P100

VoIP Phone User Manual

Model: P100
Version 1.0.3
User Manual

Index

PREFACES

About This Manual

This manual is designed to assist users in using P100 VOIP Phone. Information in this document has been carefully checked for accuracy; however, no guarantee is given as to the correctness of the contents. The information contained in this document is subject to change without notice.

Copyright Declarations

Trademarks

Products and Corporate names appearing in this manual may or not be registered trademarks or copyrights of their respective companies, and are used only for identification or explanation and to the owners' benefit, without to infringe.

Safety Instructions

The most careful attention has been devoted to quality standards in the manufacture of the P100. Safety is a major factor in the design of every set. But, safety is your responsibility too.

Use only the required power voltage. Power Input: AC 100-240V, 50-60Hz

To reduce the risk of electric shock, do not disassemble this product. Opening or removing covers may expose the P100 to hazardous voltages. Incorrect reassembly can cause electric shock when this product is subsequently used.

Never push objects of any kind into the equipment through housing slots since they may touch hazardous voltage points or short out parts those could result in a risk of electric shock. Never spill liquid of any kind on the product. If liquid is spilled, please refer to the proper service personnel.

Use only Unshielded Twisted Pair (UTP) Category 5 Ethernet cable to RJ-45 port of the P100.

Warranty

We warrant to the original end user (purchaser) that the P100 VOIP Phone will be free from any defects in workmanship or materials for a period of one (1) years from the date of purchase from the dealer. Please keep your purchase receipt in a safe place as it serves as proof of date of purchase. During the warranty period, and upon proof of purchase, should the product have indications of failure due to faulty workmanship and/or materials, we will, at our discretion, repair or replace the defective products or components, without charge for either parts or labor, to whatever extent we deem necessary to re-store the product to proper operating condition. Any replacement will consist of a new or re-manufactured functionally equivalent product of equal value, and will be offered solely at our discretion. This warranty will not apply if the product is modified, misused, tampered with, damaged by an act of God, or subjected to abnormal working conditions. The warranty does not cover the bundled or licensed software of other vendors.

Defects which do not significantly affect the usability of the product will not be covered by the warranty. We reserve the right to revise the manual and online documentation and to make changes from time to time in the contents hereof without obligation to notify any person of such revision or changes.

Note

Repair or replacement, as provided under this warranty, is the exclusive remedy of the purchaser. This warranty is in lieu of all other warranties, express or implied, including any implied warranty of merchantability or fitness for a particular use or purpose. We shall in no event be held liable for indirect or consequential damages of any kind of character to the purchaser.

To obtain the services of this warranty, contact us for your Return Material Authorization number (RMA). Products must be returned Postage Prepaid. It is recommended that the unit be insured when shipped. Any returned products without proof of purchase or those with an out-dated warranty will be repaired or replaced and the customer will be billed for parts and labor. All repaired or replaced products will be shipped by us to the corresponding return address, Postage Paid. This warranty gives you specific legal rights, and you may also have other rights that vary from country to country.

1 Introduction

This user's manual is for VoIP Phone. This user's manual will explain the keypad instruction, web configuration and command line configuration for the VoIP Phone. Before using the VoIP Phone, some setup processes are required to make the VoIP Phone work properly. Please refer to the Setup Menu for further information.

1.1 Hardware Overview

The VoIP Phone has the following interfaces for Networking, telephone interface, LED indication, and power connector.

1.1.1 Two RJ-45 Networking interface, these two interfaces support 10/100Mps Fast Ethernet. you can connect one RJ-45 Fast Ethernet port to the ADSL or Switch, and connect the other one to your computer.

1.1.2 LED Indication: There are some LED indicators in the VoIP Phone to show the functions, like speaker phone, .Rggister, ….

1.2 Software Overview

Network Protocol	**Tone**
• SIP v1 (RFC2543), v2(RFC3261) • IP/TCP/UDP/RTP/RTCP • IP/ICMP/ARP/RARP/SNTP • Fixed Client/DHCP Client/ PPPoE Client • Telnet/HTTP Server • DNS Client	• Ring Tone • Ring Back Tone • Dial Tone • Busy Tone • User Programming Tone
	Phone Function
Codec	• Volume Adjustment • Speed dial, Phone book • Flash • Speaker Phone
• G.711: 64k bit/s (PCM) • G.723.1: 6.3k / 5.3k bit/s • G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) • G.729A: 8k bit/s (CS-ACELP) • G.729B: adds VAD & CNG to G.729 • GSM: 13k bit/s	**IP Assignment** • Static IP • DHCP • PPPoE
Voice Quality	**Security**
• VAD: Voice activity detection • CNG: Comfortable noise generator • LEC: Line echo canceller • Packet Loss Compensation • Adaptive Jitter Buffer	• HTTP 1.1 basic/digest authentication for Web setup • MD5 for SIP authentication (RFC2069/ RFC 2617)
	QoS
Call Function	• ToS field
• Call Hold • Call Waiting • Call Forward • Caller ID • 3-way conference	**NAT Traversal** • STUN **Configuration**
DTMF Function	• Web Browser • Console/Telnet • Keypad
• In-Band DTMF • Out-of Band DTMF • SIP Info	**Firmware Upgrade**
SIP Server	
• Registrar Server (three SIP account) • Outbound Proxy	• TFTP • Console • HTTP • FTP

2 Keypad interface for IP Phone demo system

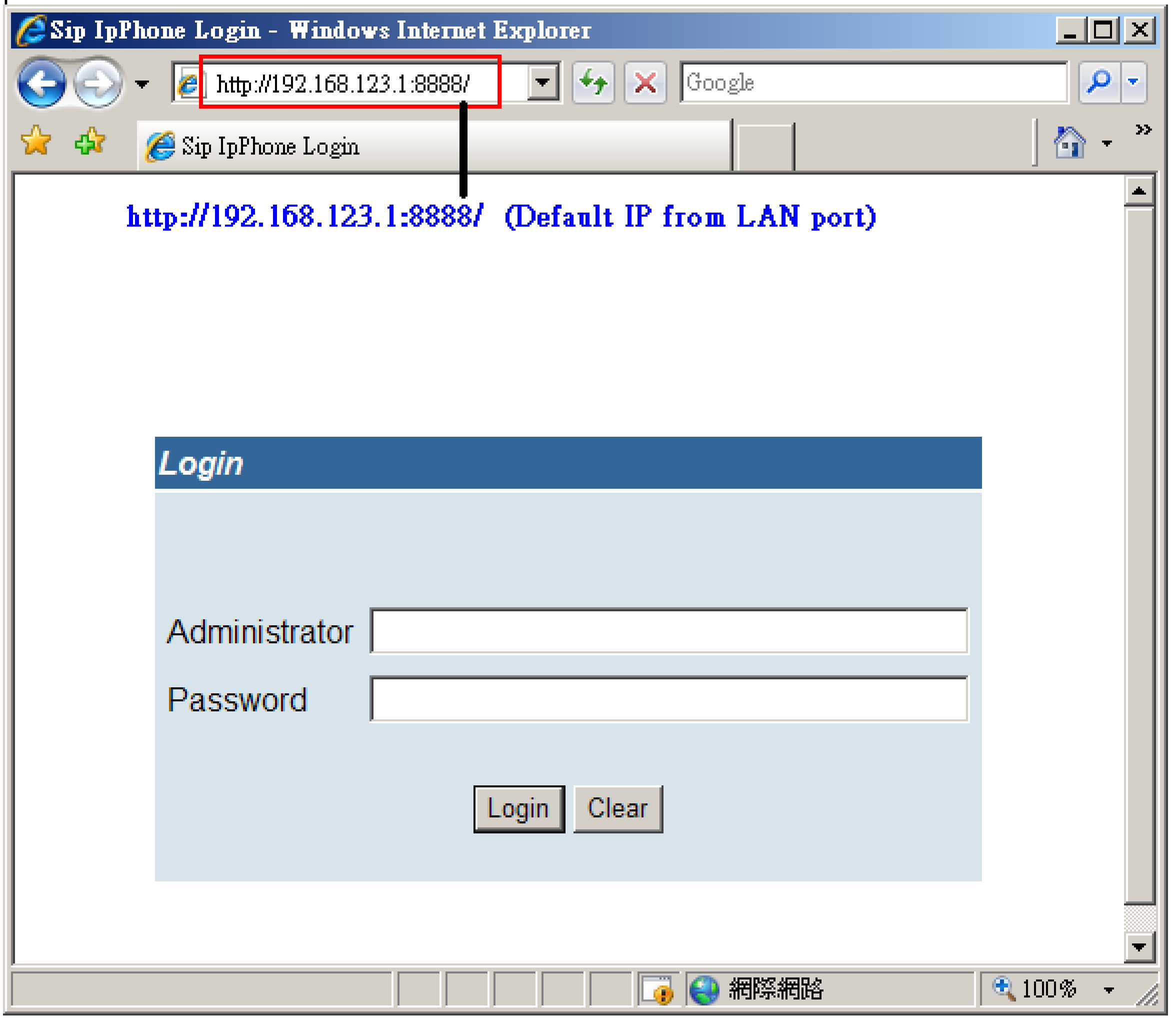
2.1 Keypad description

Key Name	Description	Note
1	"1", "-", ",", "!", "?"	
2	"2", "a", "b", "c", "A", "B", "C"	
3	"3", "d", "e", "f", "D", "E", "F"	
4	"4", "g", "h", "I", "G", "H", "I"	
5	"5", "j", "k", "l", "J", "K", "L"	
6	"6", "m", "n", "o", "M", "N", "O"	
7	"7", "p", "q", "r", "s", "P", "Q", "R", 'S"	
8	"8", "t", "u", "v", "T", "U", "V"	
9	"9", "w", "x", "y", "z", "W", "X", "Y", "Z"	
0	"0", "space"	
*	"*", "•", ":", "@"	
#	Start dialing process	
Transfer	This is "Transfer" to the other phone number	
REDIAL	This is "REDIAL" the same number again	
HOLD	This is "HOLD" function	
Mute	This is "Mute" function	
DND	This is "Reject" function	
Enter/OK	This is "OK", accept setting	
DEL	This is "Delete", Delete word or phone number	
UP/DOWN	This is Up↑ and Down↓ key	
LEFT/RIGHT	This is Left← and Right→ key	
MENU	This is the "Menu" key to set the IP Phone	
Speaker	This the Speaker Phone	
M1~M10	This is the M1 to M10, this is 10 speed dial number.	
Conf	This is three way conference function	
Volume -/+	This is volume setting	
Phone Book	This is Phone Book list	
FWD	This is "FWD" to the other phone number	

3 Setup the VoIP Phone by Web Browser

Default the IP Phone's NAT is enabled, WAN port is in DHCP Client Mode, LAN port is in DHCP Server Mode. You can connect you PC on LAN port, then you will get an IP Address from the IP Phone.

The IP Phone provides a built-in web server. You can use Web browser to configure the IP Phone. First please input the IP address **http://192.168.123.1:8888** in the Web page. Please remember to add the port number ":**8888**".



3.1 Login.

3.1.1 Please input the username and password into the blank field. The default setting is:

1. For Administrator

   Username:**admin**

   Password : **admin**

   If you use the account login, you can configure all the setting.

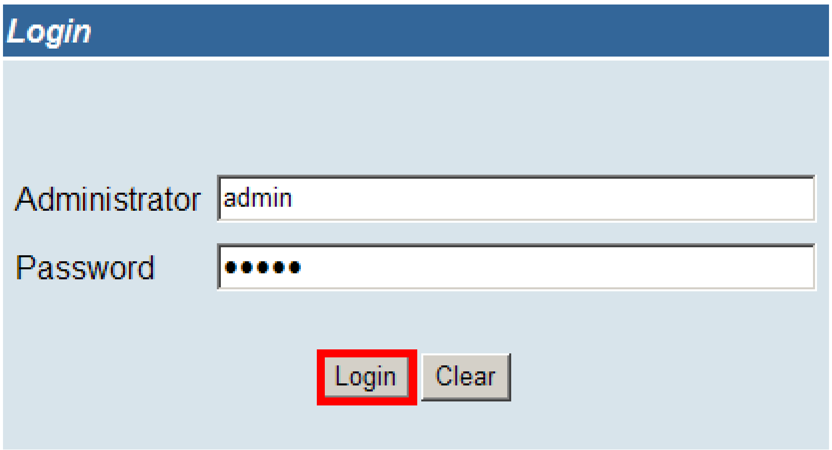
2. For normal use

   Username:**system** or **user**

   Password : **test**.

