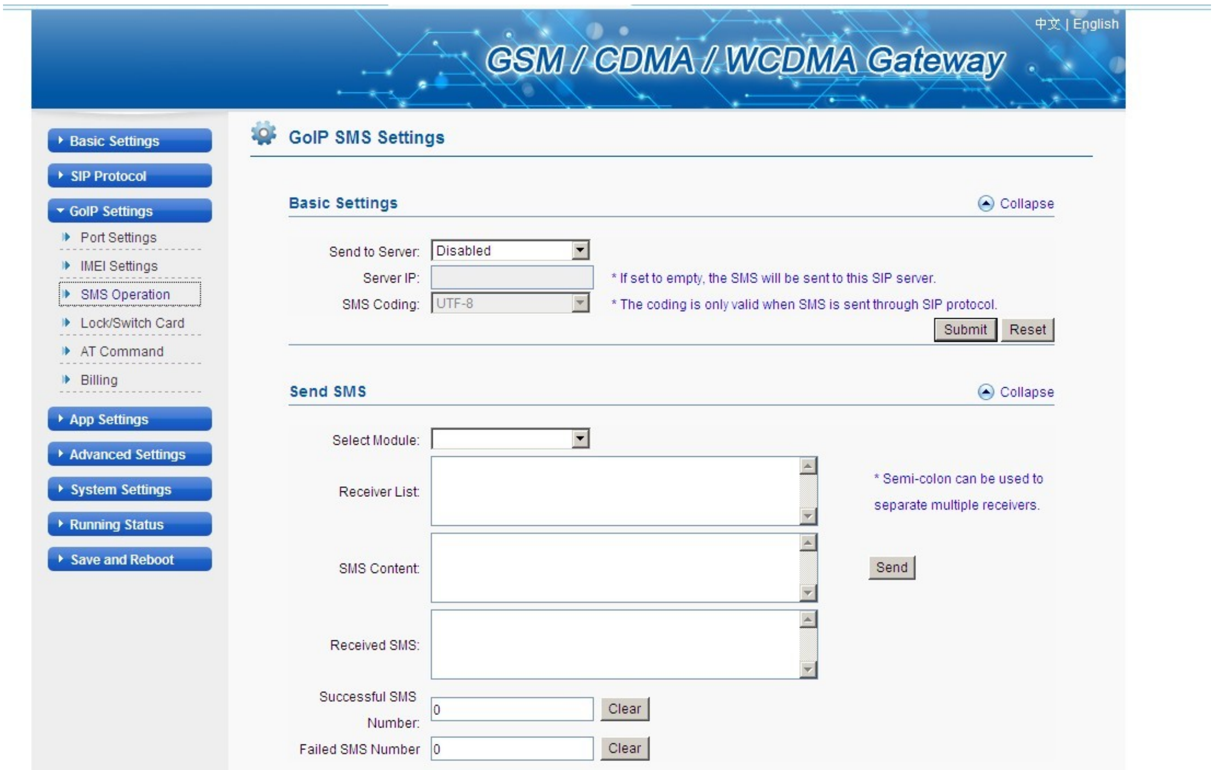
4.1 Login

Open web browser and access URL http://192.168.1.67/. The default login page will be displayed as following.

The default login account and password are:

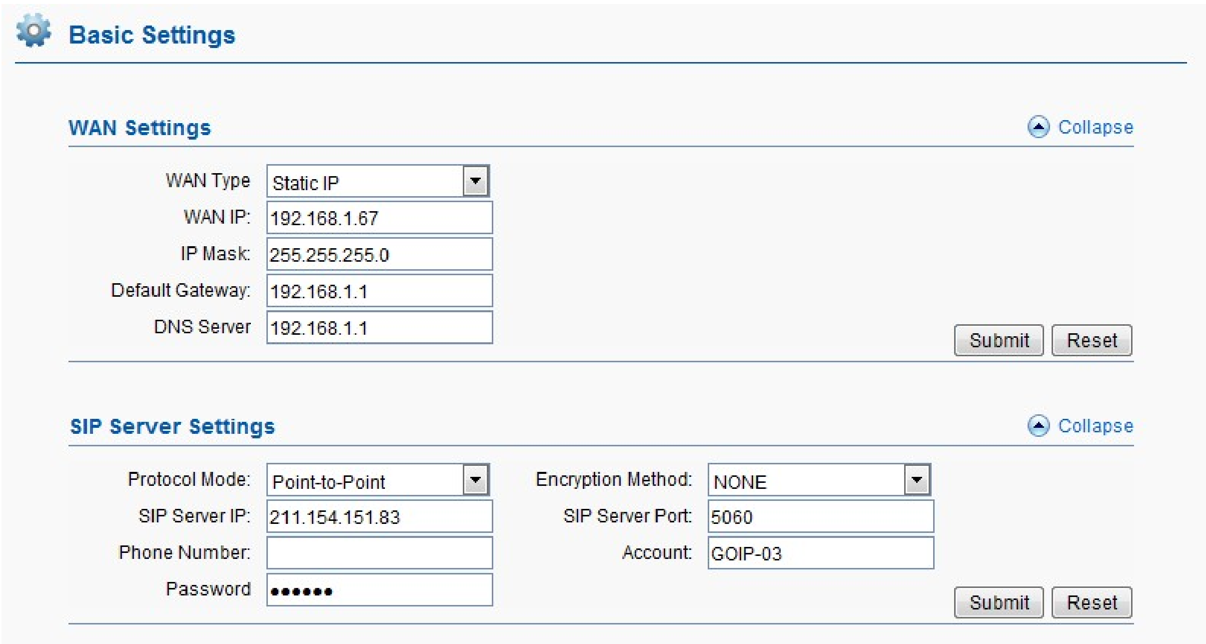
Account	root
Password	root

It is recommended to use IE or FireFox to access the web pages. After successfully logged in, the main page to set Gateway is as following:



4.2 Basic Settings

Basic Settings will be described in this paragraph. The most frequently modified parameters and most of the individual parameters are listed in this page.



SIP Server Settings is for SIP communication with IP network. Fields are specified as following:

➢ Protocol Mode: Specify SIP client working mode. Option values are

Registration/Point-to-Point. If set to Registration, SIP client will send registration messages to SIP server.

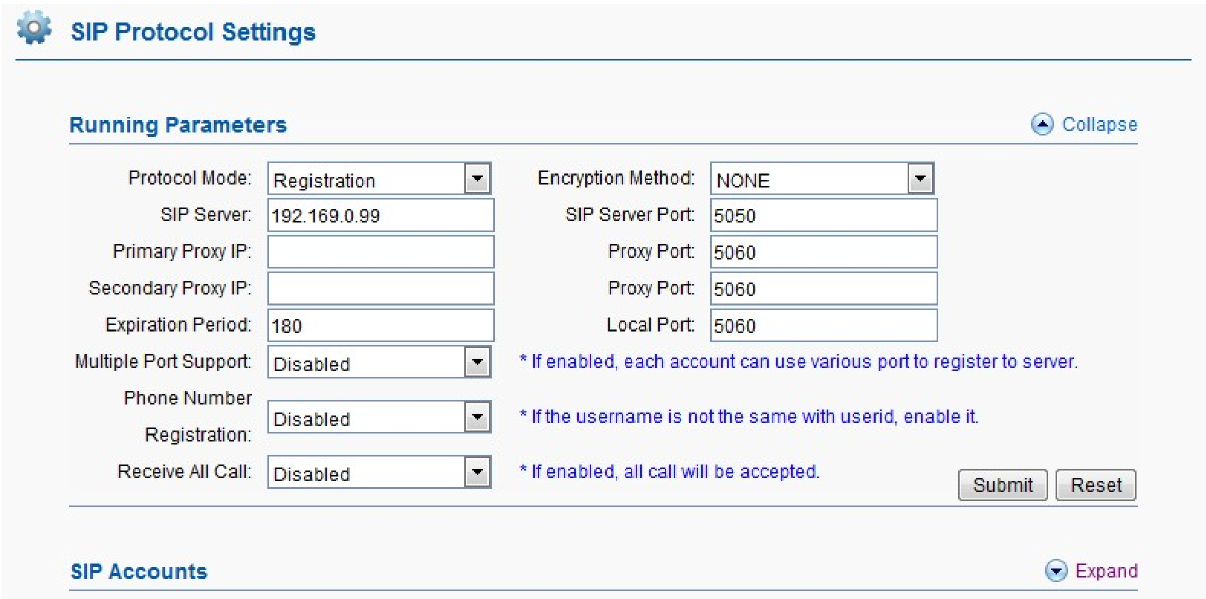
- Encryption Method: Specify the encryption method for messages between SIP server and SIP client.
- SIP Server IP: Specify the IP of SIP server.
- SIP Server Port: Specify the port of SIP server.
- Phone Number: Specify the caller phone number for SIP client. It can also be regarded as the SIP port number which can be called.
- Account: Specify the SIP account for registration.
- Password: Specify the SIP password for registration.

Note: The settings of Phone Number, Account and Password are globally active. They will apply to all SIP ports settings in SIP Protocol page.

4.3 SIP Protocol

SIP Protocol Settings will be described in this paragraph. It mainly targets to set up parameters related to SIP server, SIP account and SIP password for SIP registration.

The screenshot below shows the operation mode to set SIP running parameters.

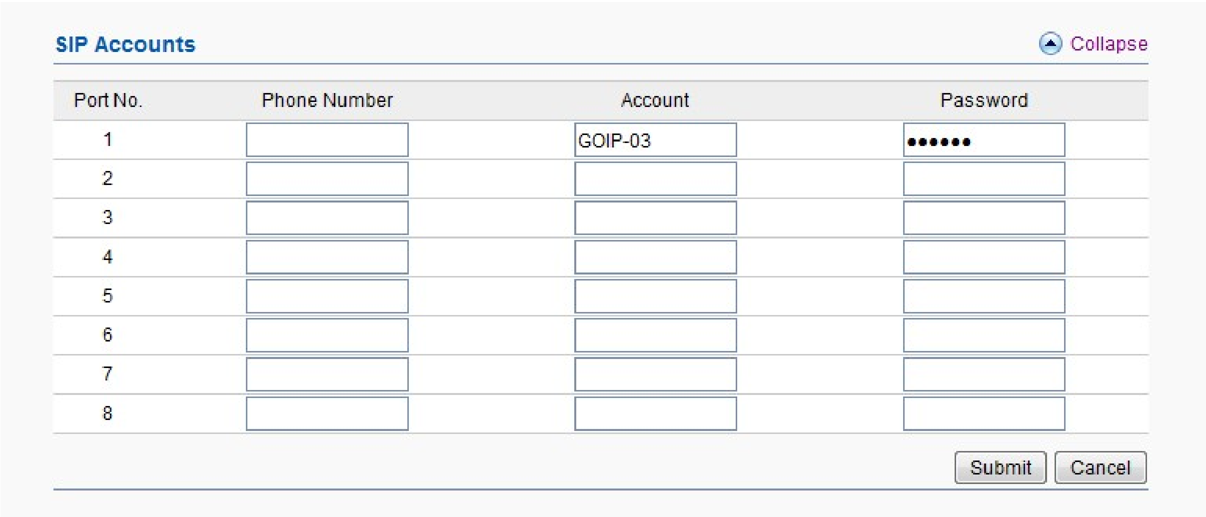


Fields are specified as following:

- Protocol Mode: It is the same as that in Basic Settings. The modification here will also apply to Basic Settings page.
- Encryption Mode: It is the same as that in Basic Settings. The modification here will also apply to Basic Settings page.
- SIP Server: Specify the domain or IP of the SIP Server.
- SIP Server Port: Specify the port of the SIP Server.

- Primary Proxy IP: Specify the primary proxy IP.
- Secondary Proxy IP: Specify the secondary proxy IP.
- Expiration Period: Specify the expiration period for registration.
- Local Port: Specify the local port used to register to SIP server.
- Multiple Port Support: Specify whether support to register to SIP server with different SIP Account.
- Phone Number Registration: Specify whether enable phone number registration or not. If set to *Enabled*, the *Phone Number* and *Account* can be different when registering to SIP Server. If set to *Disabled*, the *Phone Number* must be the same as the *Account* when registering, otherwise, the registration will fail.
- Receive All Call: Specify whether enable to receive all calls or not.

The screenshot below shows the operation mode to SIP Accounts.



If a default *Phone Number, Account and Password* is already set in Basic Settings page, all the inputs here can be left empty and the default setting will apply to all the SIP accounts. For example, if an account *GOIP-03* is set with password *888888* in Basic Settings page and all the SIP accounts here are left empty, the combination of *GOIP-03* and *888888* will be used for all SIP accounts to register to SIP Server.

There are two ways to overwrite the default SIP account setting in Basic Settings page.

1) Assign an account and password to Port 1 and leave empty for other Ports. Settings of Port 1 can also be applied to all the other Ports automatically. This is for short to set Account and Password if all the Ports use the same value.
2) Assign account and password to certain Port(s). The value of account and password can be various. In this scenario, only the Ports assigned with account and password will register to SIP Server.

