KA-2003

VoIP Phone User Manual

V.10

2006/05/30

TEL：+86-755-89962209 FAX：+86-755-84272617 E_mail: info@kawatel.com

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 • TFTP Client/DHCP Client/ PPPoE Client • Telnet/HTTP Server • DNS Client	• Ring Tone • Ring Back Tone • Dial Tone • Busy Tone • User Programming Tone
Codec	**Phone Function**
• 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	• Volume Adjustment • Speed dial, Phone book • Flash • Speaker Phone
Voice Quality	**IP Assignment**
• VAD: Voice activity detection • CNG: Comfortable noise generator • LEC: Line echo canceller • Packet Loss Compensation • Adaptive Jitter Buffer	• Static IP • DHCP • PPPoE
	Security
	• 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

TEL：+86-755-89962209 FAX：+86-755-84272617 E_mail: info@kawatel.com

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	開始撥號鍵
Flash/XFER	This is “Transfer” to the other phone number	轉接鍵
REDIAL	This is “REDIAL” the same number again	上通重播鍵
HOLD	This is “HOLD” function	通話保留鍵
Mute	This is “Mute” function	靜音鍵
DND	This is “Reject” function	拒接所有來電
Enter/OK	This is “OK”, accept setting	設定確認鍵
DEL	This is “Delete”, Delete word or phone number	刪除鍵
UP/DOWN	This is Up↑ and Down↓ key	上下鍵
LEFT/RIGHT	This is Left← and Right→ key	左右鍵
MENU	This is the “Menu” key to set the IP Phone	菜單
H-F/SPK	This the Speaker Phone	免提鍵
Line1~Line4	This is the Line1 to Line4, will be 4 line	4 線對外撥號
M1~M4	This is the M1 to M4, this is 4 speed dial number.	紀錄 4 組快速鍵
Conf	This is three way conference function	三方通話
CID/Call In	This is Incoming call list	來電紀錄
Call Out	This is out going call list	去電紀錄
Volume -/+	This is volume setting	音量鍵
Phone Book	This is Phone Book list	電話簿

2.2 Keypad Function and setting List

2.3.1 Phone Book

2.3.1.1 Search:Search Phone Book. 搜尋電話簿清單

2.3.1.2 Add entry:Add new phone number to phone book. 加入新的電話號碼

2.3.1.3 Speed dial:Add speed dial phone number to speed dial list. 加入新的速撥號碼

2.3.1.4 Erase all:Erase all phone number from Phone Book. 刪除整個電話簿

2.3.2 Call history

2.3.2.1 Incoming calls: Show all incoming call. 顯示所有來電

2.3.2.2 Dialed numbers: Show all dialed call. 顯示所有已撥號

2.3.2.3 Erase record: Delete call history. 刪除通話紀錄

1. All: Delete all call history. 刪除所有通話紀錄
2. Incoming: Delete all incoming call. 刪除所有來電紀錄
3. Dialed: Delete all dialed out call. 刪除所有撥號紀錄

2.3.3 Phone setting

2.3.3.1 Call forward

2.3.3.1.1 All Forward. 所有來電轉接

1. Activation: To Enabled/Disabled this function.
2. Number: Forward to a Speed Dial Number.

2.3.3.1.2 Busy Forward. 忙線時來電轉接

1. Activation: To Enabled/Disabled this function.
2. Number: Forward to a Speed Dial Number.

2.3.3.1.3 No Answer Forward. 無人接聽時來電轉接

1. Activation: To Enabled/Disabled this function.
2. Number: Forward to a Speed Dial Number.

2.3.3.1.4 Ring Timeout: Set the Ring times to start the no answer forward function, ex: 2 means after 2 rings then forward to the dedicated number.

2.3.3.2 Block Setting

1. All: 拒接所有來電
2. By Time: 一段時間拒接所有來電
3. Duration: Set the start time and end time to Block Setting.

2.3.3.3 Date/Time setting: Date and Time Setting. 日期時間設定功能

2.3.3.3.1 Date & Time: Set the IP Phone Date and Time. 修改日期時間

2.3.3.3.2 SNTP setting

2.3.3.3.2.1 SNTP : Enabled / Disable SNTP. 啟動/ 關閉 網路時間伺服器

2.3.3.3.2.2 Primary SNTP: Set Primary SNTP server IP address. 第一網路時間伺服器

2.3.3.3.2.3 Secondary SNTP: Set Secondary SNTP server IP address. 第二網路時間伺服器

2.3.3.3.2.4 Time zone: Set Time zone. 時區設定

2.3.3.3.2.5 Adjustment Time: Set adjustment time period. 自動對時設定

2.3.3.4 Volume and Gain

2.3.3.4.1 Handset volume: Set Handset volume from 0~15 (max.) for you to hear. 話筒音量調整

2.3.3.4.2 Speaker volume: Set Spearer phone volume from 0~15 (max.) for you to hear. 免持聽筒音量調整

2.3.3.4.3 Handset Gain: Set Handset Gain from 0~15 (max.) for the other site to haer. 話筒傳出音量調整

2.3.3.4.4 Speaker Gain: Set Spearer phone Gain from 0~15 (max.) for the other site to haer. 免持聽筒傳出音量調整

2.3.3.5 Ringer

2.3.3.5.1 Ringer volume: Ringer volume setting from 0~15 (max.). 鈴聲音量調整

2.3.3.5.2 Ringer type: Ringer tone selection from 1~4. 鈴聲旋律選擇

2.3.3.6 Auto Dial: Set Auto Dial time from 3~9 seconds.

2.3.3.7 Auto Answer: This function will active on IP Phone with FXO interface. Set Auto Answr for user can re-dial a call from IP call to PSTN call or from PSTN call to IP call.

2.3.3.8 Answer Counter: This function will active on IP Phone with FXO interface. Set Auto Answer will active after the numbers of ring.

