

# **DAG Series User Manual V1.0**



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# **Revision Records**

1. EQUIPMENT INTRODUCTION	2
1.1 Introduction	2
1.2 Equipment Structure	3
1.2.1 Front View	3
1.2.2 Rear View	4
1.3 Connect DAG serial	5
1.4 Functions and Features	6
1.4.1 Protocol standard supported	6
1.4.2 System function	6
1.4.3 Industrial standards supported	6
1.4.4 General hardware specification	6
2. BASIC OPERATIONS	8
2.1 Placing a Phone Call	8
2.1.1 Phone or Extension Numbers	8
2.1.2 Direct IP Calls	8
2.2 Call Hold	9
2.3 Call Waiting	9
2.4 Call Transfer	9
2.4.1 Blind Transfer	9
2.4.2 Attended Transfer	9
2.5 Call Features	10
2.6 Sending And Receiving Fax	10
3.CONFIGURATION GUIDE	11
3.1 Configure LAN Port's IP Address	11
3.2 Connect The DAG Serial By The Web Browser	11
3.2.1 Login	11
3.3 Configure The DAG serial Using The Web Browser	13
3.3.1 System Information	14
2.3.2 Statistics	15
3.3.3 Network Configuration	15
3.3.4 System Configuration	20
3.3.5 Digit Map	25
3.3.6 Routing Configuration	26
3.3.7 Manipulation Configuration	28
3.3.8 Advanced Configuration	30
3.3.9 Management Configuration	32
4.FAQ	35
4.1 How to get the IP address if I have modified the default IP or forgot it ?	35
4.2 Device have been connected to network physically, but the network cannot be connect	ted
or network communication is not normal	35

### **Table of Contents**

4.3 Equipment can't register	
4.4 When calling out, the callee's phone shows wrong caller ID	
4.5 When calling in, the caller always hears a busy tone	
4.6 Sudden interruption during a call	
5. GLOSSARY	

## **1. Equipment Introduction**

### **1.1 Introduction**

The DAG serial is a full feature voice and fax-over IP device that offers a high-level of integration including dual 10M/100Mbps network ports with integrated router, NAT, DHCP server, dual port FXS telephone gateway, market-leading sound quality, rich functionalities, and a compact and lightweight design. The DAG serial fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market. Moreover, it supports comprehensive voice codecs including G.711 (a/ $\mu$ -law), G.723.1 and G.729AB. A typical network diagram shows the function of DAG serial as below.





# **1.2 Equipment Structure**

### 1.2.1 Front View

### Figure 1-2-1 DAG Series Front View



Table 1-2-1 Description of DAG Front View

Interface	Description
PWR	Connecting to power adapter, DC12V.1A or 110~240VAC,50~60HZ,0.4A
WAN(RJ-45)	Connecting to internal LAN network or router
LAN(RJ-45)	Connecting to LAN port with an Ethernet cable to your PC
PHONE(RJ-11)	FXS ports should be connected to analog phones/fax machines
FXO	FXO ports should be connected to physical PSTN lines from a traditional
	PSTN PBX or PSTN Central Office

### 1.2.2 Rear View

### Figure 1-2-2 DAG Series Rear View



Table 1-2 -2 Description of DAG Series Rear View

LED	Color	Name	Status	Description	
DOWED	Carrow	Demonstration in diseases	Off	Power is off	
POWER	Green	Power status indicator	On	Power is on	
DUN	Croon	Degister indicator	Fast blinking	Register	
KUN	Gleen	Register indicator	Slow blinking	Unregister	
WANT	Vallary	WAN status in disaton	Off	Failed	
WAIN	renow	wAIN status indicator	On	Normal	
LAN	Yellow		Off	Failed	
		LAN status indicator	On	Normal	
FXS	Green	Indicate status of the respective FXS ports on	Off	Available	
		the back	On	Busy	
FXO	Green	Indicate status of the respective FXO	Off	Available	
		back	On	Busy	

### **1.3 Connection to DAG serial**

The DAG serial is easy to configure using the embedded GUI pages and the following five (5) steps.

### Five easy steps to configure the DAG serial

- 1. Connecting a standard touch-tone analog telephone (or fax machine) to first FXS port
- 2. Connecting another standard touch-tone analog telephone (or fax machine) to second FXS port or connect PSTN line to line port(FXO)
- 3. Inserting the Ethernet cable into the WAN port of DAG serial and connecting the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)
- 4. Connecting a PC to the LAN port of DAG serial
- 5. Inserting the power adapter into the DAG serial and connecting it to a wall outlet

### **1.4 Functions and Features**

### **1.4.1 Protocol standard supported**

- Standard SIP /MGCP protocol;
- Simple Traversal of UDP over NATs (STUN);
- IP Transport: RTP/RTCP
- Hypertext Transfer Protocol (HTTP);
- Dynamic Host Configuration Protocol (DHCP Server/Client);
- Domain Name System (DNS);
- ITU-T G.711A-Law/U-Law、G.723.1、G.729AB、G.168;

### 1.4.2 System function

- PLC,VAD,CNG
- DTMF mode: RFC 2833,SIP INFO and INBAND
- T.38/ Pass-Through FAX over IP
- HTTP/Telnet configuration
- Firmware upgrade by TFTP/Web
- QoS: Diffserve, TOS,802.1 P/Q VLAN tagging
- Caller ID, Call waiting, Call transfer, DND

### 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: Output: 12VDC, Input 100~240 VAC/50HZ
- Temperature:  $0 \sim 40^{\circ}$ C (operational),  $-20 \sim 70^{\circ}$ C (storage)
- Humidity: 10%~90%, no condensation
- Max power consumption: 6/8/10/15/30W
- Dimension(mm): DAG1000(1FXS):100\*68\*24, DAG1000(2FXS):160\*110\*30,

DAG1000(4FXS):195\*133\*35,DAG1000(4FXO/8FXS/8FXO):240\*150\*35, DAG2000(16FXS/FXO):440\*280\*43

 Net weight: DAG1000(1FXS):0.1kg, DAG1000(2FXS):0.25kg, DAG1000(4FXO/8FXS/8FXO):1kg, DAG2000(16FXS/16FXO):3.05kg

## 2. Basic Operations

### 2.1 Phone Call

### 2.1.1 Phone or Extension Numbers

1. Dial the number directly and wait for 3 seconds (Default "No dial timeout");

2. Dial the number directly and press #.

### Examples:

1. Dial an extension directly on the same proxy, (e.g. 8080), and then press the # or wait for 3 seconds.

2. Dial an outside number (e.g. (626) 666-8080), first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 3 seconds. Check with your VoIP service provider for further details on prefix numbers.