4.4 GoIP Settings

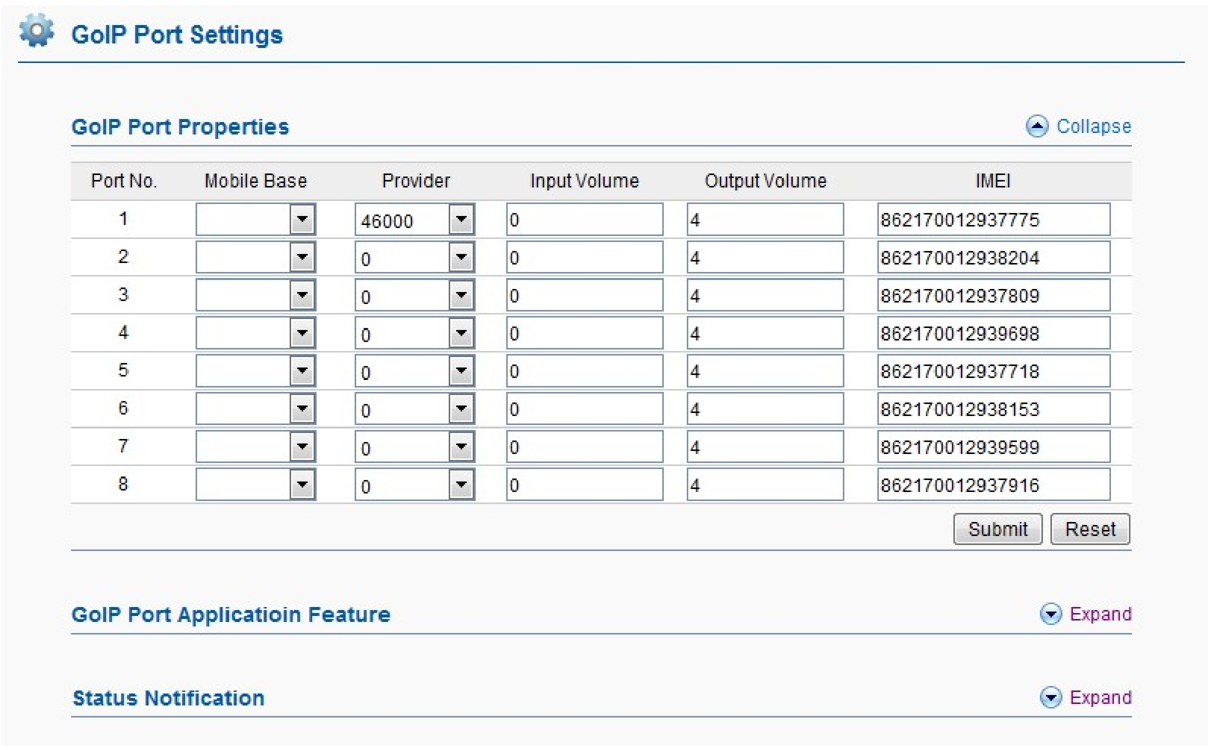
GoIP settings include:

- Port Settings
- IMEI Settings
- SMS Operation
- Lock/Switch Card
- AT Command
- Billing

These sub topics will be introduced separately below.

4.4.1 Port Settings

The screenshot below shows the operation mode to set GoIP port properties.

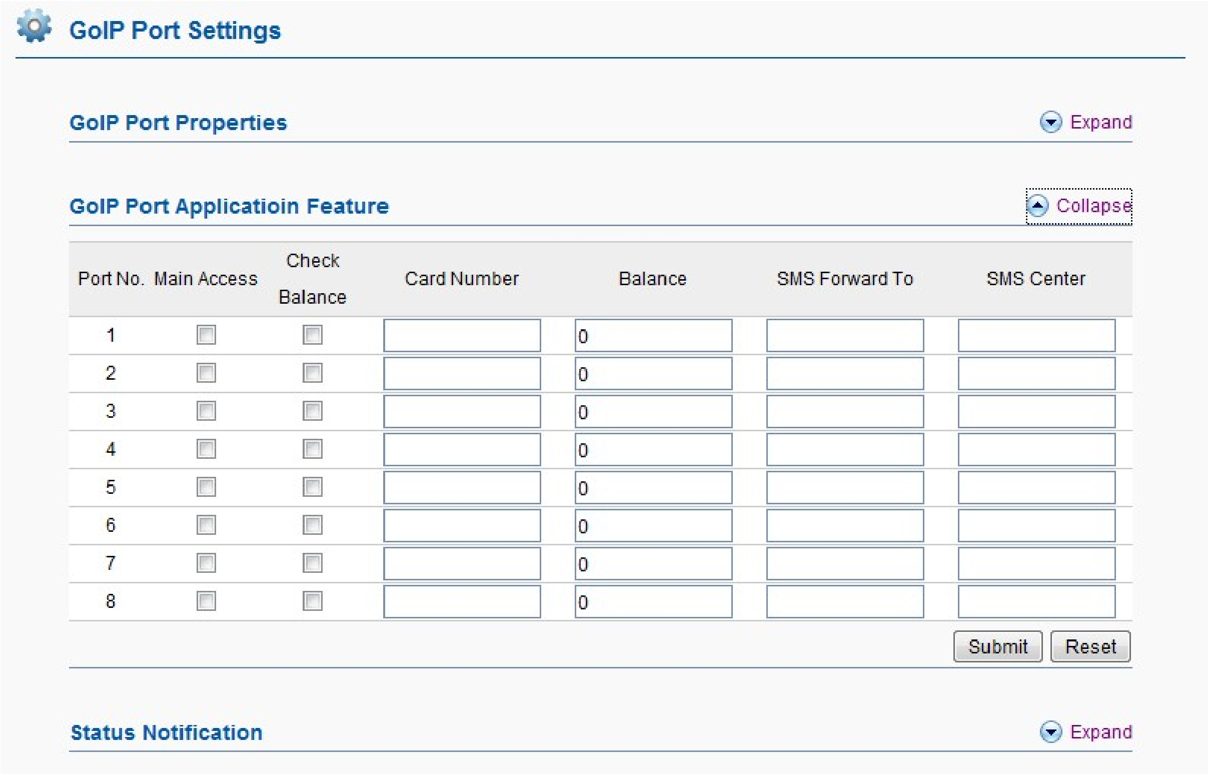


The columns are specified as following:

- Port No: The GoIP Gateway mobile port. Each port contains one or more card slots. Port No starts from 1 to 8.
- Mobile Base: Specify the mobile base.
- Provider: Specify the provider.
- Input Volume: Specify the input voice volume of this port.
- Output Volume: Specify the output voice volume of this port.

- IMEI: Specify the IMEI of this port. Any card in this port will use this specified IMEI to communicate with mobile base.

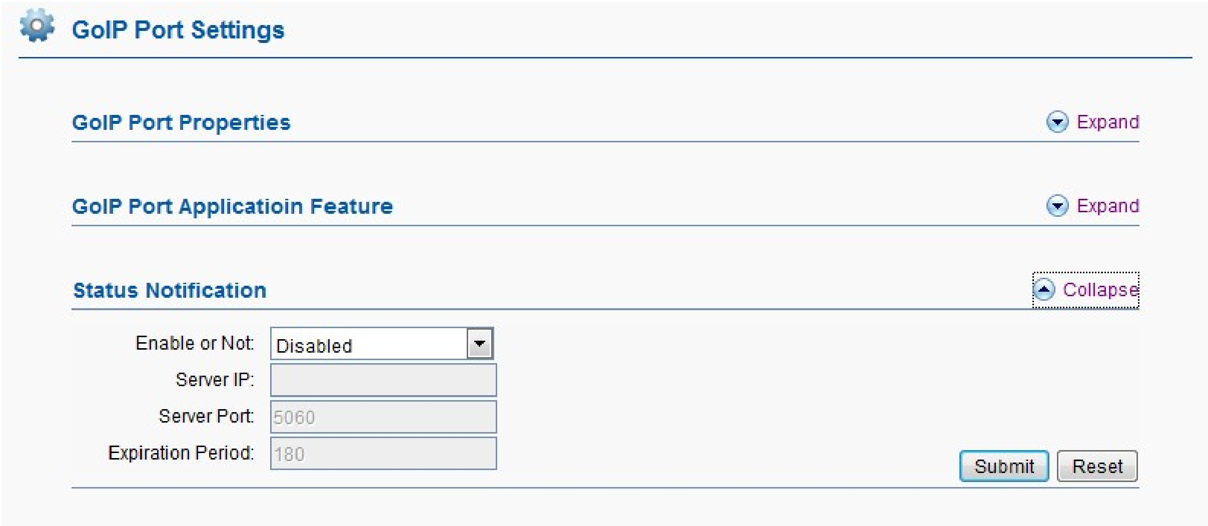
The screenshot below shows the operation mode to set GoIP port application feature.



The columns are specified as following:

- Port No: The GoIP Gateway mobile port. Each port contains one or more card slots. Port No starts from 1 to 8.
- Main Access: Specify whether the current port is used for access or not. If set to Enable, any call made to the card in this port will be redirected to the phone number on any other idle port which is not set as a main access. The any other idle port may be either on the same GoIP Gateway or on another GoIP Gateway which keeps reporting its port status to a Notification Server.
- Check Balance: Specify whether need to check card balance.
- Card Number: Specify the card number.
- Balance: Specify the balance of the card.
- SMS Forward To: Specify a receiver to which the SMS, which is sent to the card on this port, will be forwarded. The country code must be prefixed, for example, 8613512345678, while the string 86 stands for the country code of China and 13512345678 is the China Mainland mobile phone number.
- SMS Center: Specify the code of SMS Center.

The screenshot below shows the operation mode to set GoIP port notification feature.

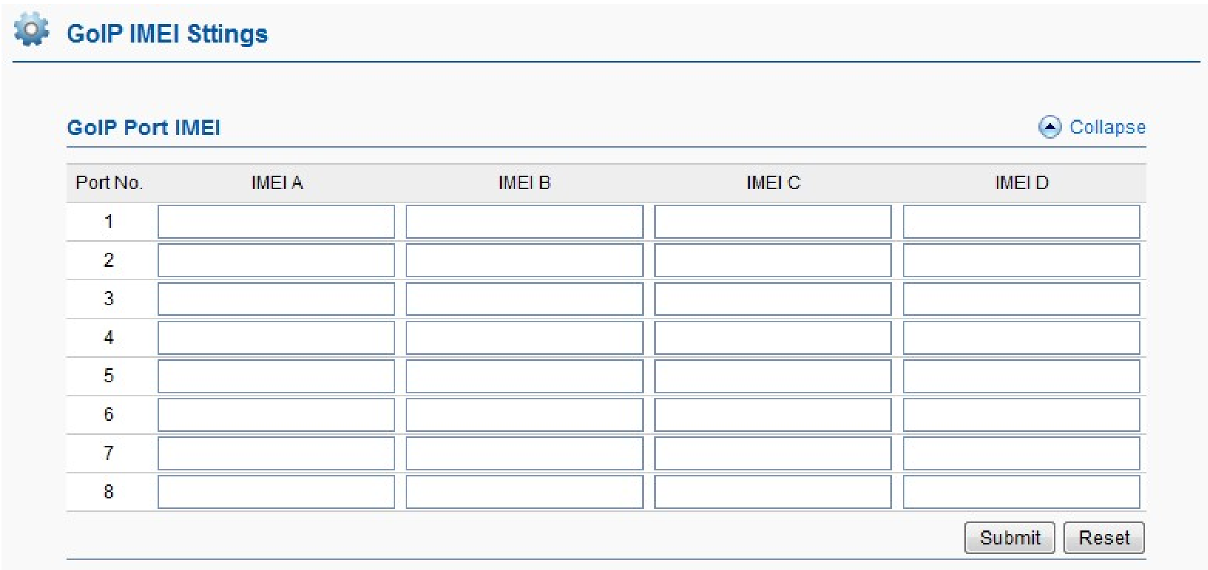


Fields are specified as following:

- Enable or Not: Specify whether enable status notification or not.
- Server IP: Specify the IP of server to which the notification is sent to.
- Server Port: Specify the port of server to which the notification is sent to.
- Expiration Period: Specify the expiration period to send status notification message.

4.4.2 IMEI Settings

The screenshot below shows the operation mode to set IMEI for each card inserted in GoIP Gateway port.



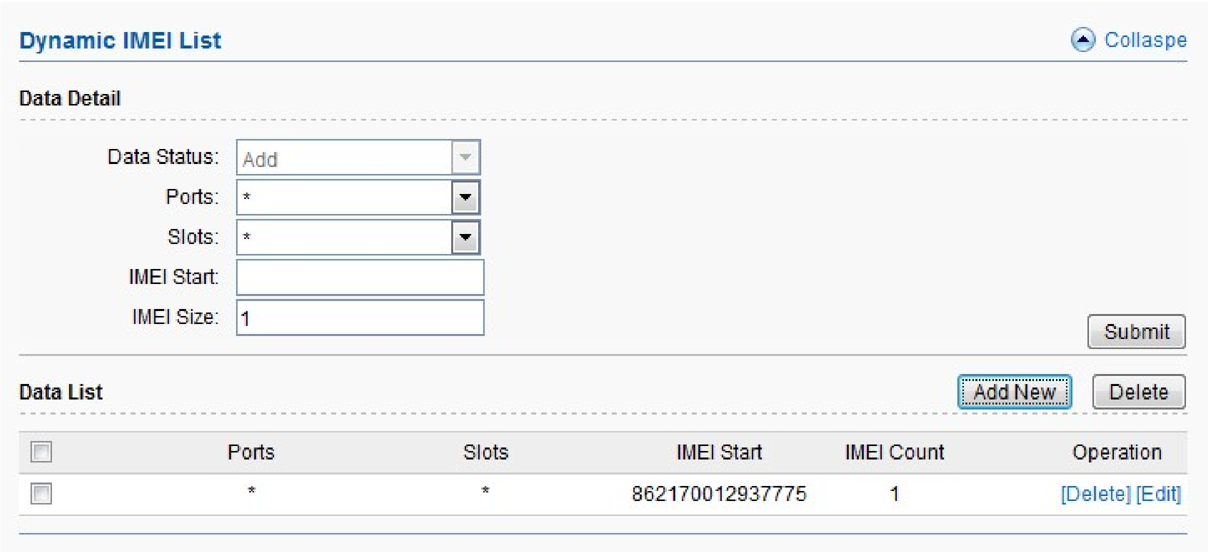
Fields are specified as following:

- Port No: The GoIP Gateway mobile port. Each port contains one or more card slots. Port No starts from 1 to 8.

- IMEI A: Specify the IMEI for card A of the port.
- IMEI B: Specify the IMEI for card B of the port.
- IMEI C: Specify the IMEI for card C of the port.
- IMEI D: Specify the IMEI for card D of the port.

The specified IMEI, instead of the default IMEI of the card, will be used for the corresponding card to communicate with mobile base.

The screenshot below shows the operation mode to set Dynamic IMEI for each card of the designated port. A group of IMEIs which are increased by 1 in numeric sequence can be set for one or all card(s) on one or all GoIP Gateway mobile port(s). If a card on a port is assigned with a group of IMEIs, it will randomly use any of the IMEI in group to communicate with mobile base.



Add New

Click button *Add New* to expand the data input area to add new data. Fields are specified as following:

- Data status: Mark the status of current data record. Option values are *Add/Edit*. Value *Add* means the data is new while value *Edit* means the data is old.
- Ports: Specify the port(s) on which the IMEI is added or modified. Option value * means that the IMEI applies to all ports.
- Slots: Specify the card(s) on which the IMEI is added or modified. Option value * means that the IMEI applies to all cards inserted in the selected ports specified by *Ports*.
- IMEI Start: Specify an initial IMEI value for the IMEI group.
- IMEI Size: Specify the size of the IMEI group.

