



DVG-6001G User Manual v1.0

Contents

| | |
|--|----|
| 1. Equipment Introduction | 3 |
| 1.1 Overview | 3 |
| 1.2 Application solutions | 3 |
| 1.3 Product appearance | 5 |
| 1.4 Function characteristics | 6 |
| 1.4.1 Protocol | 6 |
| 1.4.2 System Function | 6 |
| 1.4.3 Industrial Standards Supported | 6 |
| 1.4.4 General Hardware Specification | 7 |
| 2. Equipment Installation | 7 |
| 2.1 Installation Notice | 7 |
| 2.2 Installation Procedure | 7 |
| 2.2.1 Install SIM Card | 7 |
| 2.2.2 Antenna Installation | 7 |
| 2.2.3 Cable Connection of Equipment | 8 |
| 3. Network Configuration | 8 |
| 3.1 Preparation | 8 |
| 3.2 Attentions | 9 |
| 3.3 General Feature Codes for System Setting | 9 |
| 3.4 Static IP Configuration | 9 |
| 3.5 DHCP Configuration | 10 |
| 4. WEB Configuration | 11 |
| 4.1 Preparing | 11 |
| 4.2 WEB Landing Interface | 11 |
| 4.3 WEB Configuration | 12 |
| 4.4 System | 12 |
| 4.4.1 System Information | 13 |
| 4.4.2 Mobile Information | 13 |
| 4.4.3 SIP Information | 14 |
| 4.5 Network Configuration | 15 |
| 4.5.1 Local network | 15 |
| 4.6 Mobile Configuration | 16 |
| 4.6.1 Basic Configuration | 16 |
| 4.6.2 Mobile Configuration | 17 |
| 4.6.3 PIN Management | 20 |
| 4.6.4 SMSC | 22 |
| 4.6.5 Send Message | 22 |
| 4.6.6 USSD | 23 |
| 4.6.7 Carrier | 24 |
| 4.6.8 BCCH | 24 |

| | |
|--|----|
| 4.7 Routing Configuration | 26 |
| 4.7.1 Routing Parameter..... | 26 |
| 4.7.2 Tel->IP Routing..... | 26 |
| 4.8 Manipulation Configuration..... | 27 |
| 4.8.1 IP->Tel Destination Numbers..... | 27 |
| 4.8.2 Tel->IP Source Numbers | 29 |
| 4.8.3 Tel->IP Destination Numbers..... | 31 |
| 4.9 Option | 32 |
| 4.9.1 IP->Tel Option | 32 |
| 4.9.2 Tel->IP Operation..... | 34 |
| 4.10 IP Trunk..... | 36 |
| 4.10.1 IP Trunk..... | 36 |
| 4.10.2 IP Trunk Group..... | 37 |
| 4.11 System Configuration..... | 38 |
| 4.11.1 System Configuration..... | 38 |
| 4.11.2 Service Configuration..... | 39 |
| 4.11.3 SIP Configuration..... | 42 |
| 4.11.4 Port Configuration..... | 45 |
| 4.11.5 Digit Map | 47 |
| 4.12 Tools..... | 49 |
| 4.12.1 Firmware Upload | 49 |
| 4.12.2 IVR Voice Prompt Upload | 49 |
| 4.12.3 Data Backup..... | 50 |
| 4.12.4 Data Restore | 50 |
| 4.12.5 Syslog Parameter..... | 51 |
| 4.12.6 Login Password..... | 52 |
| 4.12.7 Factory Reset..... | 52 |
| 4.12.8 Restart | 53 |
| 5. Glossary | 54 |

1. Equipment Introduction

This chapter mainly introduces functions and structures of DVG-6001G.

1.1 Overview

DVG-6001G is a full function GSM wireless gateway based on IP. It is able to offer stable network configuration, powerful function characteristics, excellent voice quality and provide affordable VoIP solutions for operators, enterprise, SOHO, home users.

To ensure that existing in structured cabling and ensure the safety of the existing network normal operation.

1.2 Application solutions

DVG-6001G provides wireless access service for custom. The following is a typical scheme network diagram.

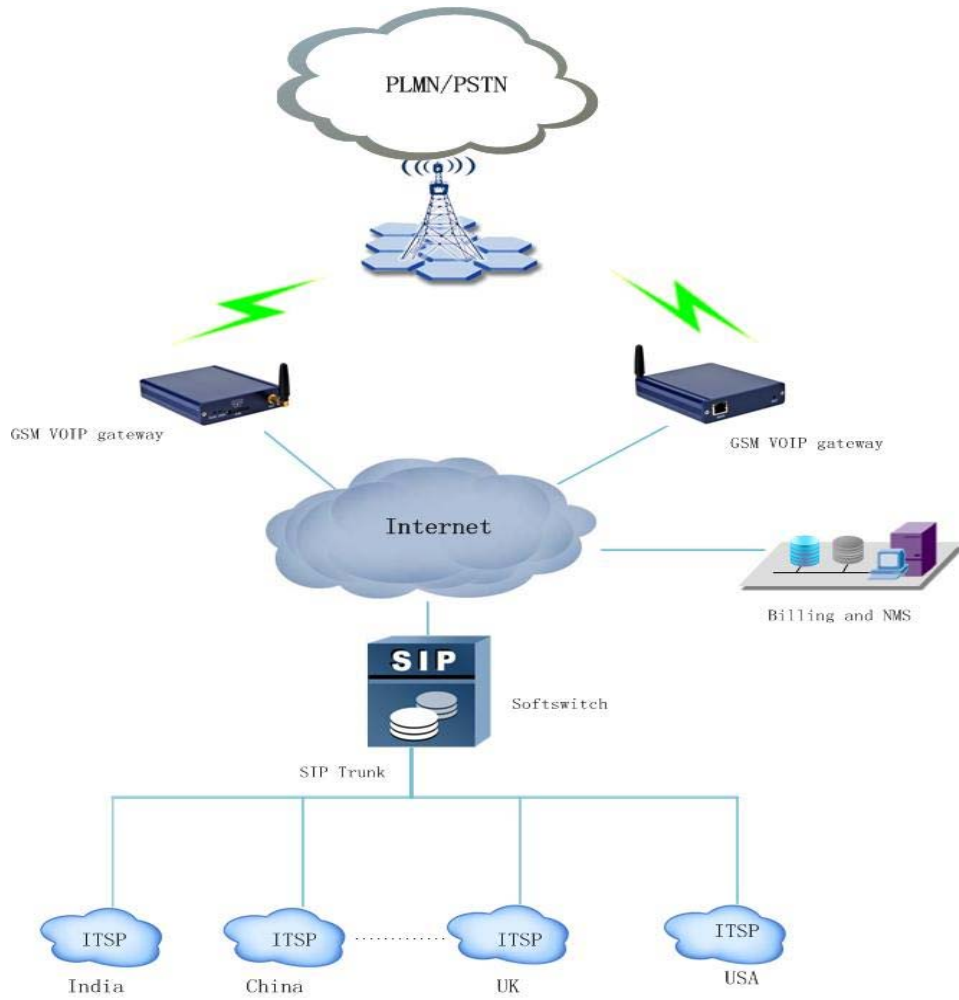


Figure 1-2-1 DVG-6001G application solution

1.3 Product appearance



Figure 1-3-1 DVG-6001G View before and after

Table 1-3-1 DVG-6001G interface introduction

| Interface | Discription |
|-----------|--|
| WAN | When connecting normal the WAN indicator will flash, while not register, please check whether the network connection is normal firstly. WAN used as a LAN under bridge mode. WAN port in transmits data at the same time the power is transmitted also, for this device provide DC power supply. |
| RST | Long click this button factory default recovery |
| SIM Slot | SIM card slot, please press the yellow button at the edge of slot, |
| ANT | External rubber lining the antenna, gain is 3dBi |

Table 1-3-2 DVG-6001G indicator light

| LED | Color | State | Description |
|-----|-------|--------------|--------------------------------------|
| RUN | Green | Slowly flash | SIP account does not have registered |
| | | Fastly flash | SIP account have registered |

| | | | |
|-----|-------|--------------|-----------------------------|
| GSM | Green | Destroy | SIP account does not have |
| | | Fastly flash | SIP account have registered |

1.4 Function characteristics

1.4.1 Protocol

- Standard SIP
- NAT
- PPPoE、HTTP、DHCP、DNS
- ITU-T G.711a-Law/μ-Law、G.723.1、G.729AB;