   If you use the account login, but you can not configure the SIP setting.

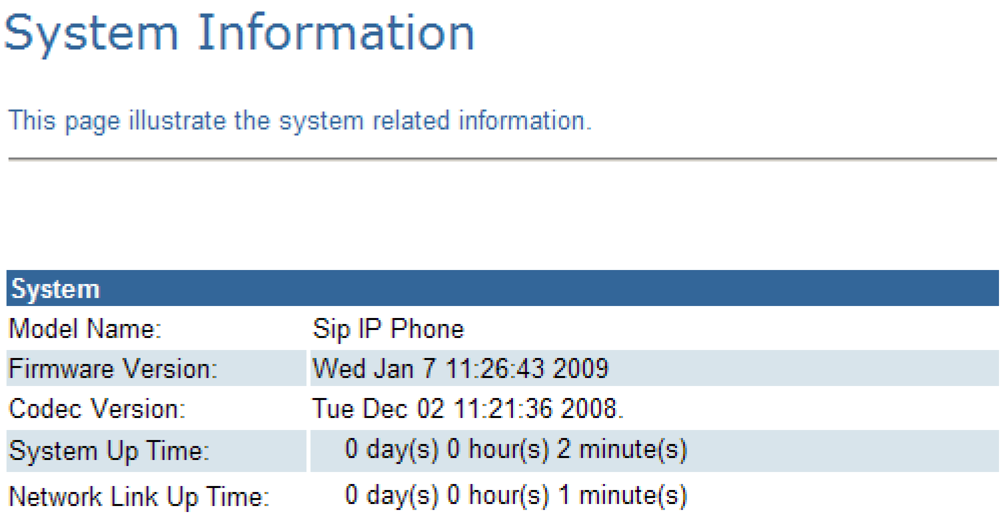
3.1.2 Click the "Login" button will move into the VOIP PHONE web based management information page.



3.2 System Info.

3.2.1 When you login the web page, you can see the VOIP PHONE current system information like firmware version, company… etc in this page.

3.2.2 Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.



3.3 Phone Book

3.3.1 In Phone Book contains Phone Book and Speed Dial Settings. You can setup the Phone Book and Speed Dial number. The Phone Book can store 140 phone numbers and the Speed Dial can store 10 phone numbers. If you want to use Speed Dial you just dial the speed dial number (from 0~9) then press "#".

Phone Book

You could add/delete items in current phone book.

Phone Book Page: page 1

Phone	Name	Number or URL	Select
0			☐
1			☐
2			☐
3			☐
4			☐
5			☐
6			☐
7			☐
8			☐
9			☐

Delete Selected | Delete All | Reset

Add New Phone

Position: (0~139)

Name:

Number or URL:

Add Phone | Reset

3.3.2 In the Phone Book function you can add/delete the phone number in the phone book list. You can input maximum 140 entries phone book list.

If you need to add a phone number into the phone book, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the "Add Phone" button.

Example :

Phone	Name	Number or URL	Select
0	2222	300@10.10.10.74	☐

If IP Phone user dial "**2222#**" the device will search phone book , If match the device will send out "300@10.10.10.74 ".

If you want to delete a phone number, you can select the phone number you want to delete then click "Delete Selected" button.

If you want to delete all phone numbers, you can click "Delete All" button.

3.3.3 The Speed Dial setting function you can add/delete Speed Dial number. You can input maximum 10 entries speed dial list.

Speed Dial Phone List

You could set the speed dial phones in this page.

Phone	Name	Number or URL	Select
0			☐
1			☐
2			☐
3			☐
4			☐
5			☐
6			☐
7			☐
8			☐
9			☐

Delete Selected | Delete All | Reset

Add New Phone

Position: (0~9)

Name:

Number or URL:

Add Phone | Reset

If you need to add a phone number into the Speed Dial list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the "Add Phone" button.

If you want to delete a phone number, you can select the phone number you want to delete then click "Delete Selected" button.

If you want to delete all phone numbers, you can click "Delete All" button.

3.4 Phone Setting

3.4.1 In Phone Setting contains Call Forward, SNTP Settings, Volume Settings, Melody Setting, Block Setting, Dial Plan Setting, Call Waiting and Soft-Key ,Hot line ,Alarm Setting functions.

3.4.2 Call Forward function: you can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

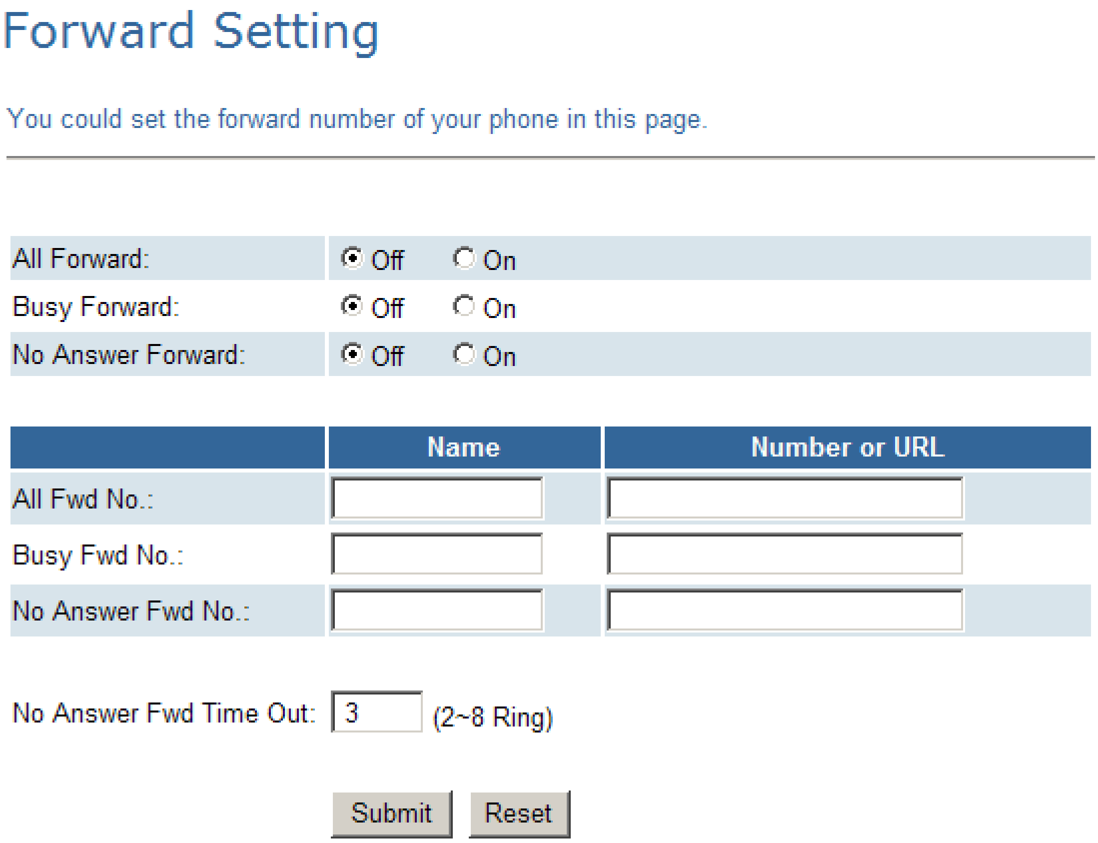
All Forward: All incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.

Busy Forward: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.

No Answer Forward: : If you can not answer the phone, the incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

When you finished the setting, please click the Submit button.

If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



SNTP Setting function: you can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button. If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will reboot automatically. And Add daylight saving set.

Daylight Saving set: on or off

DST offset: -/+,1/2

DST Start time: Jan, Feb, Mar, Apr, May, Jun, Jul, Aug, Sep, Oct, Nov, Dec。

Day of Month: 01, 02, 03, 04, 05, 06, 07, 08, 09, 10, 11, 12

Week of Month:

Last Week, Last Second Week, Week1, Week2, Week3。

Sun, Mon, Tue, Wed, Thu, Fri, Sat

Start Time: 00, 01, 02, ... ~23。

DST End Date:

Jan, Feb, Mar, Apr, May, Jun, Jul, Aug, Sep, Oct, Nov, Dec。

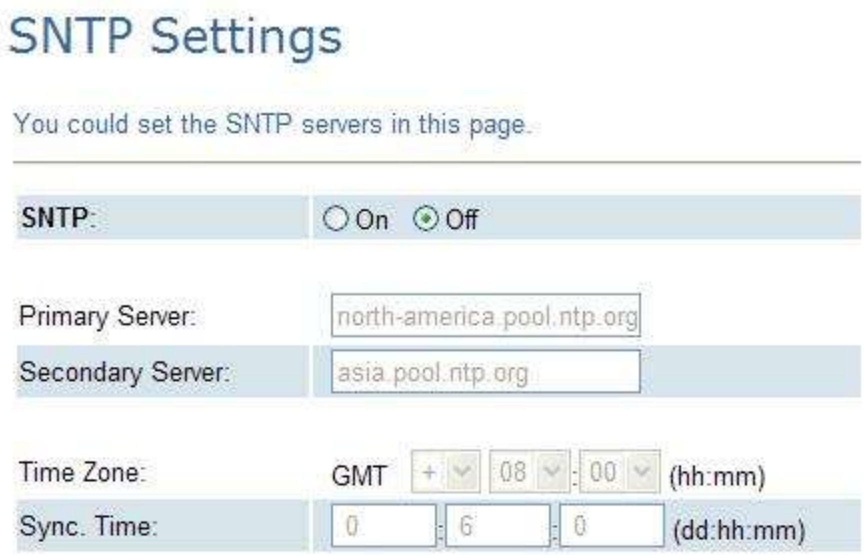
Day of Month: 01, 02, 03, 04, 05, 06, 07, 08, 09, 10, 11, 12。

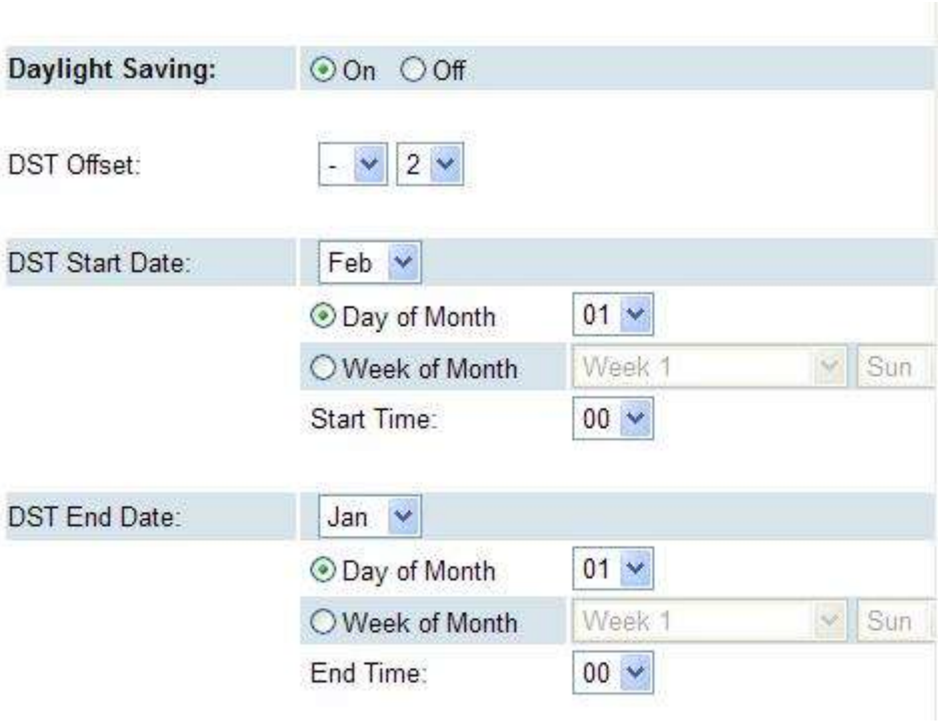
Week of Month:

Last Week, Last Second Week, Week1, Week2, Week3。

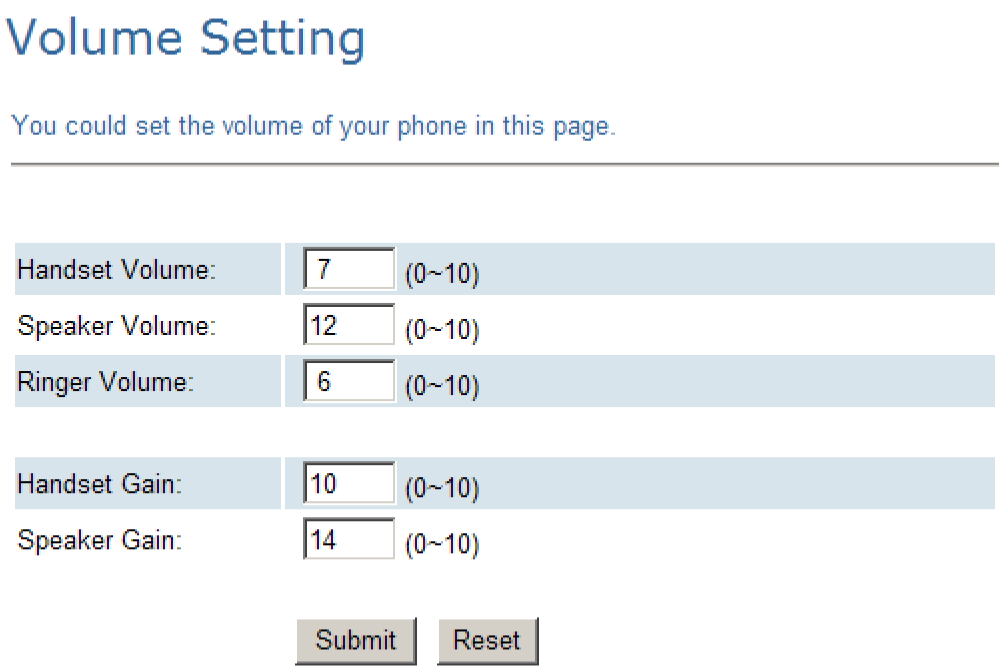
Sun, Mon, Tue, Wed, Thu, Fri, Sat

Start Time: 00, 01, 02, ... ~23。



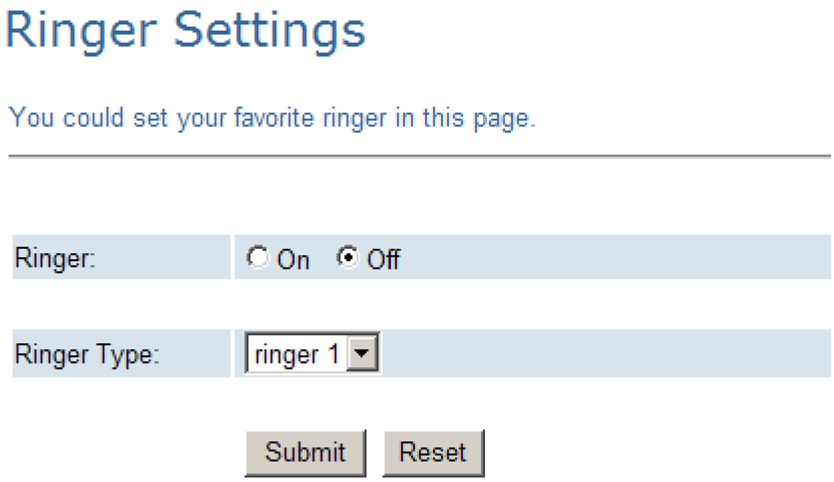


3.4.3 Volume Setting function: you can setup the Handset Volume, Ringer Volume, and the Handset Gain.



Handset Volume is to set the volume you hear from the handset.
Speaker Volume is to set the volume you hear from the speaker phone.
Ringer Volume is to set the ringer volume.
Handset Gain is to set the volume send out from from the handset.
Speaker Gain is to set the volume send out from from the micro phone.
When you finished the setting, please click the Submit button.
If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.

3.4.4 Ringer Setting: you can select the melody for the imcoming call. When you finished the setting, please click the Submit button. If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



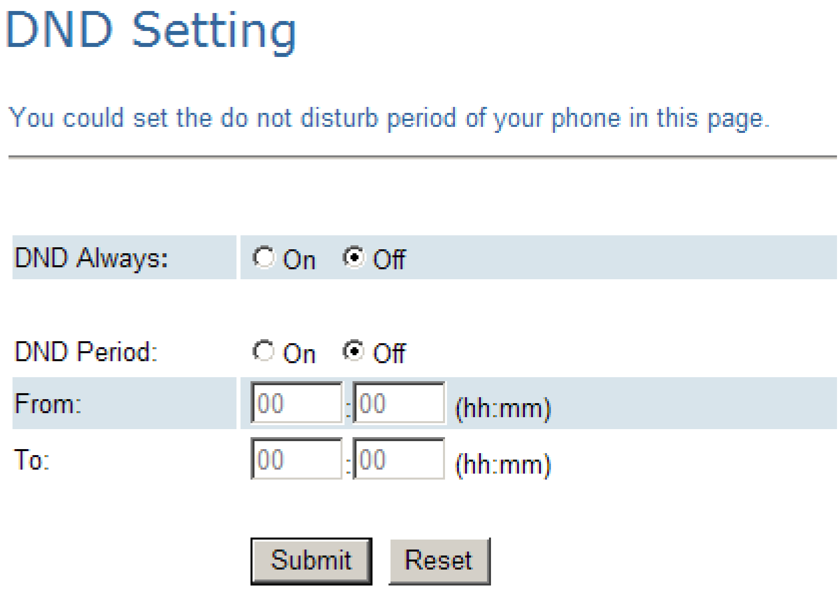
3.4.5 DND Setting: you can setup the DND Setting to keep the phone slience. You can choose Always Block or Block a period.

DND Always: All incoming call will be blocked until disable this feature.

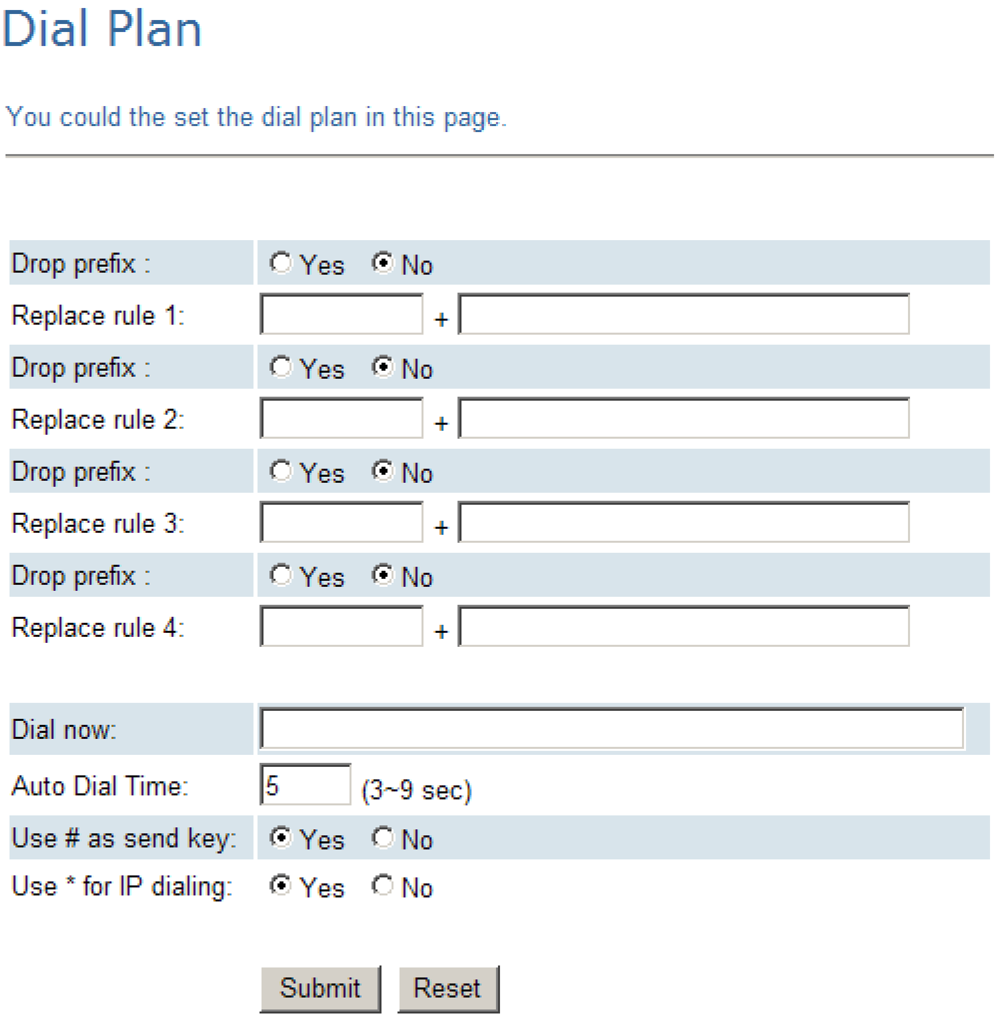
DND Period: Set a time period and the phone will be blocked during the time period. If the "From" time is large than the "To" time, the Block time will from Day 1 to Day 2.

When you finished the setting, please click the Submit button.

If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



3.4.6 Dial Plan Setting function: This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.



Symbol explan:

x or X	**0,1,2,3,4,5,6,7,8,9**
+	**or**

Replace rule:If replace prefix code is ON and prefix number is matched with rule then 005 will replace prefix.
Auto Dial Time : Stop dialing after seconds then send dial number out.
Dial Plan: When match with pattern then send dial number out but if fisrt digit is '0' then dial plan will be ignored.

Example:

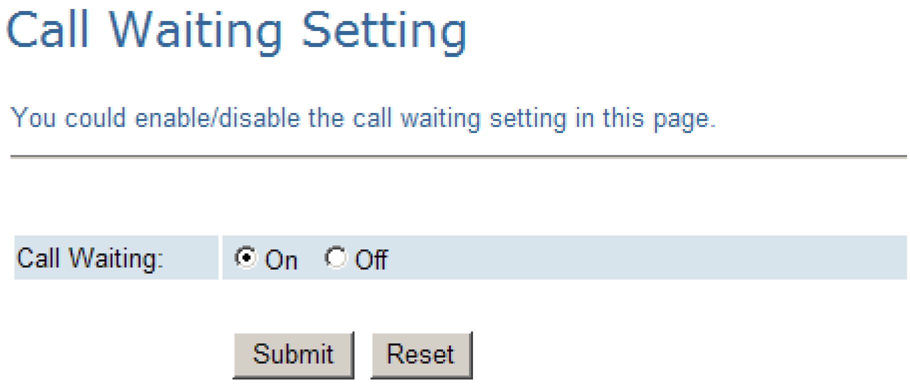
*xx	If matched with one of *00,*01....*99 then will send number out
#xx	If matched with one of #00,#01....#99 then will send number out
10x	If matched with one of 100,101....109 then will send number out
11x	If matched with one of 110,111....119 then will send number out
Xxxxxxxx	If dial with 8 digits then send number out

Auto Prefix : Number for add before dial number.
Prefix Unset Plan : When first digit or dial numeb match with pattern then ignore auto prefix.

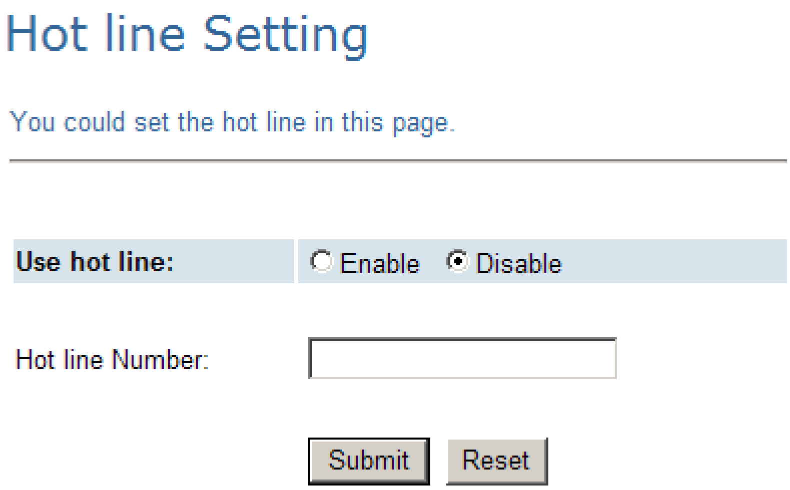
0	Ignore auto prefix if first digit is '0'
1	Ignore auto prefix if first digit is '1'
xxxxx	dial numbers are 4 digits ignore auto prefix
xxxxxx	dial numbers are 5 digits ignore auto prefix

When you finished the setting, please click the Submit button.
If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.

