

# DWG2000-8G User Manual v1.0



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# **1. Equipment Introduction**

This chapter mainly introduces functions and structures of DWG2000-8G.

#### **1.1 Introduction**

DWG2000-8G is full functions VoIP gateway based on IP and GSM wireless network, which provides a flexible network configuration, powerful features, and good voice quality. It works for carrier grade, enterprise, SOHO, residential users for cost-effective solution.

#### **1.2 Scenario of Applications of Products**

DWG2000-8G provides access of GSM network.

With the development of users and telecom service, mobile network and fixed network integration will be steadily increasing. DWG2000-8G provides high quality VoIP service which perfectly meets the requirement. This is a scenario shown as figure 1-2-1





#### **1.3 Product Appearance**

The appearance of DWG2001 shows as follow



Figure 1-3-1 Front view of DWG2000-8G

Table 1-3-1 Description of Front view

SIM Card Slot	8 SIM channel			
Indicator of Channels	Red: Indicate that occupancy status of channels Green: Indicate GSM/CDMA signal strength indicator			
PWR	Indicate the power status of SIM cards			
RUN	Indicate the run status of SIM cards			





Power Switch	Turn on or turn off the device
AC Power Input jack	8 SIM channel
WAN	WAN interface
LAN	LAN interface
RUN	Indicate the run status of module
POWER	Indicate the Power status of module
RST	Press it to reset the device to factory
Antennas	Antennas of CDMA/GSM module

Table 1-3-2 Description of Rear view

#### **1.4 Functions and Features**

#### **1.4.1 Protocol Standard Supported**

- Standard SIP and MGCP(option) protocol;
- Simple Traversal of UDP over NATs (STUN);
- Point-to-point protocol over Ethernet (PPPoE);
- Hypertext Transfer Protocol (HTTP);
- Dynamic Host Configuration Protocol (DHCP);
- Domain Name System (DNS);
- ITU-T G.711α-Law/μ-Law、G.723.1、G.729AB;

#### 1.4.2 System Function

- PLC: Packet loss concealment
- VAD: Voice activity detection
- CNG: Comfort Noise Generation
- Local/Remote SIM card work mode
- Adjustable gain of port
- DTMF adjustment
- Balance alarm
- Lock/unlock SIM/UIM
- Mobile number display rejection
- Sending/receiving SMS
- Customize IVR Recording
- White and black list
- One number access
- Open API for SMS, support USSD
- Echo Cancellation (with ITU-T G.168/165 standard)
- Automatic negotiate network
- Hotline

#### 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:AC100~240V 50/60HZ DC12V/1A
- Temperature: 0~40 °C (Operation), -20~80 °C (storage)
- Humidity: 5%~90%RH,
- Power Consumption: 5W
- Dimensions: 112(W) x76(D) x24(H) mm
- Net weight: 0.7kg

### 2. Equipment Installation

This chapter mainly introduces DWG2000-8G hardware installation and connection of equipment.

#### **2.1 Installation Notice**

DWG2000-8G uses AC power. Power supply should ensure the reliability and stability, otherwise, it may damage the SIM card or device. In addition, make sure the power supply connects to ground bar well. With right ground protect connection, that can reduce the surge voltage caused by lightning that damage the equipment, and ensure voice quality (note: when calls with irregular noise occurring, please check the power whether connect ground well). Common measures are as follows:

Making sure that all devices powered in the buildings are in accordance with NEC (National Electric Code, National Electrical Regulations) Article 250 of manual properly grounded;

Making sure that the panel of building power supply units used high-quality copper wire well connect with the ground wire, copper wire specifications shall comply with NEC Table 250-94/95 relevant provisions of the manual. Grounding cable that buried in the building field, including at least one or several 2.44m deep under the ground, or buried deeply underground at least 0.76m, with a wire around the building (see NEC manual specifications the relevant provisions of the table 250-94/95);

Setting up voltage protector between equipment and ground connected to some other computer equipments (either directly or through other devices), such as terminal or printer must also be plugged into the same surge protector.

Network interface of DWG2000-8G supports RJ45 standard with 10Mbps or 100Mbps network.

Wireless section, inserting SIM card directly, GSM channel should work properly.

#### **2.2 Installation Procedure**

The outlook of DWG2000-8G looks like a 1U chassis; to install hardware the cable is needed. After unpacking the equipment, please do follow the procedure as following steps:

#### 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.

#### Figure 2-2-1 Installing SIM card



#### 2.2.2 Antenna Installation

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

# Figure 2-2-2 Antenna Installation

#### 2.2.3 Cable Connection of Equipment

DWG2001 works in Route mode





# **3. Network Configuration**

In this chapter we will introduce the initial configuration of DWG2001 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:

- Prepare an analog telephone or mobile phone
- Make sure the gateway is power on
- Make sure the gateway is connected with the network
- Completed the SIM installation
- 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.

DWG2001 can work in two modes: route mode and bridge mode. when the gateway is under bridging mode, user should configure network parameters of WAN port; when the gateway is under the route mode, user should configure LAN port.

#### **3.3 General Feature Codes for System Setting**

Dial numbers	Features
*114#	Play the phone NO.
*115#	Check the TT number of gateway (using just when the device interconnects
*150*a#	Set IP address(static/DHCP), a can be digit 1 or 2,*150*1# is static IP
*152*a*b*c*d	Configure IP address, a, b, c, d are the four fields of IP address.
*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
*158#	Report the IP address
*157	Setting the work mode (route or bridge), * 157 * 0 # is route mode, * 157 *
*195#	Play record

Table 3-3-1 Feature codes for system setting

*198#	Clear record
*199#	Setting Record. dial*199# start record( $\leq 20$ s), then press # end the
*111#	Restart device

#### **3.4 Static IP Configuration**

This chapter introduces IP configuration of DWG2001 through calling IVR.

Assuming the IP address of a DWG2001 device is 172.16.0.100, subnet mask is 255.255.0.0, IP of

gateway is 172.16.0.1, configured as follows: Insert a SIM card into the DWG2001 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;
- 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 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\*1#", 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, which means that the gateway could not obtain a IP address successfully. Please check:

- Make sure the device have been connected to the network;
- 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 DWG2001.

#### 4.1 Preparing

WEB configuration includes the following components: network configuration, system information, mobile configuration and system configuration.

#### 4.2 Access the System Through HTTP

Enter IP address of DWG2001 in browser. The default IP of LAN port is 192.168.11.1. and the GUI shows as below:

Connect to 172.16.3	30.30 <b>? ×</b>
	G
Web Config System	
<u>U</u> ser name:	🖸 admin 💌
Password:	•••••
	$\square$ Remember my password
	OK Cancel

Figure 4-2-1 WEB log 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**

DWG2001 WEB configuration interface consists of the navigation tree and the detail configuration interfaces.

