IP Phone User's Manual

Last Update: 2007/10/05

1 Introduction

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

1.1 Hardware Overview

The IP Phone has the following interfaces for Networking, Power Connector and Optional FXO (landline Port).

Two RJ-45 networking interface

These two interfaces support 10/100Mps Fast Ethernet.
WAN: Connect to the ADSL or Router.
LAN: Connect to your PC computer. You can then connect to Internet through the Internet Sharing function of the Adapter.

1.1 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 • NAT/DHCP Server	• Ring Tone • Ring Back Tone • Dial Tone • Busy Tone • Programming Tone
	Phone Function
Codec	• Volume Adjustment • Speed dial key • Phone book • Flash
• 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	**IP Assignment**
	• Static IP • DHCP • PPPoE
Voice Quality	
• VAD: Voice activity detection • CNG: Comfortable noise generator • LEC: Line echo canceller • Packet Loss Compensation • Adaptive Jitter Buffer	**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 • IVR/Keypad
• In-Band DTMF • Out-of Band DTMF • SIP Info	**Firmware Upgrade** • TFTP • Console • HTTP
SIP Server • Registrar Server (three SIP account) • Outbound Proxy	

2 Keypad Interface

Key Name	Description
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
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
SPK	Turn on/off Speaker Phone
Line1~Line4	This is the Line1 to Line4, will be 4 line
M1~M4	This is the M1 to M4, this is 4 speed dial number.
Conf	This is three way conference function
Call In	This is Incoming call list
Call Out	This is out going call list
Volume -/+	This is volume setting

3 Setup 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 in the Web page. In the end of IP address, please add the port number ":9999". Ex: http://192.168.123.1:9999

The default LAN port IP is **192.168.123.1**, with DHCP Server enabled. The default WAN port has DHCP client enabled; the IP would be automatically assigned if it is connected to a network with DHCP server.

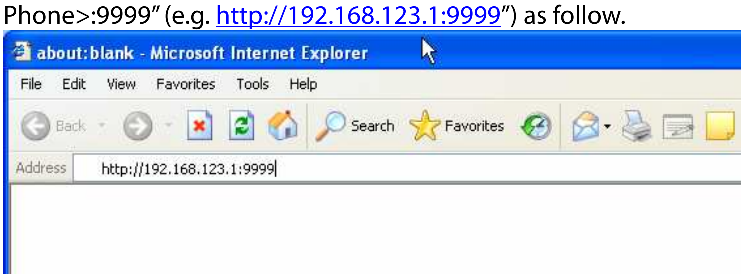
3.1 First Time Login

STEP 1 - Power Up and Connect the Phone

- Connect IP Phone WAN Port (RJ45) to a NAT Router or ADSL Modem using a Category 5 network cable. Connect IP Phone LAN Port (RJ45) to a PC using a Category 5 network cable.
- Connect the 9V Power Adapter to IP Phone Power Socket.
- The LCD Panel should then lights up, showing Date, Time and "No Service" if SIP account is not yet set up.
- Pick up the Phone, and the LCD Panel will show "IP Dialing…", and you show hear a dialing tone. If not, please check if the RJ45 WAN port is connected properly.

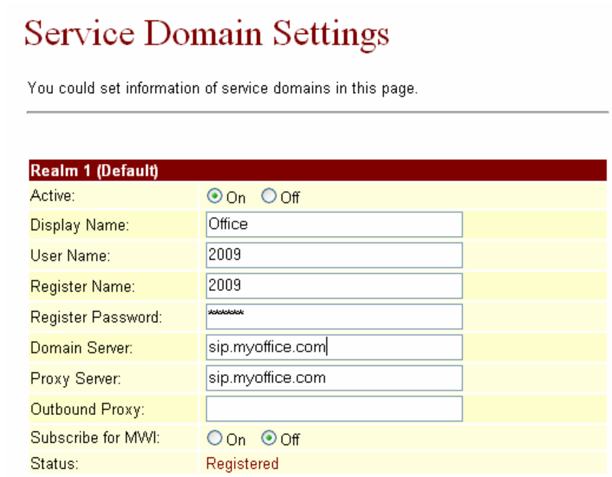
STEP 2 – Set Up the VoIP Account

- Press key MENU / 4. Network / 2. Status from the keypad to check the IP address of the Phone. The MENU key is used for escape, and the ENTER key for selection. The default IP address is 192.168.123.1.
- Inside your PC, open the browser (Internet Explorer), and enter "http://<ip address of IP Phone>:9999" (e.g. http://192.168.123.1:9999") as follow.



- Please input the username and password into the blank field. The default setting is:
The default username is: root; and the password is: test. If you use the account login, you can configure all the setting. *Please inquiry your distributor if the password is not correct.*

- Click "SIP Settings"->"Service Domain" to set up the SIP account in your IP Phone.



Please refer to Chapter 5.5 for details. **Once you change the setting in the Web Management interface, please 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.**

- The LCD panel will then show Date, Time and **registered <phone number>** after successful SIP registration.

STEP 3 – Make your First Call

- Pick up the Handset; you should now hear a dial tone.
- Press the phone number you would like to call, and ended with a '#'
  <Phone Number> + '#' (e.g. 1083#)
  That '#' key is tell the phone to dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapsed.

3.2 Default Reset from Keypads

Press MENU / 7.Administrator / 2.Default setting / 1.Load default by using Menu and arrow keys to reset back to factory defaults, and the LCD panel will start **showing Loading Program** and **System Initialized**. Please use the MENU key for escape, and the ENTER key for selection.
Press **MENU / 7.Administrator / 6.Restart** to reboot IP Phone.

3.3 Default Setting

LAN IP Address: 192.168.123.1 (LAN)
WAN IP Address: Dynamic Address/DHCP (WAN)
Web Management Page
Login Name: root
Password: test

4 Application Example

4.1 SIP-to-SIP Calling/Answering

Applications:

Both parties are registered to SIP server with either fixed real IP or private IP under NAT router. The SIP-to-SIP calling works when both calling and answering parties are registered to SIP server with given registered phone numbers.