2.3.4 Network

2.3.4.1 WAN Setup

2.3.4.1.1 IP Type

1. Fixed IP client: 以手動方式設定網路地址
2. DHCP client: 以 DHCP 方式取得網路地址
3. PPPoE client: 以 PPPoE 方式取得網路地

2.3.4.1.2 Fixed IP setting

2.3.4.1.2.1 IP Address: 此話機之網路地址設定

2.3.4.1.2.2 Subnet mask: 網路遮罩設定

2.3.4.1.2.3 Default Gateway: 網關IP地址設定

2.3.4.1.2.4 MAC address: MAC地址設定

2.3.4.1.3 PPPoE setting

2.3.4.1.3.1 User name: PPPoE使用者名稱設定

2.3.4.1.3.2 Password: PPPoE使用者密碼設定

2.3.4.2 LAN Setup

2.3.4.2.1 Bridge: 將乙太網界面設定成橋接模式

2.3.4.2.2 NAT: 將乙太網界面設定成NAT模式

2.3.4.3 DNS

2.3.4.3.1 Primary DNS: 第一DNS伺服器地址設定

2.3.4.3.2 Secondary DNS: 第二DNS伺服器地址設定

2.3.4.4 VLAN

2.3.4.4.1 Activation: 啟動/關閉 VLAN設定

2.3.4.4.2 VID:設定VID 2~4094

2.3.4.4.3 Priority: 設定優先級 0~7

2.3.4.4.4 CFI: 0~1

2.3.4.5 Status: Show WAN, LAN IP address and MAC address, 網路設定狀況, 顯示WAN口, LAN口之IP地址及MAC 地址

2.3.5 SIP Settings

If you want to use Kaypad to set the SIP setting, you have to go to item 7 (Administrator) System Authent to input the password, or you can not change the SIP setting.

2.3.5.1 Service domain

2.3.5.1.1 First realm

2.3.5.1.1.1 Activation: 第一SIP 伺服器啟動/停止
2.3.5.1.1.2 User name: SIP 使用者名稱設定
2.3.5.1.1.3 Display name: SIP 顯示名稱設定
2.3.5.1.1.4 Register name: SIP登錄名稱設定
2.3.5.1.1.5 Register password: SIP登錄密碼設定
2.3.5.1.1.6 Proxy server: SIP Proxy伺服器地址設定
2.3.5.1.1.7 Domain server: Domain伺服器地址設定
2.3.5.1.1.8 Outbound proxy: Outbound Proxy 伺服器地址設定

2.3.5.1.2 Second realm

2.3.5.1.2.1 Activation: 第二SIP 伺服器啟動/停止
2.3.5.1.2.2 User name: SIP 使用者名稱設定
2.3.5.1.2.3 Display name: SIP 顯示名稱設定
2.3.5.1.2.4 Register name: SIP登錄名稱設定
2.3.5.1.2.5 Register password: SIP登錄密碼設定
2.3.5.1.2.6 Proxy server Proxy: 伺服器地址設定
2.3.5.1.2.7 Domain server: Domain伺服器地址設定
2.3.5.1.2.8 Outbound proxy: Outbound Proxy 伺服器地址設定

2.3.5.1.3 Third realm

2.3.5.1.3.1 Activation: 第三SIP 伺服器啟動/停
2.3.5.1.3.2 User name: SIP 使用者名稱設定
2.3.5.1.3.3 Display name SIP: 顯示名稱設定
2.3.5.1.3.4 Register name: SIP登錄名稱設定
2.3.5.1.3.5 Register password: SIP登錄密碼設定
2.3.5.1.3.6 Proxy server Proxy: 伺服器地址設定
2.3.5.1.3.7 Domain server: Domain伺服器地址設定
2.3.5.1.3.8 Outbound proxy: Outbound Proxy 伺服器地址設定

2.3.5.2 Codec

2.3.5.2.1 Codec type

1. G.711 uLaw: 選擇優先用 G.711 uLaw 語音壓縮方式
2. G.711 aLaw: 選擇優先用 G.711 aLaw 語音壓縮方式
3. G.723: 選擇優先用 G.723.1 語音壓縮方式
4. G.729: 選擇優先用 G.729A 語音壓縮方式
5. G.726-16: 選擇優先用 G.726 16Kbps 語音壓縮方式
6. G.726-24: 選擇優先用 G.726 24Kbps 語音壓縮方式
7. G.726-32: 選擇優先用 G.726 32Kbps 語音壓縮方式
8. G.726-40: 選擇優先用 G.726 40Kbps 語音壓縮方式

2.3.5.2.2 VAD: Voice Active Detection Enable/Disable. 啟動/停止設定

2.3.5.3 RTP setting

2.3.5.3.1 Outband DTMF: Outband DTMF 啟動/停止設定

2.3.5.3.2 Duplicate RTP

1. No duplicate: 語音封包重送 0 次
2. One duplicate: 語音封包重送 1 次
3. Two duplicate: 語音封包重送 2 次

2.3.5.4 RPort Setting: RPort Enabled/Disabled 啟動/停止 RPORT 設定

2.3.5.5 Hold by RFC: 通話保留啟動/停止設定 (依照RFC3261標準)

2.3.5.6 Status: Show the SIP Proxy register status. You can use UP/Down key to check each Realm's status. 顯示對 SIP Proxy 的註冊狀態

1. First Realm: 第一 SIP 伺服器註冊狀態
2. Second Realm: 第二 SIP 伺服器註冊狀態
3. Third Realm: 第三 SIP 伺服器註冊狀態

2.3.6 NAT Transversal

2.3.6.1 STUN setting

2.3.6.1.1 STUN: STUN啟動/停止設定

2.3.6.1.2 STUN server: STUN伺服器地址設定

2.3.7 Administrator

2.3.7.1 Auto Config

2.3.7.1.1 Config Mode: You can select Disable/TFTP/FTP to do the auto config function. This function must work with the Auto Config Server.

2.3.7.1.2 TFTP server: Setting the TFTP server IP address.

2.3.7.1.3 FTP server: Setting the FTP server IP address.

2.3.7.1.4 FTP Login Name: Setting the login name to the FTP server.

2.3.7.1.5 FTP Password: Setting the Password to the FTP server.

2.3.7.2 Default setting: You can restore to the default setting. 還原成出廠設定值.

2.3.7.3 System Authentication: To do the SIP setting from Keypad, need to input the password first. Default is "test".

2.3.7.4 Version: This will show the system's firmware version.

2.3.7.5 Watch Dog: You can use this to enable Watch Dog function to do the debugging.

2.3.7.6 Restart: You can use this function to restart your IP Phone. 重新開機

TEL：+86-755-89962209 FAX：+86-755-84272617 E_mail: info@kawatel.com

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://192.168.123.1:9999 in the Web page. Please remember to add the port number ":9999".