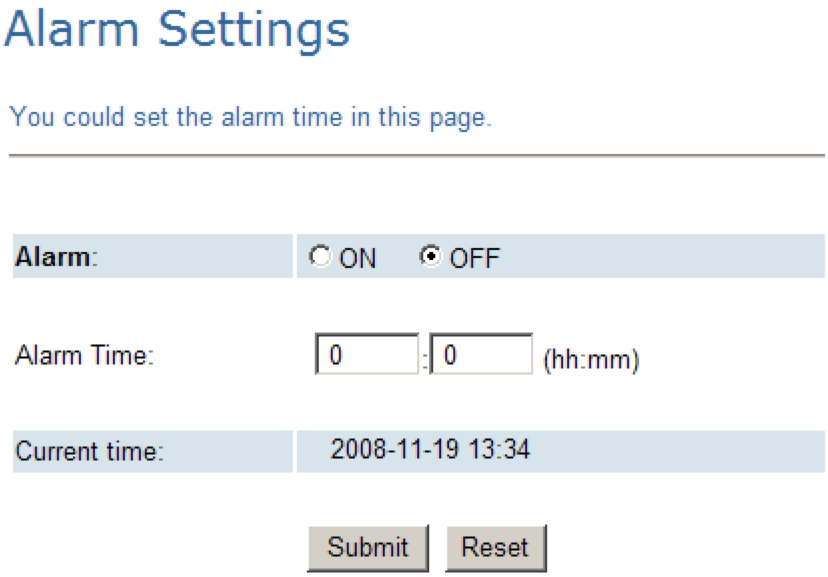
3.4.7 Call Waiting Setting function: If user doesn't want to be inform there is a new incoming call, user can set the function off. When you finished the setting, please click the Submit button. If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



3.4.8 Hotline Setting: This function is support automatic dial to your hot line number , when this setting enable device cannot aial any number.



3.4.9 Alarm setting: VOIP Phone ring time setting, when you setting time with current time are match device should produce a ring , Alarm time format is 24 hours .



3.5 Network

3.5.1 In Network you can check the Network status, configure the WAN Settings, LAN Settings, DDNS settings and VLAN Settings.

3.5.2 Network Status: You can check the current Network setting in this page.

Network Status

This page shows current status of network interfaces of the system.

System Up Time:	0 day(s) 0 hour(s) 9 minute(s)
Network Link Up Time:	0 day(s) 0 hour(s) 9 minute(s)
NAT Type:	Port restriced cone

Interface 0	
Type:	DHCP Client
IP:	10.10.10.112
Mask:	255.255.255.0
Gateway:	10.10.10.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

Interface 1	
Type:	DHCP Server
IP:	192.168.123.1
Mask:	255.255.255.0
Gateway:	192.168.123.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

3.5.3 WAN Settings: In this page you can configure the IP Phone WAN port's setting. The WAN port is for you to connect to the ADSL Router, Broadband Router. Also you can use PPPoE to get the WAN IP address from your ISP.

The IP Phone's default setting is NAT mode. If you don't need to use the NAT Mode, you can chang to Bridge Mode. If you change the setting to Bridge Mode, then the LAN setting will not effect and will be the same as WAN port.

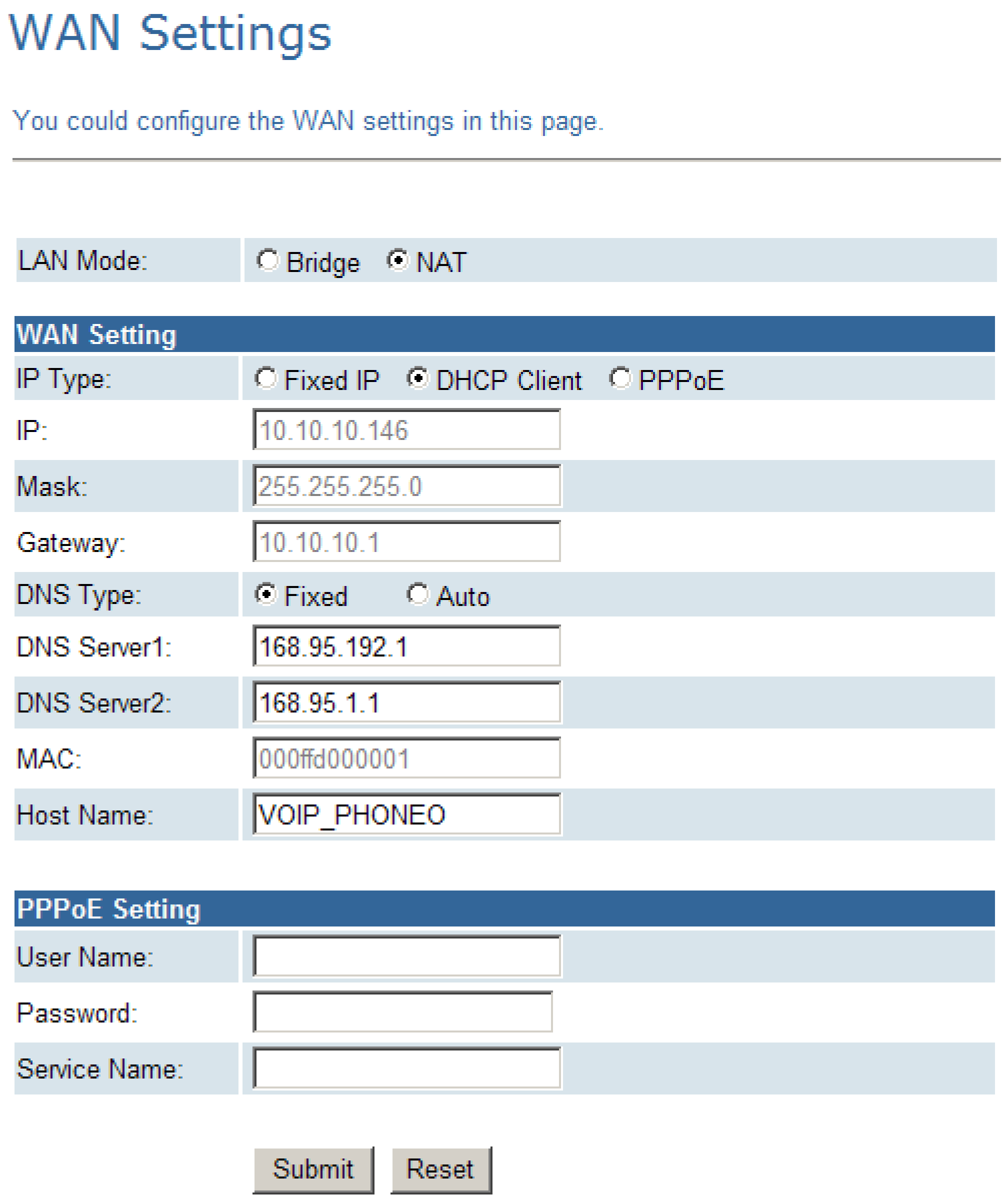
The WAN port default is DHCP Client mode, You can change the setting to Fixed IP Mode, or PPPoE Mode.

If you change the WAN port's setting to Fix IP Mode, then you have to make sure the IP address. Net Mask, Gateway, and DNS setting is suitable in your current network environment.

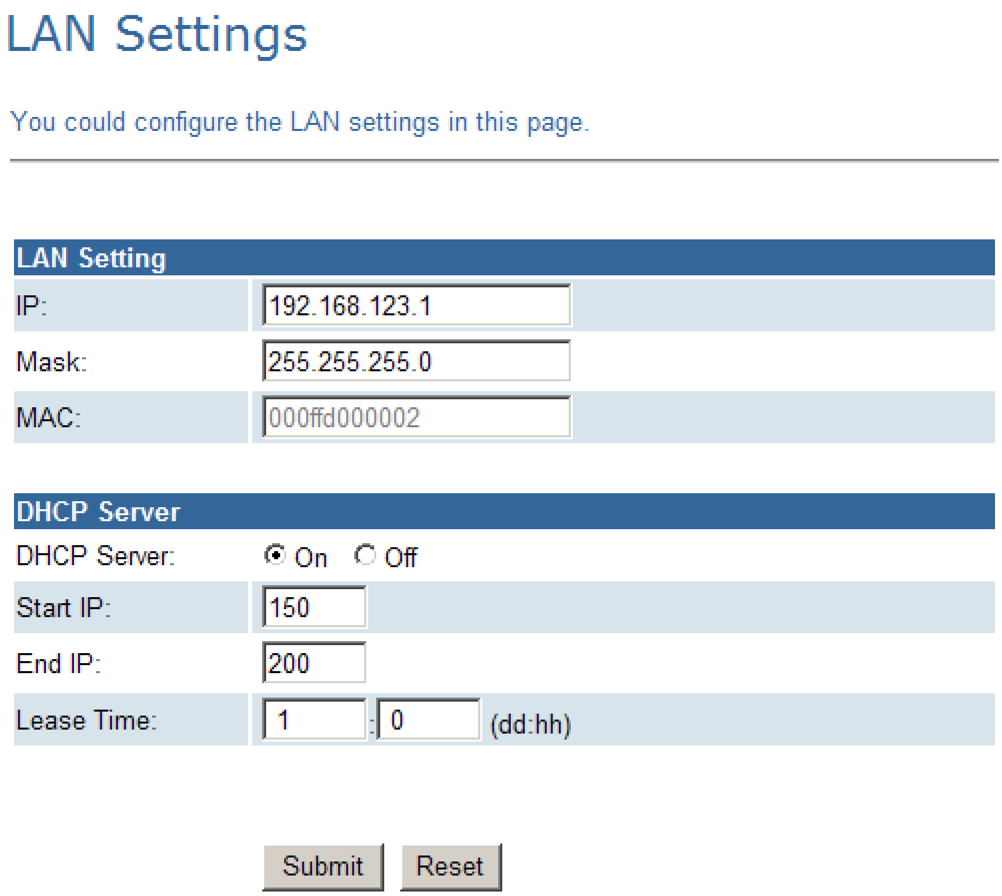
If you change the WAN port's setting to PPPoE Mode, you have to input a correct username/password to get the IP address from your Internet Service Provider.

When you finished the setting, please click the Submit button.

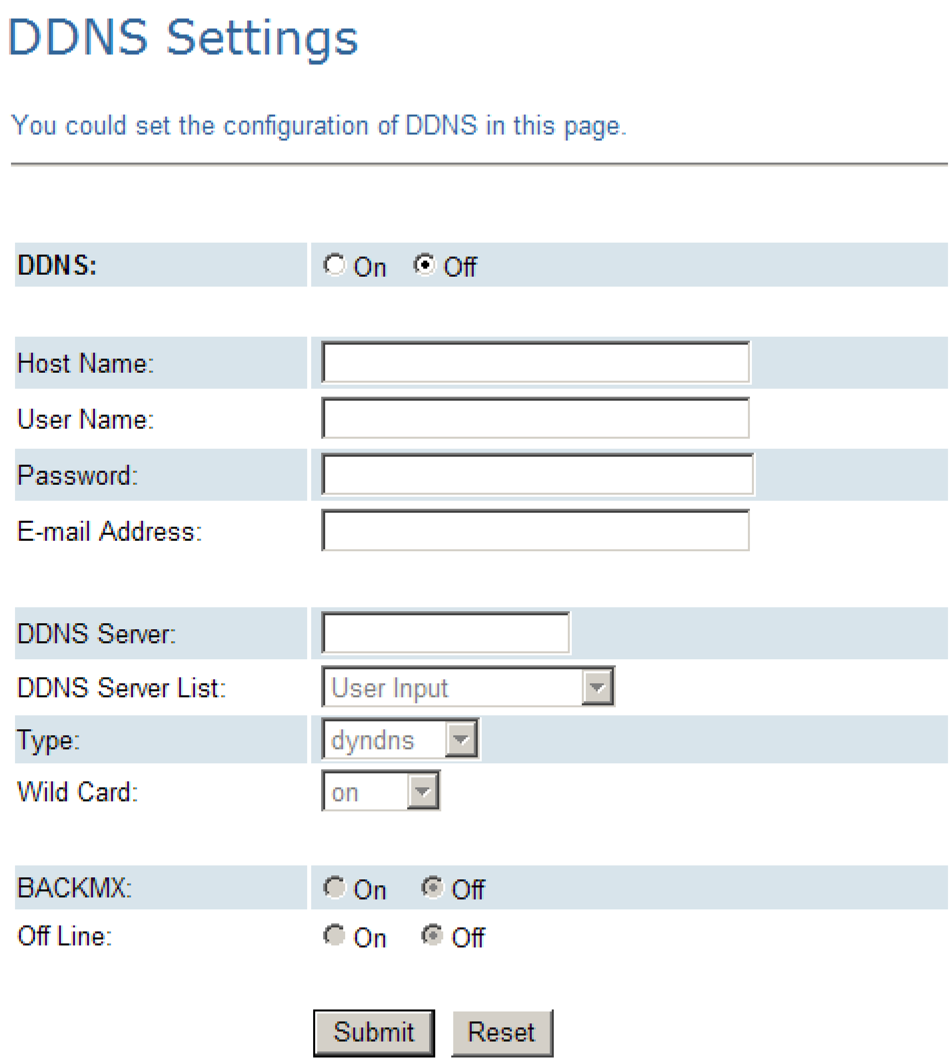
If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



LAN Settings: In this page you can configure the IP Phone LAN port's setting.
The LAN port's default IP address is 192.168.123.1, Net Mask is 255.255.255.0., and DHCP Server enabled. The start IP address if 150, end IP adress is 200. It is not necessary to change the LAN settings.
You can connect your PC to the LAN port, set your PC as DHCP Client mode, then you can get IP addreess from the TA.
When you finished the setting, please click the Submit button.
If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



3.5.4 DDNS Setting: You can configure the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button. If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



3.5.5 VLAN Setting: You can set the VLAN setting in this page. There are two parts in this page. First one is to set the packets related to the TA, and the second parts is if you use the VLAN setting in the NAT Mode.