Configurations:

- Select "DHCP Client", and bridge "ON" in the "Network / Network settings" pages,
- Remember to click the "Submit" button,
- Select Active "ON" in the "SIP settings / Service Domain" pages,
- Enter the Register Name, Register Password, Proxy Server, and Outbound Proxy,
- Select "ON" in "NAT settings / STUN setting" page, if Outbound Proxy is NOT available.
- Upon successful SIP registration, the LCD will show registered <phone number>.

Callings:

- Pick up the phone, and you should hear a dial tone.
- Press 1688# or 1688 to call the party with the registered SIP phone number 1688. Note that # key will dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapsed.

4.2 SIP to Direct IP Calling

Applications:

The application is for the calling party with ADSL connection as in either Diagrams A or B. The calling party is registered to SIP server with either fixed real IP or private IP under NAT router. The answering party is with fixed real IP.

Configurations:

- Same as in Example 4.1
- Select "ON" in "NAT settings / STUN setting" page, if Outbound Proxy is NOT available.
- Upon successful SIP registration, the LCD will show registered <phone number>.

Callings:

- Press Hand-Free key for speakerphone, and you should hear a dial tone.
- Press 211*21*191*4# or 211*21*191*4 to call the party with the real IP address of 211.21.191.4. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

4.3 Direct IP to Direct IP Calling/Answering

Applications:

The applications are for ADSL connection without NAT router as in Diagram A. Both parties are with fixed real IP. The Direct IP calling works when both calling and answering parties are with known fixed IP. SIP server registrations are not required in this application.

Configurations:

- Select "Fixed IP", and bridge "ON" in the "Network / Network settings" page
- Enter the items of IP, Subnet Mask, Gateway IP
- Click the "Submit" button.

Callings:

- Pick up the phone, and you should hear a dial tone.
- Press 211*21*191*4# or 211*21*191*4 to call the party with the real IP address of 211.21.191.4. Note that # key will dial out the number immediately. Dialing without # will not dial out until the auto dial timer (default=5 seconds) elapsed. In a moment, you should hear a ring back tone, and wait for the VoIP called party to answer.

4.4 Direct IP to Direct IP Calling within NAT Router

Applications:

For the calling party in ADSL connection with NAT router as in Diagram B, this Direct IP calling can work when the answering parties are with fixed private IP addresses within the same VPN network, or with fixed real IP addresses.

Configurations:

- Select "Fixed IP", and bridge "ON" in the "Network / Network settings" page
- Enter the items of IP, Subnet Mask, Gateway IP
- Click the "Submit" button

Callings:

- Pick up the phone, and you should hear a dial tone
- Press 192*168*1*51# or 192*168*1*51 to call the party with the private IP address of 192.168.1.51. Press 211*21*191*4 to call the party with the real IP address of 211.21.191.4. In a moment, you should hear a ring back tone, and wait for the called party to answer.

4.5 3-Way Conference Call, Call Waiting, Call Hold

4.5.1 3-Way Conference Calling Application

This is for 3-way conference call among Parties A, B, and C. Three parties are registered to SIP server with either fixed real IP or private IP. The Flash/Transfer key is used to switch to the other phone line or HOLD, and is quite useful for the 3-way conference call and the call waiting function.

Callings:

- Make a phone call from Party A to the first phone number Party B
- After the first call is established, press **HOLD key** (or Flash key) from Party A to hold the call, and Party A should hear a dial tone
- Make another phone call from Party A to the second phone number Party C
- After the second call is established, press **CONF key** from Party A to join in Party B for three-way conference call

4.5.2 Call Waiting Application:

When a new call is coming while you are talking, you can push the HOLD / FLASH key to switch to the new call. You can push the HOLD / FLASH key to switch between the two calls.

4.5.3 Call Hold Application:

You may push the HOLD key to hold the current call for a while, then push HOLD key again to resume talking.

4.5.4 Call Transfer

You can transfer the current call to another user using the Call Transfer function

- While Party A is having a phone call with Party B, and Party A would like to transfer the call to Party C
- Party A should press HOLD key, a dial tone would then be heard. Party A should then press TRANSFER key, followed by Party C's number and a "#" number
- The Party A's call would end, and the Party C's phone would ring. Once Party C picks up the call, he would speak to Party B.

Alternative Way to Call Transfer

- While Party A is having a phone call with Party B, and Party A would like to transfer the call to Party C
- Party A press TRANSFER key, a dial tone would then be heard. Party A then press Party C's number and a "#" number. Once a Waiting tone is heard, Party A should hang up the Call.
- Party C's phone would ring. Once Party C picks up the call, he would speak to Party B.

4.5.5 Call Forward

You can forward the incoming call to a third-party

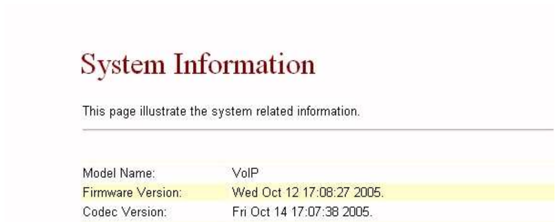
- Set up the Call Forward Number using the Web Configuration (Refer to 3.2.1) or Using the LCD

Menu (“Phone Setting” -> “Call Forward”)
- Press the Key FORWARD to enable the Call Forward

5 Phone Configuration

5.1 System Information

When you login the web page, you can see the IP Phone current system information like firmware version, company… etc in this page.
Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.