1.1 Login.

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

1. For Administrator, the username is: root; and the password is: test. If you use the account login, you can configure all the setting.
2. For normal user, the username is: system or user; and the password is: test. If you use the account login, but you can not configure the SIP setting.

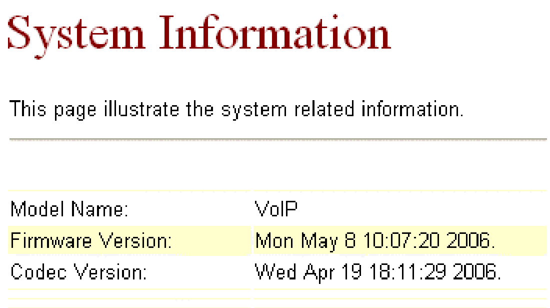
3.1.2 Click the "Login" button will move into the VOIP PHONE web based management information page.

3.1.3 If you change the setting in the Web Management interface, please do remember to click the "Submit" button in that page. After you finished the change of the setting, click the "Save" function in the left side, and click the Save Button. When you finished the setting, please click the Reboot function in the left side, and click the Reboot button in that page. After the system restart, all the setting can work properly.

3.2 System Information for the VOIP PHONE.

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

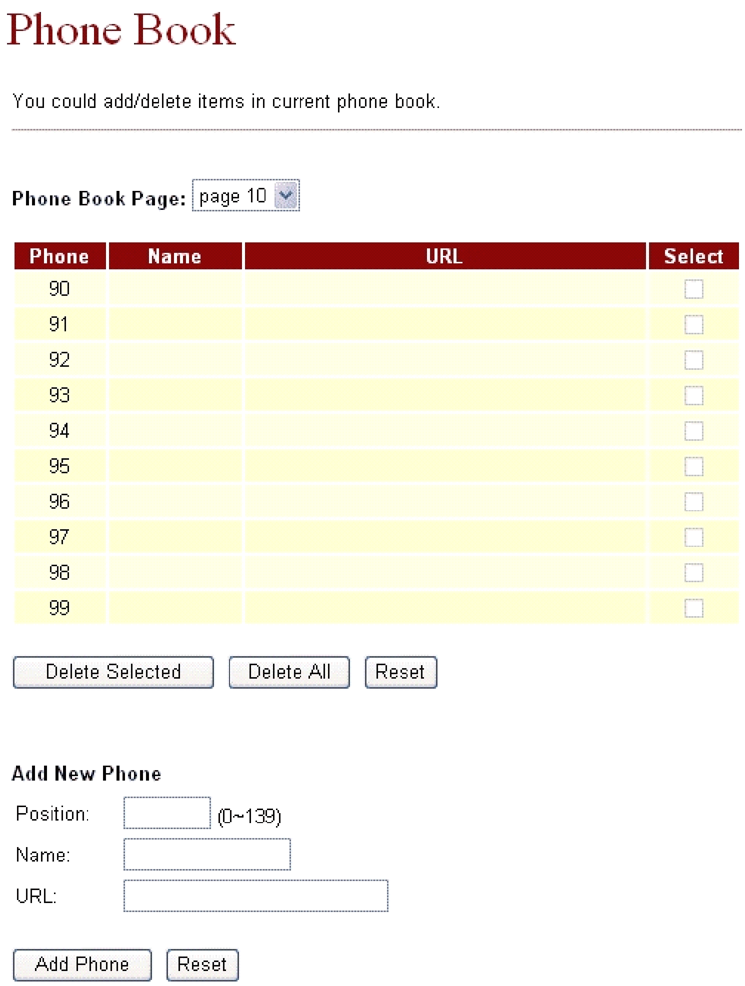
3.3.1 In Phone Book contains Phone Book and Speed Dial Settings. You can setup the Phone Book and Speed Dial number. The Phone Book can store 140 phone numbers and the Speed Dial can store 10 phone numbers. If you want to use Speed Dial you just dial the speed dial number (from 0~9) then press "#".

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.

3.3.2.1 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.

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

3.3.2.3 If you want to delete all phone numbers, you can click “Delete All” button.



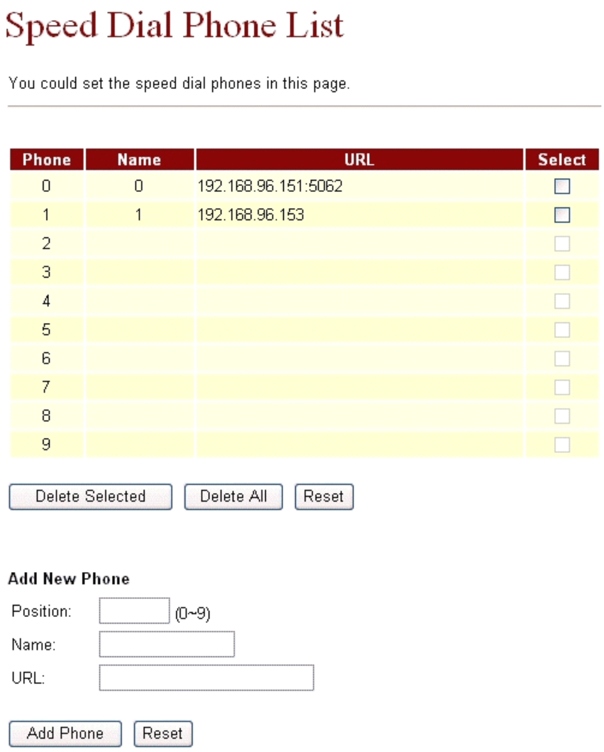
TEL：+86-755-89962209 FAX：+86-755-84272617 E_mail: info@kawatel.com

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

3.3.3.1 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.

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

3.3.3.3 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 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.

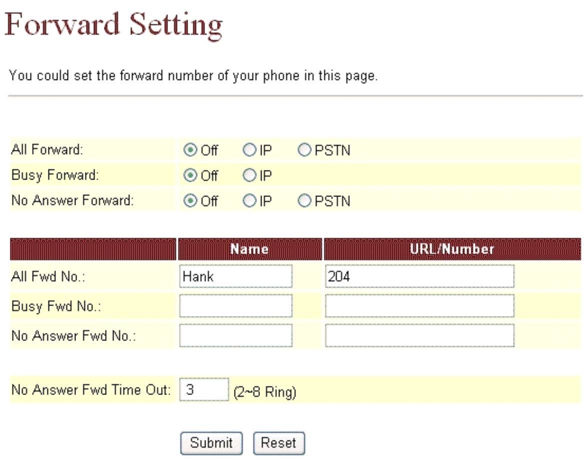
3.4.2.1 All Forward: All incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.