	System Info	ormation										
stem Information	MAC A	ddress		00-88-76-54	-32-10							
twork Configuration	Netwo	ork Mode		Bridge								
obile Configuration	Netwo	ork		172.16.33.1	0		255.255.0	0.0			5	Static IP
uting Configuration	DNS S	Server		202.96.128.	68		202.96.13	34.133				
anipulation Configuration												
peration	Syster	m Up Time		03h:09m:59	s							
rt Group Configuration	Netwo	ork Traffic S	tatistics	Received 73	22066 Bytes		Sent 3285	59 Bytes				
Trunk Configuration												
stem Configuration	Versio	n Informati	on	DWG2000-8	G Rev 9.50.42 PCB	29.1 LO	OGIC 0 BIOS	1, Built on	Sep 7	2011, 10	:07:09	
ols												
	Mobile Info	rmation										
	Port	Туре	IMSI	Status	Remaining Call	Carr	ier	Signal	ASR(9	6)ACD(s	)PDD(	Call
					Duration(min)			Quality		<u> </u>	· · ·	status
	0	GSM	460021180346188	Mobile Registere	d No Limit	CHI	AMOBILE	Tatl	0	0	0	Idle
	1	GSM	460029947244207	Mobile Registere	d No Limit	CHI	VAMOBILE	Tat	0	0	0	Idle
	2	GSM		No SIM Card	No Limit			Tall	0	0	0	Idle
	4	GSM		No SIM Card	No Limit			¥.	ŏ	ŏ	ŏ	Idle
	5	GSM	1	No SIM Card	No Limit			Taill	0	0	0	Idle
	6	GSM		No SIM Card	No Limit			Tati	0	0	0	Idle
		GSM	1	No SIM Card	No Limit			Latti	0	0	0	Idle
	0001.6											
	SIP Informa	ation										
	Port	SIP Use	r ID Reg	jister Status	Status	Port	SIP User ID	)	Reg	ister Sta	itus	Status
	0	1000	Unr	egistered	onhook	1	1000		Unr	egistere	d	onhook
	2	1000	Unr	egistered	onhook	3	1000		Unr	egistere	d	onhook
	4	1000	Unr	egistered	onnook	5	1000		Unr	egistere	d d	onnook

Figure 4-3-1 WEB introduce

Go through navigation tree, user can check, view modify, and set the device configuration on the right of configuration interface.

#### **4.4 System Information**



System Inform	nation						
MAC Add	dress		00-1F-D6-	1B-3D-02			
Network	Mode		Bridge				
Network			172.16.0.1	15	255.255.	0.0	Static IP
DNS Ser	rver		172.16.1.1				
System (	Up Time		00h:04m:1	6s			
Network	Traffic Stati	stics	Received 4	454102 Bytes	Sent 173	439 Bytes	
Version I	Information		EIA AOS 9.	50.34 PCB 64.4 L	OGIC 0 BIOS 1, Built on F	eb 18 2011, 12:00	:22
Mobile Inform	nation						
Port T	Туре	IMSI	Status		Remaining Call Duration(min)	Carrier	Signal Quality
0 0	GSM		No SI	M Card	No Limit		Taul
SIP Informatio	on						
Port	SIP User ID	þ	Register Status	Status	Status		
0			Unregistered	onhook			
				Re	fresh		

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

#### 4.4.1 System Information

	6 5		
System Information			
MAC Address	00-88-76-54-32-10		
Network Mode	Bridge		
Network	172.16.33.10	255.255.0.0	Static IP
DNS Server	202.96.128.68	202.96.134.133	
System Up Time	03h:09m:59s		
Network Traffic Statistics	Received 7322066 Bytes	Sent 32859 Bytes	
Version Information	DWG2000-8G Rev 9.50.42 PCB 2	9.1 LOGIC 0 BIOS 1, Built on Sep 7 2	011, 10:07:09

#### Figure 4-4-2 system information

#### Table 4-4-1 Description of system information

MAC Address	Displays the current MAC of the gateway, for example: 00-1F-D6-1B-3D-02
Network Mode	DWG2001 support two types network mode, which is bridge and route modes
Network	Shows IP address and subnet mask
DNS Server	Displays DNS server IP address in the same network with the gateway
System Up Time	shows the time period of the device running. For example,:1h: 20m, 24s
Traffic Statistics	Calculates the netflow, including the total bytes of message received and sent.
Version info	shows the current firmware version

#### 4.4.2 Mobile Information

#### Figure 4-4-3 Mobile information

Port	Туре	IMSI	Status	Remaining Call Duration(min)	Carrier	Signal Quality	ASR(9	%)ACD(	s)PDD(s	Call Status
0	GSM	460021180346188	Mobile Registered	No Limit	CHINAMOBILE	Taul	0	0	0	Idle
1	GSM	460029947244207	Mobile Registered	No Limit	CHINAMOBILE	Tail	0	0	0	Idle
2	GSM		No SIM Card	No Limit		Tault	0	0	0	Idle
3	GSM		No SIM Card	No Limit		Taull	0	0	0	Idle
4	GSM		No SIM Card	No Limit		Taill	0	0	0	Idle
5	GSM		No SIM Card	No Limit		Tull	0	0	0	Idle
6	GSM		No SIM Card	No Limit		Taull	0	0	0	Idle
7	GSM		No SIM Card	No Limit		Taull	0	0	0	Idle

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

Port	Numbers of ports of GSM/CDMA.
Туре	Indicates 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 / CDMA module
Remaining	Limite a call duration to the SIM card, when call duration is out of that duration,
Call Duration	the call would be discontinued. This option shows remaining talk time.
Carrier	Displays the network carrier of current SIM card.
Signal Quality	Displays the signal strength of in each channels of GSM / CDMA.

#### Table 4-4-2 Description of mobile information

#### 4.4.3 SIP Information

SIP Inform	SIP Information							
Port	SIP User ID	Register Status	Status	Port	SIP User ID	Register Status	Status	
0	1000	Unregistered	onhook	1	1000	Unregistered	onhook	
2	1000	Unregistered	onhook	3	1000	Unregistered	onhook	
4	1000	Unregistered	onhook	5	1000	Unregistered	onhook	
6	1000	Unregistered	onhook	7	1000	Unregistered	onhook	
						-		

Displays registration status information with Softswitch platform or SIP Server

	Table 4-4-3 Description of SIP information					
Port	He corresponding number of GSM channel, DWG2001 has only 1 port.					
SIP User ID	SIP registration account of the Softswitch and SIP server provided					
SIP User ID	Shows the registration status of VoIP channel, including registered and unregistered.					

#### 4.5 Network Configuration

The navigation tree of the route mode and bridge mode as below:



In the navigation tree of route mode, will have extra items of "MAC Clone","DHCP Server", "DMZ Host", "forward Rule", "Static Route".

#### 4.5.1 Local Network

Under the route mode, WAN port connects with ADSL modem, and LAN port connects with local network. It will be used as a small switch when working in brigde mode. In this situation, user just need to configure the WAN parameter and DNS. User also need configure the LAN port if working in route mode. The web interface as bellows:



Local Network	
Work Mode	C Route O Bridge
Network Configuration	
Link Speed & Duplex	Auto Detect
Obtain IP address automatically	
Use the following IP address	
IP Address	172.16.30.30
Subnet Mask	255.255.0.0
Default Gateway	172.16.0.250
C PPPoE	
Account	
Password	
DNS Server	
Obtain DNS server address automatically	
Use the following DNS server addresses	
Primary DNS Server	211.148.192.137
Secondary DNS Server	202.96.134.133

NOTE: It must restart the device to take effect.