1.4.2 System Function

- PLC、VAD、CNG
- Support local and remote SIM card register
- Adjustment port gain
- Many DTMF mode
- Single talk time limit
- Total call time limit
- Talk time remaining reset
- Balance alarm
- SIM/UIM card encryption
- SMS sending and receiving
- Customized IVR
- Black/White list
- SMS、USSD Open API interface
- Echo Cancellation (with ITU-T G.168/165 standard)
- Automatic negotiate network
- Hotline
- Automatic restart module
- Lgeneration dial

1.4.3 Industrial Standards Supported

- Stationary use environment: EN 300 019: Class 3.1
- Storage environment: EN 300 019: Class 1.2
- Transportation environment: EN 300 019: Class 2.3
- Acoustic noise: EN 300 753
- CE EMC directive 2004/108/EC
- EN55022: 2006+A1:2007

- EN61000-3-2: 2006,
- EN61000-3-3: 1995+A1: 2001+A2: 2005
- EN55024: 1998+A1: 2001+A2: 2003
- Certifications: FCC, CE

1.4.4 General Hardware Specification

- Power Supply: 44-57V, 550mA MAX
- Temperature: 0~40 ℃ (Operation) , -20~80 ℃ (storage)
- Humidity: 5%~90%RH
- Power Consumption: 6.5-13W
- Dimensions: 120(W) x90(D) x24(H) mm
- Net weight: 0.8kg

2. Equipment Installation

This chapter mainly introduces DVG-6001G hardware installation and connection of equipment.

2.1 Installation Notice

1. DVG-6001G interfaces support RJ45 10/100Mbps
2. Directly into the SIM card, GSM channel can be started work

2.2 Installation Procedure

2.2.1 Install SIM Card

When installing SIM card, opening blank panel of SIM slot, procedure shows as below:

- Push down the yellow button, the SIM slot will popup;
- Inset the SIM card to the SIM slot.

2.2.2 Antenna Installation

Take antenna connected in antenna interface of DWG which sign of "ANT" on

2.2.3 Cable Connection of Equipment

DVG-6001G works in bridge mode:



Figure 2-2-1 DVG-6001G connection

3. Network Configuration

In this chapter we will introduce the initial configuration of DVG-6001G gateway. All of the network parameters of the gateway can be configured by IVR guidance.

3.1 Preparation

Please ensure the following steps are done properly before IVR setting:

1. Prepare an analog telephone or mobile phone
2. Make sure the gateway is connected with the network
3. Completed the SIM installation
4. Make sure that the current mobile network is working

3.2 Attentions

In each step, if user hears an IVR message of “setting successful”, which means that user has finished this step successfully. However, if user hears a “setting failed” message, please check redo that step again.

DVG-6001G can work in bridge mode user should configure network parameters of WAN port.

3.3 General Feature Codes for System Setting

Table 3-3-1 Feature codes for system setting

| Dial numbers | Features |
|---------------|--|
| *114# | Play the phone NO. |
| *150*a# | Set IP address(static/DHCP), a can be digit 1 or 2,*150*1# is static IP address mode, *150*2# is DHCP mode |
| *152*a*b*c*d# | Set IP address(static/DHCP), a can be digit 1 or 2,*150*1# is |
| *153*a*b*c*d# | Configure subnet mask. a, b, c, d are the four fields of the subnet mask |
| *156*a*b*c*d# | Configure the device gateway, a, b, c, d are the four fields of the device gateway |
| *158# | Report the IP address |
| *111# | Restart |

3.4 Static IP Configuration

Assuming the IP address of a DVG-6001G device is:

IP: 172.16.0.100 subnet mask: 255.255.0.0 gateway: 172.16.1.1

configured as follows:

1. Insert a SIM card into the DVG-6001G gateway
2. The configuration mode: Dial the phone number of this SIM card. hear a message, then enter “*150*1#”, hang up when hear “ setting successful" message;

3. Configure IP address: Dial the phone number of this SIM card, hear a message, enter "* 152 * 172 * 16 * 0 * 100 #" hang up when hear "setting successful" message;
4. configure subnet mask: Dial the SIM card phone number, enter "*153*255*255*0*0#" hang up when hear "setting successful" message;
5. Configure gateway: Dial the SIM card phone number, enter "*156*172*16*0*1#" hang up when hear "setting successful" message;
6. Please wait about ten seconds when finishing the operations, restart device. dial the SIM card phone number, enter "*158#"to check the Static IP address;

3.5 3.5 DHCP Configuration

DHCP mode configure as follows:

1. Insert a SIM card into a slot, dial the SIM card number. When hearing a hint message, then enter "*150*2#", if hearing " setting successful" message, which means the DHCP is confirued successfully;
2. Restart the device, wait for 30 seconds, and then dial the SIM card telephone number, enter "* 158 #" to query the IP address;

Note: If reporting the IP address is 0.0.0.0, which means that the gateway could not obtain a IP address successfully. Please check:

1. Make sure the device have been connected to the network;
2. Make sure the DHCP Server is working. If there is no DHCP Server, please set the IP of device to static IP .

4. WEB Configuration

This chapter describes web configuration of DVG-6001G.

4.1 Preparing

DVG-6001G connected to the local area network of PC used logged in with LAN.

4.2 WEB Landing Interface

The default IP of DVG-6001G is 192.168.11.1/ 255.255.255.0. Also can modify equipment IP by IVR. Before landing please ensure that with equipment connected PC's IP address and the IP address of the equipment in the same subnet. Enter IP address of DVG-6001G in browser.



Figure 4-2-1 web landing interface

Enter username and password and then click “OK” in configuration interface. The default username and password are “admin/admin”. It is strongly recommended, change the default password to a new password for system security.

4.3 WEB Configuration

DVG-6001G web configuration interface consists of the navigation tree and the detail configuration interfaces.

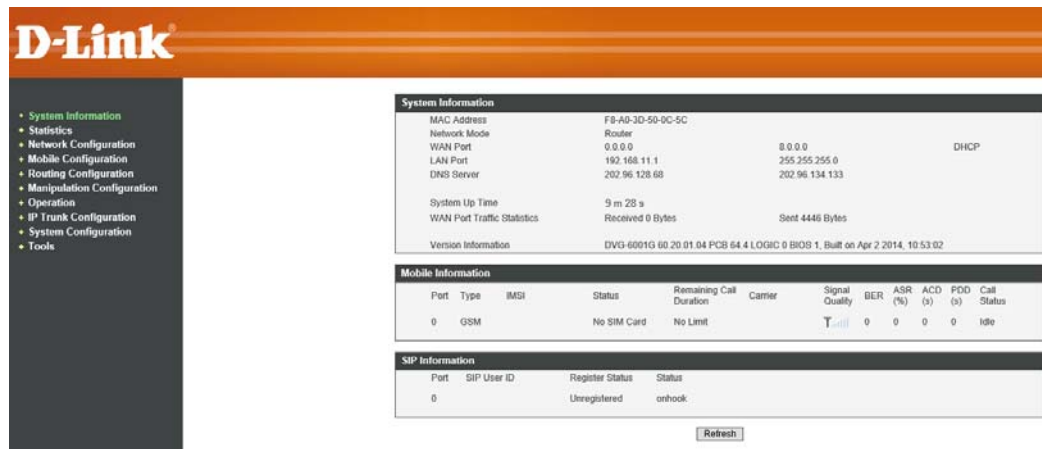


Figure 4-3-1 Web introduce

4.4 System

System information interface shows the basic information of system, Mobile information and SIP information.

| System Information | | | | | | | | | | | | |
|-----------------------------|--|----------------|--|--|--|-----------------|--|--|--|------|--|--|
| MAC Address | F8-A0-3D-50-0C-5C | | | | | | | | | | | |
| Network Mode | Router | | | | | | | | | | | |
| WAN Port | 0.0.0.0 | 8.0.0.0 | | | | | | | | DHCP | | |
| LAN Port | 192.168.11.1 | 255.255.255.0 | | | | | | | | | | |
| DNS Server | 202.96.128.68 | 202.96.134.133 | | | | | | | | | | |
| System Up Time | 10 m 28 s | | | | | | | | | | | |
| WAN Port Traffic Statistics | Received 0 Bytes | | | | | Sent 4446 Bytes | | | | | | |
| Version Information | DVG-6001G 60.20.01.04 PCB 64.4 LOGIC 0 BIOS 1, Built on Apr 2 2014, 10:53:02 | | | | | | | | | | | |

| Mobile Information | | | | | | | | | | | |
|--------------------|------|------|-------------|-------------------------|---------|----------------|-----|---------|---------|---------|-------------|
| Port | Type | IMSI | Status | Remaining Call Duration | Carrier | Signal Quality | BER | ASR (%) | ACD (s) | PDD (s) | Call Status |
| 0 | GSM | | No SIM Card | No Limit | | | 0 | 0 | 0 | 0 | Idle |

| SIP Information | | | |
|-----------------|-------------|-----------------|--------|
| Port | SIP User ID | Register Status | Status |
| 0 | | Unregistered | onhook |