### 2.1.2 Direct IP Calls

Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy.

### Elements necessary to completing a Direct IP Call:

1. Both DAG serial and other VoIP Device, have public IP addresses;

2. Both DAG serial and other VoIP Device are on the same LAN using private IP addresses;

3. Both DAG serial and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

1. Pick up the analog phone then dial "\*47"

2. Enter the target IP address.

**Note:** No dial tone will be played between step 1 and 2.

### Examples:

If the target IP address is 192.168.0.160, the dialing convention is **\*47**, then **192\*168\*0\*160**. Followed by pressing the "#" key or wait 3 seconds. Destination port is 5060.

### NOTE:

You cannot make direct IP calls between FXS0 to FXS1 since they are using same IP. It only supports the default destination port 5060. "*Disable direct IP-IP calling*" must be set to "No" in web configuration page. "*Call features enable*" must be set to "Yes" in web configuration page.

### 2.2 Call Hold

Place a call on hold by pressing the "flash" button on the analog phone (if the phone has that button).Press the "flash" button again to release the previously held Caller and resume conversation. If no "flash" button is available, use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

### 2.3 Call Waiting

Call waiting tone (3 short beeps) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the "flash" button. First call is placed on hold. Press the "flash" button to toggle between two active calls.

#### 2.4 Call Transfer

#### 2.4.1 Blind Transfer

Assume that call Caller A and B are in conversation. A wants to Blind Transfer B to C: 1. Caller A presses **FLASH** on the analog phone to hear the dial tone. 2. Caller A dials **\*87** then dials caller C's number, and then # (or wait for 4 seconds) 3. Caller A will hear the confirm tone. Then, A can hang up.

**NOTE:** *"Call features enable"* must be set to *"Yes"* in web configuration page. Caller A can place a call on hold and wait for one of three situations:

1. A quick confirmation tone (similar to call waiting tone) followed by a dial-tone. This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, Caller A can either hand up or make another call.

2. A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.

3. Continuous busy tone. The phone has timed out.

#### 2.4.2 Attended Transfer

Assume that Caller A and B are in conversation. Caller A wants to Attend Transfer B to C:

1. Caller A presses **FLASH** on the analog phone for dial tone.

2. Caller A then dials Caller C's number followed by # (or wait for 3 seconds).

3. If Caller C answers the call, Caller A and Caller C are in conversation. Then A can hang up to complete transfer.

4. If Caller C does not answer the call, Caller A can press "flash" to resume call with Caller B.

### **2.5 Call Features**

	Table 5.DAG serial Call Feature Definitions
*47	Direct IP Calling. Dial "*47" + "IP address". No dial tone is played in the
	middle.
*50	Disable Call Waiting(for all subsequent calls)
*51	Enable Call Waiting(for all subsequent calls)
*72	Unconditional Call Forward:
	Dial "*72" and then the forwarding number followed by "#". Wait for
	dial tone and hang up. (dial tone indicates successful forward)
*73	Cancel Unconditional Call Forward: To cancel "Unconditional Call
	Forward", dial "*73", wait for dial tone, and then hang up.
*78	Enable Do Not Disturb(DND): When enabled all incoming calls are
	rejected
*79	Disable Do Not Disturb(DND): When disabled, incoming calls are
	accepted.
*87	Blind Transfer
*90	Busy Call Forward: Dial "*90" and then forwarding number followed by
	"#".Wait for dial tone then hang up.
*91	Cancel Busy Call Forward. To cancel "Busy Call Forward", dial
	"*91",wait for dial tone, and then hang up.
*92	No Answer Call Forward. Dial "*92" and then the forwarding number
	followed by "#". Wait for dial tone then hang up.
*93	Cancel No Answer Call Forward. Dial "*93", wait for dial tone, and then
	hang up.
*99	Cancel all call forward
*114#	Report phone No
*158#	Report IP Address
*111#	Reset
Flash/Hook	Toggles between active call and incoming call(call waiting tone). If not in
	conversation, flash/hook will switch to a new channel for a new call.

The DAG serial supports all the traditional and advanced telephony features.

### 2.6 Sending and Receiving Fax

DAG serial supports fax in two modes: 1) T.38 (Fax over IP) and 2) fax pass through.

T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting T.38 as fax mode (default). If the service provider does not support T.38, pass-through mode may be used. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

# 3. Configuration Guide

### 3.1 Configure LAN Port's IP Address

Connect to the FXS port and then telephone set:

1. Dynamic IP address by DHCP:

```
Offhook;
```

```
Input "*150*2#";
```

Onhook;

If the equipment hint success, after 10 seconds, and restart the equipment. (Power-off then power-on)

2. Static IP address:

Offhook;

Input "\*150\*1#";

Onhook;

Then

- (1) Configure IP address: Offhook; input "\*152\*172\*16\*0\*100# "; onhook.
- (2) Configure netmask: Offhook; input "\*153\*255\*255\*0\*0# "; onhook.
- (3) Configure gateway IP address (next hop):Offhook; input "\*156\*172\*16\*0\*1# "; onhook.
- (4) if success, after 10 seconds, restart the equipment;
- 3. Query the IP address of DAG1000 : Offhook, input "\*158#"

4. If the DAG serial uses PPPoE method to get IP address, it need to configure by web browser. **NOTE**: the telephone will play voice hint "Setting successfully" if the setting step is correct.

### 3.2 Access DAG Serial By Web Browser

DAG serial has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow users to configure the DAG serial through a web browser such as Microsoft's IE.

### 3.2.1 Login

The DAG serial GUI configuration can be accessed via LAN or WAN port:

### From the LAN port

- 1. Offhook, input **\*158**# and get an IP address. The default gateway IP address: 192.168.11.1.
- 2. Directly connect a computer to the LAN port.
- 3. Open a command window on the computer.
- 4. Type in "ipconfig /release", the IP address etc. becomes 0.
- 5. Type in "ipconfig /renew", the computer gets an IP address in 192.168.11.x segment by default

6. Open a web browser, input the default gateway IP address. http://192.168.11.1. You will see the login page of the device.