Figure 4-5-4 WEB interface of Route mode



NOTE: It must restart the device to take effect.



Work Mode	Two options of route mode and bridge mode, default is bridge mode		
Link Speed & Duplex	The 5 options are "Auto Detect"," 10Mbps/Half		
	Duplex","10Mbps/Full Duplex", "100Mbps/Half Duplex" and		
	"100Mbps/Full Duplex". Default is "Auto Detect"		
Obtain IP Address	Enable the device obtain IP Address automatically or not. Default is		
Automatically	enabling		
Use the Following IP	Configure the "IP Address", "Subnet Mask" and "Default Gateway"		
Address	by manual		
РРРоЕ	Need ISP offer the account and password. Use this mode when have		
	not router in the local network.		
Obtain DNS Server	When enable the WAN port option of "Obtain DNS Server Address		
Address Automatically	Automatically", which will be enabled subsequently.		
Use the Following DNS	Fill in the IP address of "Primary DNS Server" and "Secondary DNS		
Server Addresses	Server"		
WAN /LAN Port	WAN port used for connecting the external network, LAN port used		
	for connecting local network		

Table 4-5-1 Description of Local network

#### 4.5.2 MAC Clone



MAC Clone			
	This page provide the setting	of WAN MAC Address	
	Device MAC Address: PC MAC Address:	00-1F-D6-16-03-7C 00-26-9E-94-5A-23	Restore MAC Clone MAC
	NOTE: It must	restart the device to take effect.	
		Save	

This function can prevent the device against blocking by the carriers. Enable this function on the "router mode", the device can be anti-blocked when the carrier to limit the online users by scanning the MAC address.

#### 4.5.3 DHCP Server

Under "route mode", DWG works as a router. Config DHCP service to enable the DHCP service

funcition of DWG, then DWG will works as a DHCP server.

Figure 4-5-6 Configuration of DHCP service

DHCP Server Config	
DHCP Server	C No C Yes
IP Address Pool Start	192.168.1.100
IP Address Pool End	192.168.1.199
IP Lease Time (default: 72 hours)	600965 hours
Subnet Mask (optional)	255.255.255.0
Default Gateway (optional)	192.168.1.1
Primary DNS Server (optional)	211.148.192.137
Secondary DNS Server (optional)	202.96.134.133

NOTE: The IP address in pool needs to be in the same subnet with LAN port.

Table 4-5-2 Description of DHCP Server				
IP address Pool	Determines the range of IP addess of other devices in this network			
IP Lease Time	Sets the duration for how long the IP works with the specific IP. If the			
	duration is out of the duration, the IP would be invalid			
The subnet mask,	DNS info will also be allocated to network devices automatically by			
gateway	DHCP protocol. Generally, there is no need to configure those items.			

#### 4.5.4 DMZ Host

In some conditions, certain devices in LAN network need to do two-way communication with WAN network( e.g. certain computer in LAN network need to provide multiple services to WAN network). In this situation, Configure this device as the DMZ host. Figure 4-5-7 Configuration of DMZ Host



#### 4.5.5 Forward Rules

In some conditions, certain devices in LAN network need to provide channel communication with WAN network( e.g. certain computer in LAN network need to provide FTP service of channel 21 to WAN network). In this situation, Configuring forwarding rules to this device is necessary. The difference between forwarding rules and DMZ host is, DMZ host provides several consecutive channels and communication of all protocols, while forwarding rules provides single or several channels communication based on certain protocol(TCP or UDP). If DMZ host and forwarding rules have conflicts, will be determined by forwarding rules configurations.

Forward Rule Table				
ID	Server Port	IP Address	Protocol	Enable
1			TCP 💌	
2			TCP 💌	
3			TCP 🔽	
4			TCP 💌	
5			TCP 💌	
6			TCP 💌	
7			TCP 💌	
8			TCP 💌	

Figure 4-5-8 Configuration of forwarding rules

NOTE: (1) It will not take effect while internet sharing is closed. (2) The IP address needs to be in the same subnet with LAN port. (3) "Server Port" range: 0 - 65535.

Service Port	The Service channel that should be provided to WAN network
IP address	IP address is the one of devices in LAN net work
Protocol	The service protocol(TCP or UDP)

#### 4.5.6 Static Route

Static route is the route rules in IP communication. Generally speaking, no need to config static route. Configuring static route is necessary in such conditions: when several network segaments exist in LAN network and there's certain application between these network segament. Please cancel "internet sharing" under "Network configuration" first, then configure the "static route".In commly use, please don't configure static route. If static rule is wrong, the devices may not work. Figure 4-5-9 Configuration of Static route

Static Route Table				
ID	Dest. IP Address	Subnet Mask	Nexthop	Enable
1				
2				
3				
4				
5				
6				
7				
8				

Table 4-5-4 Description of Static Route		
Dest IP Address	Destination IP address the date packets will be sended	
Subnet Mask	The Subnet Mask of Destination IP address	
Nexthop	The Next Hop IP address if want to arrive the Destination IP addres	

# Table 4-5-4 Description of Static Route

#### 4.5.7 Qos Parameter



Qos Parameter		
DSCP code point is used for diffserv setting. It utilize the first 6 bits of IP ToS. The default definition is EF(184), AF1(1), AF2(2), AF3(3), AF4(4), BE(0). You can use different DSCP for voice or data according to the network provider.		
DSCP Code/IP ToS Define	🖲 No 🔘 Yes	
Save		

If you want to use Qos,please select yes,and click save

#### 4.6 Mobile Configuration

#### 4.6.1 Basic Configuration

#### Figure 4-6-1Basic Configuration

Basic Configuration	
Dial Tone Gain (Mobile Side)	8 dB
Select Band	Default(Automatic) 💌
Pamata API Epobla	0 No. 0 Yes
Remote AFT Enable	O NO O Yes
API Server Address	0.0.0
API Server Port	0

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

#### Save

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 belows:
	GSM: 850/900/1800/1900 MHz; CDMA: 800 MHz
Remote API Enable	API is provided for third party development with DLL and IAD
	components. The API includes SMS sending and receiving, USSD sending
	and receiving. The default is "No"
API Server Address	It is the remote IP address who uses API. This is an option when selecting
API Server Port	It is the remote channel No. who uses API. This is an option when
	selecting "Yes" under "remote API enable"

#### Table 4-6-1 Description of Basic Configuration

#### 4.6.2Mobile

le Configuration	
Select Port	Port 0
Mobile Number	60
Enable Call Duration Limitation	O No O Yes
Maximum Call Duration	80 min
Free Time to Call	0 sec
Alarm Threshold (via SMS)	20 min
Mobile Number (Receiving Alarm)	15013828917
Port Description for Alarm	nate
SIM Remain Time	80 min
Restore Time	
CLIR	⊙ No C Yes
Mobile Tx Gain	4 dB
Mobile Rx Gain	4 dB
Detect Reverse Polarity	⊙ No O Yes

Figure 4-6-2 Mobile Configuration

NOTE: 1.If the duration of a call is less than 'Free Time to Call', it will be not included in 'Call Duration'. 2.Check the anti-pole signal is only effective on the CDMA.