Refresh

Figure 4-4-1 System Information

4.4.1 System Information

| System Information | | | |
|-----------------------------|--|-----------------|------|
| MAC Address | F8-A0-3D-50-0C-5C | | |
| Network Mode | Router | | |
| WAN Port | 0.0.0.0 | 8.0.0.0 | DHCP |
| LAN Port | 192.168.11.1 | 255.255.255.0 | |
| DNS Server | 202.96.128.68 | 202.96.134.133 | |
| System Up Time | 11 m 35 s | | |
| WAN Port Traffic Statistics | Received 0 Bytes | Sent 4446 Bytes | |
| Version Information | DVG-6001G 60.20.01.04 PCB 64.4 LOGIC 0 BIOS 1, Built on Apr 2 2014, 10:53:02 | | |

Figure 4-4-2 System Information

Table 4-4-1 System Information Describe

| | |
|----------------------------|---|
| MAC Address | Display the current MAC of the gateway, for example: 00-01-0C-03-A4-2F |
| Network Mode | DVG-6001G only support “bridge” mode, please reference network configure |
| Network | Display IP, subnet mask and the way of obtain IP address |
| DNS Server | Display DNS server IP address |
| System Up Time | Display the time period of the device running |
| Network Traffic Statistics | Calculates the netflow, including the total bytes of message received and sent |
| Version information | Display the version information, include: product model, firm ware version, hardware version and date compiled. |

4.4.2 Mobile Information

Display GSM channel and network status information, detailed shown as below:

| Mobile Information | | | | | | | | | | | |
|--------------------|------|------|--------|-------------------------|---------|----------------|-----|---------|---------|---------|-------------|
| Port | Type | IMSI | Status | Remaining Call Duration | Carrier | Signal Quality | BER | ASR (%) | ACD (s) | PDD (s) | Call Status |
| 0 | | | | No Limit | | | 0 | 0 | 0 | 0 | Idle |

Figure 4-4-3 Mobile Information

Table 4-4-2 Mobile Information

| | |
|------------------------------|--|
| Port | Numbers of ports of GSM |
| Type | The current type of network. Such as CDMA or GSM |
| IMSI | International Mobile Subscriber Identity, it is the uniquely identifies of SIM card |
| Status | Indicates the connection status of current GSM module |
| Remaining Call Duration(min) | Limite a call duration to the SIM card, when call duration is out of that duration, the call would be discontinued. This option shows remaining talk time. |
| Carrier | Display the network carrier of current SIM card. |
| Signal Quality | Displays the signal strength of in each channels of GSM |
| BER | Bit erro rate |
| ASR (%) | Average connection rate |
| ACD | Average call duration |
| PDD | Delay between call sent out and call connected |
| Call Status | Show the status of call, include idle, handle, hangup, call such status |

4.4.3 SIP Information

| SIP Information | | | |
|-----------------|-------------|-----------------|--------|
| Port | SIP User ID | Register Status | Status |
| 0 | 102 | Registered | onhook |

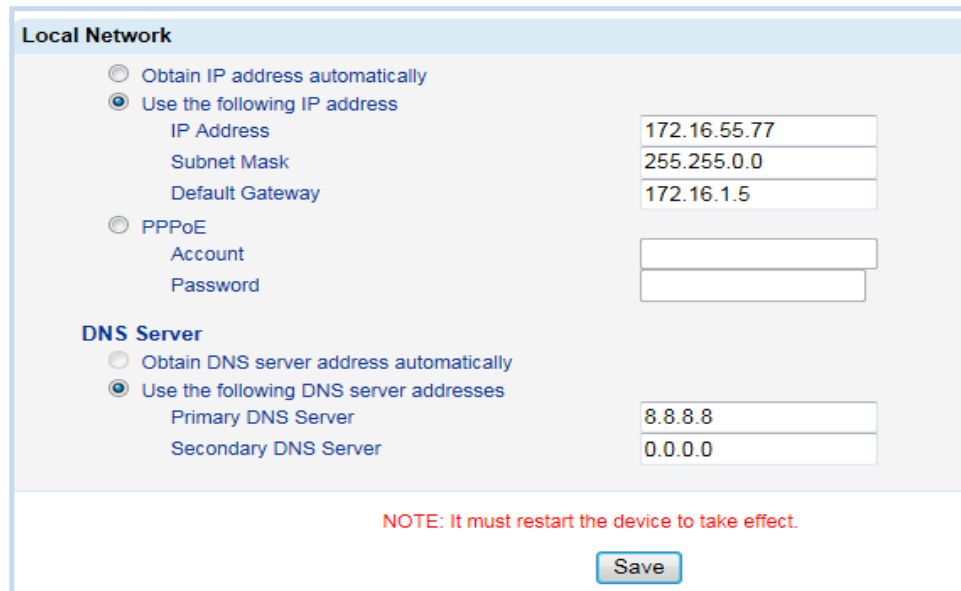
Figure 4-4-4 SIP Information

Table 4-4-3 SIP Information

| | |
|-----------------|--|
| Port | The corresponding GSM port, DVG-6001G has only 1 port |
| SIP User ID | SIP registration account of the Softswitch and SIP server provided |
| Register Status | Show the registration status of VoIP channel, including registered and unregistered. |
| Status | The status of Off-hook and hang up |

4.5 Network Configuration

4.5.1 Local network



Local Network

Obtain IP address automatically
 Use the following IP address
 IP Address: 172.16.55.77
 Subnet Mask: 255.255.0.0
 Default Gateway: 172.16.1.5

PPPoE
 Account:
 Password:

DNS Server

Obtain DNS server address automatically
 Use the following DNS server addresses
 Primary DNS Server: 8.8.8.8
 Secondary DNS Server: 0.0.0.0

NOTE: It must restart the device to take effect.

Figure 4-5-1 Local network

Table 4-5-1 Local network

| Work mode | Only Bridge mode |
|---|---|
| Obtain IP Address Automatically | After used, the IP address obtained from DHCP server |
| Use the following IP address | After used, need to manually add IP address, subnet mask and default gateway |
| PPPoE | When adopt PPPoE dial-up Internet, need to fill in account and password offered by ISP |
| Obtain DNS Server Address Automatically | DNS server complete the analytical between domain name and IP address. When enable "Obtain DNS Server Address Automatically", which will be automatically get DNS server address. |
| Use the Following DNS Server | Fill in the IP address of "Primary DNS Server" and |

| | |
|-----------|------------------------|
| Addresses | "Secondary DNS Server" |
|-----------|------------------------|

4.6 Mobile Configuration

4.6.1 Basic Configuration

Basic Configuration

Dial Tone Gain (Mobile Side) dB

Select Band ▼

Remote API Enable No Yes

API Server Address

API Server Port

API User ID

API User Password

Auto Reset Module No Yes

Counts of NO CARRIER to reset

Counts of NO DIALTONE to reset

NOTE: Option 'Reject Incoming' will be disabled, When 'yes' is checked on option 'Forward Enable'.

Figure 4-6-1 Basic Configuration

Table 4-6-1 Basic Configuration

| | |
|--------------------|--|
| Dial Tone Gain | It is the dial tone volume of call waiting, dial tone of mobile module when call out. Usually adopt the default configuration. |
| Select Band | According to carrier's band standards, standards are as below: PGSM900, DCS1800, PCS1900, EGSM900/DCS1800, GSM850/PCS1900 |
| Remote API Enable | API is provided interface for third party development with DLL and IAD components. Includes SMS/USSD sending and receiving. If want to use the client to send text message, please open API. |
| API Server Address | It is the remote IP address who uses API. This is an option when selecting "Yes" under 'remote API enable' |

| | |
|-----------------------------------|---|
| API Server Port | It is the remote channel No. who uses API. This is an option when selecting "Yes" under "remote API enable". The user can defined a not overlap with the other application port of the port number, the proposal value is 12000 |
| API User ID | Remote API user account. This is an option when selecting "Yes" under "remote API enable". |
| API User password | Remote API user password. This is an option when selecting "Yes" under "remote API enable". |
| Auto Reset Module | Open the function, in the following case module can be reboot |
| Counts of No CARRIER to reset | Continuously n times can't find operators, equipment to restart. N is 3-255. |
| Counts of No DIALTONE to reset | Continuously n times no dialtone, module to restart. N is 3-255. |