7. The default username is admin while the default password is admin too.

#### From the WAN port

The WAN port HTML configuration option is disabled by default from factory. To access the HTML configuration menu from the WAN port:

- 1. Access the Web from LAN port.
- 2. Set Work Mode to Route Mode.
- 3. Enable the "Remote Manage".
- 4. Find the WAN IP address of the equipment .example:172.16.0.177 then restart the equipment.
- 5. Access the equipment Web Configuration page by the following URI via WAN port: http:// 172.16.0.177

### 3.3 Access DAG serial by Web Browser

The web page configuration includes the following:

- System Info
   Statistics
- Network Configuration
- + System Configuration
- Digit Map
- + Routing Configuration
- Manipulation Configuration
- + Advanced Configuration
- + Management Configuration

Figure 3-3-1 Navigation tree

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

22	Kun monnauon								
<u>IIC</u>	MAC Address			0.50					
Configuration	Network Mode		UU-IF-D0-AU-U bridge	JU-FD					
onfiguration	Network		172 30 53 40		255 255 0 0	Static			
	DNS Server		202.96.128.68		202.96.134.133	otato			
onfiguration	5110 001101		202.00.120.00		202.00.101.100				
ion Configuration	System Up Du	ration	407 hour 28 m	407 hour 28 minute 19 second					
Configuration	Network Traffic	Stat.	received 7455	received 745509299 bytes sent 138966764 bytes					
nt Configuration									
	Version inform	Version information		DAG1000-4S4O Rev 20.02.01 PCB 23.1 LOGIC 0 BIOS 1, Built on Sep 1 2011, 19:09:49					
	Port Group Informa	tion							
	Port Group No.	Туре	Port Map	Primary User I	D Primary Status	Secondary User ID	Secondary Status		
	n	FXS	n	82480	Registered		UnDrogistorod		

Figure 3-3-1 Configure Interface

### 3.3.1 System Information

MAC Address		00-1F-D6-A0-0	)0-FD			
Network Mode		bridge				
Network		172.30.53.40	2	55.255.0.0	Static	
DNS Server		202.96.128.68	2	02.96.134.133		
System Up Dur	ation	407 hour 28 m	inute 19 second			
Network Traffic	Stat.	received 7455	09299 bytes s	ent 138966764 bytes		
Version informa	ation	DAG1000-4S4	O Rev 20.02.01 PCB 23	8.1 LOGIC 0 BIOS 1, Bu	ilt on Sep 1 2011, 19:09:4	9
Group Informa	tion					
Port Group No.	Туре	Port Map	Primary User ID	Primary Status	Secondary User ID	Secondary Status

Refresh

Figure 3-3-2 System Information

System information interface shows the run information and port group information.

Table 3-3-1 Description of System Information	
	_

	The device ID in HEX format. This is needed for ISP					
MAC address	troubleshooting. Note there are separate MAC addresses for the					
	WAN side and the LAN side.					
Natwork Mode	Route mode or bridge mode, if it is bridge, WAN port display					
Network Mode	Network, and the WAN port as same as the LAN port					
	Show WAN IP address of equipment,.					
	DHCP mode:					
	The equipment acquires its IP address from the first DHCP server it					
WAN mont	discovers from the LAN it is connected.					
wAN port	Static IP mode:					
	Configure the IP address, Subnet Mask, Default Router IP address,					
	DNS Server 1 (primary), DNS Server 2 (secondary) fields. These					
	fields are set to zero by default.					
L AN port	Show LAN IP address of equipment. If network mode is bridge,					
LAN port	LAN port will no display.					
DNS Server	IP addresses of primary DNS server					
System Up Duration	Time elapsed from device power on to now.					
Network Traffic Statics	Total bytes of message received and sent by network port.					
Vancion Information	Includes: product mode, software version, hardware version and					
	built time etc.					
Port Group Information	Show FXS / FXO port information					

### 2.3.2 Statistics

stics												
TCP/L	JDP Stati	stics										
TCP Pack	Send et	TCP Rec Packet	v	TCP Send Byte	TCP Red Byte	cv U P	IDP Send Packet	UDP Recv Packet		UDP Se Byte	end	UDP Recv Byte
6528		650		2742044	742044 233948		00648	32424	12145		175	2516383
Curre	nt RTP S	tatistics										
Port	Payload Type	Packet Period	Loca Port	I Peer IF	)	Peer Port	Send Packet	Recv Packet	Lo: Pa	ss cket	Jitter	Duration Time(s)
0	none	0	0	0.0.0.0		0	0	0	0		0	0
1	none	0	0	0.0.0.0		0	0	0	0		0	0
2	none	0	0	0.0.0.0		0	0	0	0		0	0
3	none	0	0	0.0.0.0		0	0	0	0		0	0
4	none	0	0	0.0.0.0		0	0	0	0		0	0
5	none	0	0	0.0.0.0		0	0	0	0		0	0
6	none	0	0	0.0.0.0		0	0	0	0		0	0
7	none	0	0	0.0.0.0		0	0	0	0		0	0

**History Call Statistics** 

#### Figure3-3-3 Statistics

Statistics option includes three sets of statistics: TCP / UDP information, RTP messages and call history information. Check the user-friendly equipment failure.

#### 3.3.3 Network Configuration

Network parameter includes: Local Network, VLAN Config, Qos Parameter, ARP Config.



Figure 3-3-4 Network Configuration

### 1. Local Network

Local Network	
Work Mode	Route mode
WAN Port Parameter	
Link speed & duplex	Auto Detect
Obtain IP address automatically	
Use the following IP address	
IP address	172.30.53.40
Subnet mask	255.255.0.0
Default Gateway	172.30.0.1
© PPPoE	
Account	
Password	
Service Name	
LAN Port Config	
Link speed & duplex	Auto Detect
IP address	192.168.11.1
Subnet mask	255.255.255.0
DNS Server	
Obtain DNS server address automatically	
Use the following DNS server addresses	
Primary DNS server	202.96.128.68
Secondary DNS server	202.96.134.133

Figure 3-3-5 Local Network

Work Mode	This parameter controls whether the device is working in NAT
WOIK WIOUC	router mode or Bridge mode.
	This option specifies the WAN port's Ip address, and its Ethernet
	network work mode.
	Link speed & duplex This option specifies the Ethernet
	network's work mode. It have: Auto Detect,10Mbps/Half
	Duplex,10Mpbs/Full Duplex,100Mbps/Half Duplex,
	100Mpbs/Full Duplex, There are three modes to operate the
	DAG serial WAN IP address, default option is Dynamically
	assigned via DHCP.
WAND	Dynamically assigned via DHCP: All the field values for the
WAN Port Parameter	Static IP mode are not used. The equipment acquires its IP
	address from the first DHCP server it discovers from the LAN it
	is connected.
	Static IP mode: Configure the IP address, Subnet Mask, Default
	Router IP address, DNS Server 1 (primary), DNS Server 2
	(secondary) fields. These fields are set to zero by default.
	Dynamically assigned via PPPoE : Set the PPPoE account
	settings. The equipment will establish a PPPoE session if any of
	the PPPoE fields is set.