Save

Mobile Number	SIM card user ID of the channel. That must be configured when "one access
	No." funciton enable.
Enable Call	This function is to limit the max call duration of channel. The max call
Duration	duration is between 1 to 65535 minutes.
Limitation	
Maximum Call	Defines a value by users. That will limit the SIM/UIM card's total call
Duration	duration. After the call duration excesses this value, no call will be initiative
	by this channel. The value range is 1-65535. If user doesn't configure this
	value, Default is no max call duration limits for this channel.
Minimum	A minimum charging time (in seconds) is defined during which no charging is
Charging Time	done at carrier side. If the conversation time is even shorter, the total call
	duration will not decrease.
Mobile Number	The mobile phone No. which used to receive the alarm SMS. Users can get
Receiving	SMS report of SIM/UIM card status(SIM Remain Time) in DWG.
Alarm)	
Alarm Threshold	When the SIM remain time is or less than this value, DWG will send the alarm

#### Table 4-6-2 Description of Mobile Configuration

(via SMS)	SMS to remind the users of the SIM remain time.
Port Description	It is the identification mark of SIM/UIM card in the designated SMS report.
for Alarm	The mobile phone No. of the SIM/UIM card is recommended to use as the
	port description for alarm, or any other string.
SIM Remain	Indicates the current sim remain time. It can't modified
Time	
Restore time	Recovers the SIM remain time to initial value, the Maximum Call Duration.
CLIR	Caller ID display restrict. This function is used to restrict the mobile phone
	No. By adding "#31" before the mobile phone ID, this funciton should be
	supported by carrier.
Mobile Tx Gain	Transits gain of the mobile module, from IP side to PSTN side.
Mobile Rx Gain	Receives gain of the mobile module from PSTN side to IP side.
Detect Reverse	This option for CMDA Reverse Polarity detection. Most CDMA operators
Polarity	don't offer polarity reverse. So VoIP to mobile DWG2000-8G will connect
1 olarity	soon. It doesn't wait mobile side answer.

#### 4.6.3 SIM/UIM Card Lock

#### Figure 4-6-3 Configuration of SIM/UIM Card Lock

SIM/UIM Card Lock	
Select Port	Port 0
SIM Card Lock PIN Code	⊙ No C Yes

Save

#### Table 4-6-3 Description of Configuration of SIM/UIM Card Lock

Select Port	Select the Channel No. which need to be locked.
SIM Card Lock	SIM card lock or unlock. Default is "No".
PIN Code	Correct PIN code is needed to lock or unlock the SIM card.

#### 4.6.4 PIN Management

PIN Management		
Select Port	Port 0	
Old PIN Code	•••••	
New PIN Code	•••••	
Confirm New PIN Code	•••••	

Figure 4-6-4 PIN Management

NOTE: PIN code can be modify, only on state that SIM card is locked.

Save

#### **Detailed description as below:**

Table 4-6-4 Description of PIN Management					
PIN	PIN is the password of SIM card personal identification. In the status of SIM				
	card locked, PIN can be modified to prevent SIM card from being stolen.				
Select Port	Selects the GSM/CDMA channel No.				
Old PIN code	The previous PIN code				
New PIN code	Inputs a new PIN code				

#### 4.6.5 SMSC

#### Figure 4-6-5 SMSC

SMSC	
Select Port	Port 0
SMSC	+8613800755500
	Save

# SMS center of mobile, theoretically, the celluar modular will automatically detect the SMSC number. This configurable option is used in a situation that the SMSC number could not detected by celluar modular. When such case happens, please contact with mobile service provider to identify the SMSC number and then add SMSC number in SMSC configurable web interface.

#### 4.6.6 SMS

Send Message			
Select Port	Random Port 💌		
Addressee Message	13888888888 Hello ! this is a test message!		
	h.		
NOTE: Length of 'Message' should be not more than 300 characters.			
	Send		

Figure 4-6-6 SMS sending

#### Configurations are as below:

Table 1 6 5	Description	ofSMS	conding
1able 4-0-3	Description	01 21/12	senaing

Select Port	Users can select a defined channel or random channel to send SMS. Input			
	the recevier's mobile phone No to send SMS.			
Addressee	Mobile phone No. of the receiver			
Message	Content of the SMS. The length is limited to 300 characters.			

#### 4.6.7 USSD

USSD (Unstructured Supplementary Service Data) is a Global System for Mobile(GSM) communication technology that is used to send text between a mobile phone and an application program in the network. Applications may include prepaid roaming or mobile chatting. USSD is similar with Short Messaging Service (SMS), but unlike SMS. USSD transactions occur during the session only. With SMS, messages can be sent to a mobile phone and stored for several days if the phone is not activated or within range.



USSD	
Port	Port 0 💌
Display	
lucus.	
input	*100*652132#
	NOTE: If you do nothing within 90s, connection will be disconnected.
	Send

Dinstar Technologies Co., Ltd.

Table 4-6-6 Description of USSD			
Port	Select the GSM channel to send USSD		
Display	Display the result of sending USSD		
Input	The area to input USSD code		

#### 4.6.8 Carrier

Figure 4-6-8select Carrier

Carrier	
Select Port	Port 0 💌
Select Mode Carrier List	Automatic  Manual

#### Save

This function is used to select carrier.

Table 4-6-6 Description of select Carrier

Select Port	Select GSM channel, default Port 0		
Select Mode	There are two mode to select carrier, automatic and manual.		
Carrier List	If you select manual mode, you can select carrier from carrier list.		

#### 4.7 Routing Configuration

#### 4.7.1 Routing Parameter



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

Table 4-7-1Description of Routing Parameter

Tel->IP Parameter	Globle parameters, it will take effect while number manipulation configured
Route calls after	The parameters indicate that the gateway will select Tel->IP routes after
manipulation	number manipulation completed
Route calls before	The parameters indicate that the gateway will select Tel->IP routes before
manipulation	number manipulation completed

#### 4.7.2 Tel->IP Routing

#### Figure 4-7-2 Tel to IP Routing

Tel->IP Routing						
	Index	Description	Source Port	Source Prefix	Destination Prefix	Destinatio n
	0	default	Any	any	any	SIP Server
	30	To vps	Port Group 31	х.	00	IP 31
	31	Carrier A to B	Port 0	013[58]	133	Port Gro

Total: 3entry 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	Description	of Tel to	<b>IP</b> Routing
			0

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 Port	It specifies the Port or Port Group which will receive the calls from PLMN

	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 indicates
	the connected number
Destination 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
Destination	Its specifies destination Port or Port Group

Figure 4-7-3 Tel to IP routing Modify

Tel->IP Routing Modify			
Index	0		
Description	default		
Source Prefix	any		
Source	Port	0	
	O Port Group	0 <all></all>	
Destination Prefix	any		
Destination	O Port	0	
	C Port Group	0 <all></all>	
	OIP	10 <other></other>	
	C IP Group	18 <asterisk></asterisk>	
	SIP Server		
	OK	Reset Cancel	

It's a default route configured in gateway. It allows any number from source port 0 send call to SIP server with any prefix.

