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Figure 1 Line drawing of 9339 VoIP Phone (Front View) 44
Figure 2 Line drawing of 9339 VoIP Phone (Bottom View & Rear View) 45

1. INTRODUCTION

9339 VoIP (Voice over Internet Protocol) Phone provides a cost-saving solution for small business/home users on their telecommunication needs. This phone is designed in a user-friendly way and is very easy to operate. It complies with the SIP (Session Initiation Protocol) protocol open standard, which can be easily used with most of the existing VoIP services.

With the built in LCD display, the user can easily configure the VoIP Phone for first time installation in a few minutes. Besides, the advanced VoIP also provides rich telephone features such as the phone book, speed dial, call history, last number redial, call forward, transfer, volume adjustment and speakerphone etc. It is your best VoIP solution in the new generation communication.

Outline drawings of 9339 VoIP Phone are shown in Figure 1 and 2.

2. FEATURES

- Standard 4 x 3 alpha-numerical keypad
- Phone book with 140 groups memory
- Speed dial 10 memories
- 64 groups incoming call memory displayed, stored
- 64 groups outgoing call memory reviewed, stored
- Handsfree/Headset mode with LED indicator
- New Calls LED Indicator
- Mute key, toggle function with LED visual indicator suppression of speech transmission. Automatically release upon off-line
- Reset Key
- Transfer calls function
- Three-way conversation
- Call waiting Type II
- Call forwarding function
- Caller ID Type I
- LCD panel consist: Dot matrix type, 5 x 8 dots, 16-character x 2-row display
- Following RFC-3261 SIP standard: Support password authentication using MD5 digest an RFC-2833 for DTMF relay

- Dynamic IP support (DHCP and PPPoE): Getting IP from DHCP server using DHCP protocol or through ADSL modem using PPPoE protocol, automatically reconnect when PPPoE lost connection

- Passing through NAT devices: Can make outgoing and incoming calls under and NAT devices (even under two layer NAT devices) when working with the specific gatekeeper/proxy devices

- Software upgrade capability

- Advanced DSP (Digital Signal Processing) technology to ensure superior audio quality: Hardware System on a Chip solution with in DSP processor makes sure the perfect voice quality

- Support G.723.1,G.726, G729A/B, G.711 (A-law/U-law) voice codec: Following ITU-T standard to support best compatibility

- Provide calls history. Recorded both answered and missed incoming calls history and dialed number history and allow users to make a direct call from the call history

- Support Silence Suppression, VAD (voice Activity Detection), CNG (Comfort Noise Generation): Silence suppression and save about half of the network bandwidth needed during normal VoIP conversation

- Call with or without gatekeeper / proxy (direct IP dialing): Following standard SIP protocols and is compatible with most of existing SIP proxy gatekeeper

- Support RFC-3261, TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, FTP, PPP, PPPoE protocols

- The WAN Port automatically works for paralleled Ethernet cable and crossed Ethernet cable

- PC Port

- Headset Port

3. KEYS AND INDICATORS DESCRIPTION

3.1. MENU

This key is used as “Menu” key to bring out the menu selections on the LCD display.

3.2. LCD DISPLAY

Menu and all status shall be displayed for users.

3.3. ENTER

When inside the menu selection/settings on the LCD display, this key is used as “ENTER” key to enter into a lower layer of menu selection or to accept the edited item’s contents.

3.4. PHONE BOOK

Users are able to store up to 140 phone numbers by pressing the “Phone Book” button. For each item of the 140 phone book numbers, the user can store both the number and the name for display.

3.5. “▲ UP” Key

When the VoIP Phone is at idle state or talking on handset or speaker, this key is used to increase the volume of the voice sound. When the VoIP Phone is entered into the menu selection, this key is used to scroll up the menu items.

3.6. “▼ DOWN” Key

When the VoIP Phone is at idle state or talking on handset or speaker, this key is used to decrease the volume of the voice sound. When the VoIP Phone is entered into the menu selection, this key is used to scroll down the menu items.

3.7. CANCEL

This key is delete word or phone number.

3.8. RINGER INDICATOR

LED will flash when there is an incoming call.

3.9. NEW CALLS LED INDICATOR

In idle state, the New Calls LED will blink to indicate a new incoming call. Check the incoming call history to cancel the blink.

3.10.HANDSFREE/HEADSET KEY WITH LED

This key is pressed to switch between the usage of the handset and the speaker devices.

3.11. MUTE KEY WITH LED

Microphone can be muted by pressing “MUTE” key and deactivated by another pressing.

3.12. HOLD

To hold the conversation during the call. This key can also be pressed to do consultant-transfer of an active call to another VoIP Phone. When the VoIP Phone is active (incoming call answered or outgoing call accepted), by pressing this key, a dial tone will be heard, then the user can key in another VoIP Phone's number to call to another party and have a conversation with him. Press Hold to reactive the original call.