4.6.2 Mobile Configuration

| Mobile State | | | | | | |
|--------------|------------------------|-----------------|---------|---------|------------------------------|------------------------|
| Port | Single Call Limitation | Call Limitation | Tx Gain | Rx Gain | Reset Module | Detail |
| 0 | No | No | 6 | 6 | Reset Module | Detail |

Figure 4-6-2 Mobile Configuration

Mobile Configuration

Select Port

Mobile Number

Step sec

Enable Call Duration Limitation of single call No Yes

Time of single call

Enable Call Duration Limitation No Yes

Auto Reset No Yes

Reset Period

Next Reset time Year Month Day Hour Min

Maximum Call Duration

Minimum Charging Time sec

Alarm Threshold (via SMS)

Mobile Number (Receiving Alarm)

Port Description for Alarm

SIM Remain Time

CLIR No Yes

Echo Suppression Level

Mobile Tx Gain dB

Mobile Rx Gain dB

Detect Reverse Polarity No Yes

Figure 4-6-3 Mobile Configuration

Table 4-6-2 Mobile Configuration

| | |
|--|---|
| Mobile Number | Corresponding port SIM card number |
| Enable Call Duration Limitation of single call | This function is to limit the max call duration of channel. Users can customize the SIM card on the single call duration; if more than the duration, call will be take out stitches. If select “Yes”, then need to set the following two options. |
| Step | Step length value range is 1-120 s, step length multiplied by time of single call just said a single call duration time allowed. |
| Time of single call | The value of limitation single call, this value range is 1-65535. step length multiplied by time of single call just said a single call |

| | |
|---------------------------------|--|
| | duration time allowed. |
| Enable Call Duration Limitation | This function is to limit the max call duration of channel. The max call duration is between 1 to 65535 minutes. |
| Auto Reset | Automatic reset talk time remaining, Let remaining call time is equal to the maximum call duration. |
| Reset Period | User defined daily, weekly or monthly reset SIM card information, that is, remaining call time is equal to the maximum call duration and start counting. |
| Next Reset time | The user defined when to begin to reset, then from the date according to reset period reset. |
| Minimum Charging Time | A single call over this time, GSM side of the operators began to collect fees, unit for seconds. |
| Alarm Threshold(via SMS) | Talk time remaining is equal to or lower than the value, the gateway to the alarm information by SMS messages to the designated mobile phone number. |
| Mobile Number (Receiving Alarm) | Receiving alarm phone number, user will received alarm message from gateway. |
| Port Description for Alarm | Alarm port information description, which will be sent to user mobile phone with alarm information. |
| SIM Remain Time | This value is multiplied by to step length is a rest call time |
| Restore Time | Restore the rest of the SIM card talk time to the maximum call duration |
| CLIR | This function is used to GSM side exhale hidden SIM card number. Adding a "#31#" in front of mobile phone number can realize the function. This function need operators support. |
| Echo Suppression Level | Control echo of call process. The higher the level, the more powerful the echo suppression. |

| | |
|-------------------------|---|
| Mobile Tx Gain | Control IP to GSM side of call the gain. Default is 6dB. |
| Mobile Rx Gain | Control GSM to IP side of call the gain. Default is 6dB. User can adjust the two gain to adjust the size of the voice. |
| Detect Reverse Polarity | To GSM module is invalid, in the role of CDMA module, the local CDMA network support open when the extremely. When not open this function, use a overtime time to report a fake the extremely, overtime time for response time delay, see business configuration parts. |

4.6.3 PIN Management

The screenshot shows a web interface titled "PIN Management". It contains a "Select Port" dropdown menu set to "Port 0". Below it, there are labels for "SIM Card Lock" and "PIN Code". The "SIM Card Lock" option has two radio buttons: "No" (which is selected) and "Yes". A text input field for the "PIN Code" is present but empty. At the bottom right, there is a "Save" button.

Figure 4-6-3 PIN Management

This screenshot is similar to Figure 4-6-3, but the "Yes" radio button for "SIM Card Lock" is selected. Additionally, a "Change PIN" button is visible next to the "Save" button at the bottom.

Figure 4-6-4 PIN Management

Table 4-6-3 PIN Code Management

| | |
|---------------|---|
| Select Port | Selection need locked channel number |
| SIM Card Lock | To prevent the SIM card is the use of others, user can lock SIM card. |
| PIN Code | Locked or unlocked SIM card need to input PIN code |
| Change PIN | Click this button to modify PIN |

Figure 4-6-5 Change PIN Code

PIN is Personal Identification Number of SIM card. Here the PIN code changed.

Figure 4-6-6 PIN code to unlock

When the PIN code three consecutively input error, system will tip input PUK yards, and reinstall new PIN code.

Table 4-6-5 PIN code to unlock

| | |
|---------------------------------|---|
| Select Port | Select GSM port needed input PUK code |
| PUK Code | PIN Unlocking Key is the PIN code unlock code. PIN code three consecutive input error, SIM card will be locked, need to unlock the PUK yards. PUK yards of input opportunity is 10 times, 10 times all lose correctly, SIM card will be locked to the permanent, that is discarded. |
| Please input new PIN code | Set a new PIN code |
| Please input new PIN code again | Again confirmed the new PIN code. |

4.6.4 SMSC

Figure 4-6-7 SMSC

Mobile phone text message center, in theory the wireless module can automatically detect the SMS center number. But when wireless module can't automatically detect the SMSC number, please contact mobile network operators, and manual Settings SMSC number.

4.6.5 Send Message

Figure 4-6-8 Send Message

Table 4-6-6 Send Message

| | |
|-------------|--|
| Select Port | From the designated port can send, also can choose random ports to send |
| Encoding | SMS code can be used in two ways, UCS2 and GSM 7bit. Editor pure English short message can use GSM 7 bit, otherwise, use UCS2. |
| To | Mobile phone number received SMS |

| | |
|---------|--|
| Message | The content of the messages, the length is not more than 300 characters. |
|---------|--|

4.6.6 USSD

The screenshot shows a web-based USSD interface. At the top, there's a header 'USSD'. Below it, there are three columns: 'Port', 'USSD Request', and 'USSD Reply'. The 'Port' field contains the number '0'. The 'USSD Request' field is empty. The 'USSD Reply' field contains the text 'not registered'. Below these fields, there is a red text note: 'NOTE: If you do nothing within 90s, connection will be disconnected.' At the bottom of the interface, there are two buttons: 'Send' and 'Exit'.

Figure 4-6-9 USSD

USSD (Unstructured Supplementary Service Data) is a new type of based on GSM network interactive data business. When using a mobile phone keyboard input some prescribed number or symbols such as * #, etc, then press the dial-up key, mobile phone will send an instruction to network. According to instructions, network choice special services to you. USSD technology used alone or in combination with the current short message technology, General Packet Radio Service GPRS (General Packet Radio Service) technology combined to provide various value-added services, such as Mobile Banking, Financial stock trading, Mobile phone calls inquires, Meteorological information prediction and query, Send and Receive Email, Flight Track, Booking Tickets Online etc.

4.6.7 Carrier

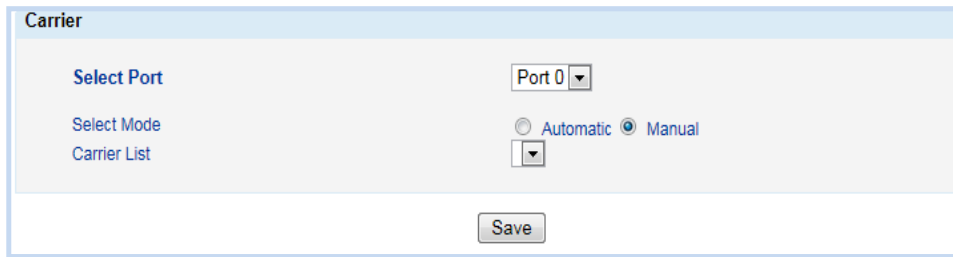


Figure 4-6-10 Carrier

Table 4-6-7 Carrier

| | |
|--------------|--|
| Select Port | Select a SIM card |
| Select Mode | There are automatic and manual two mode. Automatic mode can detect carrier automatically; manual mode will select carrier from drop-down list. |
| Carrier List | Here will list all detected operators |

4.6.8 BCCH



Figure 4-6-11 BCCH

Figure 4-6-12 BCCH

Table 4-6-8 BCCH Description

| | |
|------------------|---|
| Refresh Interval | Set BCCH parameters automatically refresh time |
| Index | Base station parameters numbers |
| MCC | Mobile Country Code, China is 460 |
| MNC | Mobile Network Code, used to distinguish between different network operators. |
| LAC | Location area number, in order to determine the position of the mobile station, each GSM PLMN coverage area is divided into many location area, location area codes (LAC) is used to identify the different location area. |
| CID | To the only to express the GSM PLMN every community, network operators should be assigned to the network of all the village a code, that is, CI, CI and LAI yards combined, used to identify each of the network and its coverage of the village BTS. |
| BCCH | Broadcast Control Channel, general information transmission, used for mobile measurement signal strength and identify district mark etc. |
| Receive Level | The base level received signal from BTS |
| Lock | Signal can be locked in a few signal good base station, selected to lock base station, click on the lock. If base station signal is very |

| | |
|--------|--|
| | poor that has locked, signal will also automatically switch to other stations. |
| Unlock | Unlock the base station that has locked. |

4.7 Routing Configuration

4.7.1 Routing Parameter

The screenshot shows a web interface for configuring routing parameters. The title is 'Routing Parameter'. Below the title, there is a label 'Tel->IP Parameter' and a dropdown menu currently showing 'Route calls before manipulation'. At the bottom of the form is a 'Save' button.