	Figure 4-7-3 Tel to IP routing Modify	
lify		

Index	30	
Description	To vps	
Source Prefix	х.	
Source	O Port	0
	Port Group	31 (Unicom)
Destination Prefix	00	
Destination	O Port	0
	C Port Group	0 <all></all>
	⊙ IP	13 <eia></eia>
	C IP Group	18 <asterisk></asterisk>
	C SIP Server	

Add a GSM to VoIP route. It indicates that the calls coming from Port Group 31<Unicom> will match the prefix "x.", "x." is a wildcard string which will match any prefix except "anonymous" calls. Meanwhile sending the calls destination IP 13<eia> if called number match with destination prefix "00".

el->IP Routing Modify		
Index	31	
Description	Carrier A to B	
Source Prefix	13[58]	
Source	Port	0 🔽
	C Port Group	0 <all></all>
Destination Prefix	133	
Destination	O Port	0
	Port Group	31 (Unicom)
	OIP	10 <other></other>
	C IP Group	18 <asterisk></asterisk>
	C SIP Server	
	OK	Reset Cancel

Figure 4-7-3	Tel to IP	routing	Modify
1  Iguit = 7 - 5	101 10 11	routing	withurity

Add GSM to GSM route, its mainly used for saving the cost between two carriers. It indecates that calls coming from Port 0 will match the prefix 13[58], "13[58]" include prefix 135 and 138, caller number can't match prefix 135 and 138 will reject by gateway. Meanwhile sending the calls to Port Group 31<Unicom> if called number match with prefix 133.

#### 4.8 Manipulaton Configuration

#### 4.8.1 IP->Tel Destination Numbers

Figure 4-8-1 IP->Tel destination numbers manipulation

1	IP->	Tel Ma	anipulation									
		Inde x	Descriptio n	Source IP	Source Prefix	Destinatio n Prefix	Destination	Stripped Digits from Left	Stripped Digits from Right	Pref ix to Add	Suffi x to Add	Numb er of Digits to Leave from Right
		0	safcom	IP Group 31	any	2547	Port Group	3	0	0		
	Total: 1entry 16entry/page 1/1page Page 1											
					Ad	d Delete	Modify					

IP->Tel destination numbers manipulation	It is an optional configuration item, and is used to add IP->Tel number change data. The IP->Tel Manipulation defined the rules of add, and deletion of called numbers, which are referenced by IP->Tel routing.					
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 IP	<ul> <li>It specifies the source IP which will send the calls to gateway</li> <li>Any: any IP address</li> <li>IP: specific an IP address</li> <li>IP Group: specific an IP group</li> </ul>					
Source Prefix	<ul> <li>All the caller number must match the source prefix. It specifies the source prefix allow to send call out</li> <li>Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.</li> <li>0xxxx: consist of some digits such as 015,08,09</li> <li>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186</li> </ul>					
Destination Prefix	<ul> <li>All the called number must match the destination prefix, the call prefix indicates the connected number</li> <li>Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.</li> <li>0xxxx: consist of some digits such as 015,08,09</li> <li>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186</li> </ul>					
Destination	Its specifies destination Port or Port Group					
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					

Table 4-8-1 Description of IP->Tel destination numbers manipulation

Add an IP->Tel Manipulation, to change the called number from 2547888888 to 07888888 Figure 4-8-2 IP->Tel destination numbers manipulation modify

Index	0	
Description	safcom	
Source Prefix	any	
Source IP	OIP	13 <mathnew></mathnew>
	IP Group	31 <allow calls=""></allow>
Destination Prefix	2547	
Destination Port	O Port	0 -
	Port Group	31 <1>
Stripped Digits from Left	3	
Stripped Digits from Right		
Prefix to Add	0	
Suffix to Add		

It indecates that calls coming from IP Group will match the prefix "any", and the called nubmer whom match with the prefix "2547" will delete 3 digits in front of it and replace it by digit "0".

#### 4.8.2 Tel->IP Source Numbers



Tel->IP Source 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					

Tel->IP destination	It is an optional configuration item, and is used to add IP->Tel number change data.					
numbers	The IP->Tel Manipulation defined the rules of add, and deletion of called numbers,					
manipulation	which are referenced by IP->Tel routing.					
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					

Table 4-8-2 Description of Tel->IP destination numbers manipulation

Source Prefix	<ul> <li>All the caller number must match the source prefix. It specifies the source prefix allow to send call out</li> <li>Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.</li> <li>0xxxx: consist of some digits such as 015,08,09</li> <li>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186</li> </ul>
Destination Prefix	<ul> <li>All the called number must match the destination prefix, the call prefix indicates the connected number</li> <li>Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.</li> <li>0xxxx: consist of some digits such as 015,08,09</li> <li>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186</li> </ul>
Destination	Its specifies destination Port or Port Group
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

Example

Add an IP->Tel Manipulation, to change the called number from 2547888888 to 07888888

Figure 4-8-4 Tel ->IP destination numbers manipulation add

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	L			

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

 OK
 Reset
 Cancel

It indecates that calls coming from IP Group will match the prefix "any", and the called nubmer whom match with the prefix "2547" will delete 3 digits in front of it and replace it by digit "0".

#### 4.8.3 Tel->IP Destination Numbers

Figure 4-8-5 Tel->IP destination numbers manipulation

Tel	->IP Des	tina	tion Numbers								
	Ind	lex	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 1	6entr	y/page 1/0page	÷ 🖵							

Add Delete Modify

Tel->IP destination numbers manipulation	It is an optional configuration item, and is used to add IP->Tel number change data. The IP->Tel Manipulation defined the rules of add, and deletion of called numbers, which are referenced by IP->Tel routing.				
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	<ul> <li>All the caller number must match the source prefix. It specifies the source prefix allow to send call out</li> <li>Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.</li> <li>0xxxx: consist of some digits such as 015,08,09</li> <li>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186</li> </ul>				
Destination Prefix	<ul> <li>All the called number must match the destination prefix, the call prefix indicates the connected number</li> <li>Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.</li> <li>0xxxx: consist of some digits such as 015,08,09</li> <li>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186</li> </ul>				
Destination	Its specifies destination Port or Port Group				
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				

Table 4-8-3 Description of Tel->IP destination numbers manipulation

#### Example

Add an IP->Tel Manipulation, to change the called number from 2547888888 to 07888888

Figure 4-8-6 Tel->IP destination numbers manipulation

>IP Destination Nun	ibers Add
Index	31
Description	
Source Prefix	
Destination Prefix	
Destination	
Desunation	
	© IP Group
	SIP Server
Stripped Digits from Le	eft
Stripped Digits from Right	
Prefix to Add	
Suffix to Add	
Number of Digits to Leave from Right	
NOTE	
NOTE: If you	I need route calls after manipulation, set the destination ip to any.
	OK Boset Cancel

It indecates that calls coming from IP Group will match the prefix "any", and the called nubmer whom match with the prefix "2547" will delete 3 digits in front of it and replace it by digit "0".

