D-Link®

DVG-6001G User Manual v1.0

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 3.5 DHCP Configuration	10
4. WEB Configuration	11
4. WEB Configuration	11 11
 4. WEB Configuration	11 11 11
 4. WEB Configuration	11 11 11 12
 4. WEB Configuration	11 11 12 12
 4. WEB Configuration	11 11 12 12 12 13
 4. WEB Configuration	11 11 12 12 12 13 13
 4. WEB Configuration	11 11 12 12 12 13 13 14
 4. WEB Configuration	11 11 12 12 12 13 13 14 15
 4. WEB Configuration 4.1 Preparing 4.2 WEB Landing Interface 4.3 WEB Configuration 4.4 System 4.4.1 System Information 4.4.2 Mobile Information 4.4.3 SIP Information 4.5 Network Configuration 4.5.1 Local network 	11 11 12 12 12 13 13 14 15 15
 4. WEB Configuration 4.1 Preparing 4.2 WEB Landing Interface 4.3 WEB Configuration 4.4 System 4.4.1 System Information 4.4.2 Mobile Information 4.4.3 SIP Information 4.5 Network Configuration 4.5.1 Local network 4.6 Mobile Configuration 	11 11 12 12 12 13 13 14 15 16
 4. WEB Configuration 4.1 Preparing 4.2 WEB Landing Interface 4.3 WEB Configuration 4.4 System 4.4.1 System Information 4.4.2 Mobile Information 4.4.3 SIP Information 4.5 Network Configuration 4.5.1 Local network 4.6 Mobile Configuration 4.6.1 Basic Configuration 	11 11 12 12 12 13 13 14 15 15 16 16
 4. WEB Configuration 4.1 Preparing 4.2 WEB Landing Interface 4.3 WEB Configuration 4.4 System 4.4.1 System Information 4.4.2 Mobile Information 4.4.3 SIP Information 4.5 Network Configuration 4.5.1 Local network 4.6 Mobile Configuration 4.6.1 Basic Configuration 4.6.2 Mobile Configuration 	11 11 12 12 13 13 14 15 16 16 17
 4. WEB Configuration 4.1 Preparing 4.2 WEB Landing Interface 4.3 WEB Configuration 4.4 System 4.4.1 System Information 4.4.2 Mobile Information 4.4.3 SIP Information 4.5 Network Configuration 4.5.1 Local network 4.6 Mobile Configuration 4.6.1 Basic Configuration 4.6.2 Mobile Configuration 4.6.3 PIN Management 	11 11 12 12 12 13 13 14 15 15 16 16 17 20
 4. WEB Configuration 4.1 Preparing 4.2 WEB Landing Interface 4.3 WEB Configuration 4.4 System 4.4.1 System Information 4.4.2 Mobile Information 4.4.3 SIP Information 4.4.3 SIP Information 4.5 Network Configuration 4.5.1 Local network 4.6 Mobile Configuration 4.6.2 Mobile Configuration 4.6.3 PIN Management 4.6.4 SMSC 	11 11 12 12 12 13 13 14 15 16 16 17 20 22
 4. WEB Configuration 4.1 Preparing 4.2 WEB Landing Interface 4.3 WEB Configuration 4.4 System 4.4.1 System Information 4.4.2 Mobile Information 4.4.3 SIP Information 4.4.3 SIP Information 4.5 Network Configuration 4.5.1 Local network 4.6 Mobile Configuration 4.6.1 Basic Configuration 4.6.2 Mobile Configuration 4.6.3 PIN Management 4.6.4 SMSC 4.6.5 Send Message 	11 11 12 12 12 13 13 14 15 15 16 16 17 20 22 22
 4. WEB Configuration 4.1 Preparing 4.2 WEB Landing Interface 4.3 WEB Configuration 4.4 System 4.4.1 System Information 4.4.2 Mobile Information 4.4.3 SIP Information 4.4.3 SIP Information 4.5 Network Configuration 4.5.1 Local network 4.6 Mobile Configuration 4.6.1 Basic Configuration 4.6.2 Mobile Configuration 4.6.3 PIN Management 4.6.4 SMSC 4.6.5 Send Message 4.6.6 USSD 	11 11 12 12 12 13 13 14 15 16 16 17 20 22 22 23
 4. WEB Configuration 4.1 Preparing 4.2 WEB Landing Interface 4.3 WEB Configuration 4.4 System 4.4.1 System Information 4.4.2 Mobile Information 4.4.3 SIP Information 4.4.3 SIP Information 4.5 Network Configuration 4.5.1 Local network 4.6 Mobile Configuration 4.6.1 Basic Configuration 4.6.2 Mobile Configuration 4.6.3 PIN Management 4.6.4 SMSC 4.6.5 Send Message 4.6.7 Carrier 	11 11 12 12 12 13 13 14 15 16 16 16 17 20 22 22 23 24