Figure 4-7-1 Rout Parameter

Table 4-7-1 Rout Parameter

| | |
|---------------------------------|--|
| Tel->IP Parameter | Routing parameter from GSM to IP |
| Route calls before manipulation | First routing, after number transformation |
| Route calls after manipulation | First number transformation, after routing |

4.7.2 Tel->IP Routing

Figure 4-7-2 Tel ->IP Routing

The screenshot shows a table titled 'Tel->IP Routing'. The table has five columns: Index, Description, Source Prefix, Destination Prefix, and Destination. The first row contains the values: 0, default, any, any, and SIP Server. Below the table, there are pagination controls showing 'Total: 1entry 16entry/page 1/1page Page 1' and three buttons: 'Add', 'Delete', and 'Modify'. A red note at the bottom reads: 'NOTE: 0 routing is not allowed to delete, only allowed to change.'

Table 4-7-2 Tel ->IP Routing

| | |
|------------------|--|
| Tel ->IP Routing | This item uses to configure incoming call routes which can be used for recieve the calls from the GSM. |
| Index | It uniquely identifies a route. Its value is assigned globally, ranging from |

| | |
|--------------------|--|
| | 0 to 31. |
| Description | It describes the route for the ease of identification. Its value is character string |
| Source Prefix | When calling number matching the prefix, this routing will take effect. Any: indicates any number 0xxxx: All of the number of begin to 0 |
| Destination Prefix | When callee number matching the prefix, this routing will take effect. |
| Destination | Specify the specific IP, IP group and SIP Server |

Click “Add” or “Modify” enter the following interface.

Figure 4-7-3 Tel->IP Routing Modify

4.8 Manipulation Configuration

4.8.1 IP->Tel Destination Numbers

| Index | Description | Source | Source Prefix | Destination Prefix | Stripped Digits from Left | Stripped Digits from Right | Prefix to Add | Suffix to Add | Number of Digit to Leave from Right |
|-------|-------------|--------|---------------|--------------------|---------------------------|----------------------------|---------------|---------------|-------------------------------------|
| 0 | 1 | Any | any | any | 0 | 0 | 6311 | --- | 0 |

Figure 4-8-1 IP->Tel Destination Numbers

Table 4-8-1 IP->Tel Destination Number

| | |
|-------------------------------------|---|
| Manipulation | This option can modify the Lord called number pass by gateway |
| Index | It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31. |
| Description | It describes the route for the ease of identification. Its value is character string |
| Source | It specifies the source IP which will send the calls to gateway |
| Source Prefix | All the caller number must match the source prefix. It specifies the source prefix allow to send call out <ul style="list-style-type: none"> • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186 |
| Destination Prefix | All the called number must match the destination prefix, the call prefix indicates the connected number <ul style="list-style-type: none"> • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186 |
| Stripped Digits from Left | It specifies the length of the digits to be deleted from left |
| Stripped Digits from Right | It specifies the length of the digits to be deleted from right |
| Prefix to Add | Add the new digits in front of the original number |
| Suffix to Add | Add the new digits at the end of the original number |
| Number of digit to leave from right | From the right side began to retain the digits |

Figure 4-8-2 IP->Tel Destination Number Add

For example, source prefix is 0123, after transform, the prefix become 23.

4.8.2 Tel->IP Source Numbers

Figure 4-8-3 Tel->IP Source Numbers

Table 4-8-3 Tel->IP Source Numbers

| | |
|-------------|---|
| Index | It is the number of the transformation and said number transformation rule label the priority. Value range is 0-31. |
| Description | It describes the route for the ease of identification. Its value is character string. |

| | |
|--------------------------------------|--|
| Source Prefix | <p>All the caller number must match the source prefix. It specifies the source prefix allow to send call out</p> <ul style="list-style-type: none"> • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186 |
| Destination Prefix | <p>All the called number must match the destination prefix, the call prefix indicates the connected number</p> <ul style="list-style-type: none"> • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186 |
| Destination | Appoint number destination: IP, IP group or SIP server |
| Stripped Digits from Left | It specifies the length of the digits to be deleted from left |
| Stripped Digits from Right | It specifies the length of the digits to be deleted from right |
| Prefix to Add | Add the new digits in front of the original number |
| Suffix to Add | Add the new digits at the end of the original number |
| Number of Digits to Leave from Right | It specifies the number of Digits to Leave from Right |

Tel->IP Source Numbers Add

Index: 31

Description: []

Source Prefix: []

Destination Prefix: []

Destination:

 IP: Any

 IP Group: []

 SIP Server

Stripped Digits from Left: []

Stripped Digits from Right: []

Prefix to Add: []

Suffix to Add: []

Number of Digits to Leave from Right: []

NOTE: If you need route calls after manipulation, set the destination ip to any.

OK Reset Cancel

Figure 4-8-4 Tel->IP Source Numbers Add

4.8.3 Tel->IP Destination Numbers

Tel->IP Destination Numbers

| Index | Description | Source Prefix | Destination Prefix | Destination | Stripped Digits from Left | Stripped Digits from Right | Prefix to Add | Suffix to Add | Number of Digits to Leave from Right |
|-------|-------------|---------------|--------------------|-------------|---------------------------|----------------------------|---------------|---------------|--------------------------------------|
| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |

Total: 0entry 16entry/page 1/0page []

Add Delete Modify

Figure 4-8-5 Tel->IP Destination Numbers

Tel->IP Destination Numbers Add

Index:

Description:

Source Prefix:

Destination Prefix:

Destination: IP IP Group SIP Server

Stripped Digits from Left:

Stripped Digits from Right:

Prefix to Add:

Suffix to Add:

Number of Digits to Leave from Right:

NOTE: If you need route calls after manipulation, set the destination ip to any.

Figure 4-8-6 Tel->IP Destination Numbers Add

Please reference Tel->IP Source Numbers. Matching rules completely the same.

4.9 Option

4.9.1 IP->Tel Option

| IP->Tel Operation | | | | | | |
|--|-----------|---------------|--------------------|-----------|-------------|-----|
| Index | Source IP | Source Prefix | Destination Prefix | Operation | Description | |
| --- | --- | --- | --- | --- | --- | --- |
| Total: 0entry 16entry/page 1/0page <input type="text" value="1"/> | | | | | | |
| <input type="button" value="Add"/> <input type="button" value="Delete"/> <input type="button" value="Modify"/> | | | | | | |

Figure 4-9-1 IP->Tel Option

Table 4-9-1 IP->Tel Option

| | |
|-------------------|--|
| IP->Tel Operation | This is an optional configuration items, when using the hotline, this item must be configured. Include: forbid call, call allowance, auto call, and password authentication. |
|-------------------|--|

| | |
|--------------------|---|
| Index | Number, value range from 0-31. |
| Source IP | <p>It specifies the source IP which will send the calls to gateway</p> <ul style="list-style-type: none"> • Any: any IP address • IP: specific an IP address • IP Group: specific an IP group |
| Source Prefix | <p>All the caller number must match the source prefix. It specifies the source prefix allow to send call out</p> <ul style="list-style-type: none"> • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6: consist of some prefix, include 136,146,156,166,176, 186 |
| Destination Prefix | <p>All the called number must match the destination prefix, the call prefix indicates the connected number</p> <ul style="list-style-type: none"> • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6: consist of some prefix, include 136,146,156,166,176, 186 |
| Operation | <p>Its specifies number analysis rule</p> <ul style="list-style-type: none"> • Forbid call • Allow call • Auto call • Password authenticate |
| Description | <p>It describes the route for the ease of identification. Its value is character string</p> |

Figure 4-9-2 IP->Tel Operation Add

For example: The above configuration said: Allow all calls from SIP server.