Click button *Submit* on the right to save the new data record.

Edit

All the records are displayed in list. Two operations are provided on the right of each record. Click *Edit* to expand the current data record to Data Detail Area which is above the Data List.

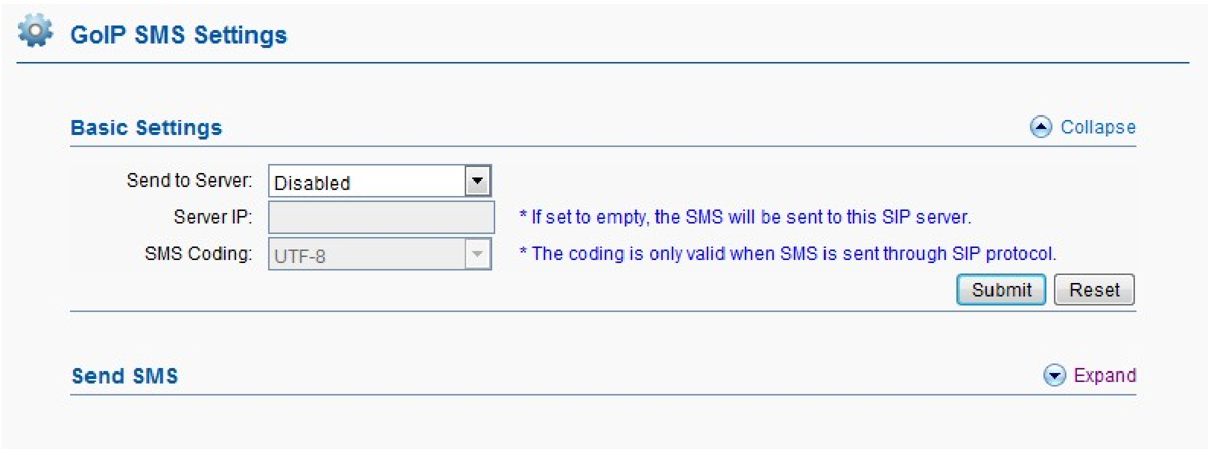
Click button *Submit* on the right to save the old data record.

Delete

Click *Delete* on the right of each record to delete the current record. A message box will be popped for delete confirmation.

4.4.3 SMS Operation

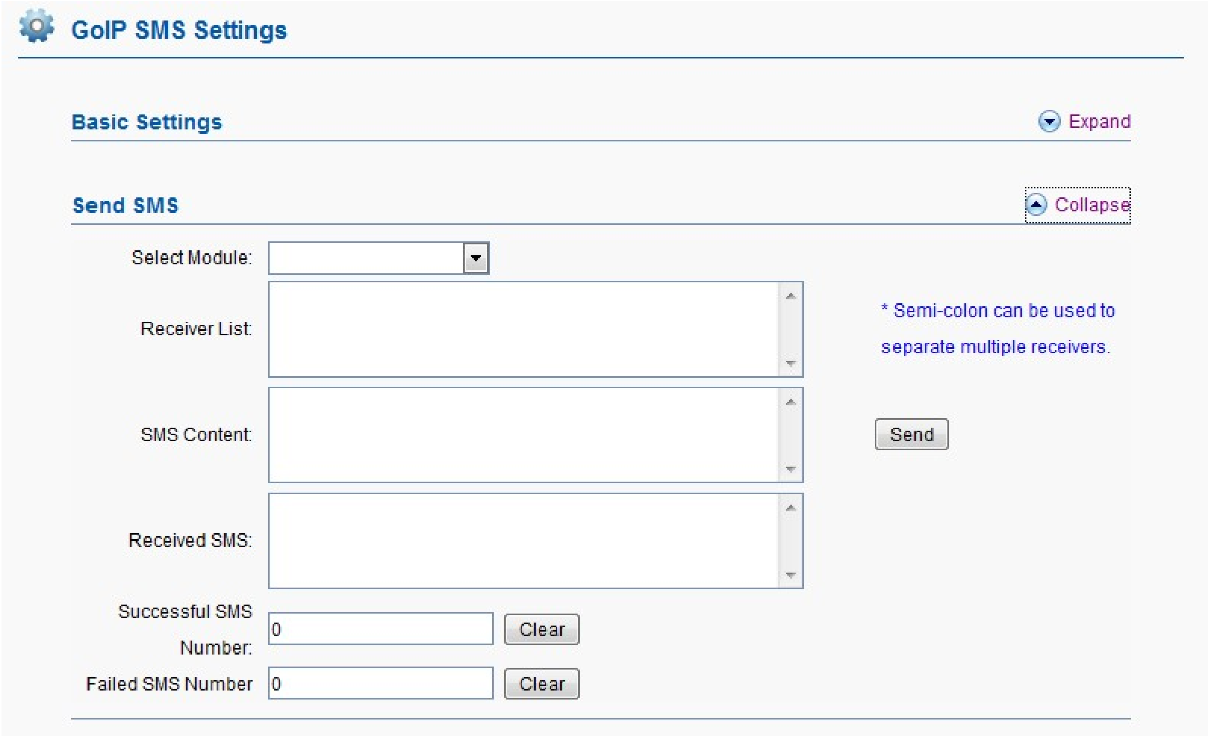
The screenshot below shows the operation mode to set SMS.



Fields are specified as following:

- Send to Server: Specify whether need to enable the functionality of sending SMS to server.
- Server IP: Specify the IP of server to which the SMS is sent. If set to empty, the SMS will be sent to SIP server.
- SMS Coding: Specify the SMS coding when SMS is sent to through SIP protocol.

The screenshot below shows the operation mode to send SMS through the GoIP Gateway.



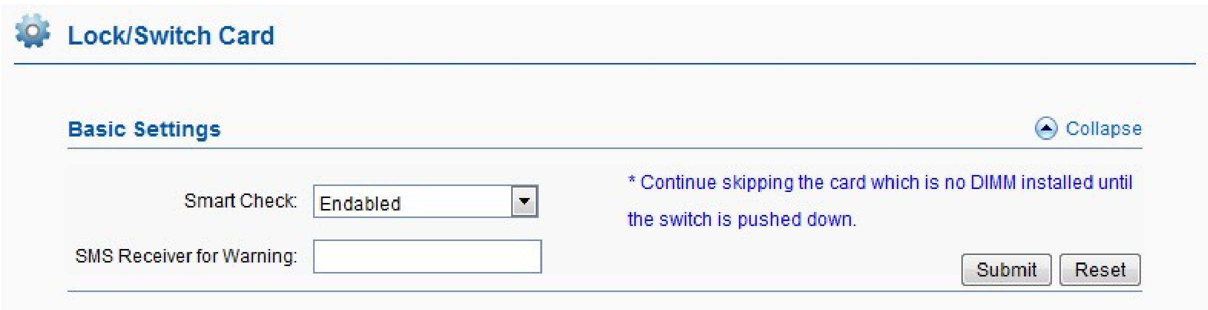
1) Select a module. The module here means GoIP mobile port and the SMS is sent out through the card which is in service on this port.
2) Input the receivers separated by semi-colon.
3) Input SMS content and click button *Send* to send out the SMS.

Field *Received SMS* is used to display the last response of the SMS sent out, if the response is not empty.

Field *Successful SMS Number* records down the total number of SMS which is successfully sent out. Field *Failed SMS Number* records down the total number of SMS which is sent failed.

4.4.4 Lock/Switch Card

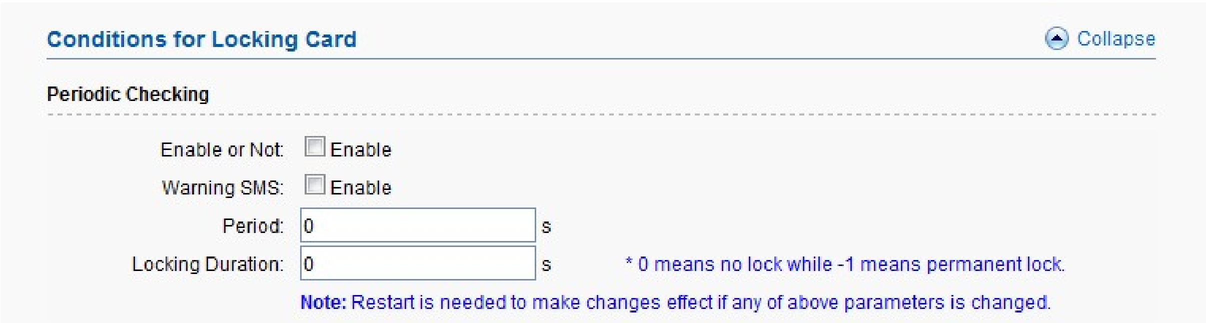
The screenshot below shows the operation mode to set globally for card lock and switch.



Fields are specified as following:

- Smart Check: Specify whether enable the smart check for cards in card slot or not. If set to Enabled, those card slots without card inserted will be skipped when system is scanning for available cards.
- SMS Receiver for Warning: Specify the receiver mobile number to receive the warning SMS for card lock or switch.

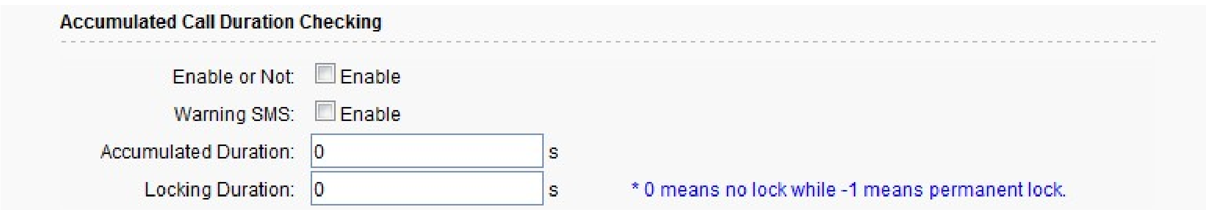
The screenshot below shows the operation mode to set conditions for locking card and switching card.



Fields are specified as following:

- Enable or Not: Specify whether enable the periodic checking or not. If set to Enable, system will check the card running time periodically. If the continuous running time reaches or exceeds the specified *Period* value, this card will be locked and the next card will come into service.
- Warning SMS: Specify whether need to send warning SMS when the card is locked.
- Period: Specify the period to check.
- Locking Duration: Specify how long the card will be locked.

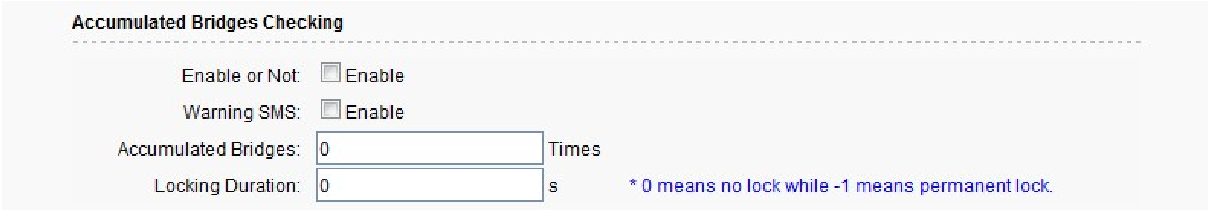
The following is to set condition of accumulated call duration.



Fields are specified as following:

- Enable or Not: Specify whether enable this condition. If set to Enable, the accumulated call duration will be used as a condition for system to check.
- Warning SMS: Specify whether need to send warning SMS when the card is locked.
- Accumulated Duration: Specify the max running duration of the card. If the accumulated running duration reaches or exceeds this value, the card will be locked if this condition is enabled.
- Locking Duration: Specify how long the card will be locked.

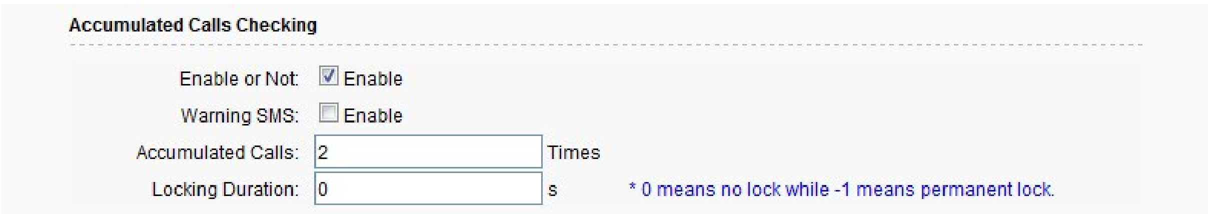
The following is to set condition of accumulated bridges.



Fields are specified as following:

- Enable or Not: Specify whether enable this condition. If set to Enable, the accumulated number of bridges will be used as a condition for system to check.
- Warning SMS: Specify whether need to send warning SMS when the card is locked.
- Accumulated Bridges: Specify the max number of bridges on this card. If the accumulated number of bridges reaches or exceeds this value, this card will be locked if this condition is enabled.
- Locking Duration: Specify how long the card will be locked.

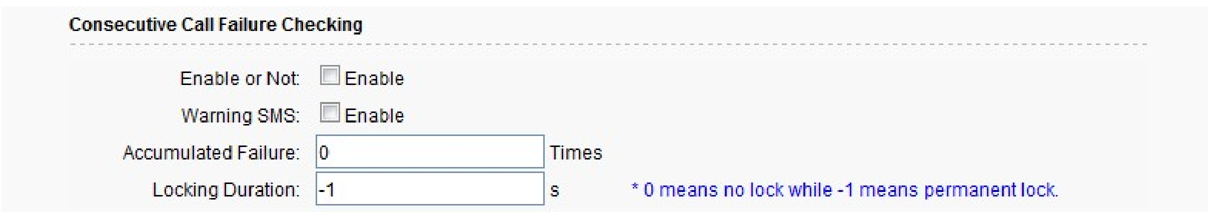
The following is to set condition of accumulated calls.



Fields are specified as following:

- Enable or Not: Specify whether enable this condition. If set to Enable, the accumulated calls will be used as a condition for system to check.
- Warning SMS: Specify whether need to send warning SMS when the card is locked.
- Accumulated Calls: Specify the max number of calls on this card. If the accumulated number of calls reaches or exceeds this value, this card will be locked if this condition is enabled.
- Locking Duration: Specify how long the card will be locked.