There are two kind of destination packets will come from the TA's WAN port, one kind of packets will go to the TA, the other will go through the LAN port to the PC.

VLAN Packets: if you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets will be check with the IP Address and the VID.

VID: You can follow your service provider to set your VID.

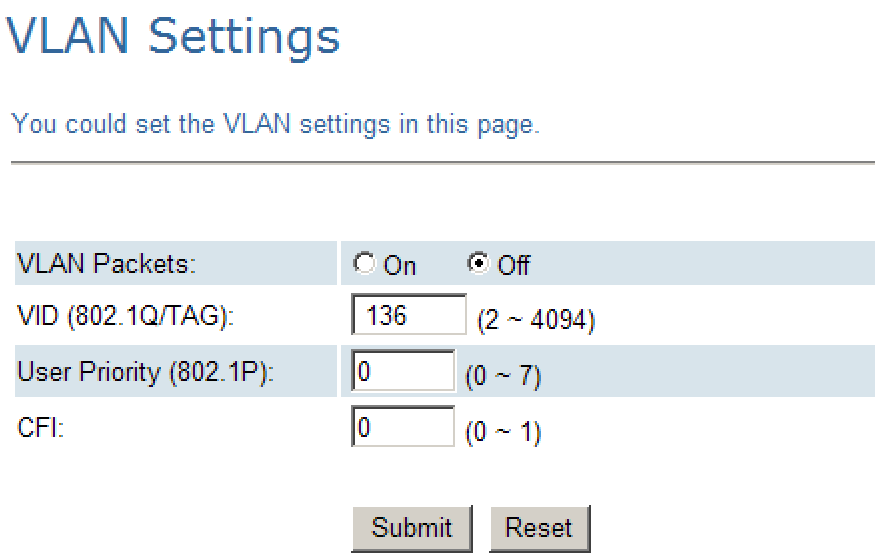
User Priority: Defines user priority, giving eight (2^3) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be defined by your service provider.

CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reason between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.

When you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets with the TA's IP address and the same VID will be accept by the TA. If the incoming packets with the TA's IP address but the different VID then the packets will be discard by the TA. The Other incoming packets with different IP address will go through the LAN port to the PC.

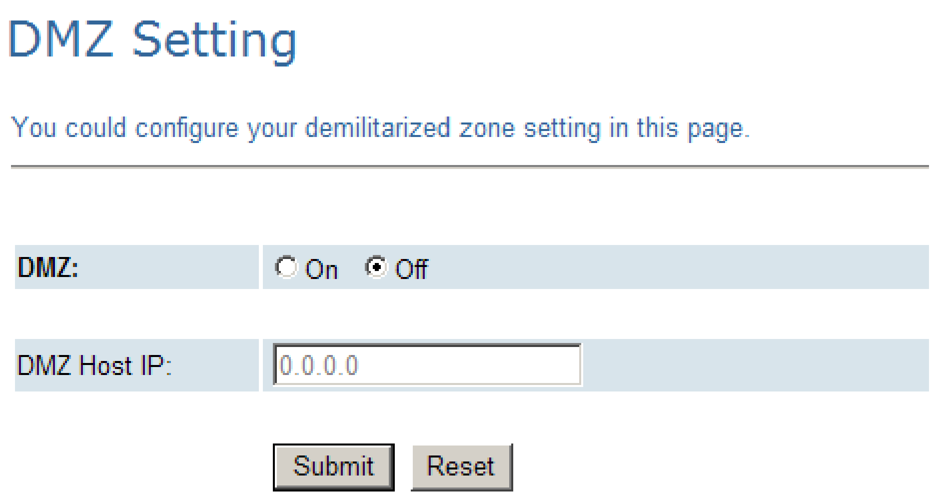
NAT VLAN Setting: When you set your device in NAT mode, the TA can help you to filter the wrong incoming packets. You can separate the other device connectd behind the TA into 4 VLAN group. You can set different VID for these 4 groups. When the incoming packets go through the TA's WAN port then the TA will check the VID, if the packets is not going to the TA(with the TA's IP address and the correct VID), and the VID is not these four VID you set, then the packets will be discard by the TA.

If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



3.5.6 DMZ Settings: In computer networks, a DMZ (Demilitarized Zone) is a computer host or small network inserted as a "neutral zone" between a company's private network and the outside public network. It prevents outside users from getting direct access to a server that has company VOIP Phone. Setting up a DMZ is very easy. If you have multiple computers, you can choose to simply place one of the computers between the Internet connection and the firewall.

If you have a computer that cannot run Internet applications properly from behind the device, then you can allow the computer to have unrestricted Internet access. Enter the IP address of that computer as a DMZ host with unrestricted Internet access. Adding a client to the DMZ may expose that computer to a variety of security risks; so only use this option as a last resort.



3.5.7 Virtual Server Setting : The device can be configured as a virtual server so that remote users accessing services such as Web or FTP services via the public (WAN) IP address can be automatically redirected to local servers in the LAN network. Depending on the requested service (TCP/UDP port number), the device redirects the external service request to the appropriate server within the LAN network. You will only need to input the LAN IP address of the computer running the service and enable it.

A Virtual Server is defined as a service port, and all requests to this port will be redirected to the computer specified by the server IP.

Virtual Server Settings

You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP (Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telent [TCP 23].

Virtual Server Page: page 1

Num	Enable	Protocol	In Port	Ex Port	Server IP	Select
0	☐					☐
1	☐					☐
2	☐					☐
3	☐					☐
4	☐					☐
5	☐					☐
6	☐					☐
7	☐					☐

Enable Selected | Delete Selected | Delete All | Reset

Enable Selected (Button): Enable the selected server.
Delete Selected (Button): Delete the selected server.
Delete All (Button): Delete all Virtual server data.
Reset (Button): Clean all data.

Add Virtual Server

Server IP:

Protocol: TCP

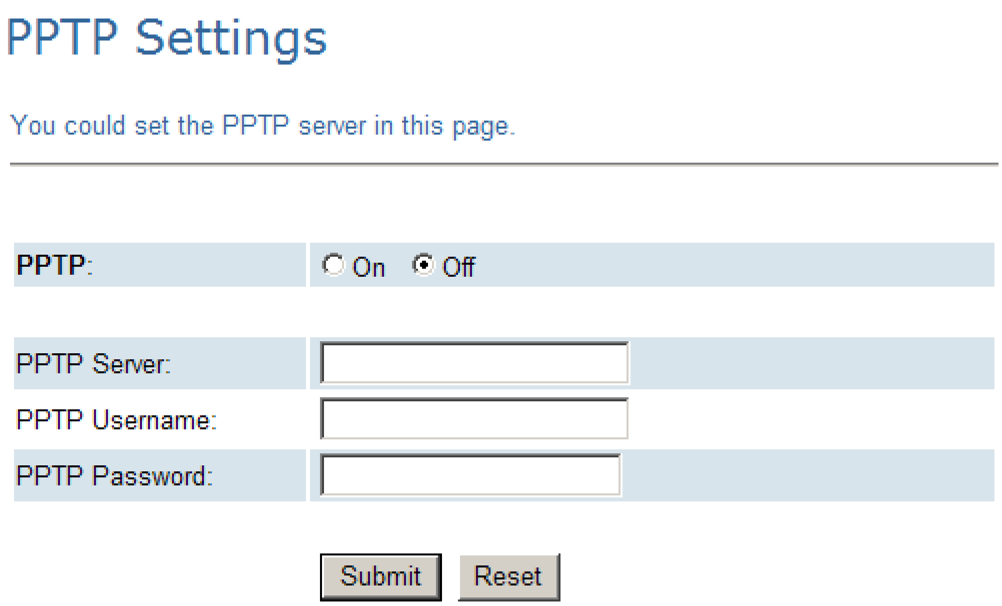
Internal Port Start: Internal Port End:

External Port Start: External Port End:

Add Server | Reset

Server IP: Displays the IP address of the server.
Protocol: Displays the TCP and UDP port information.
Internet Port Start:Display the interner port start number.
Internet Port End: Display the interner port end number.
External Port Start: Display the external port start number.
External Port End: Display the external port end number.
Add Server (Button): Add Virtual server data.
Reset (Button): Clean all data.

3.5.7 PPTP Setting:



PPTP Server: Input your PPTP server address.
PPTP Username: Input your PPTP username.
PPTP Password: Input your PPTP password.

3.6 SIP Settings

3.6.1 In SIP Settings you can setup the Service Domain, Port Settngs, Codec Settings, Codec ID Settings, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you need to setup the related informations correctly then you can register to the SIP Proxy Server correctly.

3.6.2 In Service Domain Function you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the VoIP Phone. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.

First you need click Active to enable the Service Domain, then you can input the following items:

3.6.2.1.1 Display Name: you can input the name you want to display.

3.6.2.1.2 User Name: you need to input the User Name get from your ISP.

3.6.2.1.3 Register Name: you need to input the Register Name get from your ISP.

3.6.2.1.4 Register Password: you need to input the Register Password get from your ISP.

3.6.2.1.5 Domain Server: you need to input the Domain Server get from your ISP.

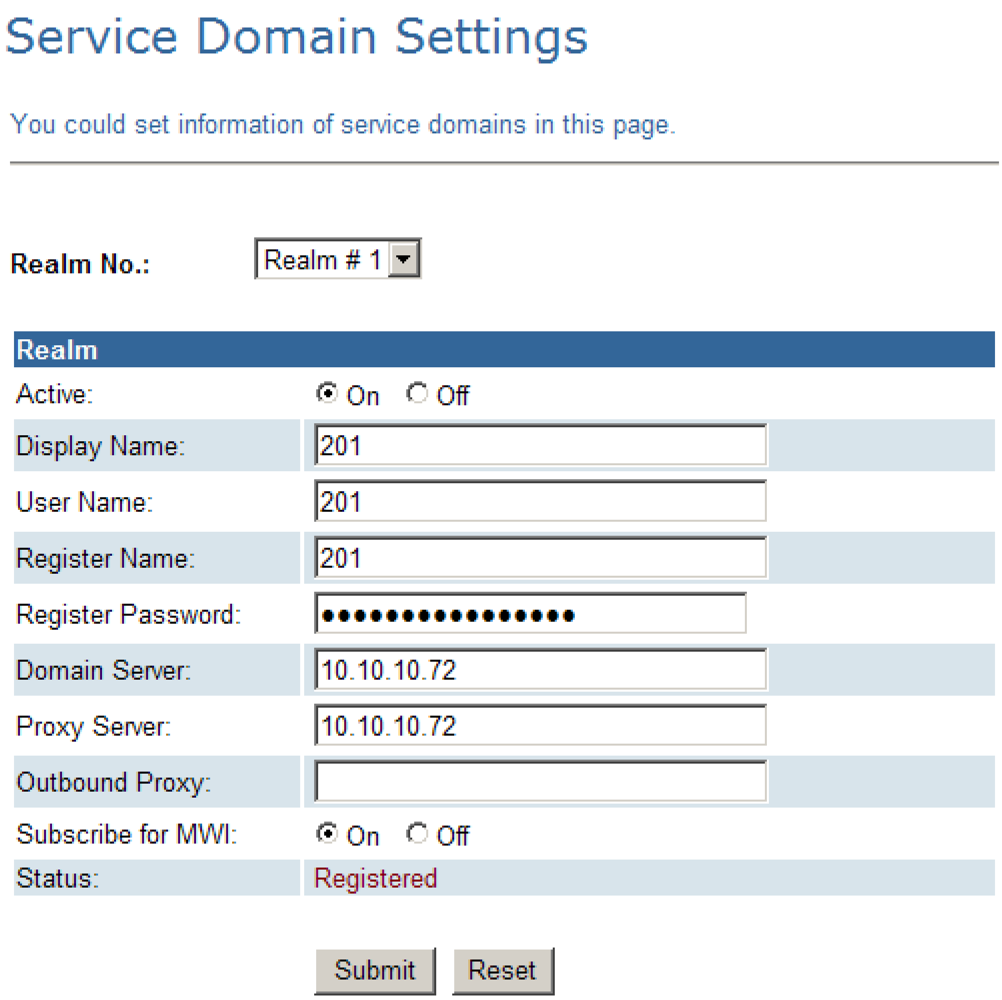
3.6.2.1.6 Proxy Server: you need to input the Proxy Server get from your ISP.