4.9.2 Tel->IP Operation

| Index | Source Prefix | Destination Prefix | Operation | Description |
|-------|---------------|--------------------|--------------------|-------------|
| 31 | any | any | Allow ,Auto Call , | a |

Total: 1entry 16entry/page 1/1page Page 1

Add Delete Modify

Figure 4-9-3 Tel->IP Operation

Table 4-9-2 Tel->IP Operation

| | |
|---------------|--|
| Index | Number, value range from 0-31 |
| Source Prefix | <p>All the caller number must match the source prefix. It specifies the source prefix allow to send call out</p> <ul style="list-style-type: none"> Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. 0xxxx: consist of some digits such as 015,08,09 1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186 |

| | |
|--------------------|---|
| Destination Prefix | <p>All the called number must match the destination prefix, the call prefix indicates the connected number</p> <ul style="list-style-type: none"> • Any: include anonymous, 0xxxx, 1[2-9]xxxx etc. • 0xxxx: consist of some digits such as 015,08,09 • 1[3-8]6: consist of some prefix, include 136,146,156,166,176, 186 |
| Operation | <p>Its specifies number analysis rule</p> <ul style="list-style-type: none"> • Forbid call • Allow call • Auto call • Password authenticate |
| Description | <p>It describes the route for the ease of identification. Its value is character string</p> |

Figure 4-9-4 Tel->IP Operation

Pictured above, allow all call generation from port dial a number to IP side.

4.10 IP Trunk

4.10.1 IP Trunk

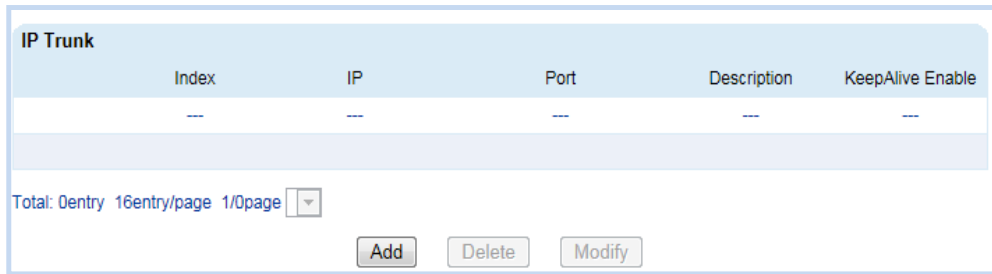


Figure 4-10-1 IP Trunk

Table 4-10-1 IP Trunk

| | |
|-------------|---|
| IP Trunk | When device connected to softswitch or SIP server, equipment and the soft switch exchange information through the IP trunk. |
| Index | Number, value range from 0-31 |
| Description | It describes the route for the ease of identification. Its value is character string |
| IP | To end the soft switch or SIP server IP |
| Port | To end the soft switch or SIP server port |

Figure 4-10-2 IP Trunk Modify

For example, No.31 trunk to connect to SIP server of 172.16.100.103:5060.

4.10.2 IP Trunk Group

Figure 4-10-3 IP Trunk Group

Table 4-10-2 IP Trunk Group

| | |
|----------------|--|
| IP Trunk Group | This configuration is optional, and is used to add the IP that have the same attributes to an IP group. The IP group will referenced by IP->Tel routing and number manipulation. |
| Index | Number, value range from 0-31 |
| Description | It describes the route for the ease of identification. Its value is character string |
| IP | It specifies the IP will add to IP group. When building the trunk later can choose some trunk form a trunk group. |

Figure 4-10-4 IP Trunk Group Add

For example, NO.31 trunk group has only one trunk NO.31, a trunk group can have multiple relay, a relay can only belong to a trunk group.

4.11 System Configuration

4.11.1 System Configuration

| System Configuration | |
|--------------------------------|--|
| Provision Configuration | |
| Primary Profile URL | <input type="text" value="http://172.16.100.139/temp/test.xml"/> |
| Secondary Profile URL | <input type="text"/> |
| Username | <input type="text" value="admin"/> |
| Password | <input type="password" value="•••••"/> |
| NTP Configuration | |
| Enable NTP | <input type="radio"/> No <input checked="" type="radio"/> Yes |
| Primary NTP Server IP | <input type="text" value="64.236.96.53"/> |
| Secondary NTP Server IP | <input type="text" value="18.145.0.30"/> |
| Time Zone | <input type="text" value="GMT-6:00 (US Central Time, Chicago)"/> |

NOTE: It must restart the device to take effect.

Figure 4-11-1 System Configuration

Network Time Protocol (NTP) is a protocol used to make computer time synchronization. It can make the computer to the server or clock source do synchronization, provide high precision accuracy of time correction.

4.11.2 Service Configuration

Service Configuration

Local Start RTP Port

Enable Silence Suppression No Yes

Call Progress Tone

Preferred Coders(in listed order)

1st

2nd

3rd

4th

Voice Frames per Tx

Do Not Answer PSTN Incoming Call for Hotline No Yes

Enable PSTN Incoming Configuration No Yes

Auto Outgoing Routing Type

IP to PSTN One Stage Dialing No Yes

Answer Delay s

Redirect Call When All Ports Busy No Yes

Play Voice Prompt for PSTN Incoming Calls No Yes

DTMF Parameter

DTMF Method

DTMF Volume

DTMF Interval ms

NAT Traversal

Other Configuration

Enable Private Service No Yes

User ID Is Phone Number No Yes

Only Accept Calls from SIP Server No Yes

Allow Call from PSTN to IP without Registration No Yes

Allow Call from IP to PSTN without Registration No Yes

Reject Anonymous Call from IP to PSTN No Yes

Use # as End Key No Yes

No Answer Timeout s

Interdigit Timeout s

Call Delay s

NOTE: 1. Answer Delay is only valid when Detect Polarity Reversal is not enabled.
2. It must restart the device to take effect.

Figure 4-11-2 Service Configuration

Table 4-11-2 Service Configuration

| | |
|-----------------------------|---|
| <p>LOCAL Start RTP PORT</p> | <p>Real-time Transport Protocol details the standard packet format on the Internet to deliver audio and video. The initial allocation of Channel when RTP voice stream transmit in the IP</p> |
|-----------------------------|---|

| | |
|--|---|
| | network. In general, using the factory default values. When there are multiple D-LINK series voice products, and the network gateway or router's NAT with loopholes, user can try changing this item. |
| Enable Silence Suppression | Silence suppression technique to ensure that only wehn talking to both sides in call to take up bandwidth and improve the utilization rate of channel. Enable the "silence suppression" almost no impact on call quality, and can save about half of the bandwidth. |
| Call progress Tone | Each country has its different call progress tone required standards, such as busy tone, ring back tones and ring tone standards, users can select the area standard from here . |
| Preferred Coders | Means the code format when Voice transfer on IP network, support PCMA, PCMU, G.723.1 andG.729AB. |
| Do Not Answer PSTN Incoming Call for Hotline | Inbound hotline immediately pick up, after waiting for VOIP side picked . |
| Enable PSTN Incoming Configuration | Select "Yes", users can configure the device through dialing feature codes. |
| Auto Outgoing Routing Type | When adopt two stage dialing, this configuration option takes effect and routing doesn't take effect. Ordinary mode means Minimum port selected. Polling means that according to the port in turn choice. |
| IP to PSTN One Stage Dialing | This function will be displayed only when select "Enable Auto Outgoing Routing" function, the User ID will be sent directly to PSTN, for example: the user calls 6715, the device will sent 6715 User ID to PSTN |
| Answer Delay | Only for CDMA effect, when IP to PSTN call and don't open the |

| | |
|---|---|
| | extremely signal detection, then call send out after a few seconds delay to connect. |
| Redirect Call When All Ports Busy | When all the GSM port occupied, this function will switch the call to another equipment and need to provide the device IP and port. |
| Play Voice Prompt for PSTN Incoming Calls | Setting is yes, when through the PSTN calls to the Channel, the device will with the clew tone, the default is "Please dial the extension User ID"; setting to No, the device will with dial tone |
| DTMF Parameter | Support RFC2833 and SIGNAL two ways. DTMF INTERVAL range is 50 ~ 800ms, DTMF VOLUME can use the default Configuration. |
| NAT Traversal | Network Address Translation is a private (keep) addresses into legitimate IP address conversion technology. Including three ways: STUN, static NAT and dynamic NAT. |
| Enable Private Service | Start with "*" the beginning of a local business, such as *158# inquiry IP address. |
| User ID is Phone Number | SIP compatibility configuration, INVITE news in the head is carry "User = Phone" parameters |
| Only Accept Calls from SIP Server | Only accept SIP server launch of call and refused to other sources of call. |
| Allow Call from PSTN to IP without Registration | Reference "Is register" of "SIP Configuration", if "Is register" setting is no, this option need set Yes ,to avoid that the devices can not call out. This option allows equipment not registered on the phone. |
| Allow Call from IP to PSTN without Registration | Reference "Is register" of "SIP Configuration", if "Is register" setting is no, this option need set Yes ,to avoid that the devices can not call out. This option allows equipment not registered on the phone. |

| | |
|--|---|
| Reject Anonymous Call from IP to PSTN | The call from IP to PSTN will be rejected. |
| Use # as END Key | General dial-up end of the logo has two kinds: 1. # operator as dial-up end, 2. Waiting for a few seconds until dial-up overtime. |
| No Answer Timeout | Call in or out over a certain time no response, the call off. |
| Interdigit Timeout | Bit of between the dialing time ,over the time will be seem as end of dial. |
| Call Delay | When a SIM card to call on the broken, in this delay time will not accept new calls, ensure the call the success rate. |