#### 4.9 Operation

#### 4.9.1 IP->Tel Operation

IP->Tel Operation								
	Index	Source IP	Source Prefix	Destination Prefix	Operation	Description		
	29	IP 13	any	any	Allow ,Need Pa	password		
	30	IP 14	2877	13[58]	Forbid ,	restrict mobile		
	31	IP 14	2877	07	Forbid ,	restrict unicom		

Figure 4-9-1 IP->Tel Operation

Total: 3entry 16entry/page 1/1page Page 1 💌

Add Delete Modify

Table 4-9-1 Description of IP->Tel Operation

	It is an optional configuration item. Operation configuration essentially involves
IP->Tel Operation	allow, barring some IP and IP Group send calls to certain numbers. It includes:
	forbid call, call allowance, auto call, and password authentication.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 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 prefix indicates
	the connected number
Destination 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
	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 string

#### Example

Index 31: barring the certain calling number from IP 14 <elastix></elastix>	
Figure 4-9-2 IP->Tel Operation Modif	ìy

Index	31		7
Source Prefix	2877		
Source IP	€ IP	14 <elastix></elastix>	_
	C IP Group	18 <asterisk></asterisk>	
Destination Prefix	07		7
Operation	Forbid Call		
	C Allow Call		
Description	restrict unicom		7

It indicates that calling party from IP 14<elastix> matched prefix 2877, and also called party matched prefix 07 are not allowed call out. The calls match this rule will be rejected by gateway. Index 29: definite a rule for IP 17<FreeSentral> that all the calls must go with valid password authentication.

Figure 4-9-3 IP->Tel Operation Modify

Index	29		
Source Prefix	any		
Source IP	€ IP	17 <freesentral></freesentral>	
	O IP Group	18 <asterisk></asterisk>	
Destination Prefix	any		
Operation	C Forbid Call		
	Allow Call		
	🗖 Auto Call 🗹	Password Authentication	
Authentication Password	•••		
Description	password		

#### 4.9.2 Tel->IP Operation

Figure 4-9-4 Tel->IP Operation

Tel->IP Op	eration				
	Index	Source Prefix	Destination Prefix	Operation	Description
Total: 0entry	16entry/page	1/0page 📃 🚽			
			Add Delete	Modify	

Tel->IP Operation	It is an optional configuration item. Operation configuration essentially involves allow, barring some IP and IP Group send calls to certain numbers. It includes: forbid call call allowance, auto call and password authentication.
Index	It uniquely identifies a route. Its value is assigned globally, ranging from 0 to 31.
	It specifies the source IP which will send the calls to gateway <ul> <li>Any: any IP address</li> </ul>
Source IP	<ul> <li>IP: specific an IP address</li> <li>IP Group: specific an IP group</li> </ul>
	All the caller number must match the source prefix. It specifies the source prefix allow to send call out
Source Prefix	<ul> <li>Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.</li> <li>0xxxx: consist of some digits such as 015,08,09</li> </ul>
	<ul> <li>1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186</li> <li>All the called number must match the destination prefix, the call prefix indicates</li> </ul>
Destination Prefix	<ul> <li>the connected number</li> <li>Any: include anonymous, 0xxxx, 1[2-9]xxxx etc.</li> <li>0xxxx: consist of some digits such as 015,08,09</li> </ul>
	1[3-8]6:consist of some prefix, include 136,146,156,166,176, 186
	Forbid call
Operation	<ul><li>Allow call</li><li>Auto call</li></ul>
	Password authenticate
Description	It describes the route for the ease of identification. Its value is character string

Table 4-9-2 Description of Tel->IP Operation

#### 4.10 IP Trunk Configuration

#### 4.10.1 IP Trunk

IP				
	Index	IP	Port	Description
	10	172.16.0.124	5060	other
	13	172.16.3.55	5060	eia
	14	172.16.0.123	5060	elastix
	17	172.16.1.123	5060	FreeSentral
	19	172.16.244.136	5060	ondo server
	31	110.164.212.105	5060	to vps

Figure	4-10-1	IP	Trunk
0			

Total: 6entry 16entry/page 1/1page Page 1 💌

Add Delete Modify

Tuble 4 10 1 Description of In Trunk	Table 4-10-1	Description	of IP	Trunk
--------------------------------------	--------------	-------------	-------	-------

IP Trunk	Add remote IP of softswitch, SIP server which will send call traffics to 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
IP	It is an interworking parameter between the remote Softswitch and the SIP server. It specifies the IP address of the peer equipment.
Port	It is an interworking parameter between the remote Softswitch and the SIP server. It specifies the SIP port number of the peer equipment

#### Example

To add a remote IP of Softswitch, set "index" to "31", SIP port number "5060"

Figure 4-10-2 IP Trunk Modify

IP Modify	
Index	31
IP	110.164.212.105
Port	5060
Description	to vps
	OK Reset Cancel

#### 4.10.2 IP Trunk Group

Index	Description	IP
18	asterisk	10,14,17,
19	all	13,19,
	18 19	18 asterisk 19 all

#### Figure 4-10-3 IP Trunk Group

#### Table 4-10-2 Description of 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	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
IP	It specifies the IP will add to IP group

#### Example

To add an IP group, set IP "10, 14, 17" to IP group 18

Figure 4-10-4 IP Trunk group modify

	10			
Description	10 Destoris	sk		
IP	asteria	Index	IP	Port
		10	172.16.0.124	5060
		13	172.16.3.55	5060
		14	172.16.0.123	5060
		17	172.16.1.123	5060
		19	172.16.244.136	5060
		31	110.164.212.105	5060

#### 4.11 System Configuration

#### 4.11.1 System Configuration

em Configuration	
Voice Prompt Language	English 💌
Provision Configuration	
Primary Provison Server IP	210.21.119.116
Primary Provison Server Port	80
Secondary Provision Server IP	218.16.122.159
Secondary provision Server port	80
Provision Check Interval	4 hours
NTP Configuration	
Enable NTP	O No O Yes
Primary NTP Server IP	64.236.96.53
Secondary NTP Server IP	18.145.0.30
Time Zone	
GMT-6:00 (US Central Time, Chicago)	<b>v</b>

Figure 4-11-1 System Configuration

NOTE: It must restart the device to take effect.

Save

Voice Prompt Language	Configure the voice prompt of DWG., e.g. configure voice prompt of IP address success or failure. DWG supports English and Chinese. Users can customize other languages. The default setting is in Chinese.
Provision configuration	Provision is used to maintain the devices. E.g. Provision can config, update and remote manage the devices in bulk.
Primary Provision Server IP	This is provided by carrier. Keep the default value if carrier don't provide this value.
Primary Provision Server Port	This is provided by carrier. Default is 80.
Secondary Provision Server	This is provided by carrier. Keep the default value if carrier don't provide this value.
Secondary Provision Server Port	This is provided by carrier. Default is 80.