3.6.2.1.7 Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

3.6.2.1.8 You can see the Register Status in the Status item. If the item shows "Registered", then your VoIP Phone is registered to the ISP, you can make a phone call direcly.

3.6.2.1.9 If you have more than one SIP account, you can following the steps to register to the other ISP.

3.6.2.1.10 When you finished the setting, please click the Submit button.



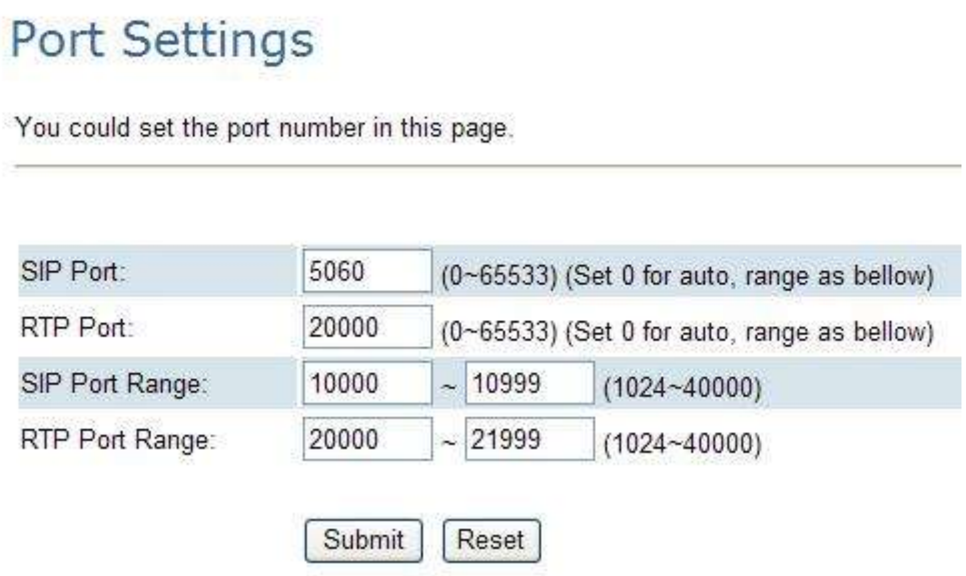
3.6.3 Port Settings: you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.

SIP Port: 5060 (0~65533, 0-> auto)。

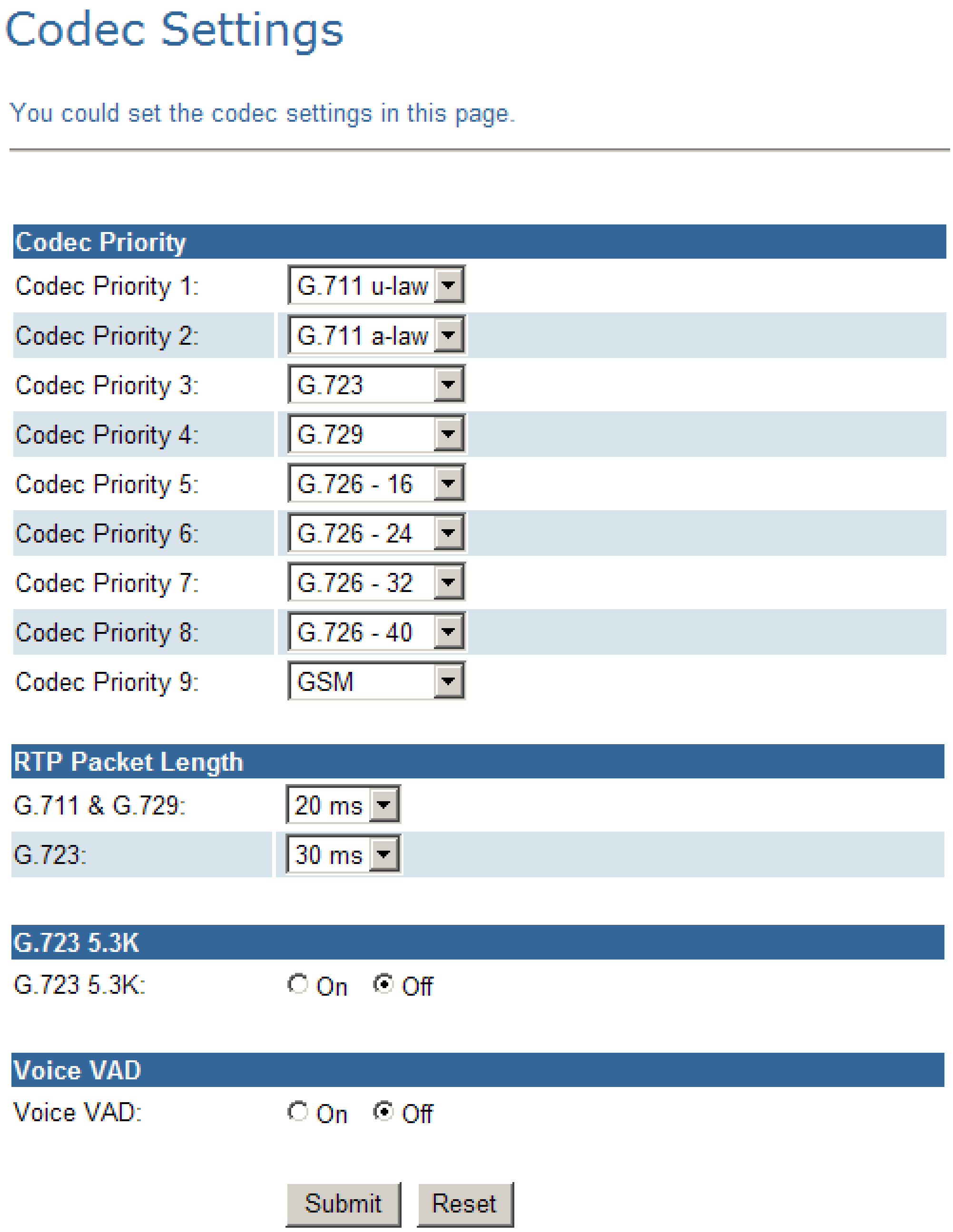
RTP Port: 20000 (0~65533, 0-> auto)。

SIP Port Range: 10000 ~ 10999 (1024 ~ 40000)。

RTP Port Range: 20000 ~ 21999 (1024 ~ 40000)。



3.6.4 Codec Settings: you can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.



3.6.5 Codec ID Settings: you can set the Codec ID to meet the other device's requirement. When you finished the setting, please click the Submit button.

Codec ID Setting

You could set the value of Codec ID in this page.

Codec Type	ID	Default Value
G726-16 ID:	23 (95~255)	☑ 23
G726-24 ID:	22 (95~255)	☑ 22
G726-32 ID:	2 (95~255)	☑ 2
G726-40 ID:	21 (95~255)	☑ 21
RFC 2833 ID:	101 (95~255)	☑ 101

Submit Reset

3.6.6 DTMF Setting: you can setup the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

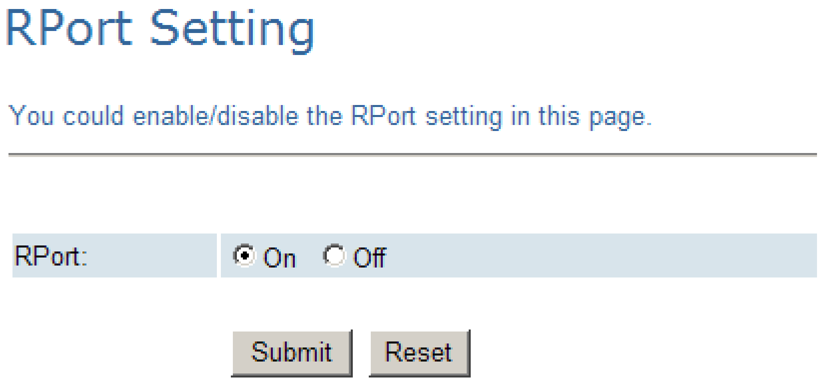
DTMF Setting

You could set the DTMF setting in this page.

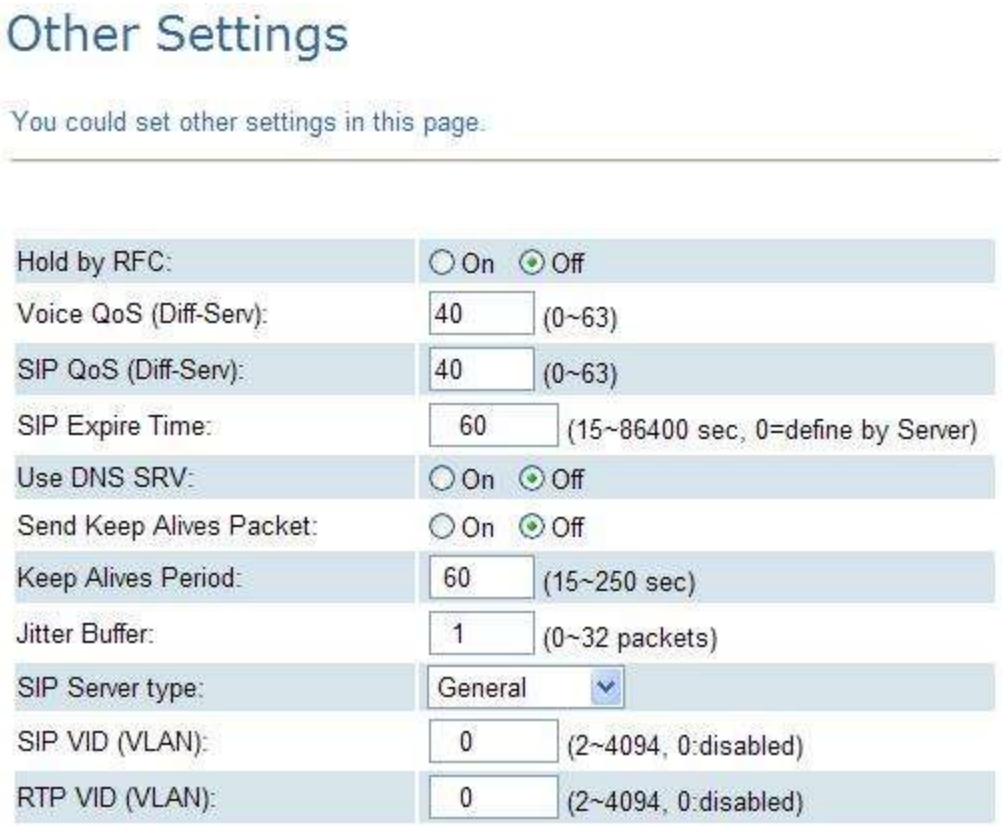
- (●) RFC 2833
- ( ) Inband DTMF
- ( ) Send DTMF SIP Info