4.11.3 SIP Configuration

SIP Configuration

SIP Proxy

SIP Server Address

SIP Server Port(default: 5060)

Check Net Status No Yes

Outbound Proxy

Outbound Proxy Address

Outbound Proxy Port

Use Random Port

Local SIP Port No Yes

Is Register

Register Interval(range: 1 - 3600s) No Yes

s

T1 ms

T2 ms

T4 ms

TMAX ms

Keepalive Interval(range:1 - 3600s) s

Enable 100rel No Yes

Refer to Use Target Contact No Yes

From Mode when Caller ID Is Available ▼

From Mode when Caller ID Is Unavailable ▼

Answer Mode ▼

183 Mode ▼

Response Code switch

| | |
|--|--|
| Response code | Response code after switch |
| <input style="width: 90%;" type="text"/> | <input style="width: 90%;" type="text"/> |
| <input style="width: 90%;" type="text"/> | <input style="width: 90%;" type="text"/> |
| <input style="width: 90%;" type="text"/> | <input style="width: 90%;" type="text"/> |

NOTE: It must restart the device to take effect.

Figure 4-11-3 SIP Configuration

Table 4-11-3 SIP Configuration

| | |
|------------------|---|
| SIP Proxy | In SIP a proxy server realize voice over IP based on the exchange. SIP server address can be IP address, can also is a domain name. |
| Outbound Proxy | Outbound Proxy is ususlly used in network with firewall/NAT. Used for processing signal and help multimedia data pass through the firewall. |
| Check Net Status | According to Keep alive interval, and constantly to the equipment to send messages, check the network connectivity. |

| | |
|---------------------------------------|---|
| SIP server Port | Local SIP listening socket, can choose the random or fixed. Random is selecting a random port when start device. Fixed port can be specified by customer. Default is 5060. |
| Is Register | DVG-6001G can work at two work mode: register and unregiser. Default is register mode. |
| Register Interval | Registration time intervals of equipment to SIP server or outbound proxy registration. |
| T1 | T1 timer of SIP protocol, default is 500ms |
| T2 | T2 timer of SIP protocol, default is 4000ms |
| T4 | T4 timer of SIP protocol, default is 5000ms. |
| TMAX | SIP compatibility configuration, after sending a SIP request, it is overtime if had not received any response in the largest waiting time of response retransmission. The largest waiting time of response retransmission double after response retransmission. |
| Keep alive Interval | Used for communication between device and SIP server and ensure the state of equipment registered. Often use the factory default. |
| Enable 100rel | SIP compatibility configuration, Used when the news comes to send 100 to each other PRACK reply. |
| Refer-to Use Target Contact | SIP compatibility configuration, fill in contact header in “Refer-to” field of SIP message. |
| From Mode when Caller ID is Available | SIP compatibility configuration, FROM field used to transfer Caller ID. Tel/User: From: caller number < sip:3001@IP>;tag=51088abb User/User: From: 3001 < sip:3001@IP>;tag=51088abb Tel/Tel: From: caller number < sip: caller number @IP>;tag=51088abb User/Tel: From: 3001 < sip: caller number @IP>;tag=51088abb |
| From Mode | SIP compatibility configuration, used for transmission ID when no Call |

| | |
|-------------------------------|--|
| when Caller ID is Unavailable | ID Numbers FROM field. Anonymous : From: <sip: Anonymous @IP>;tag=51088abb Username : From: <sip: Username @IP>;tag=51088abb |
| Answer Mode | Response way of IP to PSTN side, includes: Answered and Alerted. If select “Answered”, SIP protocol back to 200 news on the side hook; if select “Alerted”, SIP protocol back to 200 news on the side ringing. Usually keep default Settings. |
| 183 Mode | Reply 183 news after reply100 or ringback. Usually keep default settings. |
| Response Code switch | SIP compatibility configuration. Response code is SIP news code. Suchas: 183 and 100. User can modify response code in this configuration items. |

4.11.4 Port Configuration

| Port | SIP User ID | Authenticate ID | Tx Gain | Rx Gain | To VOIP Hotline | To PSTN Hotline | Auto-Dial Delay Time(s) | Detail |
|------|-------------|-----------------|---------|---------|-----------------|-----------------|-------------------------|------------------------|
| 0 | 102 | 102 | 0 | 0 | | | 3 | Detail |

Figure 4-11-4 Port Configuration

Port Configuration

Current Port Port 0 ▾

SIP User ID

Authenticate ID

Authenticate Password

Tx Gain 0dB ▾

Rx Gain 0dB ▾

To VOIP Hotline

To PSTN Hotline

Figure 4-11-5 Port Configuration

Table 4-11-4 Port Configuration

| | |
|-----------------------|---|
| Current Port | Choose the current registration port |
| SIP User ID | Used to SIP server registered the authentication, SIP registered account number is the phone part of users in SIP address, and is often used as an ID information callers, displayed in SIP software or phone on the LCD. The typical cases, SIP registered account number is a phone number or expanded the number, or a user name. |
| Authenticate ID | The authentication name is strictly to the authentication purpose, is telephone contact SIP server to verify user identity with.SIP User ID could be the same with authenticate ID, also can not. |
| Authenticate Password | SIP account register password. |
| Tx Gain | Gain from PSTN side, default is 0. |
| Rx Gain | Gain from IP side, default is 0. |
| To VOIP Hotline | PSTN side calls the port, the port immediately sent hotline number to IP side after hook. |
| To PSTN Hotline | IP side calls the port, the port immediately sent hotline number to PSTN side after hook. |

4.11.5 Digit Map

The screenshot shows a web-based configuration window titled "Digit Map". The main area contains a text input field with the value "x.T|x.#". Below the input field, a red text note states: "NOTE: Length of 'Digit Map' should be not more than 119 characters." At the bottom right of the window, there is a "Save" button.

Figure 4-11-6 Digit map

Digit Map Syntax:

1. Supported objects

Digit: A digit from "0" to "9".

Timer: The symbol "T" matching a timer expiry.

DTMF: A digit, a timer, or one of the symbols "A", "B", "C", "D", "#", or "*".

2. Range []

One or more DTMF symbols enclosed between square brackets ("[" and "]"), but only one can be selected.

3. Range ()

One or more expressions enclosed between round brackets ("(" and ")"), but only one can be selected.

4. Separator |

|: Separated expressions or DTMF symbols.

5. Subrange -

-: Two digits separated by hyphen ("-") which matches any digit between and including the two. The subrange construct can only be used inside a range construct, i.e., between "[" and "]".

6. Wildcard

x: matches any digit ("0" to "9").

7. Modifiers

.: Match 0 or more times.

8. Modifiers

+: Match 1 or more times.

9. Modifiers

?: Match 0 or 1 times.

Example:

Assume we have the following digit maps:

1. xxxxxxx | x11

and a current dial string of "41". Given the input "1" the current dial string becomes "411". We have a partial match with "xxxxxxx", but a complete match with "x11", and hence we send "411" to the Call Agent.

2. [2-8] xxxxxx | 13xxxxxxxx

Means that first is "2", "3", "4", "5", "6", "7" or "8", followed by 6 digits; or first is 13, followed by 9 digits.

3. (13 | 15 | 18)xxxxxxxx

Means that first is "13", "15" or "18", followed by 8 digits.

4. [1-357-9]xx

Means that first is "1", "2", "3" or "5" or "7", "8", "9", followed by 2 digits.

4.12 Tools

4.12.1 Firmware Upload

Firmware Upload

Send ".ldf" file from your computer to the device.