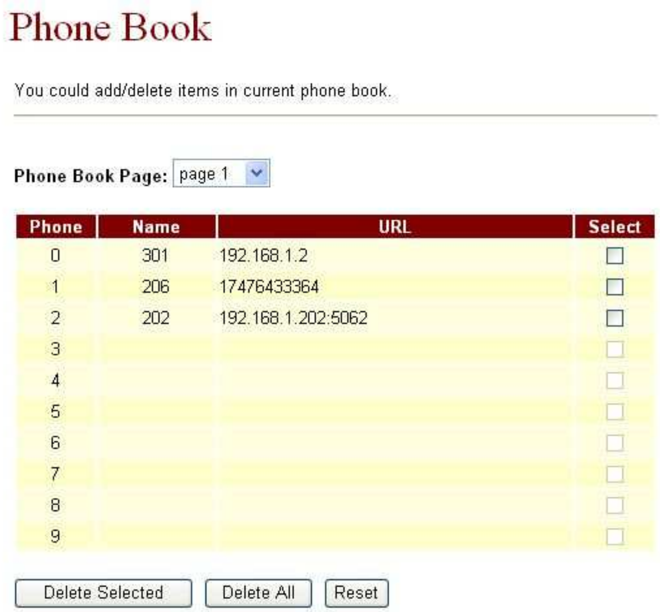


5.2 Phone Book

In Phone Book contains Speed Dial Settings. You can setup the Speed Dial number. If you want to use Speed Dial you just dial the speed dial number (from 0~9) then press “#”.

Please Book Page	Default as Page 1, ranged from Page 1 to Page 14
Phone	Phone Book Entry, ranged from 0 to 140
Name	Phone Book Entry Name. Check example for details
URL	Phone Book Entry URL: the line number or IP. Check example for details.

Example



Example 1: Name: 301, URL: 301@192.168.1.2
If user dialed [301#], the phone would dial out [192.168.1.2] actually.

Example 2: Name: 206, URL: 17476433364
If user dialed [206#], the phone would dial out [17476433364] actually.

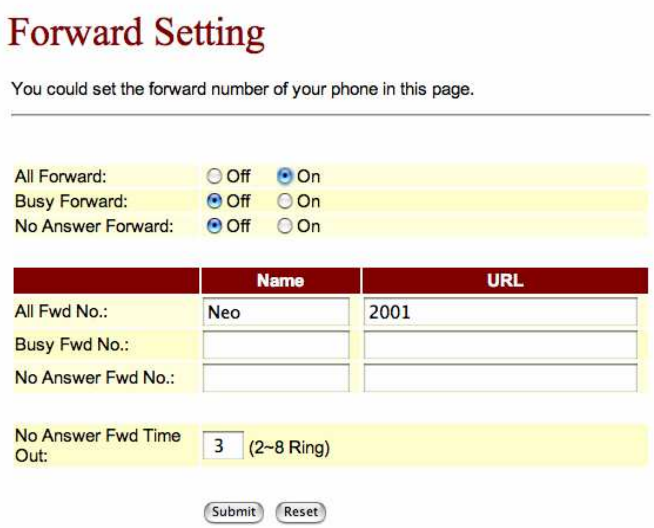
Example 3: Name: 202, URL: 192.168.1.202:5062
If user dialed [202#], the phone would dial out [192.168.1.2:5064] actually.

5.3 Phone Setting

In Phone setting contains Call Forward, SNTP Settings, Volume Settings, Block Setting, Auto Answer, Caller ID, Auto Dial Setting, Flash Time Setting and Call Waiting Setting functions.

5.3.1 Call Forward

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.



5.3.1.1 All Forward:

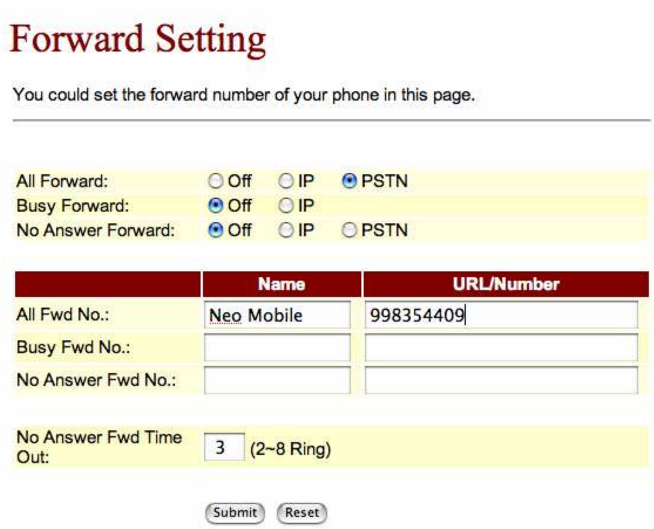
All incoming call will forward to the number you choose. You can input the name and the phone number in URL/Number field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.

5.3.1.2 Busy Forward:

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

5.3.1.3 No Answer Forward:

If you can not answer the phone, the incoming call will forward to the number you choose. 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 choose.



5.3.1.4 Call Forward from VoIP to PSTN *(Only supported by model with FXO port)*

1. Select the case to forward (e.g. All Forward), Click option of "PSTN"
2. Enter the PSTN phone number in the "All Fwd No."->"URL/Number" field, and the "Name Field"
3. Click Submit to take effect.

5.3.1.5 Call Forward from PSTN to VoIP *(Only supported by model with FXO port)*

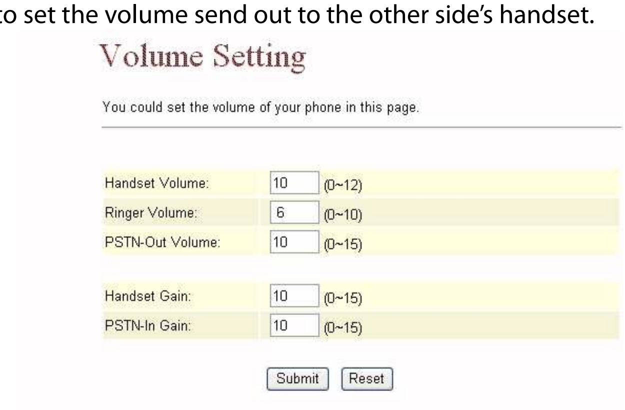
1. Select the case to forward (e.g. All Forward), Click option of "IP"
2. Enter the PSTN phone number in the "All Fwd No."->"URL/Number" field, and the "Name Field"
3. Click Submit to take effect.

5.3.2 SNTP setting

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.