Submit Reset

3.6.7 RPort Function: you can setup the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.



3.6.8 Other Settings: you can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.



3.7 NAT Trans.

3.7.1 In NAT Trans. you can setup STUN function. These functions can help your VoIP Phone working properly behind NAT.

3.7.2 STUN Setting: you can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your VoIP Phone working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

Force Public IP: On or Off。SIP change Fix IP。

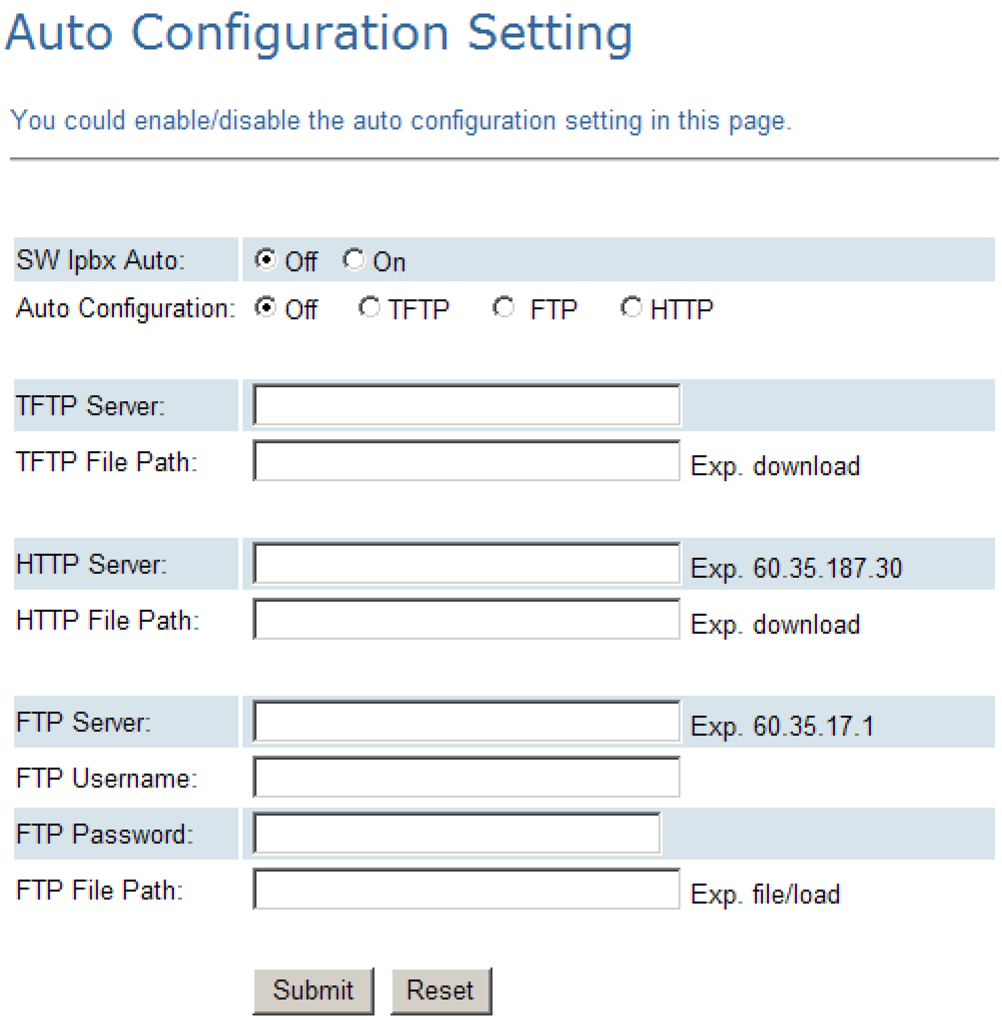
Public IP Address。Router out IP address。

Port: 5060 (80~65535)。

3.8 Others.

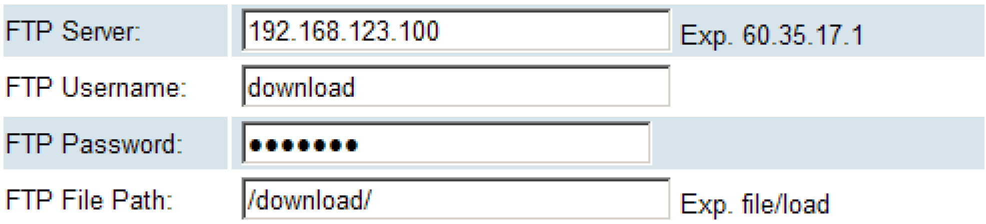
3.8.1 In Others you can setup Auto Config and ICMP Setting function. The function can configure your VoIP Phone automatically.

Auto Config: you can setup the Auto Configuration Enable/Disable and auto configuration by FTP or TFTP or HTTP. You need to select the way to do the Auto Configurationand set the Server IP address in this page. This function can automatically download the configure file to setup your VoIP Phone. When you finished the setting, please click the Submit button.



Example: Auto configuration: FTP, FTP Server: 192.168.123.100, FTP username: download, FTP password: 1234567, FTP Patch: /download/

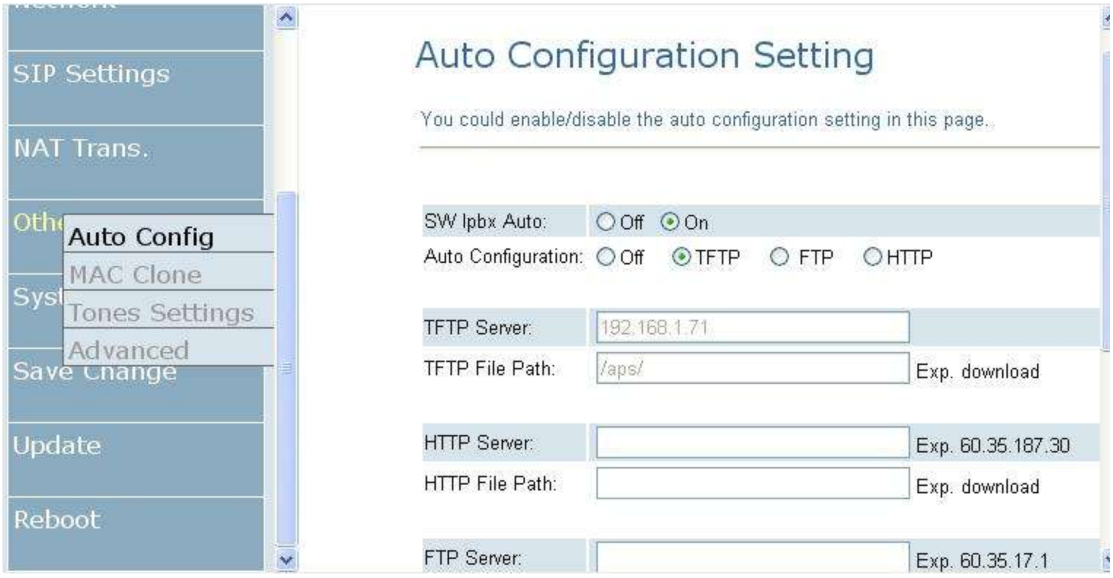
(Device will connect to the Ftp server /download folder patch and download the matching file)



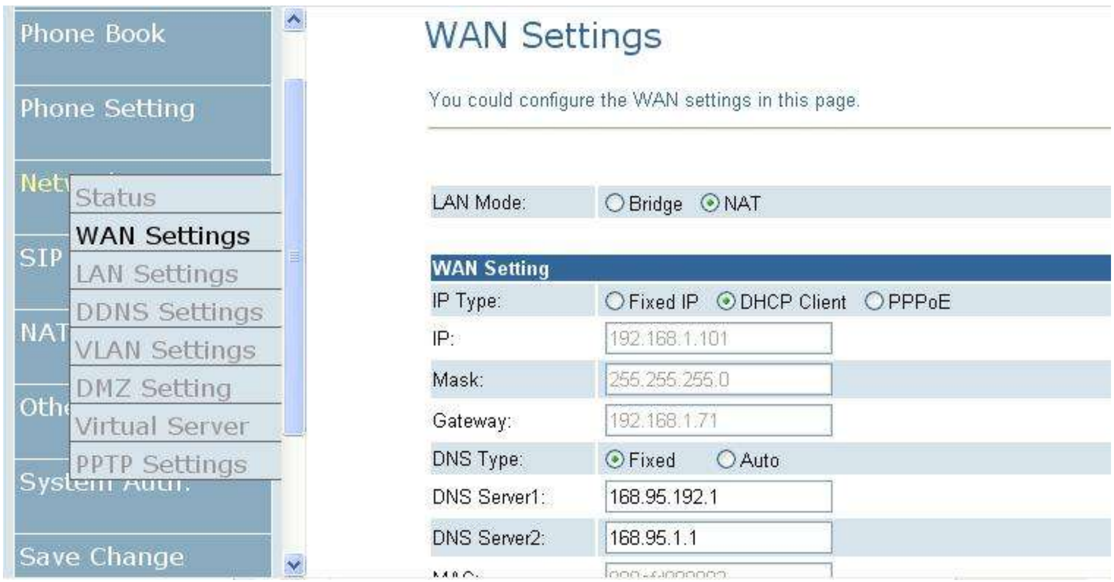
3.8.2 Auto configuration (Auto provision function) :

Step1:

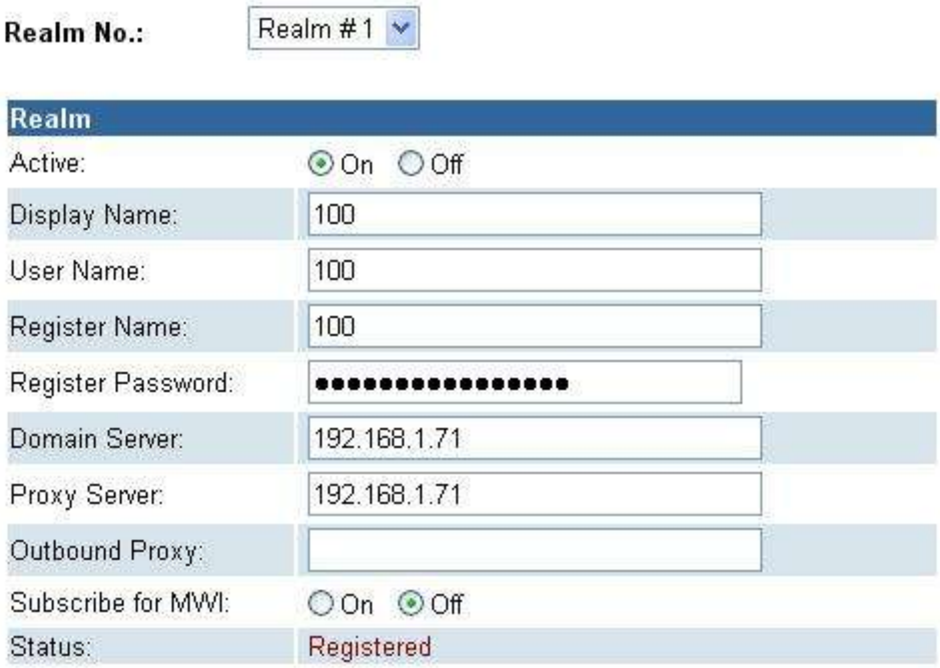
Login P100 WEB →other → Auto Configuration Setting, enable SW Ipbx Auto function & Auto Configuration TFTP bottom , next step must submit and save/reboot.



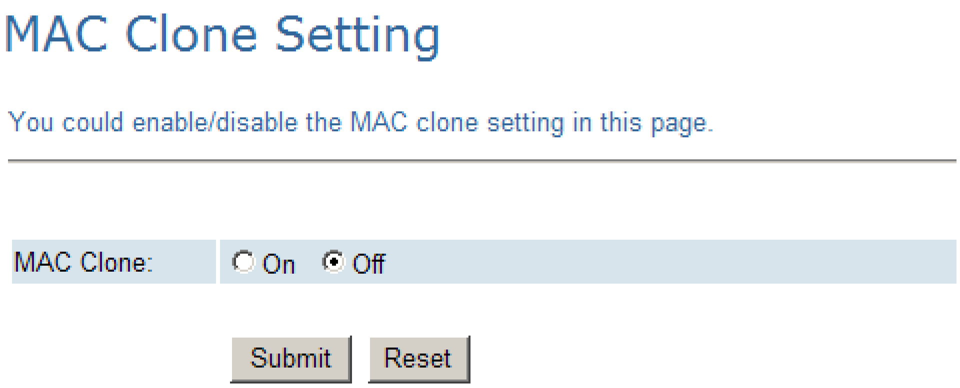
Step2:
Set and login P100 WEB, for WAN Setting IP type is DHCP Client, The iPBX404 device will auto WAN IP to P100，P100 auto get WAN IP, and then save/reboot.