	This option specifies the LAN port's ip address, and its Ethernet
	network work mode.
	Link speed & duplex:
LAN Port Config	This option specifies the Ethernet network's work mode. It have:
	Auto Detect,10Mbps/Half Duplex,10Mpbs/Full
	Duplex,100Mbps/Half Duplex,100Mpbs/Full Duplex.
	IP address: Set LAN IP address, default value is 192.168.11.1
	Subnet mask: Sets the LAN subnet mask. Default value is
	255.255.255.0
	This option specifies get the DNS server ways: Dynamically
DNS Server	assigned via DNS(WAN IP address is DHCP IP) and set static
	DNS (WAN IP address is static IP).

### 2. VLAN Parameter Config

Data VLAN	Enable
Data VLAN use the default WAN interface in this case.	
Data 802.1Q VLAN ID (0 - 4095)	0
Data 802.1p Priority (0 - 7)	0
Obtain IP address automatically	
Use the following IP address	
IP address	172.30.53.40
Subnet mask	255.255.0.0
Default Gateway	172.30.0.1
Opnamically assigned via PPPoE	
Account	
Password	
Service Name	
Voice VLAN	Enable
Voice 802.1Q VLAN ID (0 - 4095)	0
Voice 802.1p Priority (0 - 7)	0
Voice VLAN use following separate IP interface	
Obtain IP address automatically	
Use the following IP address	
IP address	
Subnet mask	
Default Gateway	
Management VLAN	Enable
Management 802.1Q VLAN ID (0 - 4095)	0
Management 802.1p Priority (0 - 7)	0
Management VLAN use following separate IP interface	
Obtain IP address automatically	
Use the following IP address	
IP address	
Subnet mask	

Note: It must restart the device to take effect.

Save

Figure 3-3-6 VLAN Parameter Config

When a network has a very wide range of applications, in order to artificially divided into different applications to different networks, to prevent them interfering with each other and to different networks with different bandwidth, customer can use the VLAN.

VLAN parameter configuration can configure three VLAN as follows.

(1)	Data VLAN	
· - /		

Data VLAN use the	Selecting "Enable" will configure Data VLAN.
default WAN interface in	
this case	
Data 802.1Q VLAN	Fill out an ID to describe a D ata VLAN group.
ID(0-4095)	
Data 802.1p Priority(0-7)	802.1 protocol to control network traffic priority, Priority from 0-7.
Obtain IP address	Select this to automatically obtain the IP of VLAN group
automatically	
Use the following IP	Fill in IP, Subnet mask and Default Gateway
address	
Dynamically assigned via	Select this when the Internet through PPPOE and complete account
PPPoE	number and password

Voice 802.1Q VLAN ID (0-4095)	Fill out an ID to describe a Voice VLAN group.	
Voice 802.1p Priority(0-7)	802.1 protocol to control network traffic priority, Priority	
	from 0-7.	
Obtain IP address automatically	Select this to automatically obtain the IP of VLAN group	
Use the following IP address	Fill in IP, Subnet mask and Default Gateway	
(3)Management VLAN		
Management 802.1Q VLAN ID	Fill out an ID to describe a Management VLAN group.	
(0-4095)		
Management 802.1p Priority(0-7)	802.1 protocol to control network traffic priority, Priority	
	from 0-7.	
Obtain IP address automatically	Select this to automatically obtain the IP of VLAN group	
Use the following IP address	Fill in IP, Subnet mask and Default Gateway	

#### (2)Voice VLAN

### 3. Qos Parameter

Qos Config	
DSCP code point is used for diffserv setting. It utilize the first 6 bits of I is EF(184), AF1(1), AF2(2), AF3(3), AF4(4), BE(0). You can use differe according to the network provider.	P ToS. The default definition ent DSCP for voice or data
DSCP code/IP ToS define	◎ no ◎ yes
Manage/signal packet: (default is 48)	0
Voice packet: (default is 48)	0
Data packet: (default is 48)	0

Figure 3-3-7 Qos Parameter

Save

DSCP code point is used for diffserv setting. It utilizes the first 6 bits of IP ToS. You can use different DSCP for voice or data according to the network provider.

### 4. ARP Config

Add ARP	
IP Address	
MAC Address	
	The IP format is: xxx.xxx.xxx.xxx The MAC format is: xx-xx-xx-xx-xx
	OK Search All
	Figure 3-3-8 ARP Config

Manually add the address resolution.

### 3.3.4 System Configuration

System parameter includes 1) System Config 2) Service Config 3) SIP Config 4) Port Config 5) Fax Config.

<ul> <li>System Config</li> <li>Service Config</li> <li>SIP Config</li> <li>Port Config</li> <li>Fax Config</li> </ul>	- System Configuration
<ul> <li>Service Config</li> <li>SIP Config</li> <li>Port Config</li> <li>Fax Config</li> </ul>	System Config
<ul> <li>SIP Config</li> <li>Port Config</li> <li>Fax Config</li> </ul>	<ul> <li>Service Config</li> </ul>
<ul><li>Port Config</li><li>Fax Config</li></ul>	<ul> <li>SIP Config</li> </ul>
<ul> <li>Fax Config</li> </ul>	<ul> <li>Port Config</li> </ul>
	<ul> <li>Fax Config</li> </ul>

Figure 3-3-9 System Configuration

#### 1. System Config

The page of system setting is mainly used to set some system parameters, including: 1)Provision Parameter; 2)NTP Parameter; 3)Time Zone

tem Config			
Provision Parameter			
Primary Profile URL			
Secondary Profile URL			
Check Interval	24	hours	
NTP Parameter			
NTP Enable	🔍 no 🔍 yes		
Primary NTP Server IP	us.pool.ntp.org		
Secondary NTP Server IP	18.145.0.30		
Time Zone GMT-6:0	0 (US Central Time, Chicago)		-

Note: It must restart the device to take effect.

	save	
	Figure 3-3-10 System Config	
Table 3-3-3 System Config		
Provision Parameter	Periodically, the server will automatically load the latest configuration	
	and version	
NTP Parameter	NTP server management system time. Select "NTP Enable" to "YES",	
	you can configure the NTP server's IP and system time.	

### 2. Service Config

In order to adapt to different environments, it is necessary to set some parameters in the service config page.