3.4.2.2 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.

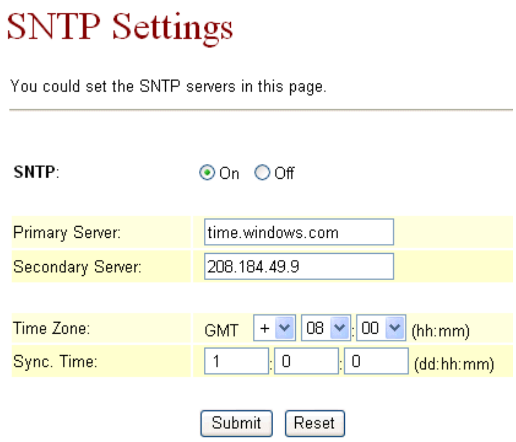
3.4.2.3 No Answer Forward: : If you can not answer the phone, the incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

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

3.4.2.5 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.



3.4.2.6 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.



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

3.4.3.1 Handset Volume is to set the volume you hear from the handset.

3.4.3.2 Speaker Volume is to set the volume you hear from the speaker phone.

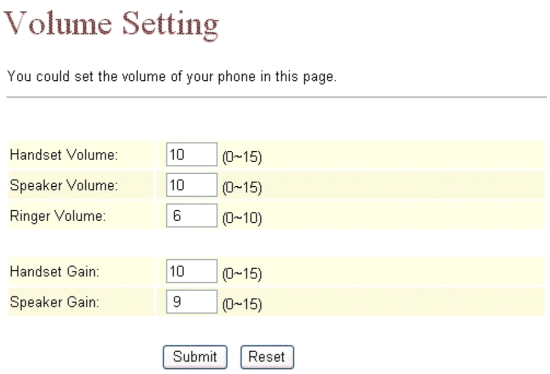
3.4.3.3 Ringer Volume is to set the ringer volume.

3.4.3.4 Handset Gain is to set the volume send out from from the handset.

3.4.3.5 Speaker Gain is to set the volume send out from from the micro phone.

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

3.4.3.7 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 Melody Setting function: 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.

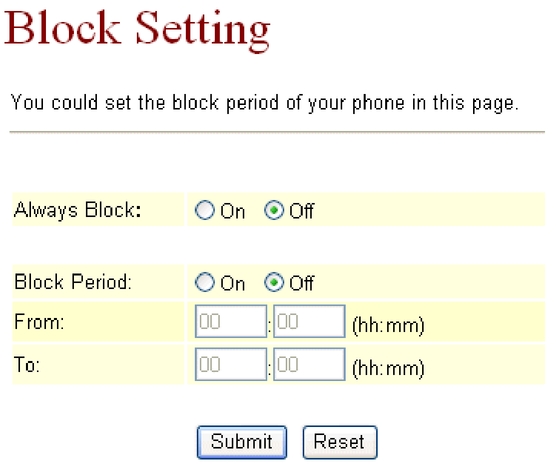
3.4.5 lock Setting function: you can setup the Block Setting to keep the phone slience. You can choose Always Block or Block a period.

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

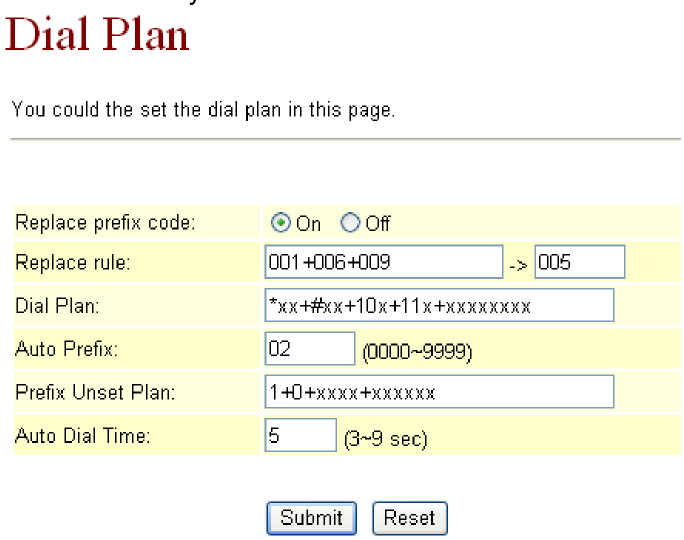
3.4.5.2 Block Period: Set a time period and the phone will be blocked during the time period. If the “From” time is large than the “To” time, the Block time will from Day 1 to Day 2.

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

3.4.5.4 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.



3.4.6.1 Symbol explan:

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

3.4.6.2 Replace rule:If replace prefix code is ON and prefix number is matched with rule then 005 will replace prefix.

3.4.6.3 Auto Dial Time : Stop dialing after seconds then send dial number out.

3.4.6.4 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

TEL：+86-755-89962209 FAX：+86-755-84272617 E_mail: info@kawatel.com

3.4.6.5 Auto Prefix : Number for add before dial number.

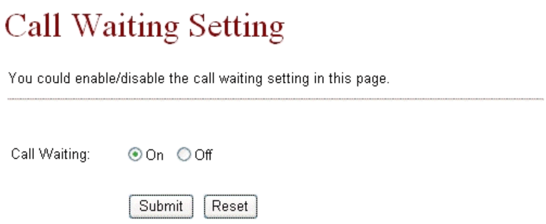
3.4.6.6 Prefix Unset Plan : When first digit or dial numeb match with pattern then ignore auto prefix.

0	Iignore 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

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

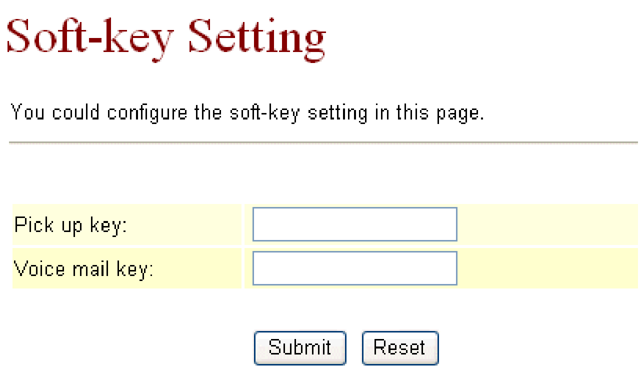
3.4.6.8 If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.