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

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



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	Description	
PLIN	Green	Slowly flash	SIP account does not have registered
NON	Green	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 used POE for power to make sure equipment stable power supply
- 2. DVG-6001G interfaces support RJ45 的 10/100Mbps
- 3. 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 stepes 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 theat 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

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

Table 3-3-1 Feature codes for system setting

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
- 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;

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- 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;
- configure subnet mask: Dial the SIM card phone number, enter "*153*255*255*0*0#"
 hang up when hear "setting successful" message;
- Configure gateway: Dial the SIM card phone number, enter "*156*172*16*0*1#" hang up when hear "setting successful" message;
- 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:

- 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;
- 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 charpter 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.

Windows Security							
iexplore.exe The server 192.168.11.1 is asking for your user name and password. The server reports that it is from GoAhead.							
User name Password Remember my credentials							
OK Cancel							

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.

D-Link										
System Information Statistics Nutwork Configuration Mobile Configuration Configuration Configuration Configuration	System Information MAC Address Network Mode WAN Port LAN Port DNS Server	F8-A0-3D-5 Router 0.0.0 192.168.11 202.96.128	8.0.0.0 255.255.2 202.96.13		DHCP					
Manipulation Configuration Operation IP Trunk Configuration System Configuration Tools Tools	System Up Time WAN Port Traffic Statistics Version Information	9 m 28 s Received 0 Bytes DVG-6001G 60 20.01 04 PCB 6		Sent 4446 Bytes 64 4 LOGIC 0 BIOS 1, Built on Apr 2 201				014, 10:53:02		
	Mobile Information Port Type IMSI	Status	Remaining Call Duration	Carrier	Signal Quality	BER	ASR (%)	ACD (s)	PDD (s)	Call Status
	SIP Information Port SIP User ID	Register Status	Status	_	1.400	Ŷ	č	•		iuv.
	ō	Unregistered	onhook Refresh	1						

Figure 4-3-1 Web introduce

4.4 System

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

SIP information.

Syste	em Info	ormation										
	MAC	Address		F8-A0-3D-5	F8-A0-3D-50-0C-5C							
	Network Mode			Router	Router							
	WAN	Port		0.0.0.0	0.0.0.0			8.0.0.0				
	LAN F	Port		192.168.11	192.168.11.1							
	DNS \$	Server		202.96.128	.68	202.9	96.134.133					
	Syster	m Up Time	9	10 m 28 s								
	WAN	Port Traffic	c Statistics	Received 0	Bytes	Sent	4446 Bytes					
	Versio	on Informat	tion	DVG-60010	G 60.20.01.04 PCB 64	.4 LOGIC 0 BI	OS 1, Built on A	Apr 2 2	014, 10	:53:02		
MODI	le Info	rmation										
	Port	Туре	IMSI	Status	Remaining Call Duration	Carrier	Signal Quality	BER	ASR (%)	ACD (s)	PDD (s)	Call Status
	_							_	(,	(-/	(-)	
	0	GSM		No SIM Card	No Limit		Taill	0	0	0	0	Idle
SIP I	SIP Information											
	Port	SIP Use	er ID	Register Status	Status							
	0			Unregistered	onhook							
	-			2								
						-						

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	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 PCE	64.4 LOGIC 0 BIOS 1, Built on Apr 2 201	14, 10:53:02

Figure 4-4-2 System Information

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	Calculates the netflow, including the total bytes of message	
Statistics	received and sent	
Version information	Display the version information, include: product model, firm	
	ware version, hardware version and date compiled.	