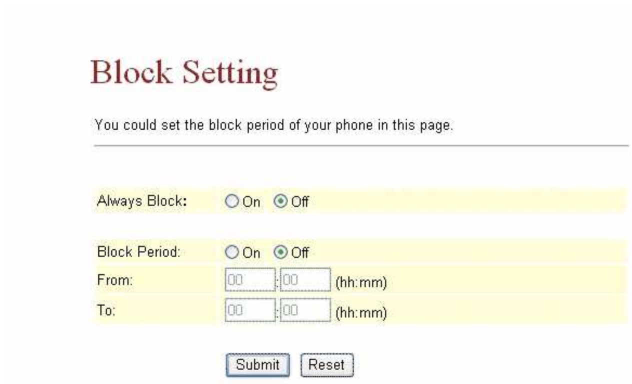
5.3.3 Volume setting

you can setup the Handset Volume, Ringer Volume, and the Handset Gain. When you finished the setting, please click the Submit button.
3.4.4.1 Handset Volume is to set the volume for you can hear from the handset.
3.4.4.2 Ringer Volume is to set the ringer volume for you can hear.
3.4.4.3 PSTN-Out Volume is to set the PSTN volume for you can hear.
3.4.4.4 Handset Gain is to set the volume send out to the other side's handset.
3.4.4.5 PSTN-In Gain is to set the volume send out to the other side's handset.



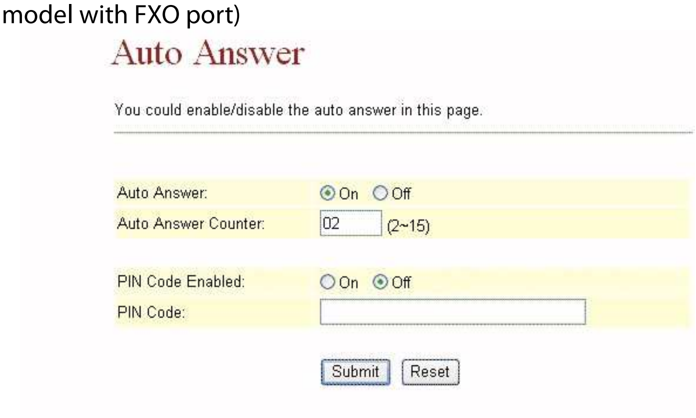
5.3.4 Block setting

You can setup the Block Setting to keep the phone silence. You can choose Always Block or Block a period.
Always Block: All incoming call will be blocked until disable this feature.
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.
When you finished the setting, please click the Submit button.



5.3.5 Auto Answer *(Only Supported by model with FXO port)*

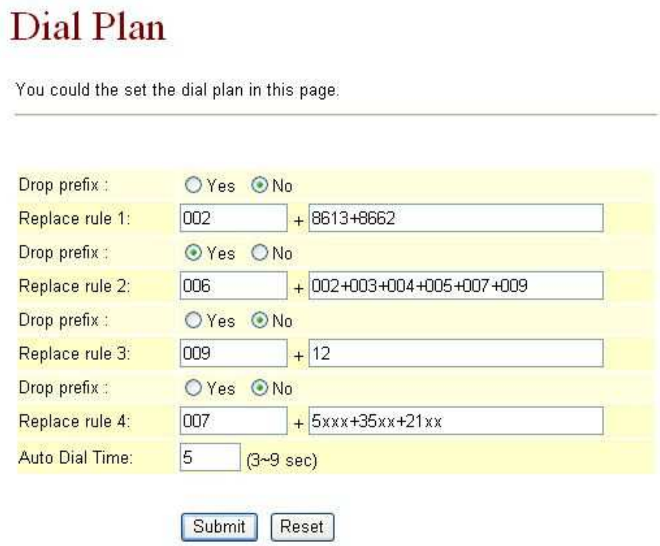
You can set the Auto Answer function to answer the incoming call by the phone. If the call is come from the IP, then the IP Phone can let user to redial the call to PSTN phone number. If the call is coming from PSTN, then the IP Phone can let user to redial to IP Phone number. Auto Answer Counter is to set after the ring counts meet the number you set then the auto answer will enable. (This feature is only supported with the model with FXO port)



5.3.6 Call Waiting Setting function

You can Enable/Disable the Call Waiting function, when you are talking with someone, there is a new incoming call, you will hear the call waiting tone.

5.3.7 Dial Plan



Example 1: Drop prefix: No, Replace rule 1: 002, 8613+8662.
1：If what we dialed begins with 8613, number "002" when is added to the beginning; so the actual number dialed would be [002+8613+xxx].
2：If what we dialed begins with 8662, number "002" when is added to the beginning; so the actual number dialed would be [002+8663+xxx].

Example 2： Drop prefix: Yes, Replace rule 2: 006, 002+003+004+005+007+009；
1：If what we dialed begins with 002, the number "002" would be replaced with "006"; so the actual number dialed would be [006+xxx].
2：If what we dialed begins with 003, the number "003" would be replaced with "006"; so the actual number dialed would be [006+xxx].

Example 3: Drop prefix: No, Replace rule 3: 009, 12.
1：If what we dialed begins with 12, the number "12" would be prefixed by number "009"; so the actual number dialed would be [009+12+xxx].

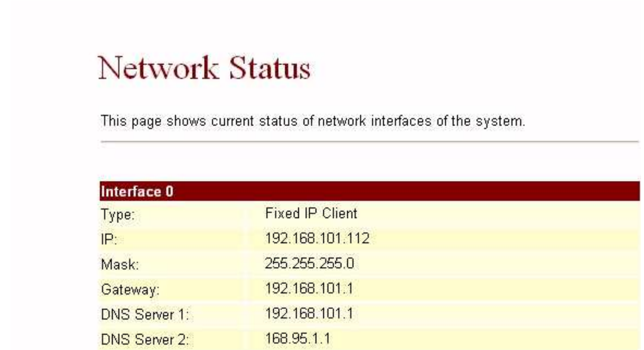
Example 4:Drop prefix: No, Replace rule 4: 007, 5xxx+35xx+21xx.
1：If what we dialed begins with 5, and followed any 3 numbers, the number would be prefixed by number "007"; so the actual number dialed would be [007+5xx].
2：If what we dialed 534, the number 5 is only followed by 2 numbers, it does not match with the dialing rule; so the actual number dialed is still [534].
3：If what we dialed begins with 35, and followed any 2 numbers, the number would be prefixed by number "007"; so the actual number dialed would be [007+35xx].
4：If what we dialed 358822, the number 5 is only followed by 2 numbers, so it does not match with the dialing rule; so the actual number dialed is still [358822].