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 Soft-key Setting function: User can define the Pick Up and Voice mail function as a foft-key. This function need to work with Proxy Server and the device also need to have the Pick Up key and voice mail key. For example, if the Server define the pick up key is *7, then user have to input the *7 in Pick up key field. When user press the IP Phone’s Pick up key, the Ip Phone will send out *7 to process the Pick up function.

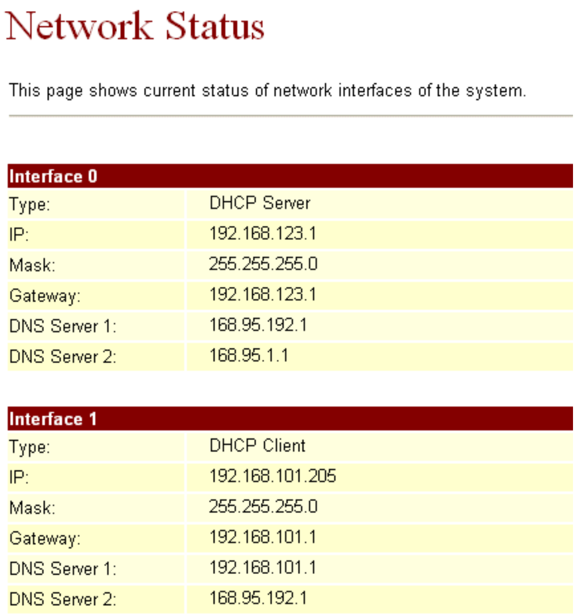
3.4.8.1 When you finished the setting, please click the Submit button. If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



3.5 etwork

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.

Interface 0	
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

Interface 1	
Type:	DHCP Client
IP:	192.168.101.205
Mask:	255.255.255.0
Gateway:	192.168.101.1
DNS Server 1:	192.168.101.1
DNS Server 2:	168.95.192.1

TEL：+86-755-89962209 FAX：+86-755-84272617 E_mail: info@kawatel.com

3.5.3 AN 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.

3.5.3.1 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.

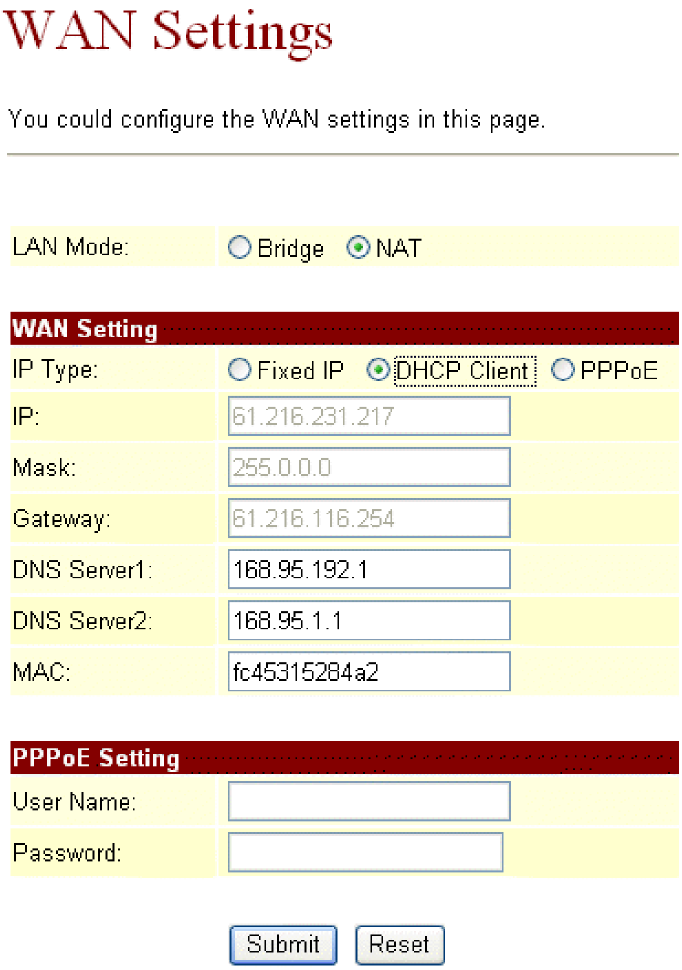
3.5.3.2 The WAN port default is DHCP Client mode, You can change the setting to Fixed IP Mode, or PPPoE Mode.

3.5.3.3 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.

3.5.3.4 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.

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

3.5.3.6 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.



TEL：+86-755-89962209 FAX：+86-755-84272617 E_mail: info@kawatel.com

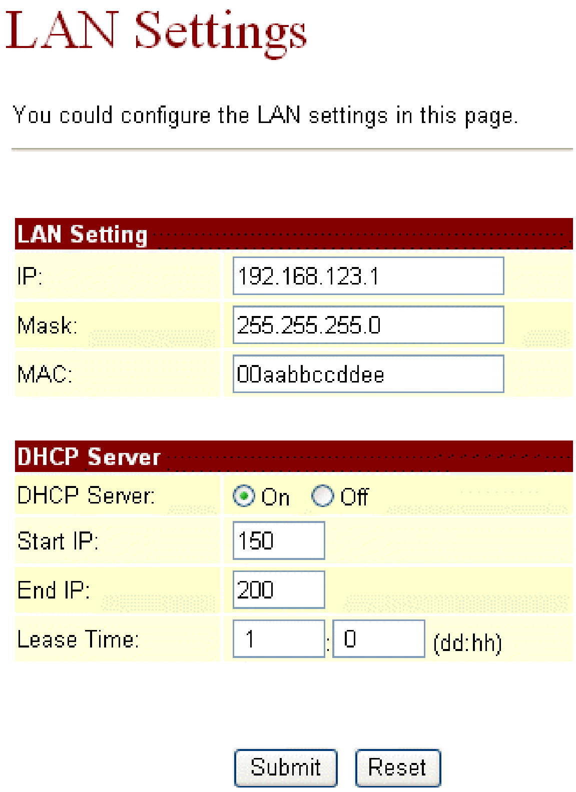
3.5.4 LAN Settings: In this page you can configure the IP Phone LAN port's setting.

3.5.4.1 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.