Table 4-4-1 System Information Describe

4.4.2 Mobile Information

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

Mob	ile Info	rmation										
	Port	Туре	IMSI	Status	Remaining Call Duration	Carrier	Signal Quality	BER	ASR (%)	ACD (s)	PDD (s)	Call Status
	0				No Limit		Tall	0	0	0	0	Idle

Figure 4-4-3 Mobile Information

Table 4-4-2 Mobile Information

Port	Numbers of ports of GSM
Туре	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	Limite a call duration to the SIM card, when call duration is out of that
Duration(min)	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 I	nforma	tion		
	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
O 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.
	Save

Figure 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	DNS server complete the analytical between
Automatically	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)	8 dB
Select Band	Default(Automatic)
Remote API Enable	
ABI Sonior Address	172 16 100 125
AFT Server Address	172.10.100.125
API Server Port	0
API User ID	-2133552688
API User Password	•••••
Auto Reset Module	© No ◉ Yes
Counts of NO CARRIER to reset	5
Counts of NO DIALTONE to reset	3
NOTE: Option 'Reject Incoming' will be disa	abled, When 'yes' is checked on option 'Forward Enable'.
	Save

Figure 4-6-1Basic 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	Acording 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	Continuously n times can't find operators, equipment to restart.
CARRIER to reset	N is 3-255.
Counts of No	Continuously n times no dialtone, module to restart. N is 3-255.
DIALTONE to reset	

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	Port 0 💌
Mobile Number	
Step	60 sec
Enable Call Duration Limitation of single call	O No O Yes
Time of single call	0
Enable Call Duration Limitation	O No O Yes
Auto Reset	💿 No 💿 Yes
Reset Period	Day 💌
Next Reset time	2002 • Year 11 • Month 30 • Day 0 • Hour 0 •
Maximum Call Duration	0
Minimum Charging Time	0 sec
Alarm Threshold (via SMS)	0
Mobile Number (Receiving Alarm)	
Port Description for Alarm	
SIM Remain Time	0
Restore Time	
CLIR	◉ No [©] Yes
Echo Suppression Level	13 💌
Mobile Tx Gain	6 dB
Mobile Rx Gain	6 dB
Detect Reverse Polarity	No O Yes
Reset Module	

Figure 4-6-3 Mobile Configuration

Table 4-6-2 Mobile Configutation

Mobile Number	Corresponding port SIM card number
Enable Call Duration	This function is to limit the max call duration of channel. Users can
Limitation of single	customize the SIM card on the single call duration; if more than
call	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	This function is to limit the max call duration of channel. The max
Limitation	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	A single call over this time, GSM side of the operators began to
Time	collect fees, unit for seconds.
Alarm Threshold(via	Talk time remaining is equal to or lower than the value, the
SMS)	gateway to the alarm information by SMS messages to the
	designated mobile phone number.
Mobile Number	Receiving alarm phone number, user will received alarm message
(Receiving Alarm)	from gateway.
Port Description for	Alarm port information description, which will be sent to user
Alarm	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#" infront of mobile phone number can
	realize the function. This function need operators support.
Echo Suppression	Control echo of call process. The higher the level, the more
Level	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	To GSM module is invalid, in the role of CDMA module, the local
	CDMA network support open when the extremely. When not open
Polarity	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