Software 浏览...

Software

NOTE: 1. The upload process will last about 60s.
2. The device will restart automatically after upload.
3. Do not shut down when the device is uploading.

Figure 4-12-1 Firmware Upload

Please consult equipment provider before upgrading, select the appropriate firmware version. Click browse choose appropriate firmware, and then click upload. Uploading please don't shut off the power, otherwise lead to paralysis of equipment.

4.12.2 IVR Voice Prompt Upload

IVR Voice Prompt Upload

Send ".wav" file from your computer to the device.

IVR Voice Prompt File for PSTN Incoming Calls 浏览...

Play IVR Voice Prompt from Default Custom

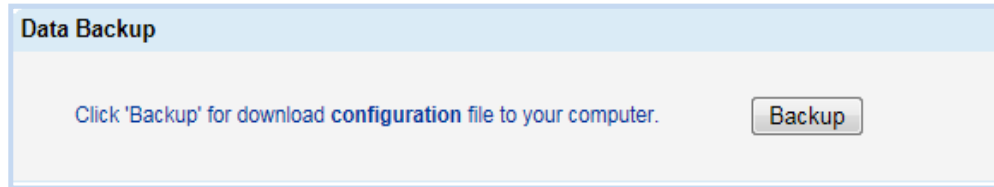
NOTE: 1. ".wav" file should be not more than 360k bytes.
2. The upload process will last about 30s.
3. The device will restart automatically after upload.

Figure 4-12-2 IVR Voice Prompt Upload

The default is when the telephone call in the PSTN, play is the default IVR "please dial the extension number", users can customize the IVR voice, and through the menu loading. Please note that loaded IVR file format must for 8000 Hz, 16 bit sampling mono wav format, and can't more than 360 KB.

4.12.3 Data Backup

Figure 4-12-3 Data Backup



When a device configuration is finished, please click data backup and saved the configuration file in reliable place. When the equipment malfunction or have other equipment needed to add, through the data restore function rapid configuration a function similar or the same equipment.

4.12.4 Data Restore

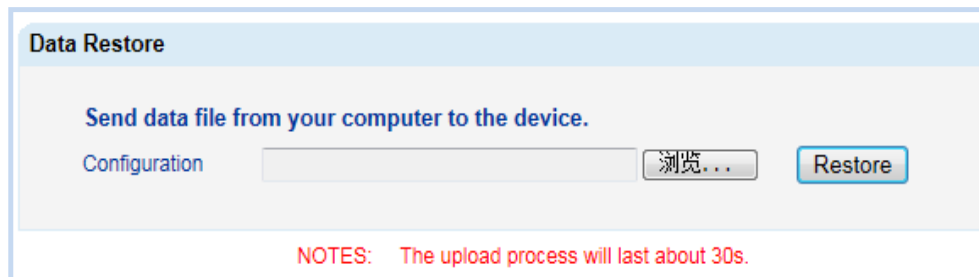


Figure 4-12-4 Data Restore

Importing backup data to equipment can save configuration time. Import will reboot. Network configuration can't through the data recovery.

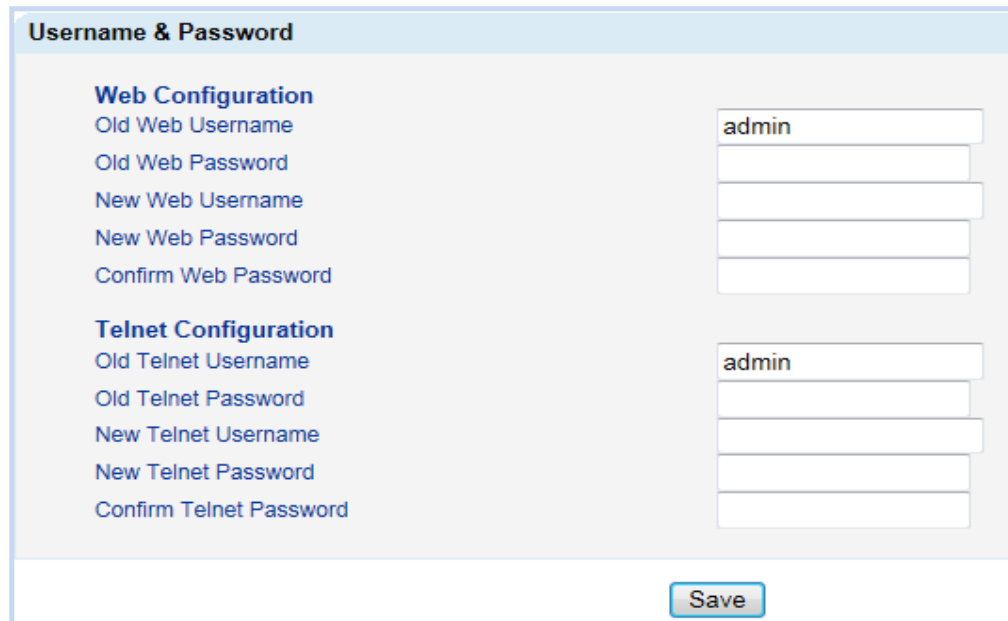
4.12.5 Syslog Parameter

Figure 4-12-5 Syslog Parameter

Table 4-12-1 Syslog Parameter

| | |
|----------------|---|
| Server Address | Syslog information will be saved in Syslog server. Fill in Syslog server IP. |
| Syslog Level | The information contained in the system logs are: NONE、DEBUG、NOTICE、WARNING、ERROR. At present, only NONE and DEBUG level effective. |
| Send CDR | If choose CDR, Syslog level should be chosen NONE. |

4.12.6 Login Password



Username & Password

Web Configuration

Old Web Username

Old Web Password

New Web Username

New Web Password

Confirm Web Password

Telnet Configuration

Old Telnet Username

Old Telnet Password

New Telnet Username

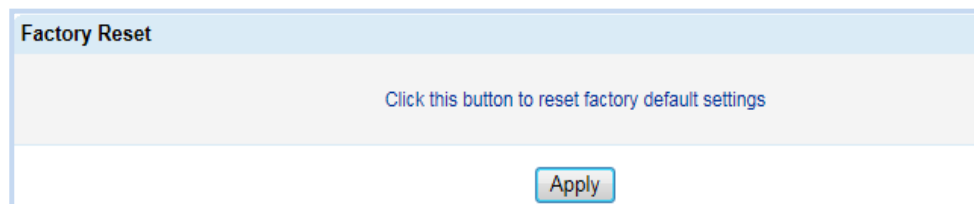
New Telnet Password

Confirm Telnet Password

Figure 4-12-6 Username and Password

The default WEB and TELNET user name/password is admin/admin,, if open the remote login please modify user name and password to prevent others to use the default user name and password to land.

4.12.7 Factory Reset



Factory Reset

Click this button to reset factory default settings

Figure 4-12-7 Factory Reset

Please use caution this operation, the operation will lead to all the parameters recovery factory state, including configuration parameters and network parameters. For safety, before reset factory, please backup configuration files. Pay attention to restore the factory default need to come into force after the restart your device.

4.12.8 Restart

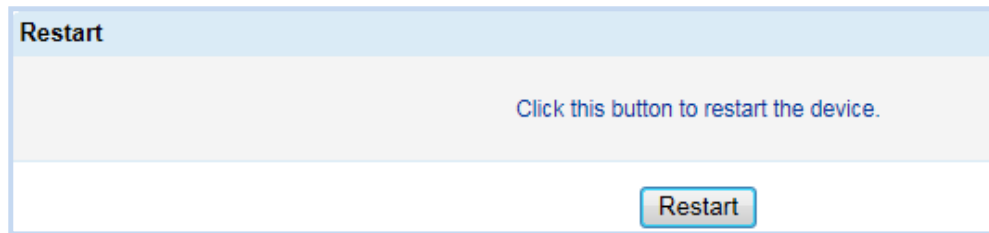


Figure 4-12-8 Restart

By clicking on the restart can remote restart your device when recovery factory defaults, data recovery or modify system parameters need to restart equipment, please try to use the WEB to restart way to restart.

5. Glossary

GSM: Global System for Mobile Communications

CDMA: Code Division Multiple Access

FMC: Fixed Mobile Convergence

SIP: Session Initiation Protocol

MGCP: Media Gateway Control Protocol

DTMF: Dual Tone Multi Frequency

USSD: Unstructured Supplementary Service Data

PSTN: Public Switched Telephone Network

STUN: Simple Traversal of UDP over NAT

IVR: Interactive Voice Response

IMSI: International Mobile Subscriber Identification Number

IMEI: International Mobile Equipment Identity

DMZ: Demilitarized Zone