#### Table 4-11-1 Description of System Configuration

Provision Check Interval	Default is 4 hours.
Enable NTP	NTP enable switch
Primary NTP Server IP	Can keep the default
Secondary Provision NTP	Can keep the default
Server IP	
Time Zone	The default is GMT +8:00, the user can adjusted accordingly
	according to their area

#### **4.11.2 Service Configuration**

Service Configuration is used for configuring voice calls and some small businesses, such as Call

Progress Tone, codec, silence suppression, \* service, the second dial and so on Figure 4-11-2 Service Configuration

Service Configuration	
Local Start RTP Port	8000
Enable Slience Supportsion	C No C Yes
Enable Sherice Supperssion	IND IN THES
Call Progress Tone	USA
Proferred Codere(in listed order)	
1st	67231
204	PCMU V
3rd	PCMA V
Voice Frames per Tx	1
Notice: The device will restart automatically when 'preferred	coders' is changed between G.723.1 and
G.729AB.	
Enable PSTN Incoming Configuration	
Enable Auto Outgoing Routing	
Auto Outgoing Routing Type	Ordinal
IP to PSTN One Stage Dialing	No C Yes
Play Voice Prompt for PSTN Incoming Calls	
Send Original Caller ID for PSTN Incoming Calls	© No C Yes
DTMF Parameter	DECOMPANIE
DTMF Method	RFC2833
RFC2833 Payload Type	101
DTMF volume	
DTMP Interval	200 mis
Enable STUN	No C Yes     No C Yes
CLID Mode	C Number C Neme
Notice: when select 'name', please insure there isn't letter in i	it
Other Configuration	
Enable Private Service	C No C Yes
User ID Is Phone Number	
Only Accept Calls from SIP Server	C No C Yes
Allow Outgoing Calls without Registration	C No C Yes
Allow Incoming Calls without Registration	C No C Yes
Allow Anonymous Outgoing Calls	C No C Yes
Reject Anonymous Incoming Calls	C No C Yes
Use # as End Key	C No C Yes
Interdigit Timeout	4 s

	Means the initial allocation of Channel when RTP voice stream transmit in				
LOCAL RTP PORT	the IP network , in general, using the factory default values. When there				
Channel	are multiple DINSTAR series voice products, and the network gateway				
	or router's NAT with loopholes, user can try changing this item				
Enable Silence	Enable the "silence suppression" almost no impact on call quality, and can				
Suppression	save about half of the bandwidth.				
	Each country has its different call progress tone required standards, such				
Call Progress Tone	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, support				
	PCMA, PCMU, G.723.1 and G.729AB.				
Preferred Coders	Note: when the preferred codec switch between G.723.1 and G.729AB,				
	System will automatically reset				
Enable PSTN	Many when call from DCTN side you can dial the function have for				
Incoming	Means when call from PSIN side, you can dial the function keys for				
Configuration	check number, setting IP and so on function				
Enable Auto Outgoing	Means when call out , whether by ordinal or polling pick to Select a				
Pouting	Channel, this feature are generally used for when use the same SIP User				
ID to register or use as trunking mode					
ID to DSTN One Stage	This function will be displayed only when select "Enable Auto Outgoing				
Dialing	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				
Play Voice Prompt for	Setting is yes, when through the PSTN calls to the Channel, the device				
PSTN Incoming Calls	will with the clew tone, the default is "Please dial the extension User				
	ID"; setting to No, the device will with dial tone				
	For Example, the phone A from PSTN side call				
Send Original Caller	DWG2001/DWG2000-4G/DWG2000-8G Channel SIM card				
ID for PSTN Incoming	corresponding User ID, the Channel's SIP User ID is C, Channel hook and				
	then call B, when "Send Original Caller ID for PSTN Incoming" setting				

Table 4-11-2 Description of Service Configuration

	is Yes, the Caller User ID that send to B will be A, when "Send Original
	Caller ID for PSTN Incoming" setting is No, the caller User ID that sent to
	B will be C(except for anonymous outgoing )
	DWG2001/DWG2000-4G/DWG2000-8G support RFC2833 and SIGNAL
DTMF	two ways. DTMF INTERVAL range is 50 ~ 800ms, DTMF VOLUME can
	use the default Configuration
	(Simple Traversal of UDP over NATs, NAT's UDP simple cross) is a
	network protocol. It is allowed to stay behind the NAT (or multiple NAT)
	client part to identify their clients' public address, found himself after
Enable STUN	what Type of NAT and NAT for a particular Channel is bound to a local
	Internet terminal Channel. This information is used for two host to set up
	UDP communication behind the same NAT router. The agreement
	defined by the RFC 3489
	Select the name under the special needs, the most common way is use the
CLID Mode	default User ID
Allow Outgoing Calls	Refer to "SIP Configuration" -> "Is register" . If "Is register" setting is no,
without Registration	this option need set Yes ,to avoid that the devices can not call out
Allow Incoming Calls	Refer to "SIP Configuration" -> "Is register" . If "Is register" setting is no,
without Registration	this option need set Yes ,to avoid that the devices can not call in
Allow Anonymous	
Outgoing Calls	The incoming anonymous calls will be rejected
	In General, SIP phones are based on # as the end, if this option is set to
Use # as End Key	No, the dial-up will end expires dial-up time
Inter digit Timeout	Bit of between the dialing time ,over the time will be seem as end of dial

#### 4.11.3 SIP Configuration

rigue - ri 5 bir Comgutation				
SIP Configuration				
SIP Proxy				
SIP Server Address	172.16.119.119			
SIP Server Port(default: 5060)	5060			
Outbound Proxy				
Outbound Proxy Address				
Outbound Proxy Port	5060			
Use Random Port	⊙ No C Yes			
Local SIP Port	5060			
Is Register	C No O Yes			
Register Interval(range: 1 - 3600s)	1800 s			
T1	500 ms			
T2	4000 ms			
Τ4	5000 ms			
TMAX	32000 ms			
Keepalive Interval(range:0 - 3600s,0 means disable)	10 s			
Enable 100rel	⊙ No C Yes			
Refer to Use Target Contact	⊙ No C Yes			

Figure 4-11-3 SIP Configuration

Used for Configuring VoIP channel, add SIP Registry Platform and local SIP SIP Channel, and configure SIP protocol and other related information Configuration Used for configure SIP server address and Channel, the address can be IP SIP Server Address, also can be a domain name (DNS should to be able to resolution), Address the details please advisory the service provider Port default setting is 5060. For details, please consult the service provider SIP Proxy Port Outbound proxy, it mainly used in firewall / NAT environment. That make the Outbound Proxy signaling and media streams are able to penetrate the firewall, the details Address please advisory the service provider **Outbound Proxy** Outbound proxy port number, the details please advisory the service provider Port Use Random Set the local monitor SIP port(fixed or random), random is every time you Port start the device will random Select a free SIP port Monitor Default set yes, if you want the device can make a call without register, set Is Register No, Also enable the "Allow Outgoing Calls without Registration" and "Allow Incoming Calls without Registration" function Register Interval Means how often the equipment will register once to the SIP server/proxy T1 Used to define the SIP protocol T1 timer value, default is 500ms T2 Used to defines the SIP protocol timer values, default value is 4000ms Т3 Used to define the T2 timer value in SIP protocol, the default is 5000ms

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Keep alive	Used to keep communicate between equipment and the SIP server that make
Interval	default values

#### 4.11.4 Port Configuration

Port Configuration is used to configure ports' gain, Off-hook Auto-Dial, etc.