5.4 Network

In Network you can check the Network status, configure the Network Settings and DDNS settings.

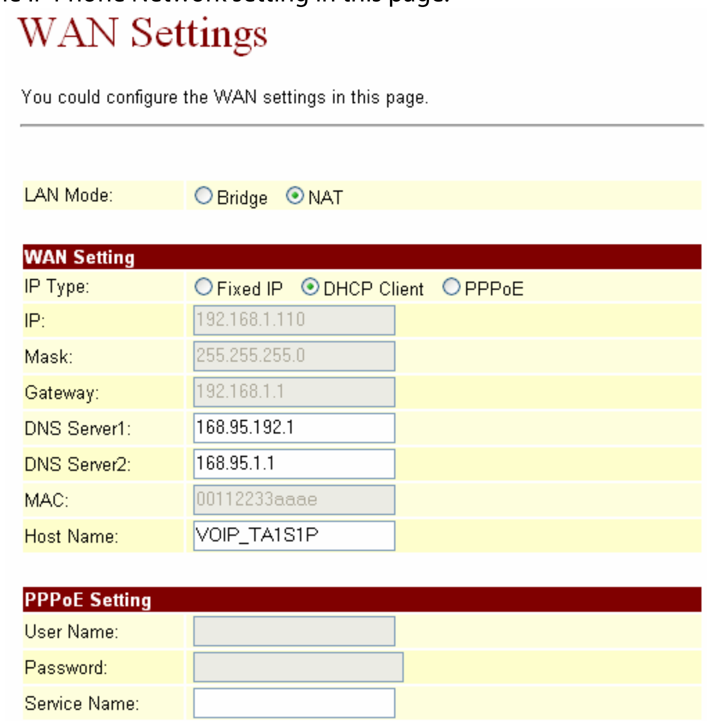
5.4.1 Network Status

You can check the current Network setting in this page.



5.4.2 WAN Settings

You can configure the IP Phone Network setting in this page.



5.4.2.1 Mode

There are two mode for the WAN port: Bridge or NAT.
Bridge mode: The two WAN and LAN Ethernet ports will be bridged and transparent.
NAT mode: The embedded NAT will be enabled.

The IP type for IP Phone is default at Fixed IP (192.168.1.100). You may select a proper IP type for your network requirements.

5.4.2.2 NAT Settings

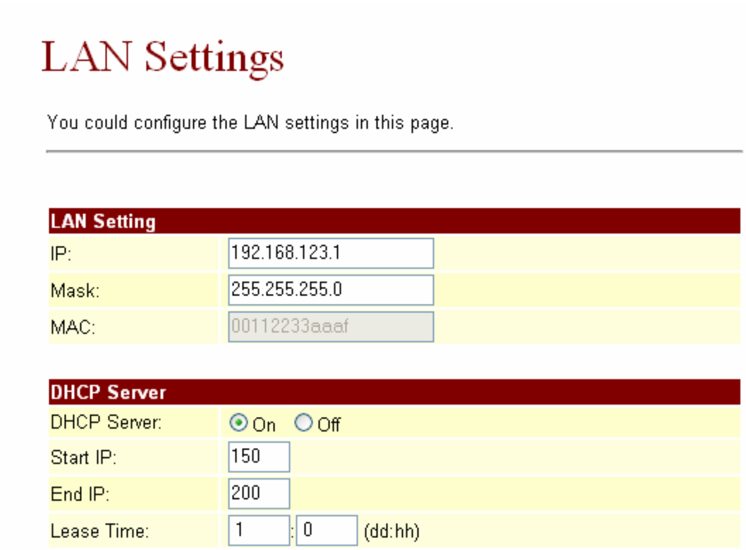
To enable embedded NAT, you must set mode of WAN port to "NAT". This embedded NAT is useful for ADSL users without NAT router, and it separates the WAN port from the LAN port to perform router IP address translation.
For the WAN port, please select a proper IP type:
DHCP Client(default): Automatically get an assigned IP from the your router. Please select this if there is a router in your network.
PPPoE: Most commonly way to connect to your Internet Service Provider through ADSL. Enter the given username and password from the Internet Service Provider.

Fixed IP: Assign an Fixed IP to the Adapter.

MAC settings for LAN and WAN have been preprogrammed and must be different from each other. Please leave it default unless necessary.

After you finished the setting, please click the "Submit" button.

5.4.3 LAN Settings



5.4.3.1 LAN Setting

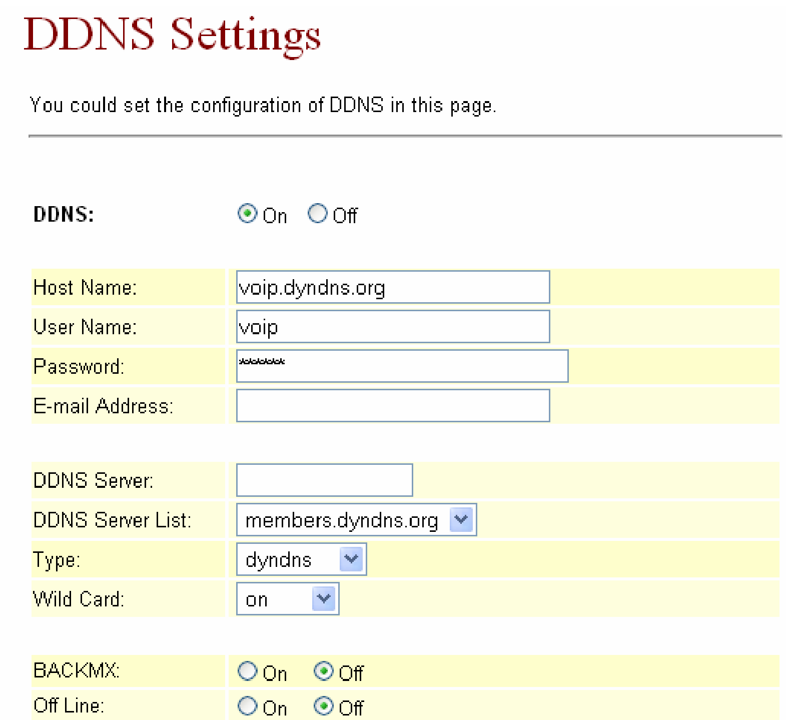
IP: The IP Address of the LAN Port. Default is 192.168.123.1