Service config	
RTP start port	8000
	<b>N</b> 44
silence suppression enable	≪ no⊙yes
Call progress tone	UBA
8LIC setting	UBA
Hook Flash Deteot	
Hock Hesh Close	≪ no © yes
NDX INC	400 115
Preferred Vocoder(In listed order)	
Choice 0	G.729AB
Choice 1	PCMU 🔽
Choice 2	PCMA
Volce frames per TX	2
Notice: The device will restart automatically when preferred vocoder is changed between 0.723.1 and 0.728AB	L.
FXO Parameter	
FXD keep onhook until called offhook	® no© yes
FXO config enable	C no 8 yes
FXO round robin enable	C no 🕷 yes
FXD round robin type	Poli
FXO port 1 stage calling enable	© no≪iyes
FXO is detect polarity reversal	® no © yes
FXO Answer Delay	5 s
Play hint to FXO enable	© no⊛ yes
Send real caller ID enable	® no© yes
Tone disconnect enable	C no 6 yes
Current disconnect enable	® no© yes
FXO silence timeout	600
DTME Paramatar	
DTMF refraites	BIGNAL
DTMF volume	OdB V
DTMF send interval	200 ms
8TUN enable	® no© yes
	Muselee
Notice: if 'came' is calented, please ensure there is no letter in it	Number
8DP parameter when hold	sendonly 🔽
Other config	
Episity reversal epishie	8 m 8 m
Califacturar enable	No no Cityes
Diverse contra englia	Concernent and the second s
Disable direct IB-IP callon	Concernences
Liker (D is nhose number	C no my pesa Ri no fil mare
Colvercent server call in	non u yea Non Al yea
Alow make call without register	n nu v jes N na <sup>(1)</sup> vez
Allow answer cell without register	and a second and a second a s
Send Announces	n nu u yea Mara Ali war
Relact accounts call	n nu u yea Mina Ali war
Lise ± as dial key	e na v jez
No dial fimenit	4 no 10 yes
No Answer Timeout	55 5
Ring Timeout	55 5
No Reply Forwarding Timeout	33 5

Note: It must restart the device to take effect.

Figure 3-3-11 Service Config

save

RTP start port	Defines the local RTP-RTCP port for listening and transmission.		
Silence Suppression	Endpoints sending audio as an RTP stream are not required to send		
	packets during silent periods. The capability to stop sending RTP		
	packets during silent periods is known as "Silence Suppression" or		
	VAD (Voice Activity Detection).		
Call Progress Tone	Configure ring or tone frequencies according to preference. By default		
	tones are set to USA frequencies. Frequencies should be configured		
	with known values to avoid uncomfortable high pitch sounds.		
SLIC Setting	Dependent on standard phone type (and location).		
Hook Flash Detect	Time period when the cradle is pressed (Hook Flash) to simulate		
	FLASH. To prevent unwanted activation of the Flash/Hold and		
	automatic phone ring-back, adjust this time value.		
Preferred Vocoder	The equipment supports up to 4 different vocoder types including		
	PCMU, PCMA, G.723.1, G.729AB.The user can configure vocoders in		
	a preference list that will be included with the same preference order in		
	SDP message. The first vocoder is entered by choosing the appropriate		
	option in "Choice 0". The last vocoder is entered by choosing the		
	appropriate option in "Choice 3".		
FXO Parameter	Many FXO configuration parameters to configure the FXO port		
DTMF	Flexible DTMF transmission method, user interface of in-audio,		
	RFC2833/INBAND/SIGNAL.		
SDP parameter when	When the call was hold ,the invitation of SDP the parameters can be		
hold	carried inactive or sendonly.		
STUN	IP address or port of the STUN server		
Incoming display	Caller ID There are two options: the name and number		
Polarity Reversal			
I blanty Reversal	Default is No. If set to Yes, polarity will be reversed upon call		
Tolarity Reversar	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination.		
Send Flash Event	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination. Default is NO. If set to yes, flash will be sent as DTMF event		
Send Flash Event Call Features	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination. Default is NO. If set to yes, flash will be sent as DTMF event Default is Yes. (If Yes, call features using star codes will be supported		
Send Flash Event Call Features	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination. Default is NO. If set to yes, flash will be sent as DTMF event Default is Yes. (If Yes, call features using star codes will be supported locally)		
Send Flash Event Call Features Direct IP-IP Calling	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination. Default is NO. If set to yes, flash will be sent as DTMF event Default is Yes. (If Yes, call features using star codes will be supported locally) Default is NO		
Send Flash Event Call Features Direct IP-IP Calling Send Anonymous	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination. Default is NO. If set to yes, flash will be sent as DTMF event Default is Yes. (If Yes, call features using star codes will be supported locally) Default is NO Default is NO. If this parameter is set to "Yes", users ID will be sent as		
Send Flash Event Call Features Direct IP-IP Calling Send Anonymous	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination. Default is NO. If set to yes, flash will be sent as DTMF event Default is Yes. (If Yes, call features using star codes will be supported locally) Default is NO Default is NO. If this parameter is set to "Yes", users ID will be sent as anonymous; essentially block the Caller ID from displaying.		
Send Flash Event Call Features Direct IP-IP Calling Send Anonymous Reject Anonymous Call	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination. Default is NO. If set to yes, flash will be sent as DTMF event Default is Yes. (If Yes, call features using star codes will be supported locally) Default is NO Default is NO. If this parameter is set to "Yes", users ID will be sent as anonymous; essentially block the Caller ID from displaying. Default is NO. If set to yes, incoming calls with anonymous Caller ID		
Send Flash Event Call Features Direct IP-IP Calling Send Anonymous Reject Anonymous Call	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination. Default is NO. If set to yes, flash will be sent as DTMF event Default is Yes. (If Yes, call features using star codes will be supported locally) Default is NO Default is NO. If this parameter is set to "Yes", users ID will be sent as anonymous; essentially block the Caller ID from displaying. Default is NO. If set to yes, incoming calls with anonymous Caller ID will be rejected with 486 busy message.		
Send Flash Event Call Features Direct IP-IP Calling Send Anonymous Reject Anonymous Call No Dial Timeout	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination. Default is NO. If set to yes, flash will be sent as DTMF event Default is Yes. (If Yes, call features using star codes will be supported locally) Default is NO Default is NO. If this parameter is set to "Yes", users ID will be sent as anonymous; essentially block the Caller ID from displaying. Default is NO. If set to yes, incoming calls with anonymous Caller ID will be rejected with 486 busy message. Default is 4 seconds		

Table 3-3-4 Service Config

### 3. SIP Config

It is used to set the local SIP port, the SIP server address and port. If an outbound proxy is required, please fill in the IP address and port of the outbound proxy; otherwise, just keep the default settings.