PIN Management	
Select Port	Port 0 💌
SIM Card Lock PIN Code	No Ves
	Save

Figure 4-6-3 PIN Management

PIN Management	
Select Port	Port 0 ▼ ◎ No ◎ Yes
PIN Code	
	Save Change PIN

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

Change PIN	
Old PIN Code New PIN Code Confirm New PIN Code	
	Save Back

Figure 4-6-5 Change PIN Code

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

PIN Management	
Select Port	Port 0
Please Input PUK Code:	
Please Input New PIN Code:	
Please Input New PIN Code Again:	
	Save

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
	I IIV COUC IO UIIIOCK

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	Set a new PIN code
code	
Please input new PIN	Again confirmed the new PIN code.
code again	

4.6.4 SMSC

SMSC	
Select Port	Port 0 💌
SMSC	
	Save

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

Send Message	
Select Port	Random Port
Encoding	UCS2 -
To Message	
	NOTE: Length of 'Message' should be not more than 300 characters.
	Send

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.				
То	Mobile phone number received SMS				

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

4.6.6 USSD

USSD			
Port	USSD Request	USSD Reply	
0	A 7	not registered	A T
		NOTE: If you do nothing within 90s, connection will be disconnected.	

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

Carrier		
Select Port	Port 0	
Select Mode Carrier List	C Automatic Manual	
	Save	

Figure 4-6-10 Carrier

Table 4-6-7 Carrier

Select Port	Select a SIM card				
Select Mode	There are automatic and manial two mode. Automatic mode can				
	detect carrier automaticly; manual mode will select carrier from				
	drap-dowm list.				
Carrier List	Here will list all detected operators				

4.6.8 BCCH

BCCH																		
		0			1			2			3			4			5	
Port	LAC	CID	dbm	LAC	CID	dbm	LAC	CID	dbm	LAC	CID	dbm	LAC	CID	dbm	LAC	CID	dbm
0																		
-																		
						F	Refresh In	terval			5	s						
						ſ	Refresh	1	Auto	Refresh	Ste	n Refres	h					
						F	Refresh In Refresh	iterval	Auto	Refresh	5 Sto	s op Refresi	h					

Figure 4-6-11 BCCH

BCCH							
	Refresh Inte	erval			5 s		
	Auto Refre	esh			Stop Refres	sh	
	Index	MCC	MNC	LAC	CID	BCCH	Receive Level
			Refresh	Lock	UnLock	Back	

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.
ВССН	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

Routing Parameter	
Tel->IP Parameter	Route calls before manipulation
	Save

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

Tel->IP R	outing							
	Index	Description	Source Prefix	Destination Prefix	Destination			
	0	default	any	any	SIP Server			
Total: 1entry	Total: 1entry 16entry/page 1/1page Page 1 💌							
Add Delete Modify								
		NOTE: 0 routing is not allowed to delete, only allowed to change.						

Table 4-7-2 Tel ->IP Routing

Tel ->IP	This item uses to configure incoming call routes which can be used for
Routing	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	
Description	string	
	When calling number matching the prefix, this routing will take effect.	
Source Prefix	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.

Tel->IP Routing Add	
Index Description Source Prefix Destination Prefix Destination	31 IP IP IP SIP Group SIP Server
	OK Reset Cancel

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

4.8 Manipulation Configuration

4.8.1 IP->Tel Destination Numbers

ſ	IP->Te	el Destin	ation 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
l		0	1	Any	any	any	0	0	6311		0
	Total: 16	entry 16er	ntry/page 1/1page	e Page 1 💌							
l					Add	Delete	Modify				

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

Table 4-8-1 IP->Tel D	estination 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			
Index	from 0 to 31.			
Description	It describes the route for the ease of identification. Its value is			
Description	character string			
Source	It specifies the source IP which will send the calls to gateway			
	All the caller number must match the source prefix. It specifies the			
	source prefix allow to send call out			
Source Prefix	• 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			
	All the called number must match the destination prefix, the call			
Destination	prefix indicates the connected number			
Destination	• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.			
Flelix	• 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	It most for the length of the disite to be deleted from left			
from Left	It specifies the length of the digits to be deleted from left			
Stripped Digits	Termeri Caradha Ingella Cala di State da ha dalata di Cara si da			
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	From the right side began to retain the digits			
right				