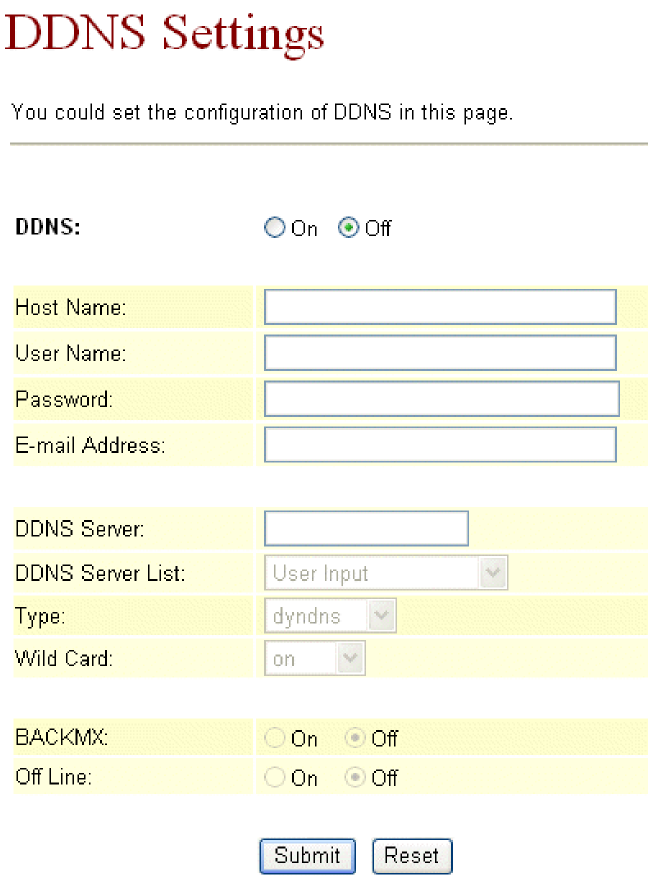
3.5.4.2 You can connect your PC to the LAN port, set your PC as DHCP Client mode, then you can get IP addreess from the TA.

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

3.5.4.4 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 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.



3.5.6 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.

3.5.6.1 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.

3.5.6.2 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.

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

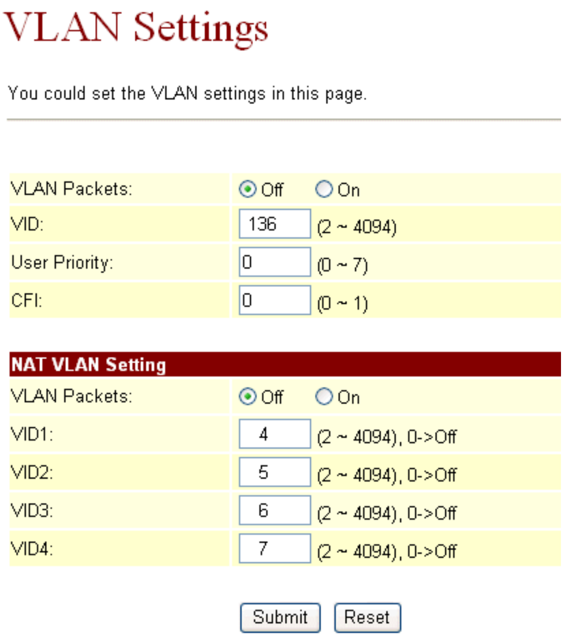
3.5.6.4 User Priority: Defines user priority, giving eight (2^3) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be defined by your service provider.

3.5.6.5 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.

3.5.6.6 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.

3.5.6.7 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.

3.5.6.8 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.



TEL：+86-755-89962209 FAX：+86-755-84272617 E_mail: info@kawatel.com

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.

3.6.2.1 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.

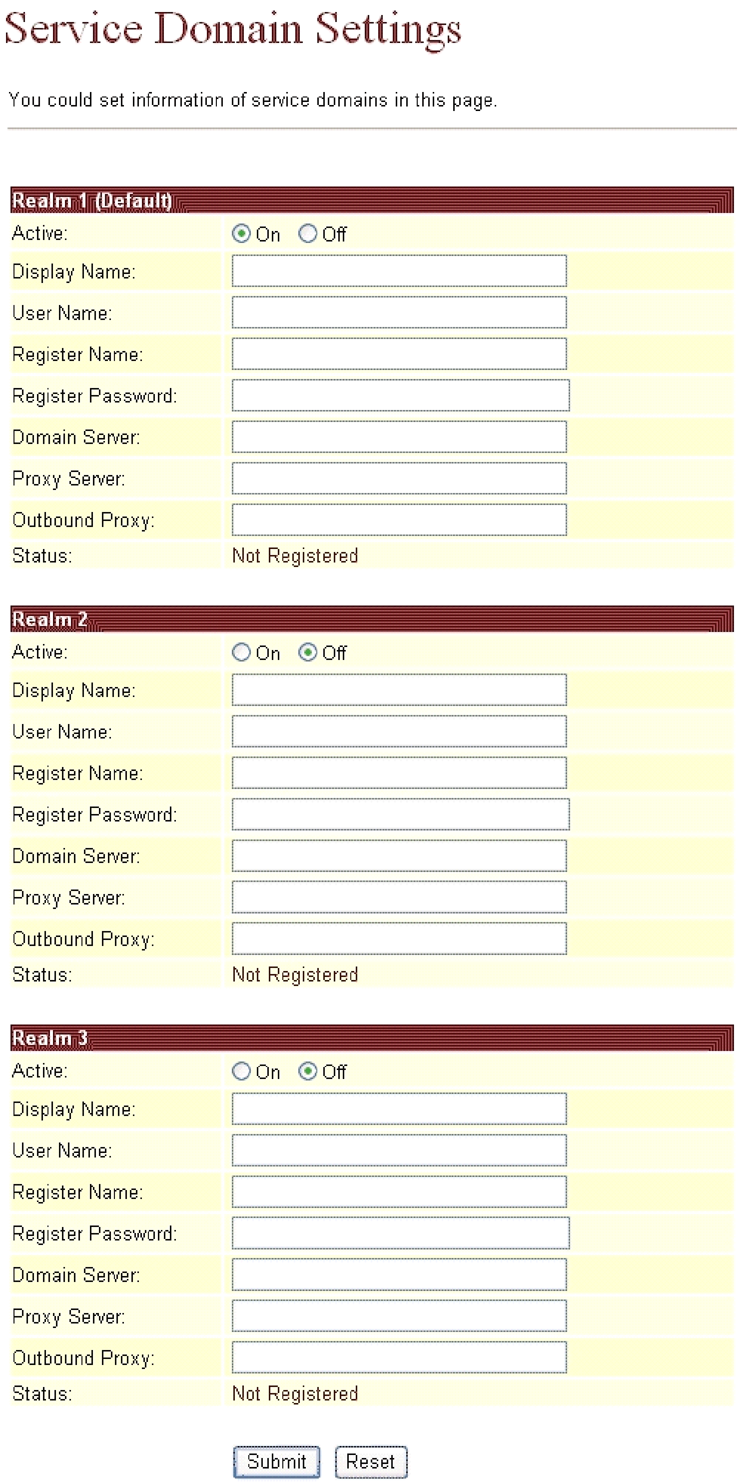
3.6.2.1.6 Proxy Server: you need to input the Proxy Server get from your ISP.