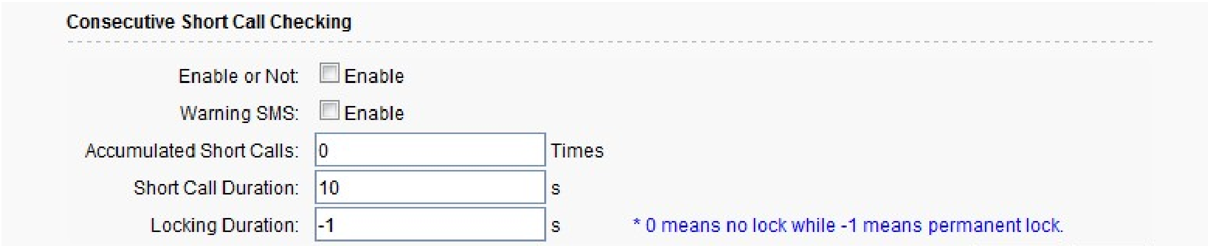
The following is to set condition of consecutive failed calls.



Fields are specified as following:

- Enable or Not: Specify whether enable this condition. If set to Enable, the accumulated number of failed calls will be used as a condition for system to check.
- Warning SMS: Specify whether need to send warning SMS when the card is locked.
- Accumulated Failure: Specify the max number of consecutive failed calls on this card. If the number of consecutive failed calls reaches or exceeds this value, the card will be locked if this condition is enabled.
- Locking Duration: Specify how long the card will be locked.

The following is to set condition of consecutive short calls.

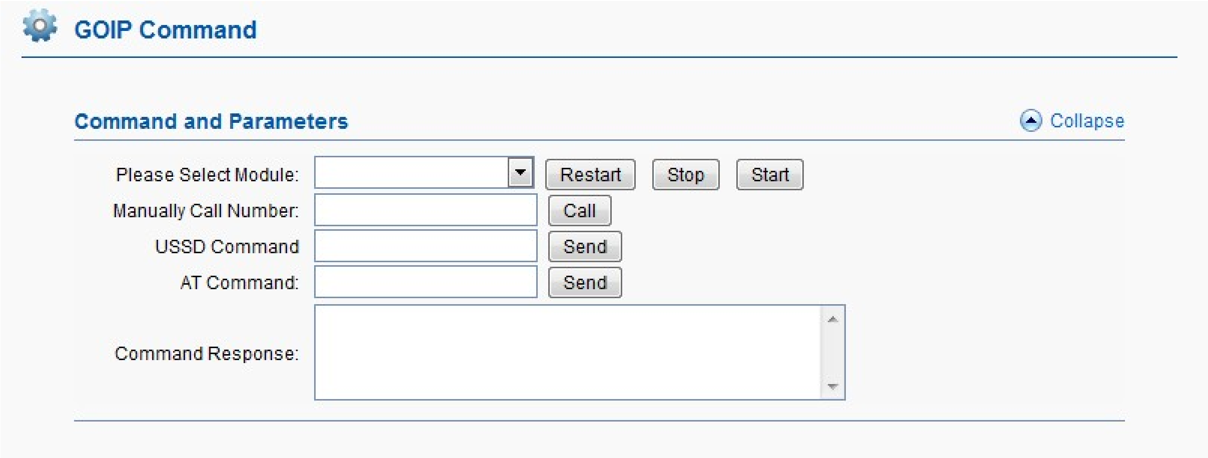


Fields are specified as following:

- Enable or Not: Specify whether enable this condition. If set to Enable, the accumulated short calls will be used as a condition for system to check.
- Warning SMS: Specify whether need to send warning SMS when the card is locked.
- Accumulated Short Calls: Specify the max number of consecutive short calls on this card. If the number of consecutive short calls reaches or exceeds this value, the card will be locked if this condition is enabled.
- Short Call Duration: Specify the call duration to recognize a short call. Any call whose duration is less than this value will be regarded as an short call.
- Locking Duration: Specify how long the card will be locked.

4.4.5 AT Command

The screenshot below shows the operation mode to send AT command to GoIP Gateway.



The module here means the GoIP mobile port.

Button *Restart* is used to restart this module.

Button *Stop* is used to stop this module.

Button *Start* is used to start this module.

Button *Call* is used to dial out through the selected module manually.

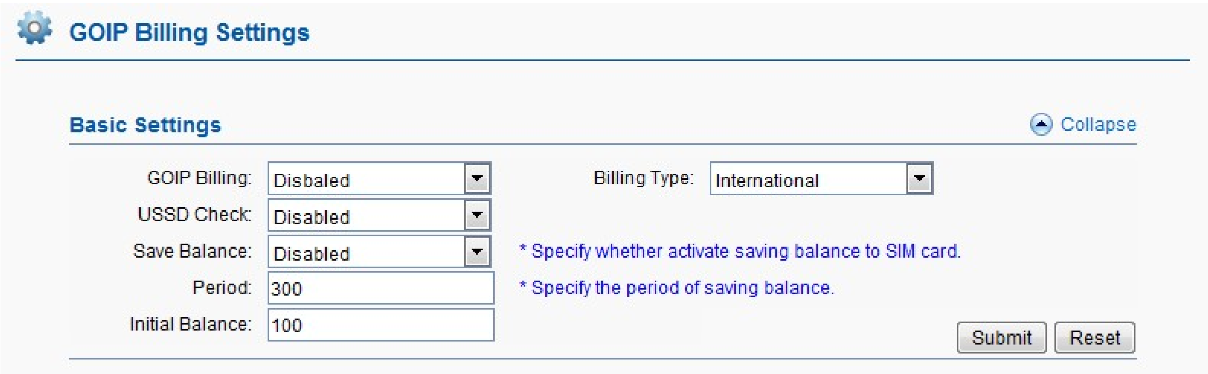
For the USSD command, please refer to the local carrier standard.

For AT command, please refer to appendix 1.

Field Command Response is used to display the response of last command. After send a command, a re-enter of this page is needed to see the command response.

4.4.6 Billing

The screenshot below shows the operation mode to set GoIP billing. A smart billing server for mobile port is embedded in GoIP Gateway.



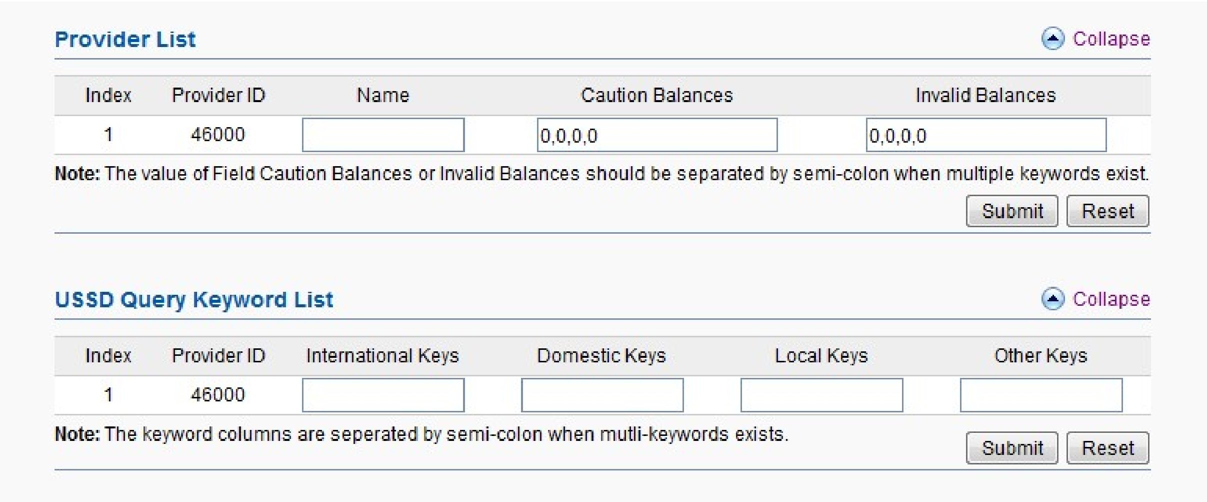
Fields are specified as following:

- GoIP Billing: Specify whether enable GoIP billing or not. If set to Enabled, system will bill the outbound calls for the port which has been assigned with billing tariffs.
- Billing Type: Specify the type to get balance through USSD. Option values are International/Internal/Local Net/Other. Each optional value maps to the corresponding USSD Keyword in USSD Query Keyword List. The balance is checked base on the USSD keyword which is mapped by this selected

choice. This field takes effect only when both *GoIP Billing* and *USSD Check* are set to *Enabled*.

- USSD Check: Specify whether enable to get balance through USSD check or not. This field takes effect only when *GoIP Billing* is set to *Enabled*.
- Save Balance: Specify whether need to save the current balance to card periodically. If set to Enabled, the balance will be updated to card periodically or after a call is released.
- Period: Specify the period to save balance.
- Initial Balance: Special the initial balance for a new card.

The screenshot below shows the operation mode to set *Caution Balances*, *Invalid Balances* and *USSD Keyword List*. These settings are used for both getting balance through USSD and billing GoIP calls. The provider ID is detected by GoIP Gateway automatically. For a new Gateway without any card inserted, there may be no records in the two lists.



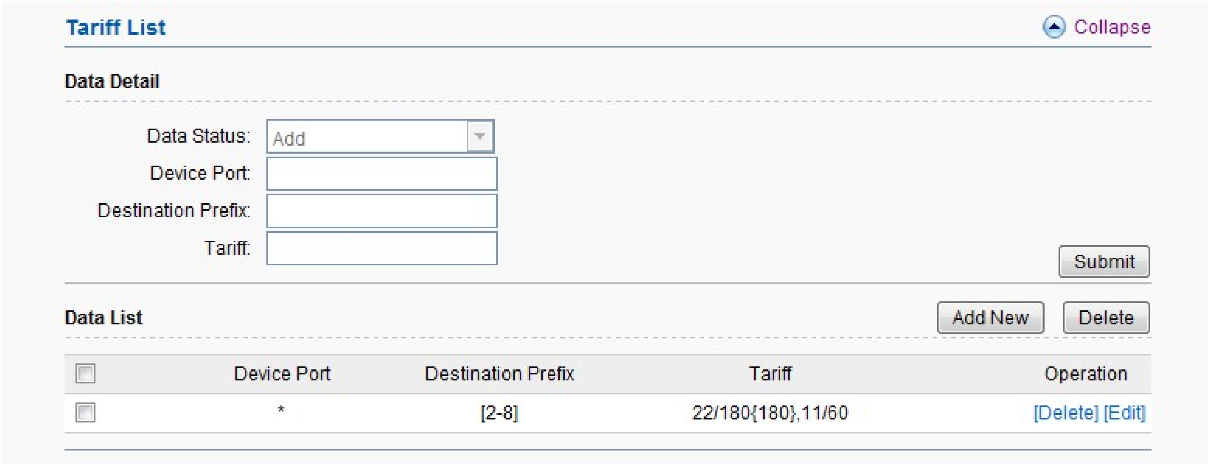
Fields are specified as following:

- Name: Specify the provider name.
- Caution Balances: Specify four balances separated by semi-colon. Each balance corresponds to one Keyword from left *International Keys* to right *Other Keys* in Keyword List table. Each caution balance is a threshold for system to get card balance through USSD if current balance is less than this threshold. However, the final card balance is based on only one type which is specified by field *Billing Type* in the part of Basic Settings.
- Invalid Balances: like Caution Balances, it specifies the threshold for system to disable the card if current balance is less than this threshold.
- International Keys: Specify keyword for system to analyze the international balance data after sending USSD command to carrier mobile network.
- Domestic Keys: Like International Keys, it specifies keyword for system to analyze the domestic balance data after sending USSD command to carrier mobile network.
- Local Keys: Like International Keys, it specifies keyword for system to

analyze the local balance data after sending USSD command to carrier mobile network.

- Other Keys: Like International Keys, it specifies keyword for system to analyze other balance data after sending USSD command to carrier mobile network.

The screenshot below shows the operation mode to set billing tariff.



Add New

Click button *Add New* to expand the data input area to add new data. Fields are specified as following:

- Data status: Mark the status of current data record. Option values are *Add/Edit*. Value *Add* means the data is new while value *Edit* means the data is old.
- Device Port: Specify the GoIP mobile port on which this tariff will take effective. If set to *, this tariff record will apply to all ports. The single port number can be an integer from 1 to 8.
- Destination Prefix: Specify the destination prefix used to bill call. If this prefix is best matched with a destination of an outgoing call from the port(s), the corresponding tariff will be chosen to bill the call. The prefix can be a regular expression. For example, [2-8] matches any phone number which starts with digit 2 to 8. And [0-9] matches all phone numbers.
- Tariff: Specify the tariff detail. Multiple billing stage tariffs are supported. Each stage can be assigned with a different tariff. Comma is used to separate multiple billing stage tariffs. For example, the tariff can be set to *22/180{180},11/60*. The value 22/180 means 22 will be charged per 180 seconds, while the value 11/60 means 11 will be charged per 60 seconds. The whole tariff means:
    - If call duration is within 180 seconds, 22/180 will be used to bill this call;
    - If call duration is greater than 180, the billing will contain two parts. 22/180 will be used to bill the call for the first 180 seconds, and 11/60 will be used to bill the call for the rest durations.

Click button *Submit* on the right to save the new data record.

Edit

All the records are displayed in list. Two operations are provided on the right of each record. Click *Edit* to expand the current data record to Data Detail Area which is above the Data List.

Click button *Submit* on the right to save the old data record.

Delete

Click *Delete* on the right of each record to delete the current record. A message box will be popped for delete confirmation.

Another shortcut button is also provided on the top right of Data List to delete multiple selected records in batch. A message box will be popped for confirmation of batch delete.