IP->Tel Destination Numb	ers Add			
Index	31			•
Description	del-01			1
Source Prefix	0123			1
Source	 IP IP Group SIP Server 	Any	•	
Destination Prefix	any]
Stripped Digits from Left Stripped Digits from Right	2			
Prefix to Add				
Suffix to Add Number of Digits to Leave from Right				
	ОК	Reset	Cancel	

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
---------------	--------	---------

Tel	Tel->IP Source Numbers										
	In	ndex	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	: Oentr	ry 16er	ntry/page 1/0pa	ge 💌							
					Add	Delete	Modify				

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

	All the caller number must match the source prefix. It specifies the	
	source prefix allow to send call out	
Source Prefix	• 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	
	All the called number must match the destination prefix, the call	
Destination	prefix indicates the connected number	
Destination	• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.	
Prefix	• 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	Te anno 10 an de a bana de a Calenda da La de la del teta de Carana al de	
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	It specifies the number of Digits to Leave from Right	
from Right		

Tel->IP Source Numbers Add		
Index	31	
Description	51	
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

ľ	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: 0er	ntry 16ent	try/page 1/0pag	e 🔻							
					Add	Delete	Modify				

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

Tel->IP Destination 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 af	ter manipulation, set the destination ip to any.
	ОК	Reset Cancel

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 Ope	IP.>Tel Operation						
	Index	Source IP		Source Prefix	Destination Prefix	Operation	Description
Total: 0entry	16entry/page	1/0page					
			Add	Delete	Modify		

Figure 4-9-1 IP->Tel Option

Table 4-9-1 IP->Tel Option

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

Index	Number, value range from 0-31.
	It specifies the source IP which will send the calls to gateway
Source ID	• Any: any IP address
Source IP	• IP: specific an IP address
	• IP Group: specific an IP group
	All the caller number must match the source prefix. It specifies the
	source prefix allow to send call out
Source Prefix	• 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
	All the called number must match the destination prefix, the call
Destination	prefix indicates the connected number
Destination	• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.
Pielix	• 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
	Its specifies number analysis rule
	• Forbid call
Operation	• Allow call
	• Auto call
	Password authenticate
Description	It describes the route for the ease of identification. Its value is
Description	character string

IP->Tel Operation Add	
Index	31
Source Prefix	any
Source IP	 □ IP Any □ IP Group ■ SIP Server
Destination Prefix	any
Operation	Forbid Call Allow Call Auto Call Password Authentication
Description	a
	OK Reset Cancel

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

Tel->IP O	Tel->IP Operation						
	Index	Source Prefix	Destination Prefix	Operation	Description		
	31	any	any	Allow ,Auto Call ,	а		
Total: 1entr	y 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
	All the caller number must match the source prefix. It specifies the source
	prefix allow to send call out
Source Prefix	• 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

	All the called number must match the destination prefix, the call prefix
Destination	indicates the connected number
Destination	• Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.
Prenx	• 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
	Its specifies number analysis rule
	• Forbid call
Operation	• Allow call
	• Auto call
	Password authenticate
Description	It describes the route for the ease of identification. Its value is character
Description	string

Tel->IP Operation Add	
Index Source Prefix	30
Destination Prefix Operation	Forbid Call Allow Call Auto Call Password Authentication
Description	
	OK Reset Cancel

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

IP Trunk					
	Index	IP	Port	Description	KeepAlive Enable
Total: 0entry	16entry/page 1/0)page 🔽			
		Add	Delete Modify		

Figure 4-10-1 IP Trunk

Table 4-10-1 IP Trunk

ID Trunk	When device connected to softswitch or SIP server, equipment and the soft	
IF ITUIK	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	

IP Trunk Add	
Index	31 💌
IP	172.16.100.103
Port	5060
Description	X-lite
KeepAlive Enable	
	OK Reset Cancel

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

IP Trunk Group			
	Index	Description	IP
Total: 0entry 16entry/page 1/0pa	ge		
	Add	Delete Modify	

Figure 4-10-3 IP Trunk Group

Table 4-10-2 IP Trunk Group

	This configuration is optional, and is used to add the IP that have the same	
IP Trunk Group	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	
ID	It specifies the IP will add to IP group. When building the trunk later can	
11"	choose some trunk form a trunk group.	