5.4.3.2 DHCP Server

There is built-in DHCP Server in the Phone Adapter. The default is enabled.
Start IP/End IP: The range of IP to be assigned. E.g. Assign IP from 192.168.1.150 to 192.168.1.200
Lease Time: The lease time of each DHCP assignment

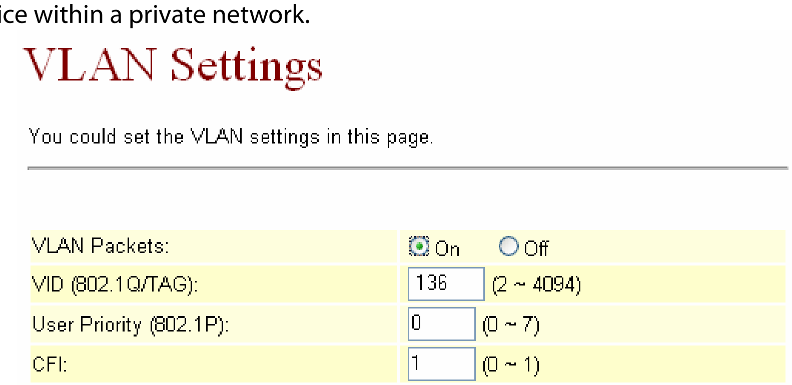
5.4.4 DDNS Setting

You can configure the DDNS setting in this page. You need to have the DDNS account and input the information 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.



5.4.5 VLAN Setting

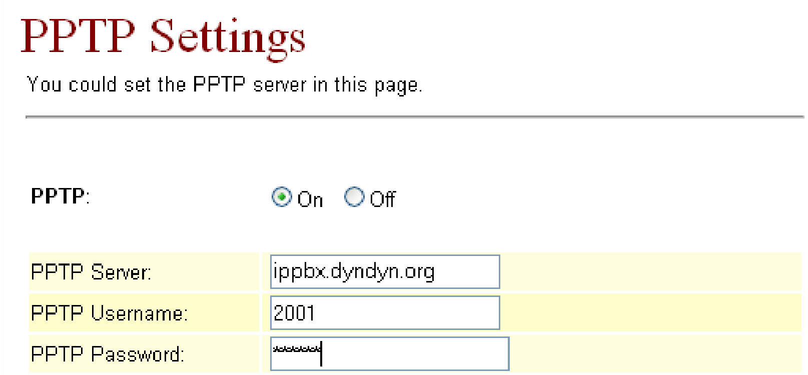
Our Phone Adapter support VLAN. Note that building a VLAN network requires a switch support VLAN. VLAN can be used to separate VOIP traffic from other DATA traffic. Then, user can configure the VLAN switch to handle the VOIP VLAN tag in higher priority. This is one of the ways to guarantee Quality-of-Service within a private network.



5.4.6 PPTP Setting

You can connect the Phone Adapter to a VPN (PPTP) Server. Using VPN, user can connect to a SIP Server (SIP Proxy or IP PBX) inside a private network, or making a secure phone call through the Adapter.

PPTP Server: VPN Server to connect.
PPTP Username/Password: Username and Password for the VPN account of the VPN Server

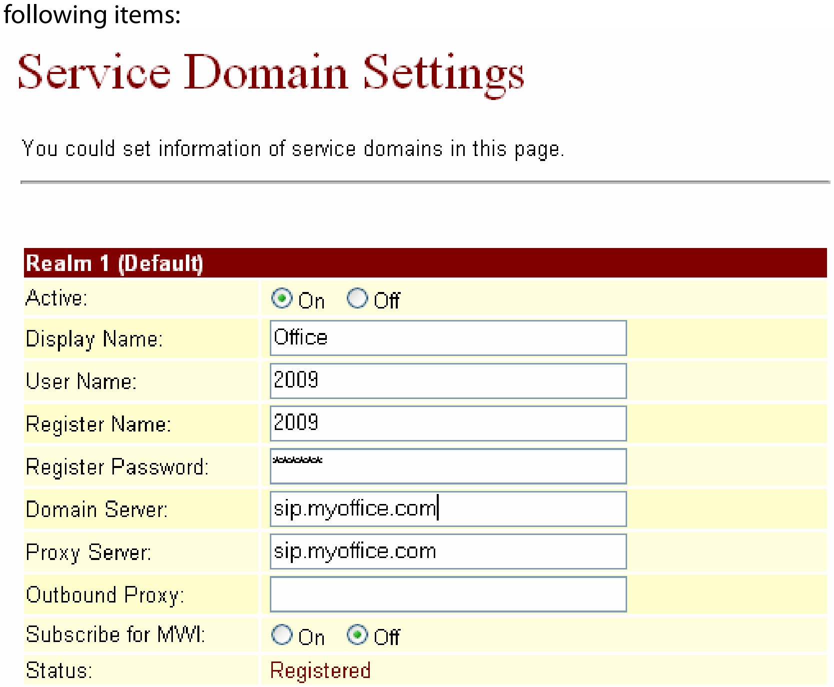


5.5 *SIP Settings*

You can setup the Service Domain, Port Settings, Codec Settings, RTP Setting, RPort Setting and Other Settings for SIP Proxy Server registrations in this page.