4.5 Application Settings

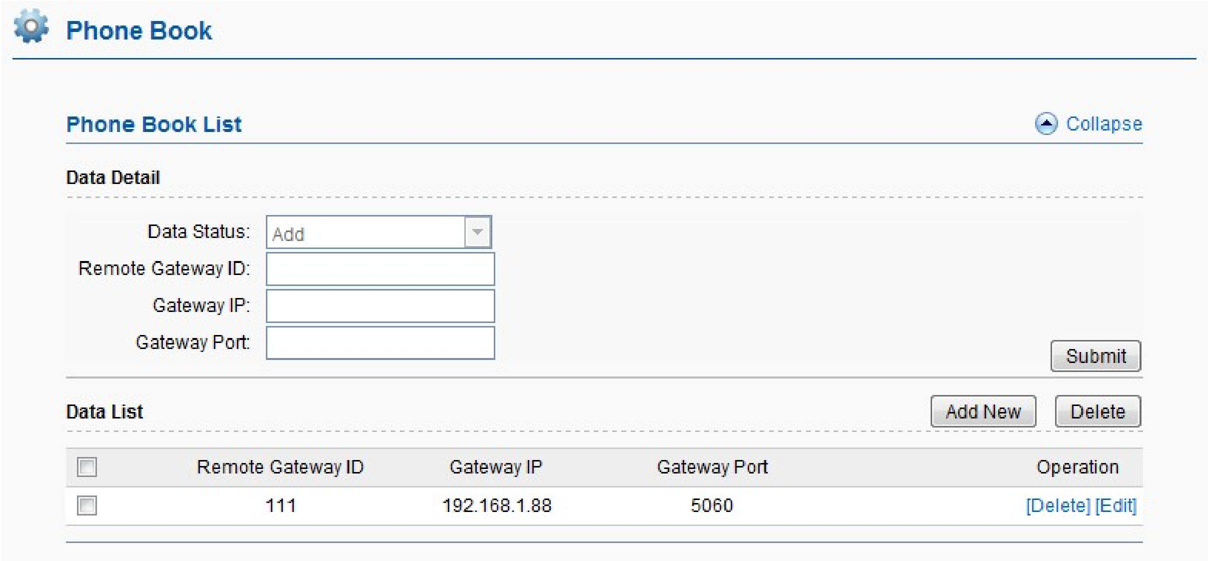
Application Settings focus on the business feature. It includes:

- Phone Book
- Dial Pattern
- Dial Prefix Manipulation
- Local Billing

These sub topics will be introduced separately below.

4.5.1 Phone Book

The screenshot below shows the operation mode to set phone book. Phone book is a list contains the relationship between destination phone prefix and gateway information.



Add New

Click button *Add New* to expand the data input area to add new data. Fields are specified as following:

- Data status: Mark the status of current data record. Option values are *Add/Edit*. Value *Add* means the data is new while value *Edit* means the data is old.
- Remote Gateway ID: Specify the prefix of destination number for outbound call to IP. If a destination is best matched with any prefix in phone book, the destination call will be routed to the IP gateway specified by *Gateway IP* and *Gateway Port*.
- Gateway IP: The remote gateway IP.
- Gateway Port: The remote gateway port.

Click button *Submit* on the right to save the new data record.

Edit

All the phone book records are displayed in list. Two operations are provided on the right of each record. Click *Edit* to expand the current data record to Data Detail Area which is above the Data List.

Click button *Submit* on the right to save the old data record.

Delete

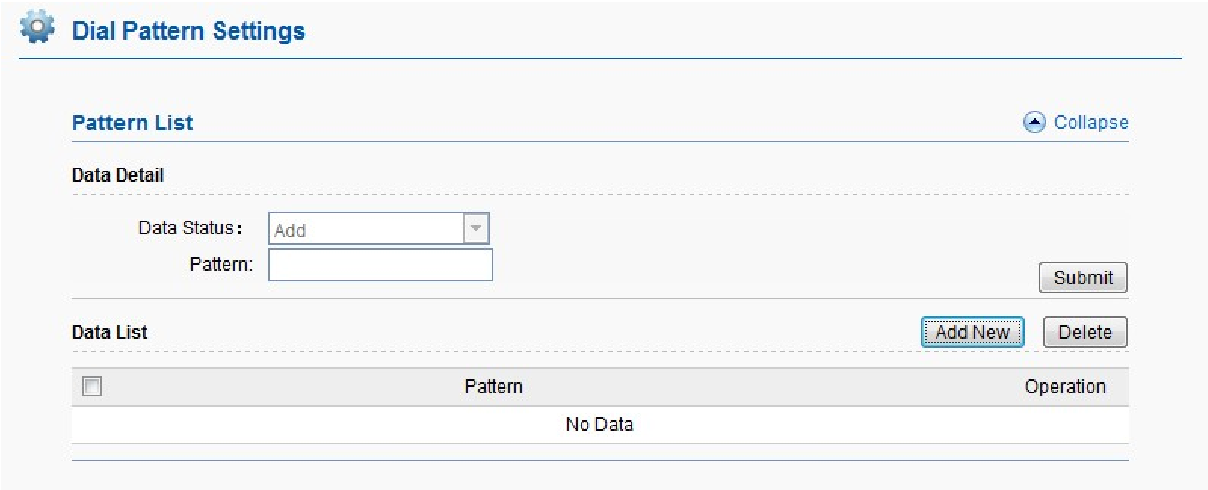
Click *Delete* on the right of each record to delete the current record. A message box will be popped for delete confirmation.

Another shortcut button is also provided on the top right of Data List to delete multiple selected records in batch. A message box will be popped for confirmation of batch delete.

Note: The operations *Add New/Edit/Delete/Batch Delete* mentioned in the following paragraph are almost the same as phone book.

4.5.2 Dial Pattern

The screenshot below shows the operation mode to set dial patterns.



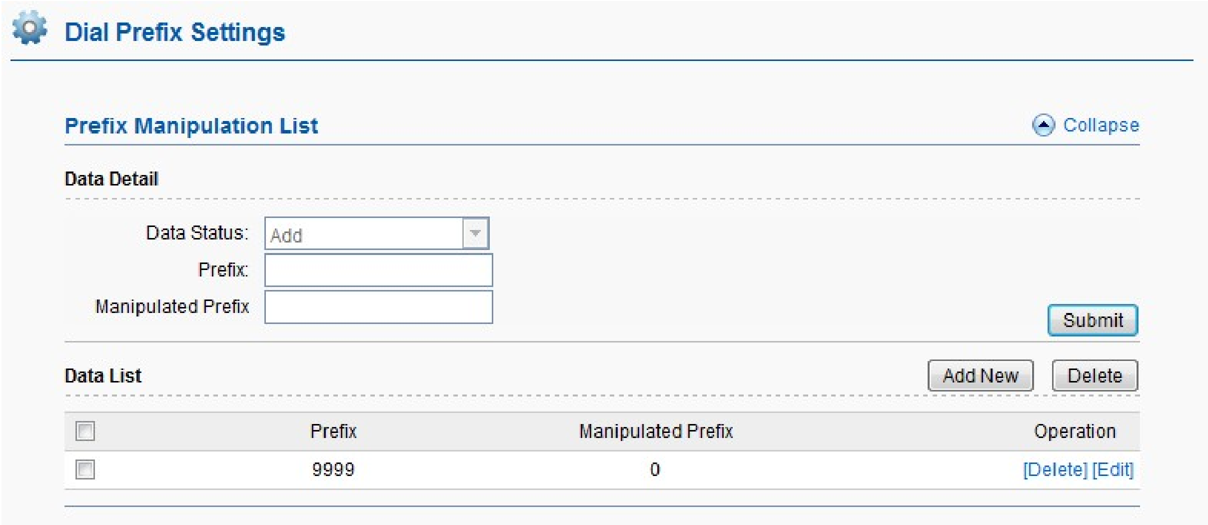
Dial pattern is a string which specifies digits and length of the digits of the dialed number. Generally, the pound sign # is used as the termination of number input for dialing. However, if patterns are specified and system detects the dialed number matches any of the patterns, it will stop collecting input and send out the collected number to dial even though no pound sign is encountered.

The dial pattern string is a normal regular expression, for example:

The pattern *90[1-4]* means the dialed number starts with 90 and the last digit is any of 1/2/3/4. So any of the input 901, 902, 903 or 904 is acceptable.

4.5.3 Dial Prefix Manipulation

The screenshot below shows the operation mode to set manipulation for dial prefix.



Fields are specified as following:

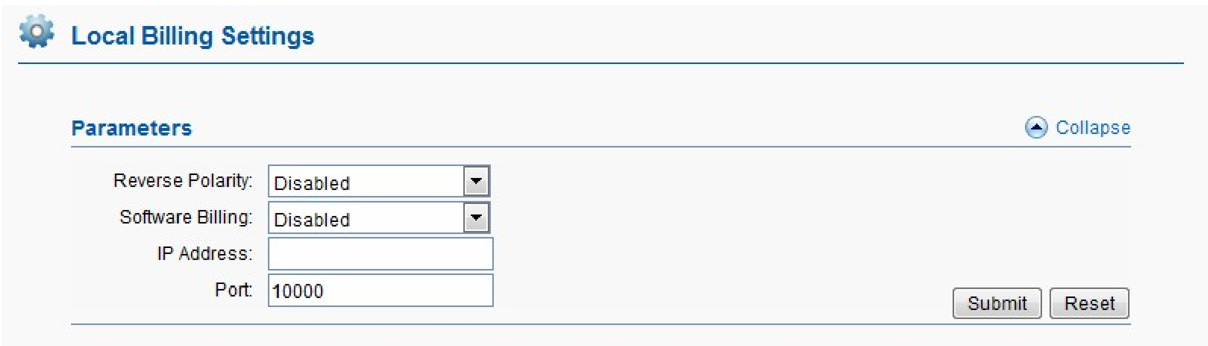
- Prefix: The original prefix in phone number.
- Manipulated Prefix: Specify the digits with which the value specified by *Prefix* will be substituted.

Take the value in screenshot as an example, the prefix 9999 in dialed number will be substituted with 0. That's to say, if 999988760101 is input to dial, the final number dialed out is 088760101.

Note: the manipulation is executed after pattern is matched.

4.5.4 Local Billing

The screenshot below shows the operation mode to set local billing settings.



Fields are specified as following:

- Reverse Polarity: Specify whether need to enable feature of reverse polarity or not.
- Software Billing: Specify whether need to enable software billing or not.
- IP Address: Specify the IP of billing server.
- Port: Specify the port of billing server.

4.6 Advanced Settings

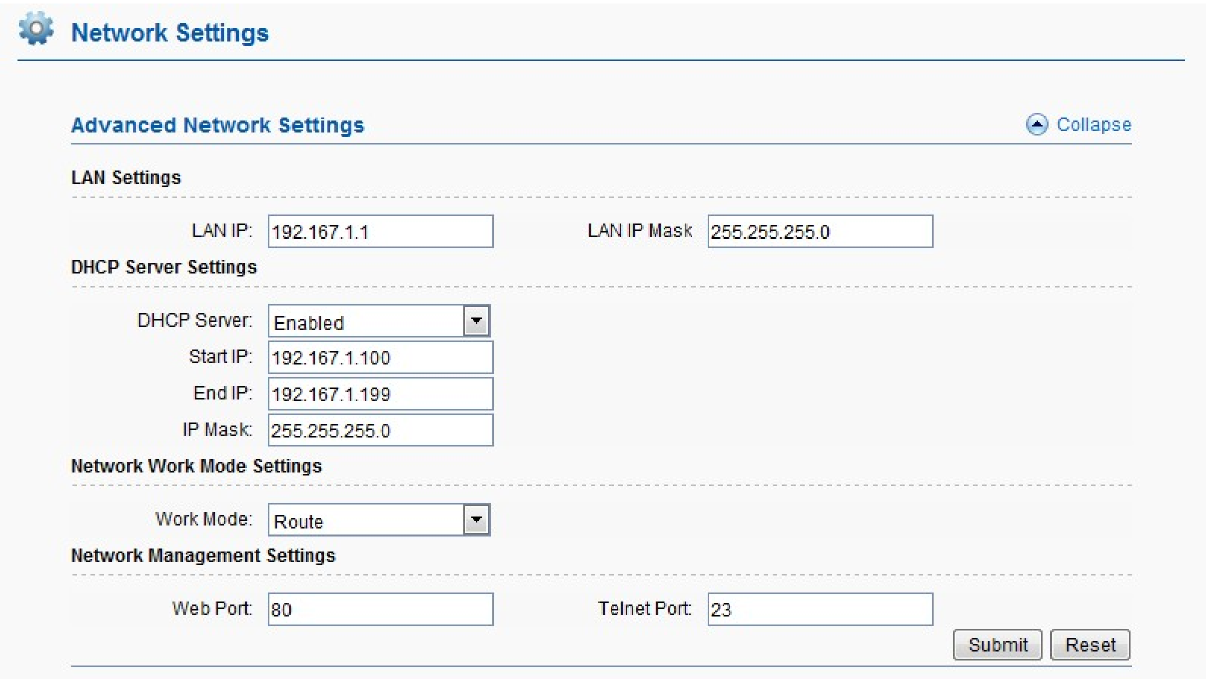
Advanced Settings focus on the high level usage of GoIP Gateway. It includes:

- Network
- Voice and Codec
- Analog Port

These sub topics will be introduced separately below.

4.6.1 Network

The screenshot below shows the operation mode to set advanced network. The difference between advanced network and basic network is that the prior focuses on those functionalities whose settings are seldom modified. Generally the default settings for advanced network are already suitable for system running.



LAN port is used for PC to connect GoIP Gateway directly without any other route. The default LAN IP is *192.167.1.1*. If a PC is connected to LAN port of a GoIP Gateway, it needs the same sub-network to access GoIP Gateway directly. For example, the PC IP is 192.167.1.10. Administrator can login web pages through URL: *http://192.167.1.1/*.

DHCP server is used to automatically assign an IP address to a computer or other network devices which is connected to LAN port of GoIP Gateway. If a computer successfully obtains an IP from DHCP server, its DNS should be manually set to the actual DNS value.