IP Trunk Group Add		
Index Description	31	
IP	Index IP 31 172.16.100.103	Port 5060
	OK Reset Can	cel

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

Provision Configuration	
Primary Profile URL	http://172.16.100.139/temp/test.xml
Secondary Profile URL	
Username	admin
Password	••••
NTP Configuration	
Enable NTP	No Yes
Primary NTP Server IP	64.236.96.53
Secondary NTP Server IP	18.145.0.30
Time Zone GMT-6:00 (US Centra	I Time, Chicago)
NOTE: It must	t 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

vice Configuration	
Local Start RTP Port	8000
Enable Slience Suppression	No O Yes
Call Progress Tone	USA 🔹
Preferred Coders(in listed order)	
IST	G.729AB -
2nd	
3rd	PCMA
4th	G./23.1 💌
Voice Frames per Tx	2
Do Not Answer PSTN Imcoming Call for Hotline	O No O Yes
Enable PSTN Incoming Configuration	No Yes
Auto Outgoing Routing Type	Polling 💌
IP to PSTN One Stage Dialing	No Yes
Answer Delay	5 s
Redirect Call When All Ports Busy	No Ves
Play Voice Prompt for PSTN Incoming Calls	© No ● Yes
DTMF Parameter	
DTMF Method	SIGNAL 💌
DTMF Volume	0dB 💌
DTMF Interval	200 ms
NAT Traversal	Disable
Other Configuration	
Enable Private Service	No Yes
User ID Is Phone Number	No Ves
Only Accept Calls from SIP Server	No O Yes
Allow Call from PSTN to IP without Registration	O No O Yes
Allow Call from IP to PSTN without Registration	O No O Yes
Reject Anonymous Call from IP to PSTN	No O Yes
Use # as End Key	O No O Yes
No Answer Timeout	55 s
Interdigit Timeout	4 s
Call Delay	10 s
NOTE: 1. Answer Delay is o	only valid when Detect Polarity Reversal is not en
2. It must restart the	device to take effect.
Sa	ave

Figure 4-11-2 Service Configuration

Table 4-11-2 Service Configuration

LOCAL Start DTD	Real-time Transport Protocol details the standard packet
DOPT	format on the Internet to deliver audio and video. The initial
FORI	allocation of Channel when RTP voice stream transmit in the IP

	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.
	Silence suppression technique to ensure that only wehn talking
	to both sides in call to take up bandwidth and improve the
Enable Silence	utilization rate of channel. Enable the "silence suppression"
Suppression	almost no impact on call quality, and can save about half of the
	bandwidth.
	Each country has its different call progress tone required
Call progress Tone	standards, such as busy tone, ring back tones and ring tone
	standards, users can select the area standard from here .
	Means the code format when Voice transfer on IP network,
Preferred Coders	support PCMA, PCMU, G.723.1 and G.729AB.
Do Not Answer PSTN	
Do Not Answer PSTN Imcoming Call for	Inbound hotline immediately pick up, after waiting for VOIP
Do Not Answer PSTN Imcoming Call for Hotline	Inbound hotline immediately pick up, after waiting for VOIP side picked .
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming Configuration	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing feature codes.
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming Configuration	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing feature codes. When adopt two stage dialing, this configuration option takes
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming Configuration Auto Outgoing Routing	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing feature codes. When adopt two stage dialing, this configuration option takes effect and routing doesn't take effect. Ordinary mode means
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming Configuration Auto Outgoing Routing Type	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing feature codes. 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
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming Configuration Auto Outgoing Routing Type	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing feature codes. 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.
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming Configuration Auto Outgoing Routing Type	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing feature codes. 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. This function will be displayed only when select "Enable Auto
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming Configuration Auto Outgoing Routing Type IP to PSTN One Stage	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing feature codes. 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. This function will be displayed only when select "Enable Auto Outgoing Routing" function, the User ID will be sent directly to
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming Configuration Auto Outgoing Routing Type IP to PSTN One Stage Dialing	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing feature codes. 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. 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
Do Not Answer PSTN Imcoming Call for Hotline Enable PSTN Incoming Configuration Auto Outgoing Routing Type IP to PSTN One Stage Dialing	Inbound hotline immediately pick up, after waiting for VOIP side picked . Select "Yes", users can configure the device through dialing feature codes. 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. 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