SIP Config	
Primary SIP Server	470 40 400 00
Primary SIP Server Address	1/2.16.100.08
Primary SIP Server Port (Default: 5060)	5060
Register Interval (Default: 1800)	1800 s
Secondary SIP Server	
Secondary SIP Server Address	
Secondary SIP Server Port (Default: 5060)	5060
Register Interval (Default: 1800)	1800 s
Outbound proxy	
Outbound proxy address	
Outbound proxy port	5060
SUBSCRIBE for MWI(Message Waiting Indicator)	no  ves
VM user ID	
Use random port	● no ◎ ves
local sip port	5060
Is register	© no ⊛ yes
DNS query type	A query
DNS refresh interval (range:0 - 60,000min, 0 means disable)	0 min
T1	500 ms
T2	4000 ms
T4	5000 ms
TMAX	32000 ms
Keepalive interval(range:0 - 3600s,0 means disable)	10 s
100rel enable	🖲 no 🗇 yes

#### Note: It must restart the device to take effect.

#### save

#### Figure 3-3-12 SIP Config

#### Table 3-3-5 SIP Config

Primary SIP Server	Fill the SIP Server IP address, Port(default:5060), Register
	Interval(default:1800s)
Secondary SIP Server	The SIP server as a backup
Outbound proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or
	Session Border Controller. Used by equipment for firewall or NAT
	penetration in different network environments. If symmetric NAT is
	detected, STUN will not work and only outbound proxy can correct the
	problem.
Subscribe for MWI	Default is No
VM user ID	Voice Mail user ID
Use random port	If selected "Yes", the port number will be randomly generated
Local sip port	If "Use random port" option was set to "No", this option will be used.
	Default is 5060.
Is register	Default is Yes, if it was set No, the stand will not register the SIP
	server, and the RUN LED will quick blinking (resisted status is Yes)
Keepalive interval	This parameter specifies how often the equipment sends a blank UDP
	packet to the SIP server in order to keep the "hole" on the NAT open.
	Default is 10 seconds.

### 4. Port Config

This page is used to config the account and password of DAG serial, call transfer, call waiting and do-not-disturb service. Port 0 maps to FXS0, while Port n maps to FXS n.

Port Config	
Current Port	Port 0
Tx Gain	0dB
Rx Gain	0dB
Offhook Auto-Dial	
Enable DND	● No <sup>©</sup> Yes
Enable Caller-ID	◎ No   Yes
Note: The parameters below won't take effect if the port wa Primary SIP Account SIP User ID Authenticate ID Authenticate Password	s added into hunting group.
Secondary SIP Account	
SIP User ID	
Authenticate ID	
Authenticate Password	
Call Forwarding Unconditional	
Call Forwarding Busy	
Call Forwarding No Reply	
Disable Call Waiting	No
Disable Call Waiting Tone	No ○ Yes

Save

Figure 3-3-13 Port Config

Table 3-3-6 Port Config

Current Port	Select configure a SIP trunk port
TX/RX Gain	Handset volume adjustment. RX is for receiving volume, TX is
	for transmission volume. Default values are -6dB for both
	parameters. Loudest volume: +6dB Lowest volume: -6dB.
Offhook Auto-Dial	Fill in the number of offhook auto-dial
Enable DND	Default is No. When enabled all incoming calls are rejected.
Enable Caller-ID	Default is Yes
SIP User ID	SIP count number
Authenticate ID	SIP user name which registers to soft switch/SIP server
Authenticate password	SIP password which registers to soft switch/SIP server
Call Forwarding Unconditional	All incoming calls are transferred appointed phone.
Call Forwarding Busy	Call will transfer appointed phone when Busy.
Call Forwarding No Reply	Call will transfer appointed phone when nobody answer the
	call.
Disable Call Waiting	Default is No. User can use * code to use this feature per call
	basis.
Disable Call Waiting Tone	Default is No. Indicates an incoming call, default is 2 short
	beeps.

### 5. Fax Config

Fax Config	
Fax mode	T.38
Fax tone detection mode	Automatic
Enable ECM	no O yes
Fax rate	14400 bps

Note: It must restart the device to take effect.

#### save

#### Figure 3-3-14 Fax Config

### Table 3-3-7 Fax Config

Fax mode	T.38 (Auto Detect) FoIP by default, or Pass-Through
Fax tone detection mode	Default is Automatic. There are callee and caller two options
ECM	Error Correction Mode
Fax rate	The equipment support most 14400bps least 2400bps fax rate. The
	default value is 14400bps.

### 3.3.5 Digit Map

Digit Map	
Digit Map	x.# x.T
	NOTE: Length of 'Digit Map' should be not more than 120 characters.
	Save
Digit Map Syntax:	
<ol> <li>Supported objects Digit: A digit from Timer: The symbol " DTMF: A digit, a tip</li> <li>Range [] One or more DTMF synonly one can be self</li> <li>Range () One or more expressionly one can be self</li> <li>Separator [: Separator</li> <li>Subrange Two digits separated expression construct, i.e., bef</li> <li>Wildcard x: matches any digit</li> <li>Modifiers : Match 1 or more f</li> <li>Modifiers ?: Match 0 or 1 tim</li> </ol>	a "0" to "9". T" matching a timer expiry. mer, or one of the symbols "A", "B", "C", "D", "\$", or "*". mbols enclosed between square brackets ("[" and "]"), but ected. sions enclosed between round brackets ("(" and ")"), but ected. sions or DTMF symbols. ed by hyphen ("-") which matches any digit between and The subrange construct can only be used inside a range etween "[" and "]". t ("0" to "9"). times. times. ees.

#### Figure 3-3-15 Digit Map

In the "digit map" set dial rules and dialing rules can't exceed 120 characters in length.

### **3.3.6 Routing Configuration**

### **1. Routing Parameter**

Routing Parameter		
IP in Parameter	Route calls before manipulation	
Tel in Parameter	Route calls before manipulation	•
	Save	

Figure 3-3-16 Routing Parameter

This option determines the following routing of call take effect before or after manipulation.