5.5.1 Service Domain

You may register up to three SIP accounts in the IP Phone. You can call your friends via firstly enabled SIP account and receive the phone calls from all the three SIP accounts. It can support up to 3 SIP accounts, allowing user to register on different service providers. Click "Active" ON to enable the Service Domain, then enter the following items:



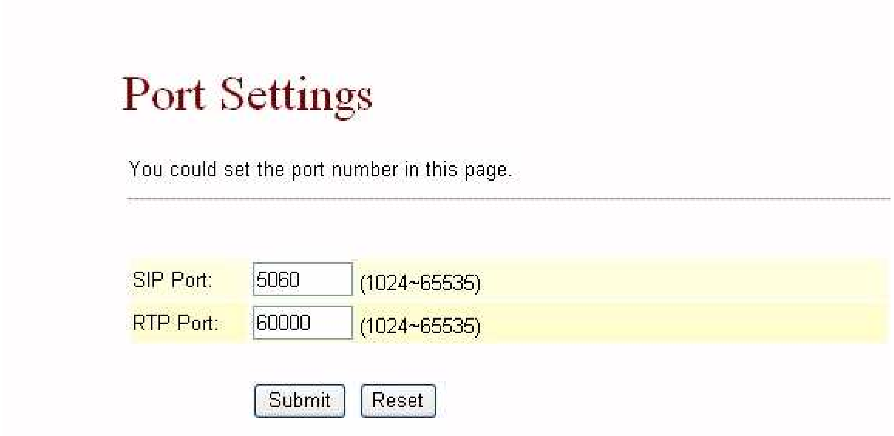
Realm (1 ~ 3)
Display Name: Enter the name you want to display. You can choose any name you like.
User Name: Enter the User Name given by your ITSP or IP PBX.
Register Name: Enter the Register Name given by your ITSP or IP PBX.
Register Password: Enter the Register Password given by your ITSP or IP PBX.
Domain Server: Enter the Domain Server given by your ITSP or IP PBX.
Proxy Server: Enter the Proxy Server given by your ITSP or IP PBX.
Outbound Proxy: Enter the Outbound Proxy of ITSP or IP PBX. If not provided or if it is same as Proxy Server, just leave it blank
Subscribe for MWI: Click On if your ITSP or IP PBX supports Voice Mail notification

When it shows "Registered" in the Register Status, it indicates a successful registration to the ITSP, and the "PHONE" LED will start flashing. The IP Phone is then ready for VoIP call.

If you have more than one SIP account, please follow the steps to register to other ITSPs. After you finished the setting, please click the "Submit" button.

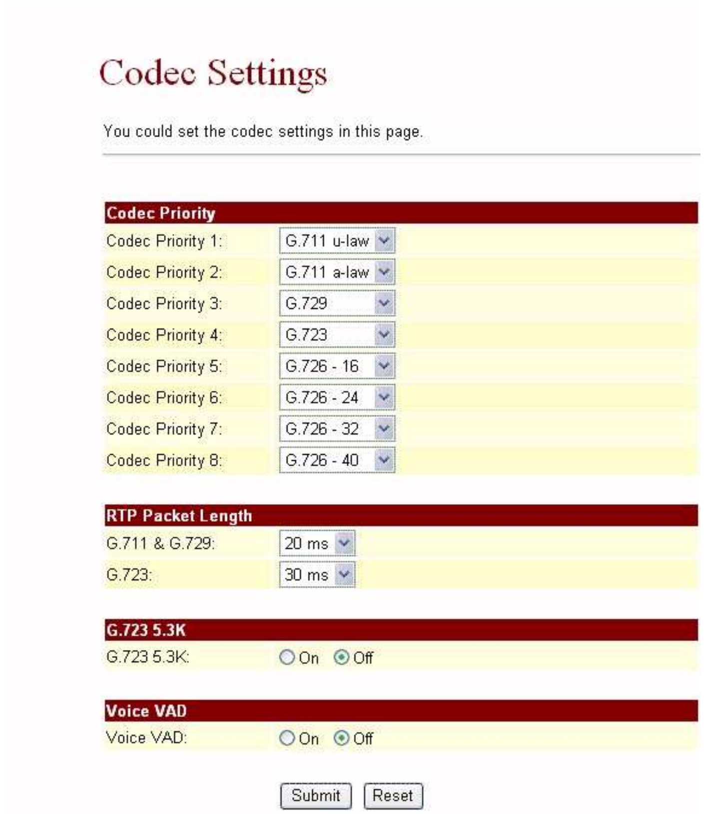
5.5.2 Port Settings

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.



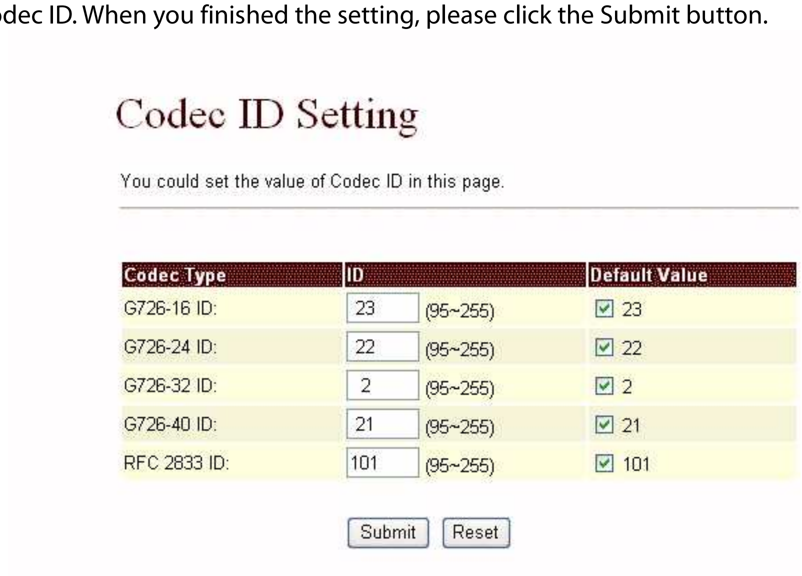
5.5.3 Codec Settings

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.