	extremely signal detection, then call send out after a few		
	seconds delay to connect.		
Redirect Call When All	When all the GSM port occupied, this function will switch the		
	call toanother equipment and need to provide the device IP and		
Folts Busy	port.		
Play Voice Prompt for	Setting is yes, when through the PSTN calls to the Channel, the		
Play voice Prompt for	device will with the clew tone, the default is "Please dial the		
FSTN meening Cans	extension User ID"; setting to No, the device will with dial tone		
	Support RFC2833 and SIGNAL two ways. DTMF INTERVAL		
DTMF Parameter	range is $50 \sim 800$ ms, DTMF VOLUME can use the default		
	Configuration.		
	Network Address Translation is a private (keep) addresses into		
NAT Traversal	legitimate IP address conversion technology. Including three		
	ways:STUN, static NAT and dynamic NAT.		
Enable Drivete Service	Start with "*" the beginning of a local business, such as		
Ellable Filvate Service	*158# inquiry IP address.		
User ID is Phone	SIP compatibility configuration, INVITE news in the head is		
Number	carry "User = Phone" parameters		
Only Accept Calls from			
Only Recept Cans nom	Only accept SIP server launch of call and refused to other		
SIP Server	Only accept SIP server launch of call and refused to other sources of call.		
SIP Server	Only accept SIP server launch of call and refused to other sources of call. Reference "Is register" of "SIP Configuration", if "Is register"		
SIP Server Allow Call from PSTN	Only accept SIP server launch of call and refused to other sources of call. Reference "Is register" of "SIP Configuration", if "Is register" setting is no, this option need set Yes ,to avoid that the devices		
SIP Server Allow Call from PSTN to IP without	Only accept SIP server launch of call and refused to other sources of call. 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		
SIP Server Allow Call from PSTN to IP without Registration	Only accept SIP server launch of call and refused to other sources of call. 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.		
SIP Server Allow Call from PSTN to IP without Registration	Only accept SIP server launch of call and refused to other sources of call. 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. Reference "Is register" of "SIP Configuration", if "Is register"		
SIP Server Allow Call from PSTN to IP without Registration Allow Call from IP to	Only accept SIP server launch of call and refused to other sources of call. 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. Reference "Is register" of "SIP Configuration", if "Is register" setting is no, this option need set Yes ,to avoid that the devices		
SIP Server Allow Call from PSTN to IP without Registration Allow Call from IP to PSTN without Registration	Only accept SIP server launch of call and refused to other sources of call. 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. 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		