Figure 4-11-4 Port Con	nfiguration
------------------------	-------------

Port Configuration	
All ports register used same user ID	● No C Yes
Current Port	Port 0 💌
SIP User ID Authenticate ID Authenticate Password	20313229 123 •••
Tx Gain Rx Gain	0dB  -2dB
Offhook Auto-Dial Auto-Dial Delay Time	3 s

Save
------

Port Configuration	Used to configure ports' gain, Off-hook Auto-Dial, etc.
ALL ports register	The default is not, each SIP account, if set yes ,all the port will use user
used same user ID	ID of port0, when the call ring in sequence
SIP User ID	Is the account used for registration, equipment port's unique identifier,
	"Authenticate ID" is equivalent to show the name, "Password" is register
	Password, which no passward can no fill, the details please contact the
	service provider
Tx Gain	Refers to the call volume that from himself during a call to the end users,
	adjust the "Tx Gain" will affect the voice volume of the end user, the
	default value is 0
Rx Gain	Refer to the call volume from the remote end user to ourself volume,
	adjust the "gain acceptance" will affect the voice volume we will heard,
	the default value is 0.
Offhook Auto-Dial	Hotline service.when PSTN part client calls to this port, will auto forward
	to the hotline User ID. If no need this feature, just left it blank
Auto-Dial Delay Time	Offhook Auto-Dial delay time, the range is 0-10 seconds

#### Table 4-11-3Description of Port Configuration

#### 4.11.5 Digit Map

Figure	4-1	1-5	Digit	man
IIguiv	- I I	1 2	DIGIL	mup

Digit Map			
Digit Map	x.Tlx.#		

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

#### 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. xxxxxx | 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

Equipment upgrades can through Dinstar's softswitch platform, when the device disconnects with Dinstar's softswitch platform or some special circumstances. The firmware is able to upload locally.

Figure 4-12-1 Firmware upload

Firmware Upload				
Send "Idf"	file from your compute	er to the device.		
Software			Browse Upload	
	NOTE: 1. The upload proc	cess will last about 60s	5.	
	2. The device will r	estart automatically aft	ter upload.	

Select the upgrade program under correct directory services, and then click upload will complete upgrade the firmware.

NOTE: during the upgrade process, please do not swtich off the power supply, equipment may paralyze.

#### 4.12.2 IVR Voice Prompt Upload

By default, when PSTN call incoming, the system will play the default IVR, and also the user can load custom IVR.

Figure 4-12-2 IVR	Voice Prompt Upload	
VR Voice Prompt Upload		
Send "wav" file from your computer to the de	evice.	
IVR Voice Prompt File for PSTN Incoming Calls		Browse Upload
Play IVR Voice Prompt from	● Default ● Custom	Save
NOTE: 1. The upload process will last about 30s. 2. Once uploading successfully, the next u	Iploading operation will be only	available after about 30s.

NOTE: the customize voice files can be recorded using Windows recording programs, the sound format is 8000Hz, 16 bit sampling in mono, with WAV format, size of files can not exceed 190KB

#### 4.12.3 Data Backup



Data Backup	
Click 'Backup' for download configuration file to your computer.	Backup

Click 'Backup' for download configuration file to your computer.

#### 4.12.4 Data Restore



Data Restore				
Send data file from your computer to the device.				
Configuration	浏览···· Restore			
	NOTES: The uplead presses will lest shout 20a			

Send data file from your computer to the device

#### 4.12.5 Syslog Parameter

Figure 4-12-5 Syslog parameter

Syslog Parameter	
Enable Syslog	🖲 no 🔘 yes
Server Address	
Syslog Level	NONE
Send CDR	● no <sup>©</sup> yes
	Save

Table 4-12-1 Description of Systog I arameter		
Enable SysLog	select yes to enable syslog client function	
Server Address	Fill in the Syslog server IP Address here	
Syslog Level	There are five level of syslog:NONE, DEBUG, NOTICE, WARNING, ERROR, we urge you to select DEBUG.	
Send CDR	If you select yes, DWG will send CDR to syslog server.	

|--|

#### 4.12.6 Login Password



Username & Password	
Web Configuration	
Old Web Username	admin
Old Web Password	
New Web Username	
New Web Password	
Confirm Web Password	
Telnet Configuration	
Old Telnet Username	admin
Old Telnet Password	
New Telnet Username	
New Telnet Password	
Confirm Telnet Password	
	Save

When using web or telnet Configuration, please enter default user name and password. User can modify the login name and password.

#### 4.12.7 Factory Reset





Be careful do this operation, after restore factory setting, all the parameters will be changed to the factory default.

#### 4.12.8 Restart



When system restarts, user click RESET button on the web.

#### 5. FAQ

5.1 Device have been connected to network physically, but the network cannot be connected or network communication is not normal

1) Make sure the network cable is ok or not, can through view the device WAN port or LAN port indicator light to determine the physical connection is working or not;

2) Make sure the connected network devices (router, switch or hub) support 10M/100M adaptive, if not, connect the Equipment directly to PC, landing WEB and in the "local connection" Configuration interface Select the correct Ethernet Work Mode;

3) Check the Network Configuration, if the Configuration is incorrect, please re-Configuration. If you are using DHCP mode, check DHCP Server is working properly;

4) Check whether there is a LAN device conflict with the exists IP ADDRESS.

5.2 Equipment can not register

If the Run LED does not flash mean unregistered

1) Check the network connection is working (see above section), whether the Configuration is correct;

2) Check whether the LAN firewall setting is inappropriate (such whether limit the network communication); If it is, there are two ways to try to resolve;

3) Ask network administrators to open limitation with the equipment's network communications (it is a special equipment, not afraid of virus attacks); (2) try to enable the equipment tunnel (through the WEB for Configuration, Also, please NOTE, open the tunnel will impact voice quality, Please do not enable the tunnel as far as possible, reference WEB Configuration Interface Description section);

4) Check whether the Local Network to the SIP PROXY platform network environment is relatively poor or not, and if so, please check Local Network or contact the service provider;

5) if go through those steps, the device still be in trouble, please contact the equipment provider;

5.3 When calling out, the callee's phone shows wrong caller ID:

1) Ask the callee checks whether the device is failure or device battery power is low

2) Make sure the callee has been subscribed called User ID display service

3) If only part of the caller User ID with this problem, please contact the telecom carrier.

5.4 when calling in, the caller always hears a busy tone Make sure Enable DND(Do-not-Disturb) in system

5.5 sudden interruption during a call

1) make sure whether is human error caused the problem

2) Make sure with the account balance or lack of disruption caused the call disconnected

3) Make sure whether there is interference with the fax tone or equipment busy tone, these interference may lead to calls dropped

4) Make sure whether the LAN equipment such as gateway or router fails, user can try to restart the gateway or router

5.6 voice single-pass, double-barrier or poor quality

1) Make sure the equipment is working properly with grounded power

2) Check the device network connection is in working status

3) Ask network administrators to open limitation with the equipment's network communications (it is a special equipment, not afraid of virus attacks); (2) try to enable the equipment tunnel (through the WEB for Configuration, Also, please NOTE, open the tunnel will impact voice quality, Please do not enable the tunnel as far as possible, refer WEB Configuration Interface Description section)

4) Make sure the LAN equipment is working, user can try to restart the gateway or router to solve the problem

5) Check whether there is more than one DINSTAR series products in LAN network: some gateways or routers, processing network packet is vulnerable (for example, to multiple network devices or the same protocol network communication, NAT allocated the same conversion communications Channel). If there is such a case, suggest replacing a router or specify each voice gateway with different LOCAL RTP PORT Channel (refer to the base WEB Configuration interface section)

6) Check the equipment network environment for the softswitch platform, monitor the network condition, make sure the network is solid

#### 6. 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