### 2. IP in Routing

IP ir	n Routing					
	Index	Description	Source IP	Source Prefix	Destination Prefix	Destination
	0	toPort0	SIP Server	any	82480	Port Group 0
	31	default	Any	any	any	Port Group 0
			_			

Total: 2entry 16entry/page 1/1page Page 1 -

Add Delete Modify

Figure 3-3-17 IP in Routing

30		-
IP Trunk	Any 🔻	
SIP Server		
Port	0 🔻	
Port Group	0 <port0></port0>	
	30 IP Trunk SIP Server Port Port	30 ○ IP Trunk Any ▼ ◎ SIP Server ○ Port 0 ▼ ◎ SIP Server



i iguie 5 5 10 ii m Routing / iuu
-----------------------------------

Index	Routing priority :1-30
Description	Description the routing
Source Prefix	Source number Prefix
Source	IP Trunk/SIP Server, any means wildcard string
Destination Prefix	Destination number Prefix
Destination	Select a single port or port group

### 3. Tel in Routing

	Index	Description	Source Port	Source Prefix	Destination Prefix	Destination n				
	31	default	Any	any	any	SIP Serve				
tal: 1entry	16entry/page	1/1page Page 1 💌								
		A	dd Delete	Modify						
Figure 3-3-19 Tel in Routing										

ort ort Group	Any 0 <port0></port0>		•			
ort ort Group	Any 0 <port0></port0>		•			
ort ort Group	Any 0 <port0></port0>		▼ ▼			
ort Group	0 <port0></port0>		-			
ort	0		-			
ort Group	0 <port0></port0>		-			
P Trunk			-			
IP Server						
	ort ort Group P Trunk IP Server	ort 0 ort Group 0 <port0> P Trunk P Server</port0>	ort 0 ort Group 0 <port0> P Trunk IP Server</port0>	ort 0   ort Group 0 <port0>  Trunk  P Server</port0>	ort 0  ort Group 0 <port0> Trunk P Server</port0>	ort 0  ort Group 0 <port0> Trunk P Server</port0>

NOTES: 'Destination Prefix' or 'Source Prefix' field: 'any' means wildcard string.

Figure 3-3-20 Tel in Routing Add

Table 3-3-9 Tel in Routing Add
--------------------------------

Index	Routing priority :1-30
Description	Description the routing
Source Prefix	Source number Prefix
Source	Select a single port or port group
Destination Prefix	Destination number Prefix
Destination	Select a single port or port group, IP Trunk/SIP Server

### 3.3.7 Manipulation Configuration

### **1. IP in Destination Numbers**

IP	in Destir	nation Numbe	rs								
	Inde x	Description	Source	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
Tota	I: Oentry	16entry/page 1/	Opage 🔽								



Figure 3-3-21 IP in Destination Numbers

P in Destination Numbers	s Add		
Index	31		-
Description			
Source Prefix			
Source	IP Trunk	Any 👻	
	SIP Server		
Destination Prefix			
Destination	Port	Anv	
	Port Group	0 <port0></port0>	
Stripped Digits from Left			
Stripped Digits from			
Right			
Prefix to Add			
Suffix to Add			
Number of Digits to			$\neg$
Leave from Right			

NOTE: 'Destination Prefix' or 'Source Prefix' field: 'any' means wildcard string.



Figure 3-3-22 IP in Destination Numbers Add

Table 3-3-10 II III Desultation numbers Adu
---

Index	Routing priority :1-30
Description	Description the routing
Source Prefix	Source number Prefix
Source	IP Trunk/SIP Server, any means wildcard string
Destination Prefix	Destination number Prefix
Destination	Select a single port or port group
Stripped Digits from Left	Remove the called number digits from the left
Stripped Digits from Right	Remove the called number digits from the right
Prefix to Add	Add a number prefix
Suffix to Add	Add a number suffix
Number of Digits to Leave	Starting from the right to retain the called number digits
from Right	

#### 2. Tel in Source Numbers

Tel	Tel in Source Numbers										
	Inde x	Description	Source	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	I: Oentry	16entry/page 1/0	page 🖵								

Add Delete Modify

Figure 3-3-23 Tel in Source Numbers

el in Source Numbers A	bid		
			_
Index	31		1
Description			]
Source Prefix			]
Source	Port	Any	-
	Port Group	0 <port0></port0>	
Destination Prefix			]
Destination	Port	0	-
	Port Group	0 <port0></port0>	
	IP Trunk	Any	
	SIP Server		
Stripped Digits from Left			]
Stripped Digits from Right			]
Prefix to Add			]
Suffix to Add			1
Number of Digits to Leave from Right			]

NOTE: 'Destination Prefix' or 'Source Prefix' field: 'any' means wildcard string.

OK	Reset	Cancel

Figure 3-3-24 Tel in Source Numbers Add

Configuration parameters are the same with "IP in Source Numbers Add".

#### **3.Tel in Destination Numbers**

Tel in Dest	ination Numbe	rs								
Inde x	Description	Source	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/	)page 🔽								
				Add	Delete	odify				

Figure 3-3-25 Tel in Destination Numbers

Configuration parameters are the same with "IP in Source Numbers Add".

### 3.3.8 Advanced Configuration

### 1. Port Group

Port	Group							
For	Index	Description	Primary SIP User ID	Primary Authenticate ID	Secondary SIP User ID	Secondary Authenticate ID	Port	Select Mode
	0	port0	82480	82480			0,	Cyclic Ascending
Total: '	Fotal: 1entry 16entry/page 1/1page Page 1         Add       Delete       Modify         Figure 3-3-26 Port Group							
Po	rt Grou	ıp Add						
	Index			7			•	

Index	7		-
Description			
Primary SIP User ID			
Primary Authenticate ID			
Primary Authenticate Password			
Secondary SIP User ID			
Secondary Authenticate ID			
Secondary Authenticate Password			
Select Mode	Ascending		-
Port	Port 0(FXS)	Port 1(FXS)	
	Port 2(FXS)	Port 3(FXS)	
	Port 4(FXO)	Port 5(FXO)	
	Port 6(FXO)	Port 7(FXO)	

OK Reset

Reset Cancel

Figure 3-3-27 Port Group Add

Table 3-3-11 Port Group Add

Index	Priority of the port group :1-7, 1 is highest priority value.
Description	Descript the port group
Primary/Secondary SIP User ID	SIP count number
Primary/Secondary Authenticate ID	SIP user name which registers to soft switch/SIP server
Primary/Secondary Authenticate	SIP password which registers to soft switch/SIP server
Password	
Select Mode	There four options: Ascending, Cyclic Ascending,
	Descending, Cyclic Descending
Port	Port0-port7, By checking the different ports to form a port
	group

### 2. IP Trunk

Index	63	•
Port		
Description		
	OK Reset Cancel	

Index	64 Priority values from 0-63
IP	Set port IP
Port	Set port number
Description	The description of IP trunk

Backup

### 3.3.9 Management Configuration

### 1. Firmware Upload

Send "ldf" f	ile from your computer to the device.
Software	浏览 Upload
	NOTE: 1. The upload process will last about 60s. 2. The device will restart automatically after upload. 3. Do not shut down when the device is uploading.
	Figure 3-3-29 Firmware Upload
	Table 3-3-13 Firmware Upload Add
	-
oftware	Click "Browse" to select the firmware, and then click "Upload".