Reject Anonymous Call	The call from IP to PSTN will be rejected.	
from IP to PSTN		
	General dial-up end of the logo has two kinds: 1. # operator	
Use # as END Key	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	
Interdigit Timeout	end of dial.	
Call Dalar	When a SIM card to call on the broken, in this delay time will	
Call Delay	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 Outbound Proxy	172.16.33.50 5060 ◎ No [©] Yes
Outbound Proxy Address Outbound Proxy Port	5060
Use Random Port Local SIP Port	No <a>Ves 5060
Is Register Register Interval(range: 1 - 3600s)	No Yes 180 s
T1 T2 T4 TMAX	500 ms 4000 ms 5000 ms 32000 ms
Keepalive Interval(range:1 - 3600s) Enable 100rel Refer to Use Target Contact From Mode when Caller ID Is Available From Mode when Caller ID Is Unavailable Answer Mode 183 Mode	100 s No Yes No Yes Tel/User Anonymous Answered Immediately
Response Code switch Response code	Response code after switch
NOTE: It must restart the de	evice to take effect. ave

Figure 4-11-3 SIP Configuration

Table 4-11-3 SIP Configuration

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

	Local SIP listening socket, can choose the random or fixed. Random
SIP server Port	is selecting a random port when start device. Fixed port can be
	specified by customer. Default is 5060.
In Descinter	DVG-6001G can work at two work mode: register and unregister.
is Register	Default is register mode.
Register	Registration time intervals of equipment to SIP server or outbound
Interval	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.
	SIP compatibility configuration, after sending a SIP request, it is
	overtime if had not received any response in the largest waitting time of
IMAX	response retransmission. The largest waitting time of response
	retransmission double after response retransmission.
Keep alive	Used for communication between device and SIP server and ensure the
Interval	state of equipment registered. Often use the factory default.
Enchla 100rol	SIP compatibility configuration, Used when the news comes to send
Enable Toorer	100 to each other PRACK reply.
Refer-to Use	SIP compatibility configuration, fill in contact header in "Refer-to"
Target Contact	fieldof SIP message.
	SIP compatibility configuration, FROM field used to transfer Caller ID.
From Mode	Tel/User: From: caller number <sip:3001@ip>;tag=51088abb</sip:3001@ip>
when Coller	User/User: From: 3001 <sip:3001@ip>;tag=51088abb</sip:3001@ip>
	Tel/Tel: From: caller number <sip: caller="" number<="" td=""></sip:>
ID IS AVAIIADIE	@IP>;tag=51088abb
	User/Tel: From: 3001 <sip: @ip="" caller="" number="">;tag=51088abb</sip:>
From Mode	SIP compatibility configuration, used for transmission ID when no Call

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

4.11.4 Port Configuration

P	ort List								
	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	102	
Authenticate ID	102	
Authenticate Password	•••••	
Tx Gain Rx Gain	0dB ▼ 0dB ▼	
To VOIP Hotline		
To PSTN Hotline		
	Save Back	

Figure 4-11-5 Port Configuration

Table 4-11-4	Port C	onfiguration
--------------	--------	--------------

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

Digit Map	
Digit Map	x.T x.#
	NOTE: Length of 'Digit Map' should be not more than 119 characters.

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 "xxxxxx", 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 "Idf" file for Software Software	rom your computer to the device.	浏览	Upload Download
NOT	 E: 1. The upload process will last about 60s. 2. The device will restart automatically after 3. Do not shut down when the device is uplo 	upload. oading.	

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 de	evice.				
IVR Voice Prompt File for PSTN Incoming Calls		浏览… Upload			
Play IVR Voice Prompt from	Oefault Custom	Save			
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.

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4.12.3 Data Backup

	Figure 4-12-5	Data Backup	
Data	Backup		
	Click 'Backup' for download configuration file to y	our computer.	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

Data Restore			
Send data file fro	om your com	puter to the device. 〔 浏览	Restore
	NOTES:	The upload process will last about 30s.	

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

Syslog Parameter	
Enable Syslog	● no ○ yes
Server Address	
Syslog Level	NONE
Send CDR	no yes

Figure 4-12-5 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	admin
Telnet Configuration Old Telnet Username Old Telnet Password New Telnet Username New Telnet Password Confirm Telnet Password	admin
	Save

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
	Apply

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.

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4.12.8 Restart

Restart	
	Click this button to restart the device.
	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