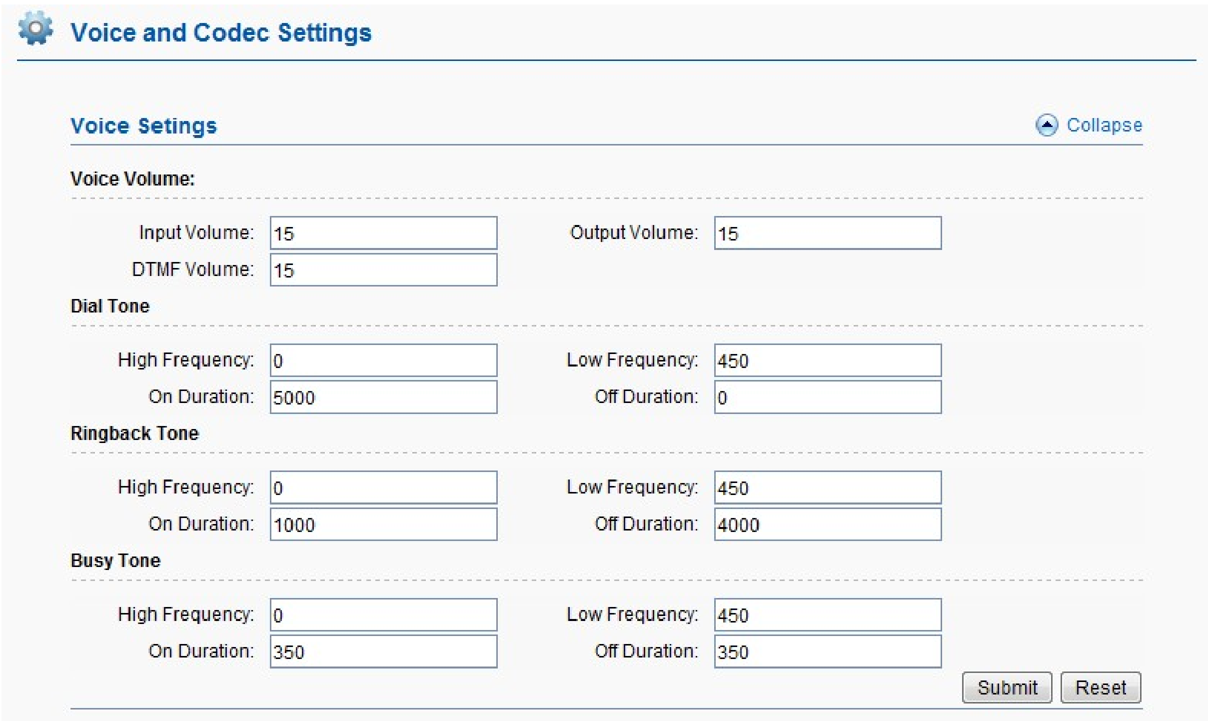
There are three working modes provided for network: Route/Hub/Disabled.

The default port of web server is 80. The field *Web Port* is used to set another different port for web server. For example, if field *Web Port* is set to 8080 and a PC is connected to the LAN port of GoIP Gateway with IP 192.167.1.10, the web pages then should be accessed through URL: http://192.167.1.10:8080/ from this computer.

The field *Telnet Port* is used to change the default port of telnet service.

4.6.2 Voice and Codec

The screenshot below shows the operation mode to set voice feature which only applies to analog FXO and FXS Gateway.



Voice Volume is used to specify the input voice volume, output voice volume and DTMF tone volume. The acceptable value for volume is an integer no less than 10 and no greater than 40.

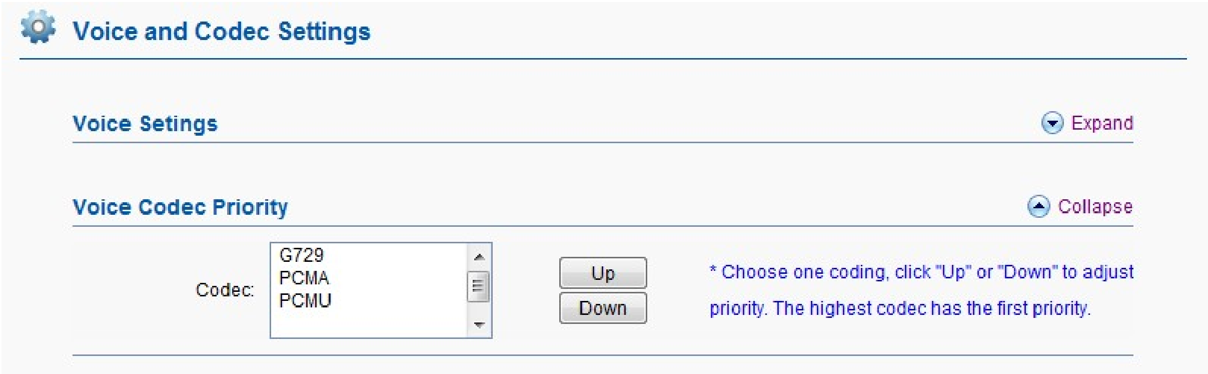
The Dial Tone is sent to a customer or operator to indicate that the receiving end is ready to receive dial pulses or DTMF signals. It is used in all types of dial offices when the customer's or operator's dials produce dial pulses.

A Ring Back tone (or ringing tone) is an audible indication that is heard on the telephone line by the caller while the phone they are calling is being rung. It is normally a repeated tone, designed to assure the calling party that the called party's line is ringing.

The Busy Tone indicates that the called customer's line has been reached but that it is busy, being wrong, or on permanent signal. When an operator applies a busy signal, it is sometimes called a busy-back tone. Line Busy Tone is a Low Tone that is on and off every 0.5 second.

The settings of Dial Tone, Ring Back Tone and Busy Tone depend on area. The default settings for Asia are shown in the screenshot above for reference.

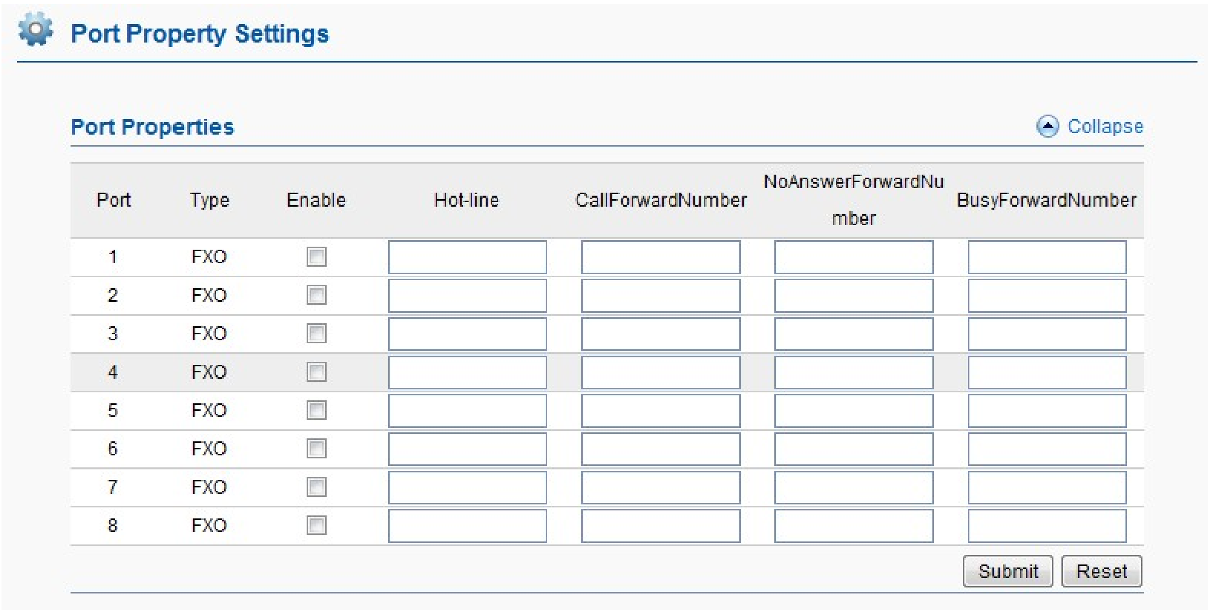
The screenshot below shows the operation mode to set codec priority.



Three codec types are provided to adjust GoIP Gateway to different network environment. The top codec will be chosen to use by default. G729 uses the least bandwidth.

4.6.3 Analog Port

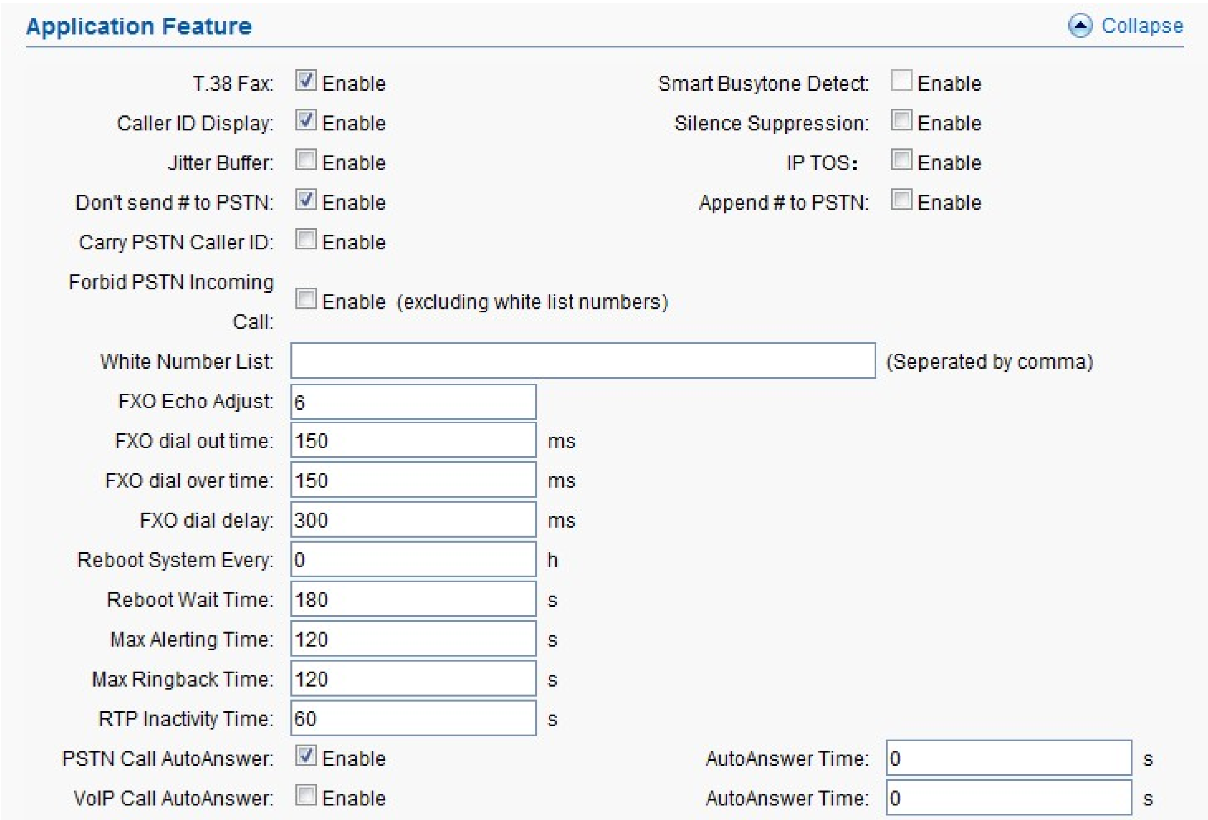
The screenshot below shows the operation mode to set analog port attribute.



The columns are specified as following:

- Port: The analog port sequence from 1 to 32.
- Type: Option values are FXO/FXS. (GoIP port is currently treated as FXO.)
- Enable: Specify whether enable or disable this port.
- Hot-line: Specify a phone number for short. If the port type is FXO, any extern call to this port will be redirected to this hot line. If the port type is FXS, a call will be dialed out automatically when the telephone connected to this port is picked up.
- CallForwardNumber/NoAnserForwardNumber/BusyForwardNumber: These parameters are designed to be used with a third party system.

The screenshot below shows the operation mode to set application feature for analog port.



The fields are specified as following:

- T.38 Fax: Specify whether enable T.38 fax or not.
- Smart Busytone Detect: Specify whether enable Smart Busy Tone Detect or not. It is enabled by default and can't be modified.
- Caller ID Display: Specify whether enable Caller ID Display or not.
- Silence Suppression: Specify whether enable Silence Suppression or not.
- Jitter Buffer: Specify whether enable Jitter Buffer or not.

- IP TOS: Specify whether enable IP TOS or not.
- Don't send # to PSTN: Specify whether need to remove the last digit # to PSTN if the last digit of numbers is #. If set to Enable, the last # will be removed.
- Append # to PSTN: Specify whether need to send an extra # to PSTN after the normal digital numbers are sent.
- Carry PSTN Caller ID: Specify whether need to carry PSTN caller ID to system.
- Forbid PSTN Incoming Call: Specify whether need to prevent the PSTN incoming call. If set to enable, any of the incoming calls whose callerid is not in the white list specified in *White Number List* will be prevented.
- White Number List: Specify a caller number list separated by comma. It is used in combination with *Forbid PSTN Incoming Call* to let the specified caller pass through and continue the incoming call flow if *Forbid PSTN Incoming Call* is set to Enable.
- FXO Echo Adjust: Specify the adjustment for FXO Echo.
- FXO dial out time: Specify the duration for a single digit is pressed down. The recommended value is an integer between 50 and 300.
- FXO dial over time: Specify the time interval between two digits. The recommended value is an integer between 80 and 300.
- FXO dial delay: Specify the time interval between phone pick up and playing back dial tone. The recommended value is an integer between 200 and 800.
- Reboot System Every: Specify the interval for system to reboot automatically.
- Reboot Wait Time: Specify the wait time before system reboot. It mainly used for system to gracefully shutdown.
- Max Alerting Time: Specify the max alerting duration.
- Max Ringback Time: Specify the max ringback duration.
- RTP Inactivity Time: Specify the max duration of silence from FXO. System will hang up the call automatically if the silence duration reaches or exceeds this value.
- PSTN Call AutoAnswer: Specify whether need to auto-answer the call which is from PSTN. If set to Enable, *AutoAnswer Time* can be used to specify the delay to automatically answer the incoming PSTN call.
- VoIP Call AutoAnswer: Specify whether need to auto-answer the call which is from IP network. If set to Enable, *AutoAnswer Time* can be used to specify the delay to automatically answer the incoming IP call.
- DTMF Mode: Specify the DTMF mode. Option values are RFC2833/Inband/SIP INFO.
- RFC2833 Payload Type: Specify the RFC2833 DTMF Payload Type. Only valid when *DTMF Mode* is set to RFC2833. The default value is 101.
- G729 or G723.1: Specify the analogy port voice codec.
- RTP Ptime: Specify the interval of RTP packages.

- Fax Rate: Specify the fax rate. Option values are 14400/12000/9600/7200/4800/2400.