Figure 3-3-30 Data Backup

Click the Backup, and save the configuration file in your PC.

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

### 3. Config Restore

Send data file from y	our computer to th	e device.		
Configuration			浏览	Restore

Figure 3-3-31 Data Restore

Click "Browse" to select the Configuration file, and then click "Restore".

### 4.System Log

Syslog Config	
Enable Syslog	no
Server Address	
Syslog Level	NONE
Send CDR	◉ no ♡ yes
	Caus

Figure 3-3-32 Syslog Config

Table	3-3-14	Syslog	Config
rabic	5-5-14	by slog	Conng

Enable Syslog	Default is No
Server Address	Storage system log server address
Syslog Level	At present only two options(None and Debug) are available
Send CDR	Default is No

### 5. Ping Test

Ping Test	
Ping Destination	
Number of Ping(1-100)	4
Ping Packet Size(56-1024 bytes)	56

#### Start Stop

Information		
	A	

#### Figure 3-3-33 Ping Test

#### Table 3-3-15 Ping Test

Ping Destination	Destination IP address
Number of Ping(1-100)	Number of ICMP packets
Ping Packet Size(56-1024 bytes)	Length of Packet

### 6. Tracert Test

Tracert Test	
Tracert Destination	
Max Hops of Tracert(1-255)	30
	Start Stop
Information	
	·
	*

#### Figure 3-3-34 Tracert Test

Fill "Tracert Destination IP" and "Max hops of tracert(1-255)", then click "start", tracking information will be displayed in the space below.

### 7. Login Password

Modify login user name and password to avoid unauthorized access to your DAG serial . The default username is admin while the default password is admin too.

Username & Password	
Web Orefe	
vveb Config	
Old web username	admin
Old web password	
New web username	
New web password	
Confirm web password	
Telnet Config	
Old telnet username	admin
Old telnet password	
New telnet username	
New telnet password	
Confirm telnet password	

#### save

Figure 3-3-35 Login Password

### 8. Factory Reset

Factory Reset	
	Click this button to reset factory default settings
	Apply

Figure 3-3-36 Factory Reset

Click "Apply" to restore the factory settings.

#### 9. Restar

Restart	
	Click this button to restart the device.
	Restart

Figure 3-3-37 Restar

Click the "Save" button in the Configuration page to save the changes to the equipment configuration. The following screen confirms that the changes are saved. If the changes need restart, reboot or power cycle the equipment to make the changes take effect.

# 4. FAQ

### 4.1 How to get the IP address if I have modified the default IP or forgot it ?

Customers have two ways to get the IP address.

1) Press the RST button, then customer can retain the default IP of LAN port.

2) Analog telephone by dialing "\*", re-set the IP address, refer to 3.1.

4.2 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 work states of physical connection;

2) Make sure the connected network devices (router, switch or hub) support 10M/100M adaptive. Else, connecting the Equipment directly to PC and landing WEB, then in the "local connection" Selecting the correct Ethernet Work Mode;

3) Check the Network Configuration, if the Configuration is incorrect, please re-Configuration. If customers are using DHCP mode, so check whether DHCP Server work properly;

4) Check whether there is a LAN port conflict with the existing IP address.

### 4.3 Equipment can't register

If the Run LED flashes slowly ,it means 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:

(1) 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;

4.4 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.

4.5 When calling in, the caller always hears a busy tone

Make sure enable DND(Do-not-Disturb) in system

**4.6** 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 please download tutorials from our official website .

### 5. Glossary

**ADSL** Asymmetric Digital Subscriber Line: Modems attached to twisted pair copper wiring that transmit from 1.5 Mbps to 9 Mbps downstream (to the subscriber) and from 16 kbps to 800 kbps upstream, depending on line distance.

**AGC** Automatic Gain Control is an electronic system found in many types of devices. Its purpose is to control the gain of a system in order to maintain some measure of performance over a changing range of real world conditions.

**ARP** Address Resolution Protocol is a protocol used by the Internet Protocol (IP) [RFC826], specifically IPv4, to map IP network addresses to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when IPv4 is used over Ethernet

**CODEC** Abbreviation for Coder-Decoder. It's an analog-to-digital (A/D) and digital-to-analog (D/A) converter for translating the signals from the outside world to digital, and back again.

**CNG** Comfort Noise Generator, generate artificial background noise used in radio and wireless communications to fill the silent time in a transmission resulting from voice activity detection. **DATAGRAM** A data packet carrying its own address information so it can be independently routed from its source to the destination computer

**DECIMATE** To discard portions of a signal in order to reduce the amount of information to be encoded or compressed. Lossy compression algorithms ordinarily decimate while sub-sampling. **DNS** Short for *Domain Name System* (or *Service* or *Server*), an Internet service that translates *domain names* into IP addresses

**DSP** Digital Signal Processor. A specialized CPU used for digital signal processing. Allywll products all have DSP chips built inside.

**DTMF** Dual Tone Multi Frequency. The standard tone-pairs used on telephone terminals for dialing using in-band signaling. The standards define 16 tone-pairs (0-9, #, \* and A-F) although most terminals support only 12 of them (0-9, \* and #).

**FXO** Foreign exchange Office. An FXS device can be an analog phone, answering machine, fax, or anything that handles a call from the telephone company like AT&T. They should also operate the same way when connected to an FXS interface. • An FXS interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards. • FXS is complimentary to FXS (and the PSTN).

**FXS** Foreign exchange Station. An FXS device has hardware to generate the ring signal to the FXS extension (usually an analog phone). • An FXS device will allow any FXS device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone. • The FXS Interface connects to FXS devices (by an FXS interface, of course).

**DHCP** The *Dynamic Host Configuration Protocol* (DHCP) is an Internet protocol for automating the configuration of computers that use TCP/IP. DHCP can be used to automatically assign IP addresses, to deliver TCP/IP stack configuration parameters such as the subnet mask and default router, and to provide other configuration information such as the addresses for printer, time and news servers.

**ECHO CANCELLATION** Echo Cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call.

In addition to improving quality, this process improves bandwidth savings achieved through silence suppression by preventing echo from traveling across a network. There are two types of echo of relevance in telephony: acoustic echo and hybrid echo. Speech compression techniques and digital processing delay often contribute to echo generation in telephone networks.

H.323 A suite of standards for multimedia conferences on traditional packet-switched networks.

**HTTP** Hyper Text Transfer Protocol; the World Wide Web protocol that performs the request and retrieve functions of a server

**IP** Internet Protocol. A packet-based protocol for delivering data across networks **IP-PBX** IP-based Private Branch Exchange