Step3: The iPBX404 device will auto to P100 success a register account & iPBX404 register status.



3.8.3 MAC Clone Setting: MAC clone function changes the Mac address of workstation in a LAN address with that are intentical to WAN port,



3.8.4 Tone Settings: Configure your tones settings in this page.

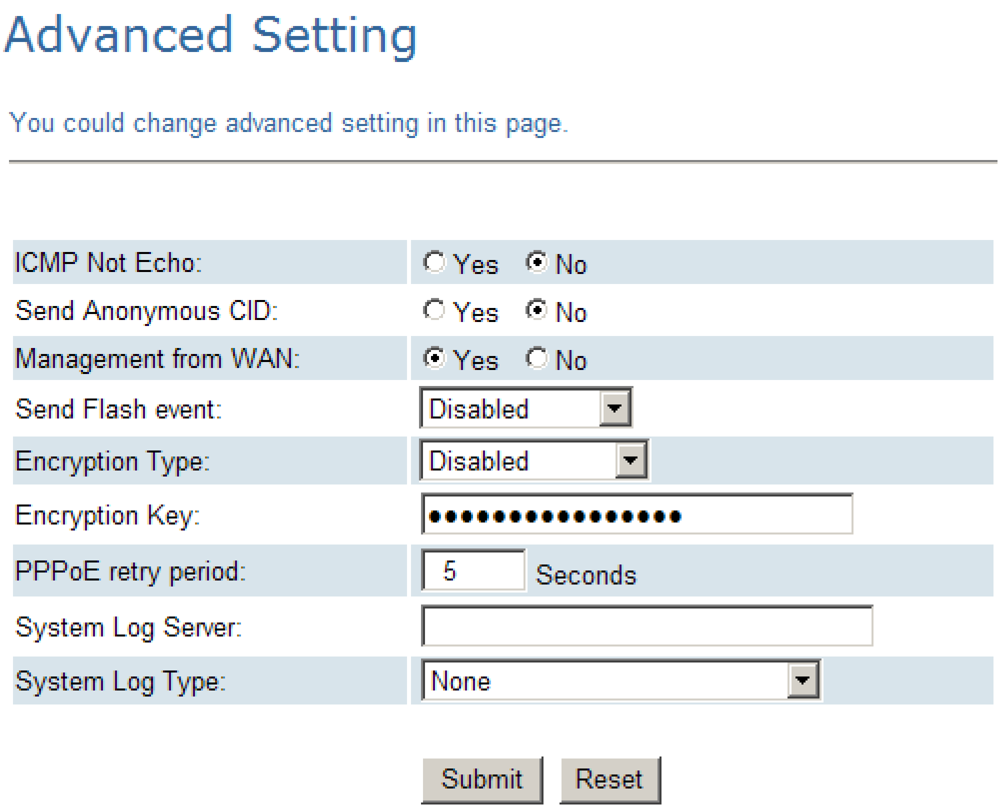
Tones Settings

You could configure your tones settings in this page.

	Dial Tone	Ring Back Tone	Busy Tone	Congestion Tone	Ring Tone	Call Waitting Tone
Cadence On:	☐	☑	☑	☑	☑	☑
Hi-Tone Freq.:	440	480	620	620	480	440
Lo-Tone Freq.:	350	440	480	480	440	350
Hi-Tone Gain:	4522	2261	2261	2261	15360	2261
Lo-Tone Gain:	2261	2261	2261	2261	15360	1130
On Time 1:	0	200	50	30	200	30
Off Time 1:	0	400	50	20	400	20
On Time 2:	0	0	0	0	0	30
Off Time 2:	0	0	0	0	0	400
On Time 3:	0	0	0	0	0	0
Off Time 3:	0	0	0	0	0	0

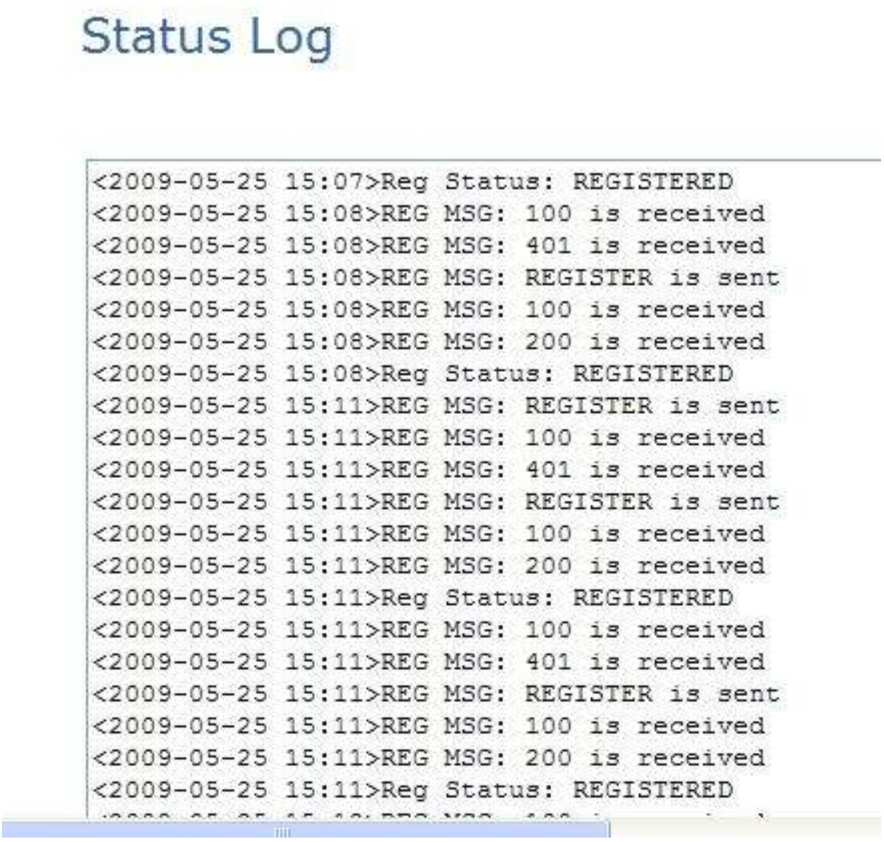
Submit Reset

3.8.5 ICMP Setting: you can setup the ICMP echo Enable/Disable in this page. This function can disable echo when someone ping this device, it can avoid haker try to attack the device. When you finished the setting, please click the Submit button.



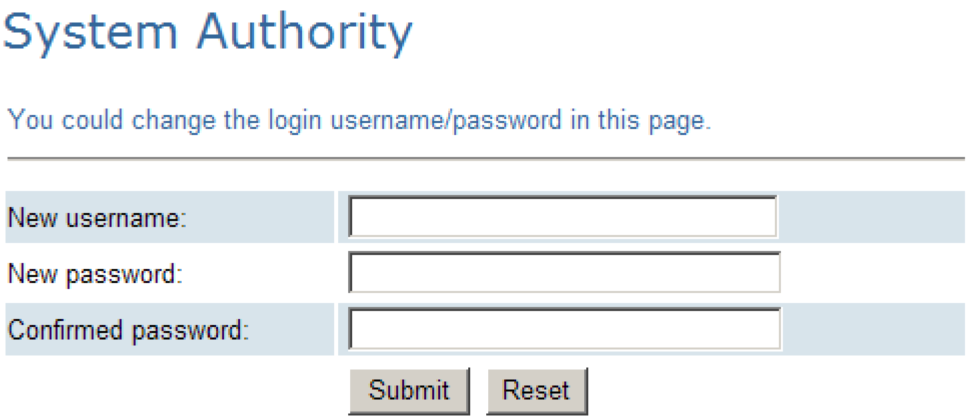
ICMP Not Echo: Not be responded for all ping messages. Default is no.
Send Anonymous CID: Send Caller id to anonymous. Default is no.
Send Flash event: Provides two methods for sending DTMFEven or SIP info messages.
Encryption Type:
Encryption Key:
PPPoE retry period: Specifies the time taken to redial when PPPoE dialing fails. Default is 5 Seconds. Time range is 5-255
System Log Server: Specifies the location of the system log server where log information will be stored.
System Log Type: Specifies the format or system log messages, for to choose from; none , call statics , debug information and both.

3.8.6 Status Log: Add Check system status & data.



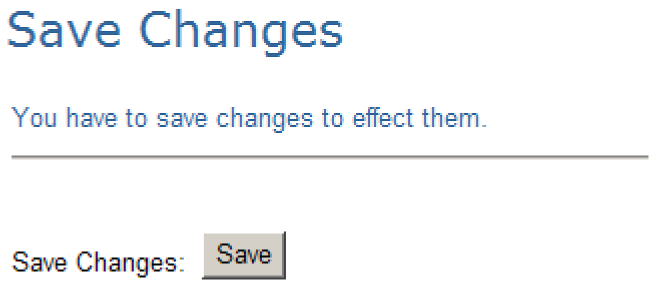
3.9 System Auth.

3.9.1 In System Authority you can change your login name and password.



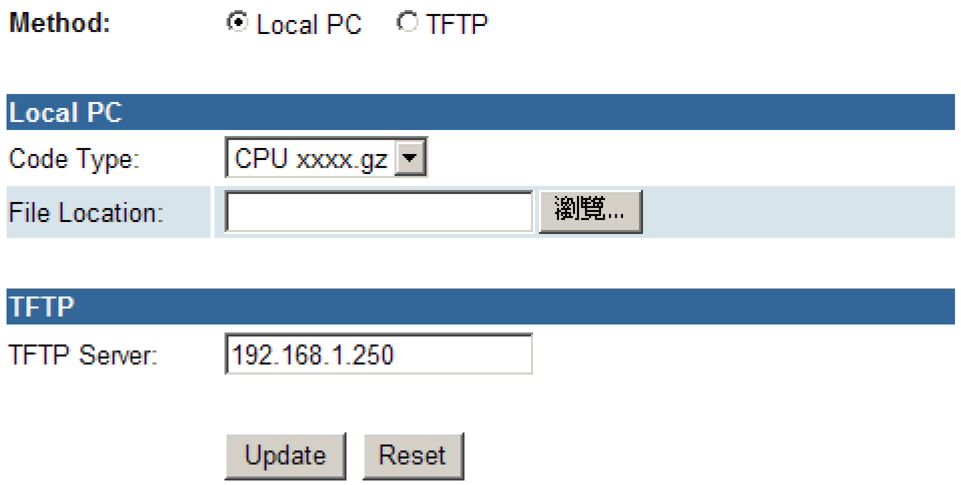
3.10 Save Change

3.10.1 In Save Change you can save the changes you have done. If you want to use new setting in the VoIP Phone, You have to click the Save button. After you click the Save button, the VoIP Phone will automatically restart and the new setting will effect.

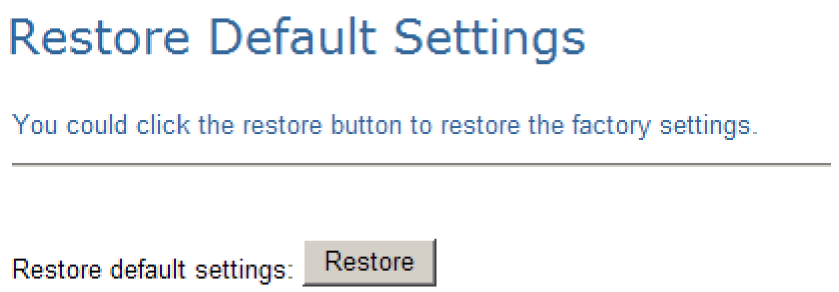


3.11 Update

3.11.1 In Update you can update the VoIP Phone's firmware to the new one or do the factory reset to let the VoIP Phone back to default setting.
3.11.2 In New Firmware function you can update new firmware via HTTP in this page. You can ugrade the firmware by the following steps:
Select the firmware code type, Risc or DSP code.
Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.
Select the correct file you want to download to the VoIP Phone then click the Update button.



3.11.3 In Default Setting you can restore the VoIP Phone to factory default in this page. You can just click the Restore button, then the VoIP Phone will restore to default and automatically restart again.



3.12 Reboot

3.12.1 Reboot function you can restart the VoIP Phone. If you want to restart the VoIP Phone, you can just click the Reboot button, then the VoIP Phone will automatically.