4.7 System Settings

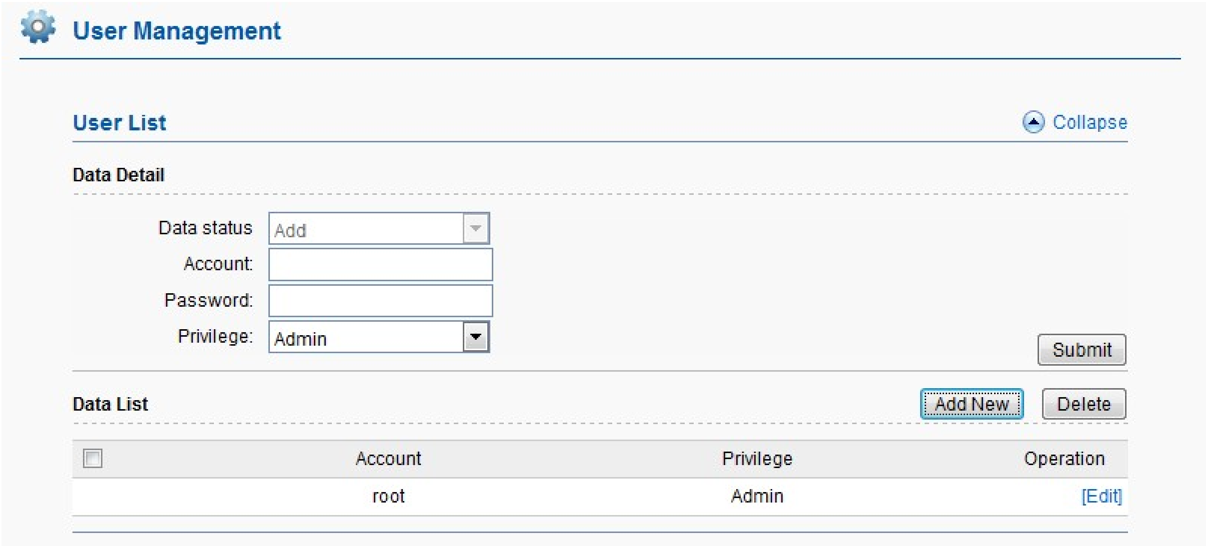
System Settings include:

- User Management
- Remote Management
- System Update

These sub topics will be introduced separately below.

4.7.1 User Management

The screenshot below shows the operation mode to manage system user.



Default User

The default system user account is root. This account can't be deleted and only *Password* and *Privilege* can be modified for this account.

Add User

Click button *Add New* to expand the data input area to add new data. Fields are specified as following:

- Data status: Mark the status of current data record. Option values are *Add/Edit*. Value *Add* means the data is new while value *Edit* means the data is old.
- Account: The user account used to login web system. The account value can not be modified after save.
- Password: The password used to login web system.

- Privilege: The privilege of user. Option values are *Admin*/*User*.

Click button *Submit* on the right to save the new data record.

Edit User

All the user records are displayed in list. Two operations are provided on the right of each record. Click *Edit* to expand the current data record to Data Detail Area which is above the Data List.

Click button *Submit* on the right to save the old data record.

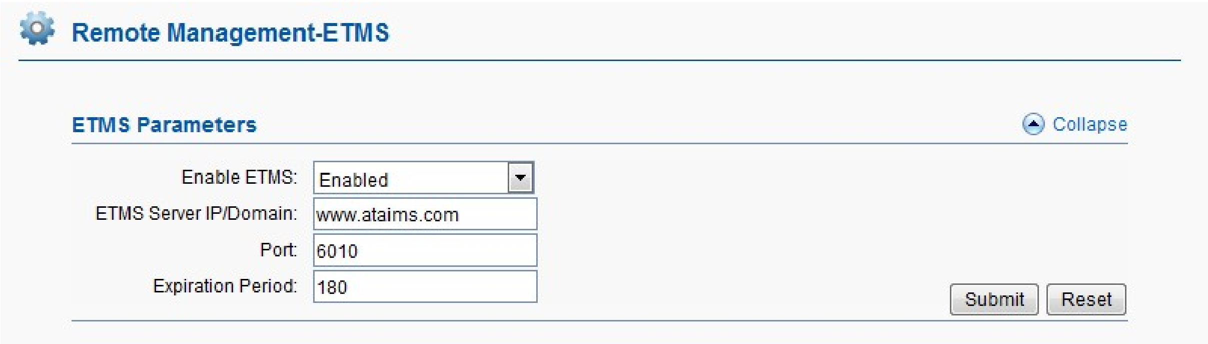
Delete User

Click *Delete* on the right of each record to delete the current record. A message box will be popped for delete confirmation.

Another shortcut button is also provided on the top right of Data List to delete multiple selected records in batch. A message box will be popped for confirmation of batch delete.

4.7.2 Remote Management

The screenshot below shows the operation mode for remote management.



Remote Management is used to manage the GoIP Gateways located in other physical locations. Network must be available for the gateway to communicate with ETMS Server.

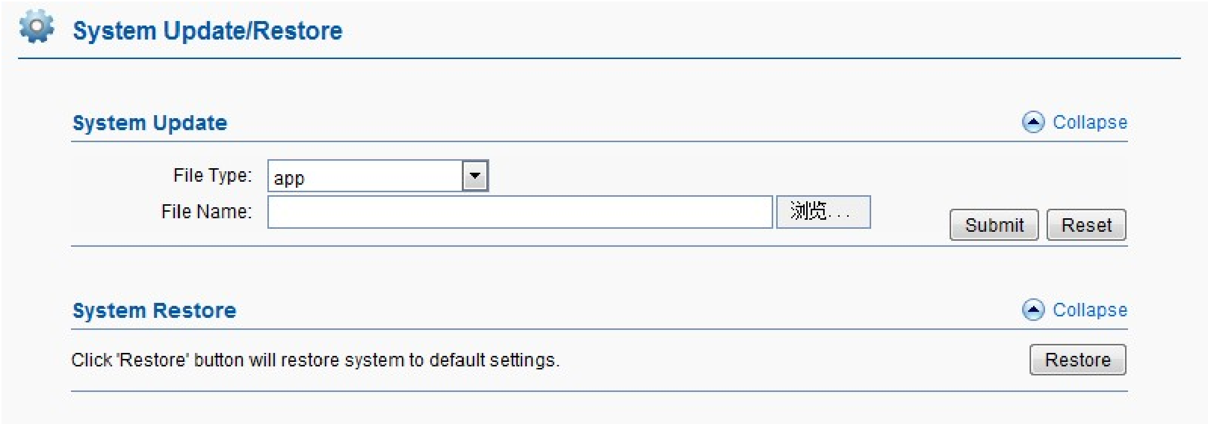
If ETMS is enabled and correctly set, the GoIP will register to EMTS server and set up the connection between itself and ETMS server. Administrator can login ETMS server and monitor all the registered GoIP Gateways. Commands can also be sent from ETMS server to certain gateway for management.

The configuration fields are specified as following:

- Enable ETMS: Specify whether enable ETMS registration or not. Option values are *Enabled*/*Disabled*.
- ETMS Server IP/Domain: Specify the ETMS Server address. Either IP or Domain is a valid input.
- Port: Specify the ETMS server port.
- Expiration Period: Specify the expiration period for registration to ETMS server.

4.7.3 System Update

The screenshot below shows the operation mode for system update or restore.



System Update

The content for system update includes:

- app
- appcfg
- dspapp
- h323cfg723
- dspboot
- cntrmd
- usrrmd
- pswrmd
- rngrmd
- mac0
- mac1
- vcfg
- lic
- usrdef

The configuration fields are specified as following:

- File Type: Specify the content to update. Option values are listed above.
- File Name: Specify the content file name. Click button *Browser* and then select the target file from the popped file selection window.

System Restore

System restore is used to restore the system to default settings. A message box will be popped for the confirmation of restore.

4.8 Running Status

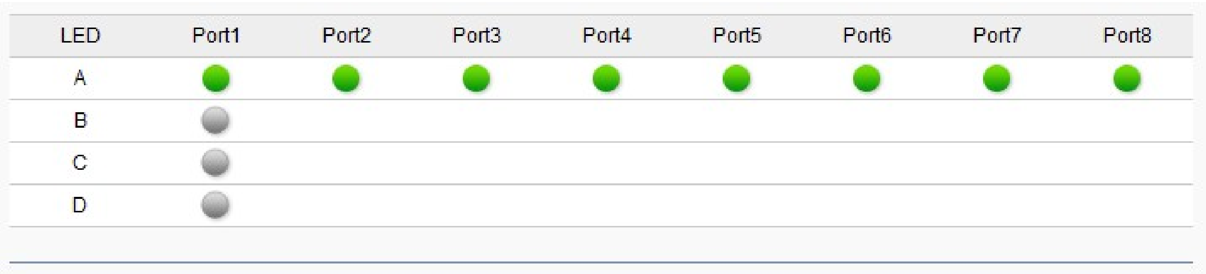
Running Status includes:

- Port Status
- Call Status
- System Status
- Call Statistics

It is used to monitor real-time situation for calls, GSM ports and equipment hardware status.

4.8.1 Port Status

The screenshot below shows the GoIP Gateway port **LED** status. Different LED color stands for different port status.



Note:

LED A/B/C/D displays in accordance with the lights on the front board of GoIP Gateway. Port 1 to 8 relate to the physical port of GoIP Gateway. The following table shows the relationship between LED color and port status.

LED Color	Empty				
Status	No Card	Card Stands By	Card in Service	Card in Calling	Balance is not enough

The screenshot below shows the port status in detail.

GoIP Port Status — Collapse

Port No.	Provider	Module Detected	Card Detected	Signal Strength	SMS Count
1	46000	Yes	Yes	26	0
2	46000	Yes	Yes	24	0
3	46000	Yes	Yes	27	0
4	46000	Yes	Yes	23	0
5	46000	Yes	Yes	22	0
6	46000	Yes	Yes	25	0
7	46000	Yes	Yes	20	0
8	46000	Yes	Yes	24	0

The status columns are specified as following:

- Port No: The physical port sequence from 1 to 8.
- Provider: The mobile provider that system detects.
- Module Detected: Specify whether port module has been detected or not. Option values are *Yes*/*No*.
- Card Detected: Specify whether card is detected or not. Option values are *Yes*/*No*.
- Signal Strength: Specify the mobile signal strength.
- SMS Count: Shows how many SMS has been sent since the last start up of system.

The screenshot below shows the registration status of SIP account.

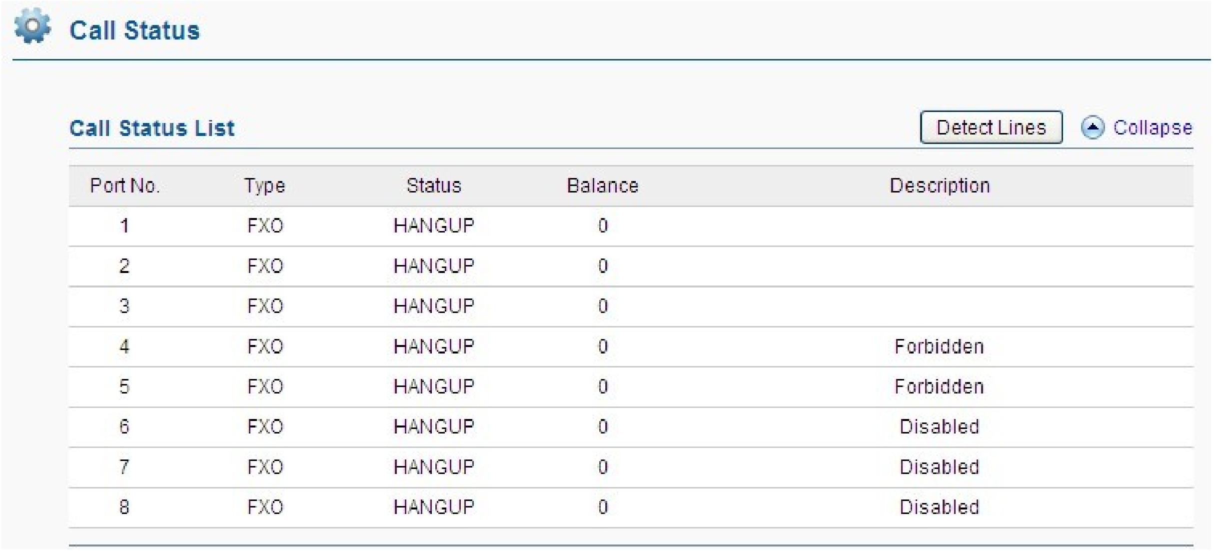
SIP Client Status — Collapse

Protocol Type:	SIP	Operation Mode:	
Server IP:	211.154.151.83	Server Port:	5060
Running Status:	Enabled	Registration Status:	1,2,3,4 OK

Pay attention to the field *Registration Status* which reports the registration result of SIP account. Take "1,2,3,4 OK" as an example, it stands for SIP account 1/2/3/4 are successfully registered to Server.

4.8.2 Call Status

The screenshot below shows the live call status.



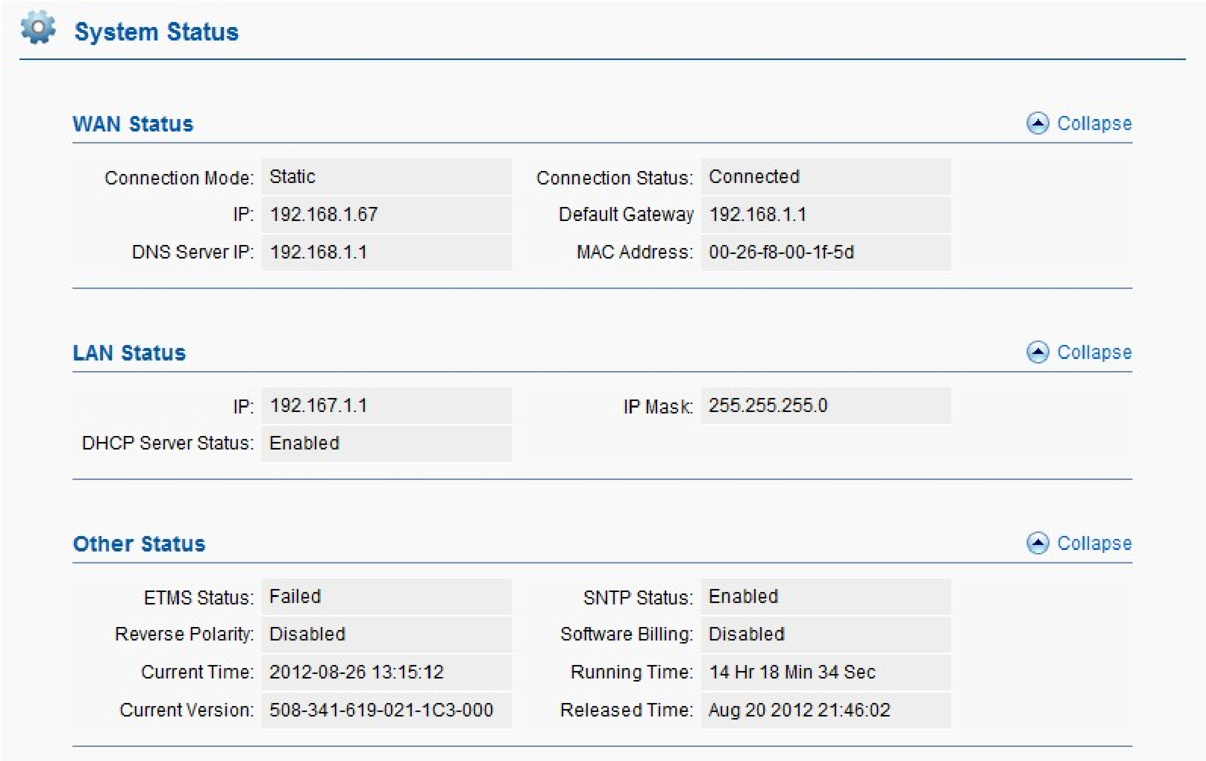
Port No.	Type	Status	Balance	Description
1	FXO	HANGUP	0	
2	FXO	HANGUP	0	
3	FXO	HANGUP	0	
4	FXO	HANGUP	0	Forbidden
5	FXO	HANGUP	0	Forbidden
6	FXO	HANGUP	0	Disabled
7	FXO	HANGUP	0	Disabled
8	FXO	HANGUP	0	Disabled

The status columns are specified as following:

- Port No: The physical port sequence from 1 to 8.
- Type: FXO or FXS. Currently GoIP is regarded as FXO.
- Status: Specify the call status.
- Balance: Specify the current balance of the card in this port.
- Description: Specify the card status.