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.

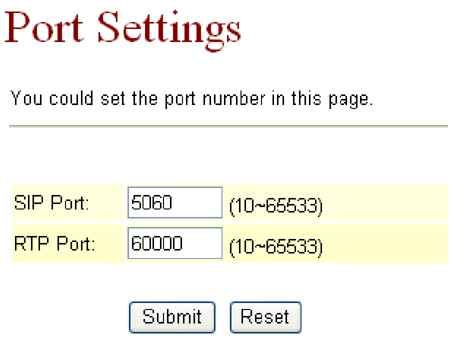
3.6.2.1.8 You can see the Register Status in the Status item. If the item shows “Registered”, then your VoIP Phone is registered to the ISP, you can make a phone call direcly.

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

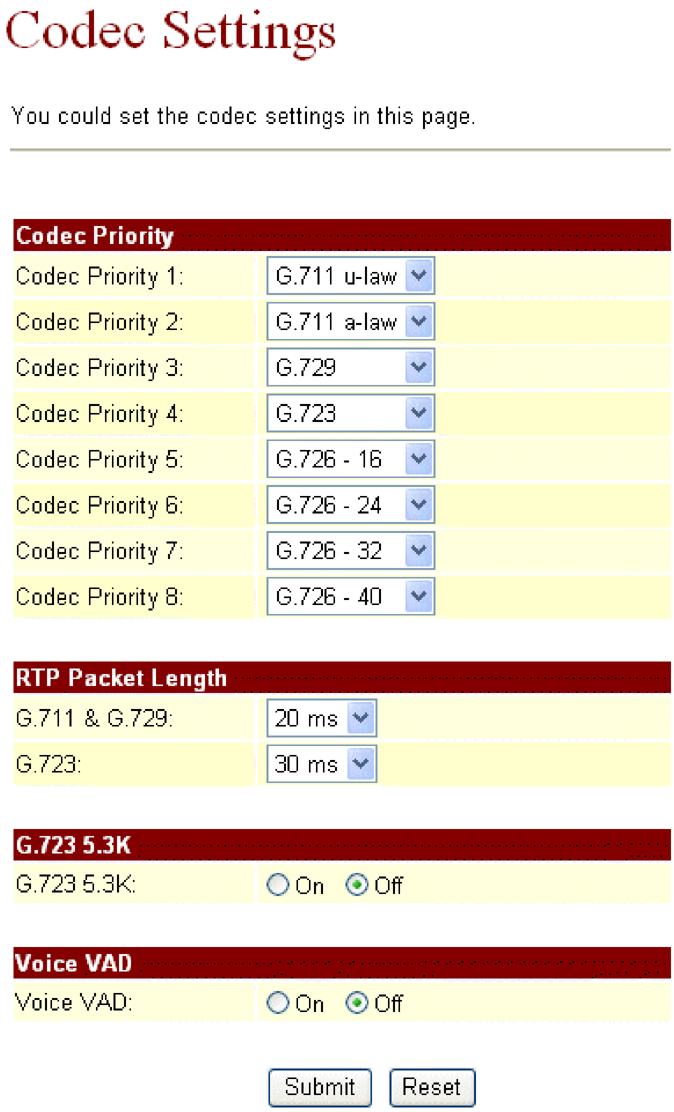
3.6.2.1.10 When you finished the setting, please click the Submit button.


Service Domain Settings
You could set information of service domains in this page.
Realm 1 (Default)
Active:
On
Off
Display Name:
User Name:
Register Name:
Register Password:
Domain Server:
Proxy Server:
Outbound Proxy:
Status:
Not Registered
Realm 2
Active:
On
Off
Display Name:
User Name:
Register Name:
Register Password:
Domain Server:
Proxy Server:
Outbound Proxy:
Status:
Not Registered
Realm 3
Active:
On
Off
Display Name:
User Name:
Register Name:
Register Password:
Domain Server:
Proxy Server:
Outbound Proxy:
Status:
Not Registered
Submit
Reset

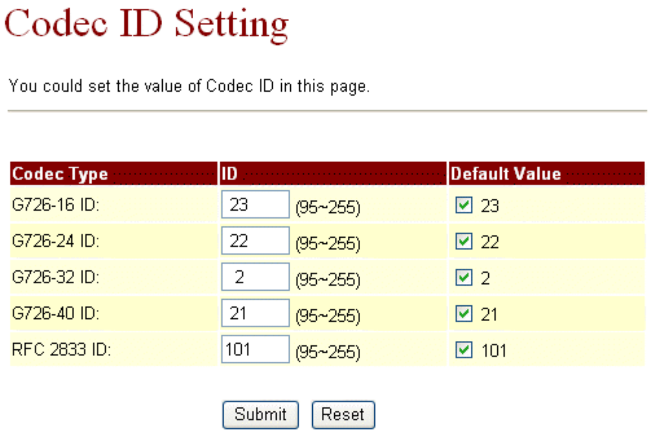
3.6.3 ort Settings: you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.



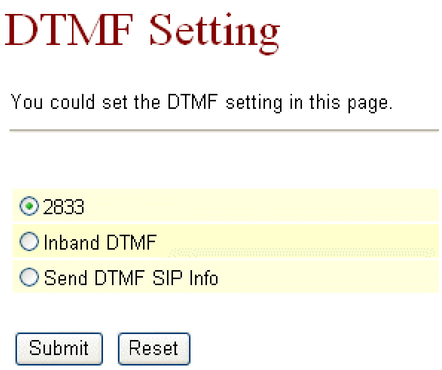
3.6.4 Codec Settings: you can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.