5.5.4 Codec ID Setting

Sometimes 2 VoIP devices with different Codec ID will cause the interoperability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use, then you can change your Codec ID. When you finished the setting, please click the Submit button.



5.5.5 DTMF Setting

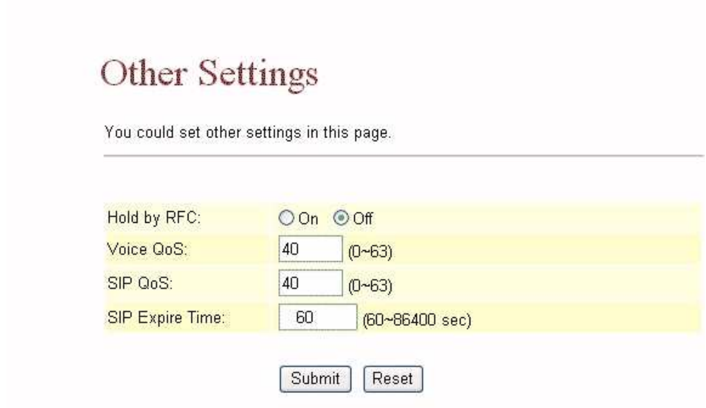
You can setup the InBand DTMF, 2833 Out-Band DTMF and Send DTMF SIP Info Enable/Disable in this page. Please configure this based on your VoIP Provider’s recommendation. When you finished the setting, please click the Submit button.

5.5.6 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.

5.5.7 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.



5.6 NAT Trans

You can setup STUN function. These functions can help your IP Phone working properly behind NAT.

STUN Setting

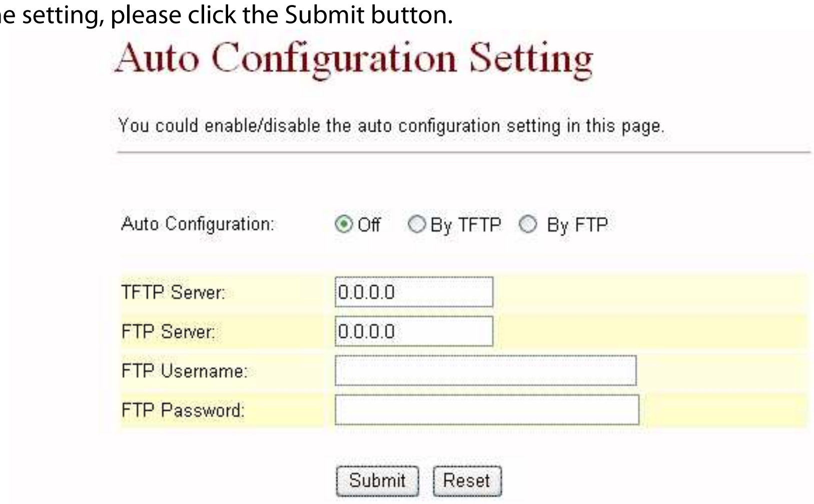
You can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your IP Phone working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

5.7 Others

You can setup Auto config, PTT Setting and ICMP Setting function. The function can configure your VoIP Phone automatically.

5.7.1 Auto Configuration Setting

You can setup the Auto Configuration Enable/Disable and auto configuration TFTP 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.



5.7.2 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 hacker try to attack the device. When you finished the setting, please click the Submit button.

5.8 System Auth

You can change your login name and password. . When you finished the setting, please click the Submit button.

5.9 Save Change

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

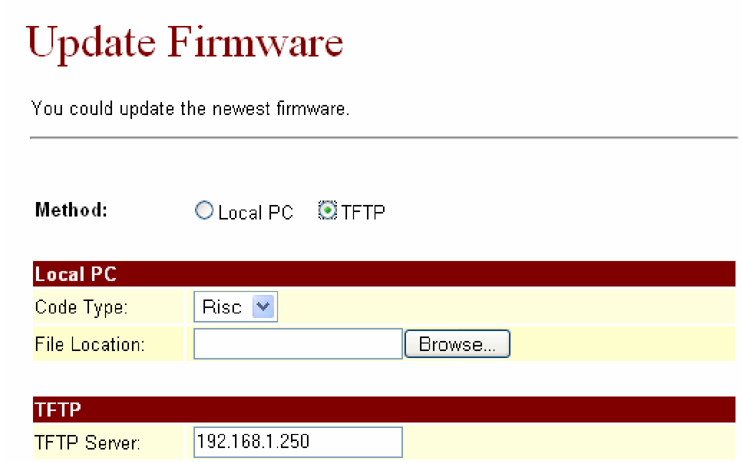
5.10 Update

You can update the IP Phone's firmware to a updated version or do the factory reset to let the IP Phone back to default setting.

5.10.1 Update New Firmware

You can update new firmware via HTTP in this page. You can upgrade 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 IP Phone then click the Update button.



5.10.2 Default Setting

You can restore the IP Phone to factory default in this page. You can just click the Restore button, then the IP Phone will restore to default and automatically restart again.

5.11 Reboot

5.11.1 Reboot

To reboot the IP Phone, you can reboot you the Reboot function under Phone MENU, or you can reboot using the web management page.

6 Troubleshooting Configuration

6.1 The LCD is showing "Ethernet Error!"

Make sure the WAN port of IP Phone is connected properly to your ADSL Modem or Router. Power Reset and check again.

6.2 The LCD is showing "No Service"

"No Service" indicates that the IP Phone is not successfully registered to the VoIP Service.

- Check if the Account information is set up correctly. Goto "SIP Settings"->"Service Domain" of the Web Management Interface. Make sure all the information is entered properly, especially is the password. Press Submit and Reboot the IP Phone
- Make sure the IP Phone can connect to the Internet. Check if your ADSL Modem is working properly.
- Call your VoIP Service provider, to check if there is any problem with their server.

6.3 DO NOT HEAR DIAL TONE?

No dial tone happens when there is Internet connection problem or when the Service registration failed. Please check the LCD display, and refer to 6.1 and 6,2.