4.8.3 System Status

The screenshot below shows the system status. It includes WAN status, LAN status and others. The reported information can help you get the system status detail in a fast, simple way.



System Status

WAN Status — Collapse

Connection Mode:	Static	Connection Status:	Connected
IP:	192.168.1.67	Default Gateway	192.168.1.1
DNS Server IP:	192.168.1.1	MAC Address:	00-26-f8-00-1f-5d

LAN Status — Collapse

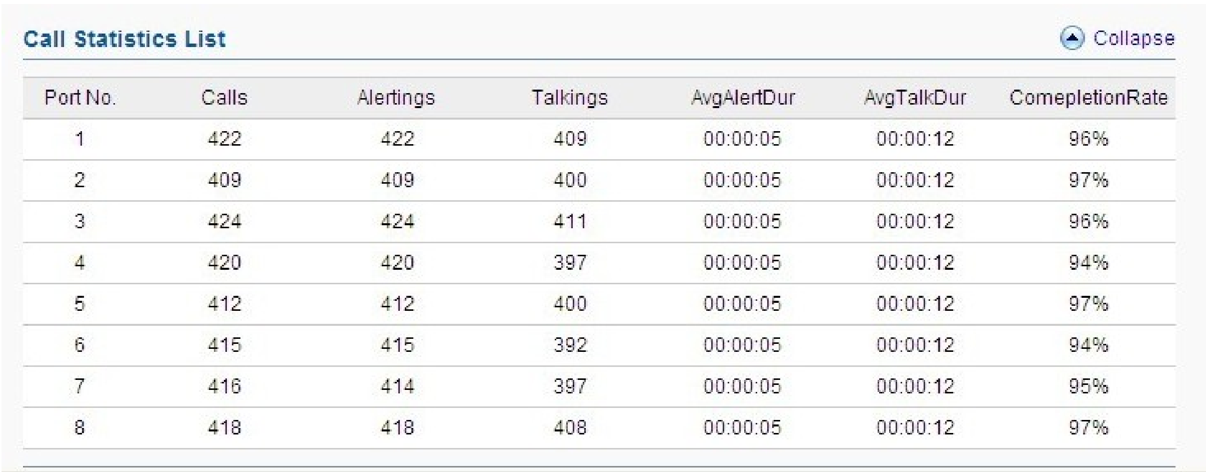
IP:	192.167.1.1	IP Mask:	255.255.255.0
DHCP Server Status:	Enabled		

Other Status — Collapse

ETMS Status:	Failed	SNTP Status:	Enabled
Reverse Polarity:	Disabled	Software Billing:	Disabled
Current Time:	2012-08-26 13:15:12	Running Time:	14 Hr 18 Min 34 Sec
Current Version:	508-341-619-021-1C3-000	Released Time:	Aug 20 2012 21:46:02

4.8.4 Call Statistics

The screenshot below shows the call statistics information for analysis.



Call Statistics List — Collapse

Port No.	Calls	Alertings	Talkings	AvgAlertDur	AvgTalkDur	ComepletionRate
1	422	422	409	00:00:05	00:00:12	96%
2	409	409	400	00:00:05	00:00:12	97%
3	424	424	411	00:00:05	00:00:12	96%
4	420	420	397	00:00:05	00:00:12	94%
5	412	412	400	00:00:05	00:00:12	97%
6	415	415	392	00:00:05	00:00:12	94%
7	416	414	397	00:00:05	00:00:12	95%
8	418	418	408	00:00:05	00:00:12	97%

The status columns are specified as following:

- Port No: The physical port sequence from 1 to 8.
- Calls: Specify the total calls made out from this port since the last start up of system.
- Alertings: Specify the total number of responded alerting message for all the calls made.
- Talkings: Specify the total number of answer from destination for all the calls made.
- AvgAlertDur: Specify the average duration to receive the response of alerting message.

- AvgTalkDur: Specify the average duration of talking between caller and callee.
- CompletionRate: Specify the percentage of successful call for which there is a responded alerting messaged returned.

A total summary is displayed at the bottom of the table.

4.9 Save and Reboot

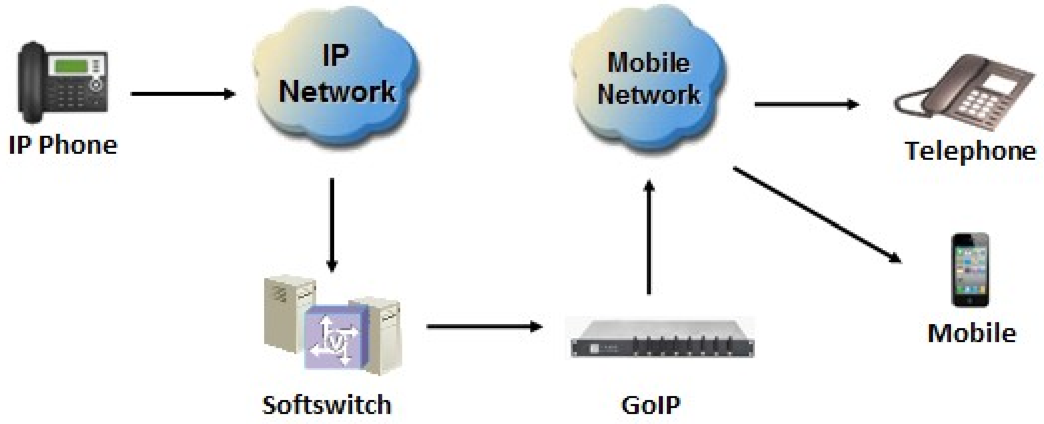
Generally, any modification should require the reboot of GoIP Gateway to bring the modification into effect. However, single Save without Reboot is also frequently used to save the modifications which will be effective on next reboot of GoIP Gateway.

The screenshot above shows the operation buttons. Button *Save* is used to save all the modifications while button reboot is used to save modifications first and then reboot device immediately.

5 Typical Used Scenario

This chapter presents some typical used scenarios for reference.

5.1 Landing from IP to Mobile Network

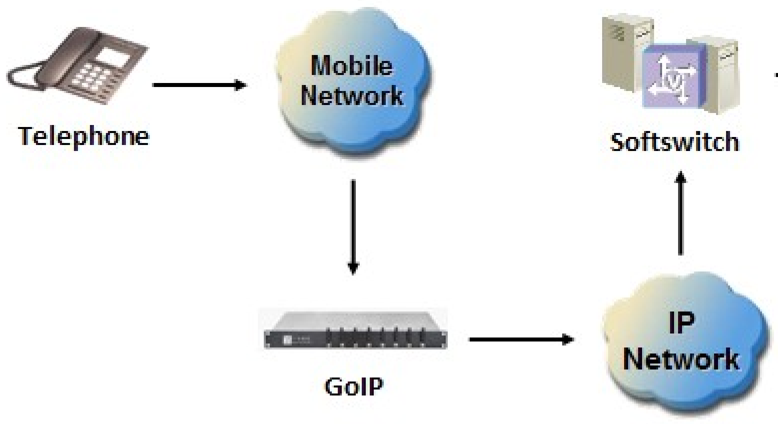


GoIP Gateway is now used more and more for telephone carriers to land their IP calls to mobile network. It plays the role of converting IP telephone signal to GSM telephone signal, relaying the media stream between IP network and Mobile network.

GoIP Gateway can be placed either in the LAN of Softswitch server or in public network environment which can be accessed by Softswitch server through public IP in different physical location.

Note: If the GoIP Gateway is placed in local LAN and accessed by Softswitch from another public IP, the functionality of **Media Relay MUST be enabled** to make sure the voice is working in full duplex mode. The following figure shows the Media Relay setting of VOS Softswitch.

5.2 Access from Mobile Network to IP



GoIP Gateway can be used as the access from mobile network to IP. Any call made to the mobile card inserted into GoIP Gateway will be routed to IP network and connected to Softswitch server. The Softswitch server can redirect the caller to final destination user.

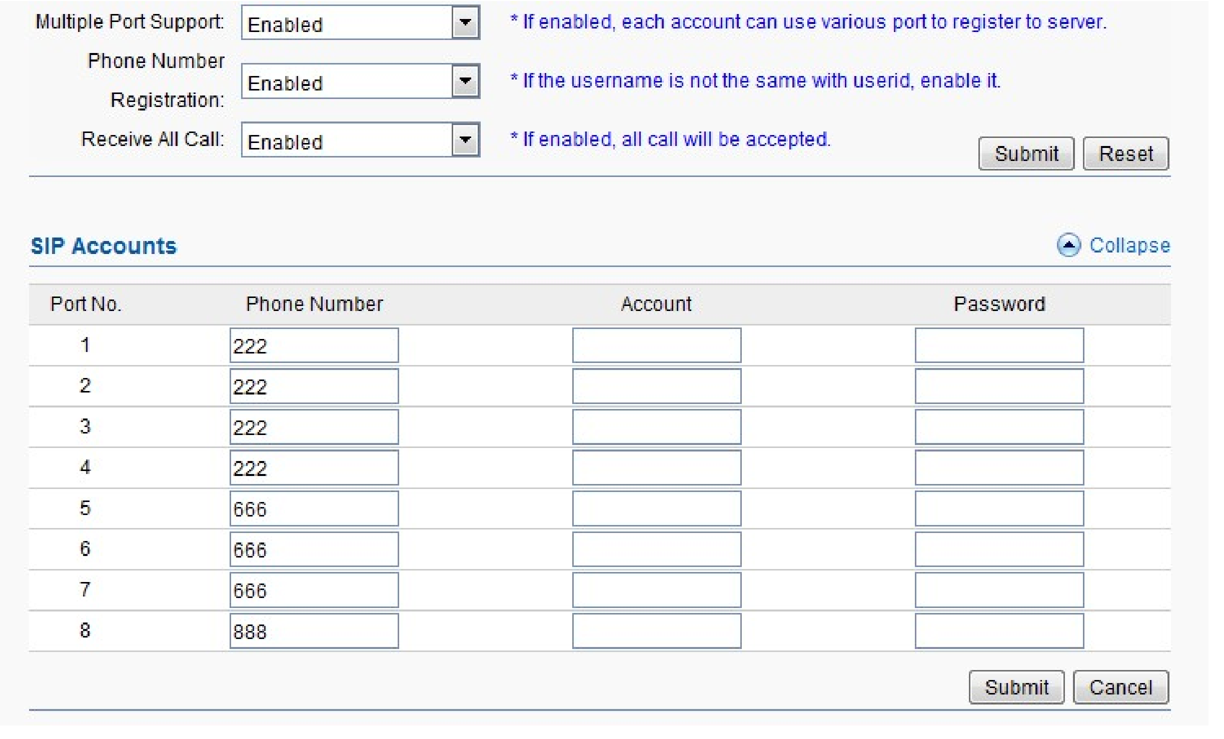
Note: If the GoIP Gateway is placed in local LAN and accessed by Softswitch from another public IP, the functionality of **Media Relay MUST be enabled** to make sure the voice is working in full duplex mode.

6 FAQ

This chapter presents the most frequently encountered issues and corresponding solutions.

6.1 How to designate a port for outbound call

Sometimes the outbound call to mobile network is required to be called through one or any one of a designated group of GoIP Gateway ports in order to reduce the cost. Here is an example to show you how to complete the settings for GoIP Gateway to make outbound call through a designated port.



Note:

1) Field *Multiple Port Support* must be enabled.

2) Field *Phone Number Registration* must be enabled.

3) Column *SIP Phone Number* in SIP Account list must be set. The Phone Number can be regarded as the SIP port phone number and can be called by other parties. This is the key point to outbound through a designated port.

4) Based on the example from the figure above, SIP Port 1 to 4 are grouped with phone number 222, SIP Port 5 to 7 is grouped with phone number 666 and SIP Port 8 is grouped with phone number 888. However, only one port exists in the third group.

   a) The outbound call whose destination is prefixed with 222, such as 22213512345678 will be routed to any of the SIP Port 1 to 4.

   b) The outbound call whose destination is prefixed with 666, such as 66613512345678 will be routed to any of the SIP Port 5 to 7.

c) The outbound call whose destination is prefixed with 888, such as 88813512345678 will be routed to SIP Port 8.

7 Appendix 1

This chapter shows the AT commands in detail.

惠州市粤讯网络科技有限公司
Huizhou YueXun Network Technology Co.,Limited

粤訊國際信息有限公司
YX International Information Co.,Limited

Email：

feekin@yx19999.com

Tel：

+86-400-777-0752（China）

+86-752-2777557; +86-752-2777558（Huizhou China）

+86-755-33599922（Shenzhen China）

+86-20-82404686 (Guangzhou China)

Website:

www.yx19999.net www.yx19999.com