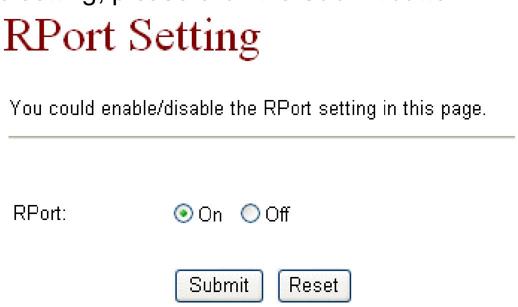
3.6.5 odec ID Settings: you can set the Codec ID to meet the other device's requirement. When you finished the setting, please click the Submit button.



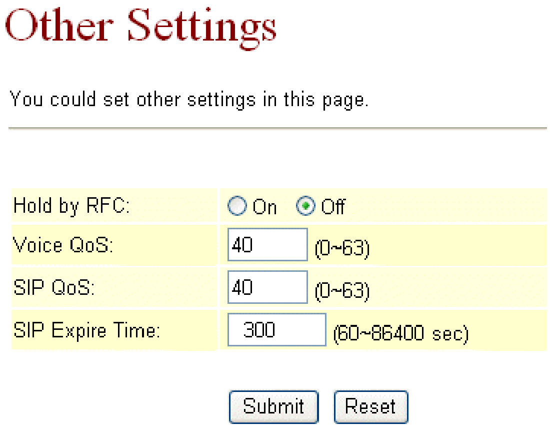
3.6.6 DTMF Setting: you can setup the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.



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.



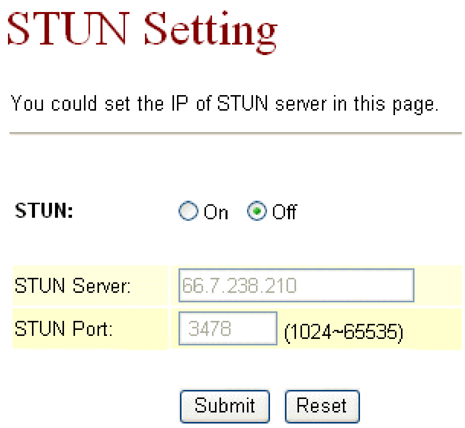
3.6.8 ther Settings: you can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value 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.

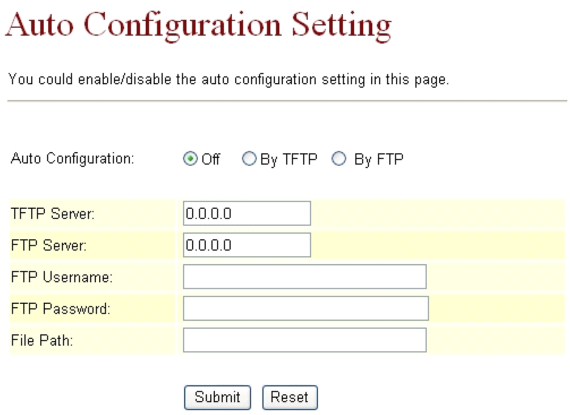
3.7.2 STUN Setting: you can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your VoIP Phone working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.



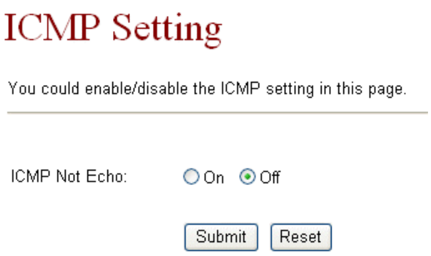
3.8 Others.

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

3.8.2 Auto Config: you can setup the Auto Configuration Enable/Disable and auto configuration by FTP or TFTP. 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.

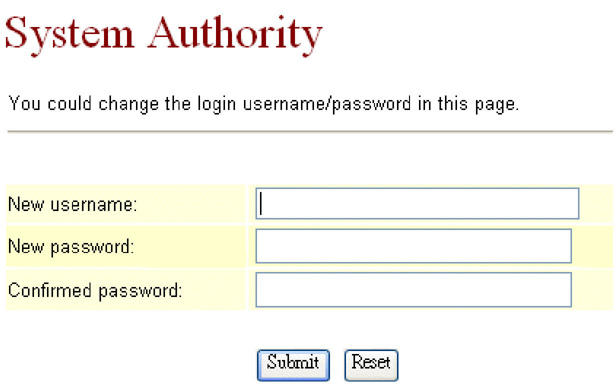


3.8.3 ICMP Setting: you can setup the ICMP echo Enable/Disable in this page. This function can disable echo when someone ping this device, it can avoid haker try to attack the device. When you finished the setting, please click the Submit button.



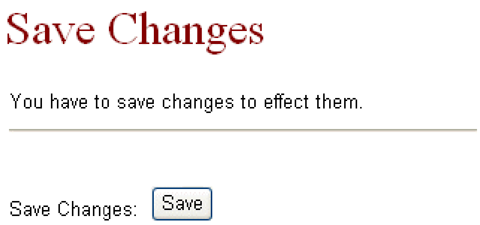
3.9 System Auth.

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



3. 10 ave 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

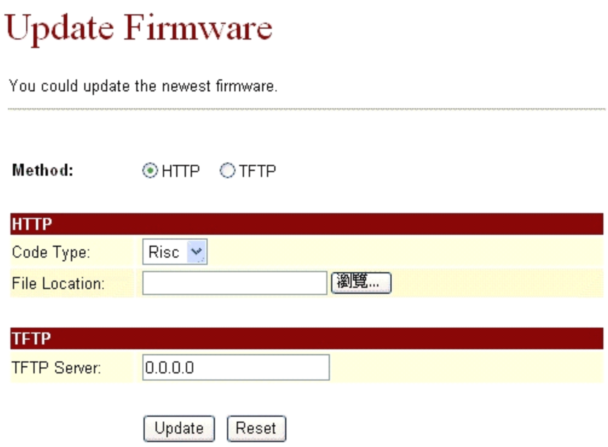
3. 11. 1 In Update you can update the VoIP Phone's firmware to the new one or do the factory reset to let the VoIP Phone back to default setting.

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:

3. 11. 2. 1 Select the firmware code type, Risc or DSP code.

3. 11. 2. 2 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.

3. 11. 2. 3 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.