3.13.REDIAL

When the VoIP Phone is off-hooked and this key is pressed immediately, the last dialed number will be called out right away.

3.14.CONFERENCE

In the telephone conversation process, press " Hold" key, then the Hold side puts through to the third party and press "CONFERENCE" key to make a conference call.

3.15. TRANSFER

This key is pressed to transfer an active call to another VoIP Phone. When the VoIP Phone is active (incoming call answered or outgoing call accepted), by pressing this key, a dial tone will be heard, then the user can key in another VoIP Phone's number to transfer the call to another party.

When a call is incoming and this VoIP Phone is ringing, by pressing this "Transfer" key and then press another VoIP Phone's number can transfer the call immediately to another party without answering the call.

3.16. SPEED DIAL

Users are able to store 10 specific phone numbers in the slots of M1 - M10. Users are able to make a speed dial call to the specific party by pressing the speed dial key from M1 – M10.

3.17.RESET

This key is underneath the Speed Dial Memory Card, which use when the IP phone restore to the default settings.

4. TECHNICAL SPECIFICATIONS

4.1. General

The outer dimensions of 9339 VoIP Phone with the handset resting on the base unit is:

Width	:	213mm
Length	:	206mm
Height	:	92mm

The weight of the whole unit is 0.8 Kg.

4.2. Operating Temperature and Humidity

Temperature	:	0°c - 40°c
Humidity	:	90% RH max, No dew forming

4.3. Ringer

4.3.1 Ringer Volume	: Ringer volume setting from 0~10 (max)
4.3.2 Ringer Type	: Ringer Tone selection from 1~4.

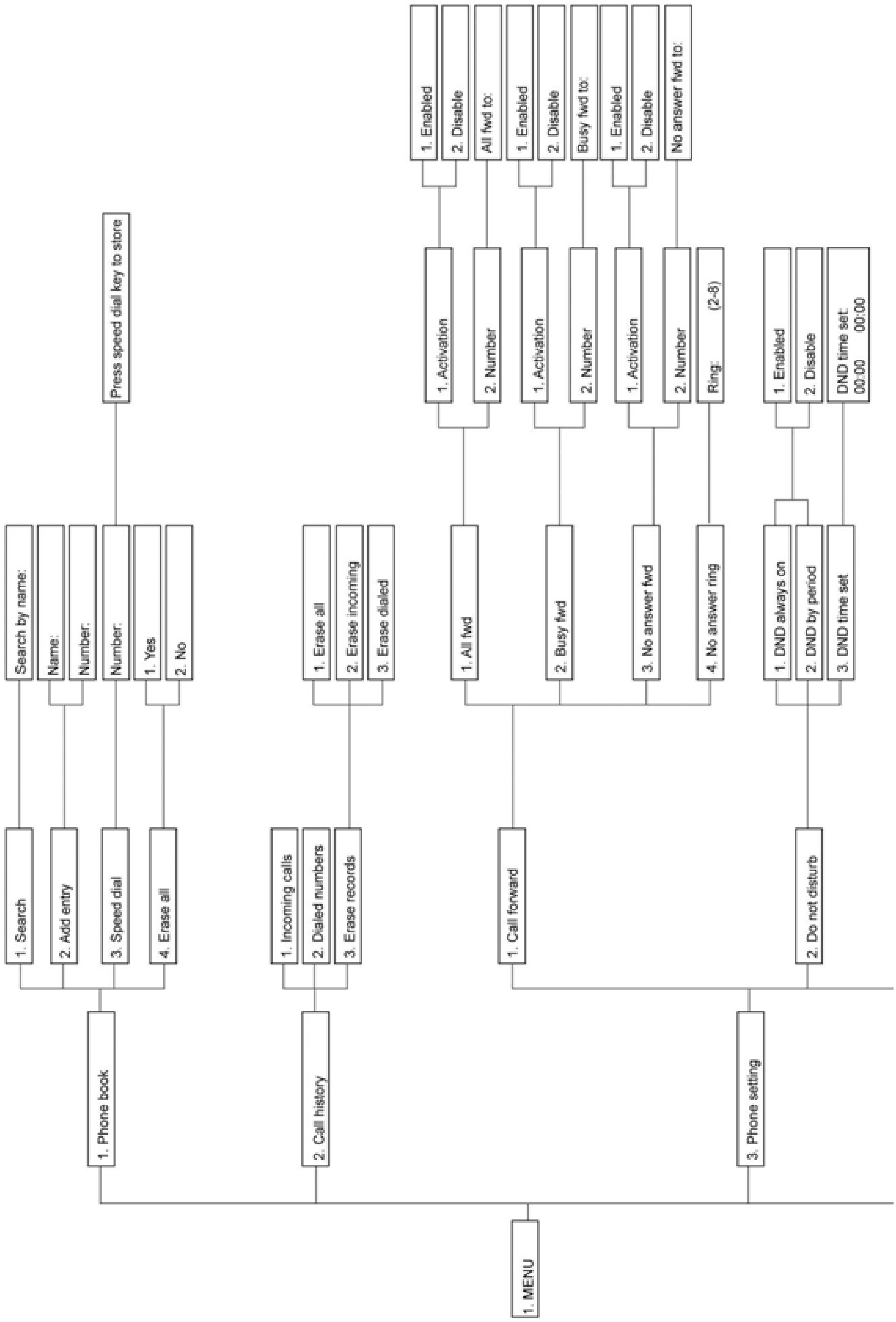
4.4. Others

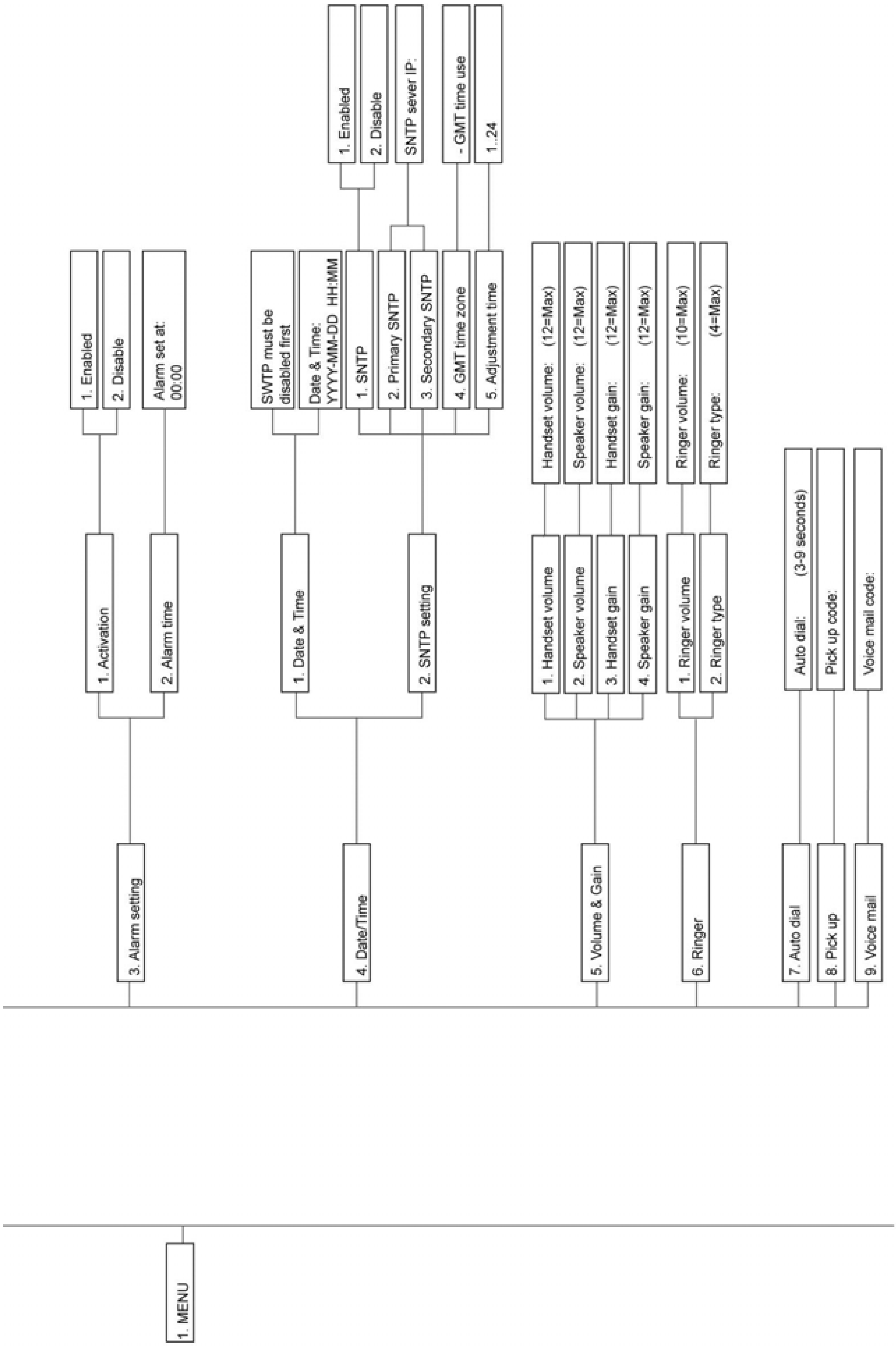
Specification	Description
PC Port	1xRJ45 10/100 Base-T Ethernet, line auto-sensing/switching.
WAN Port	1xRJ45 10/100 Base-T Ethernet, line auto-sensing/switching.
LCD display	2 rows each of 16 characters
Universal Switching Power Adaptor	Input: 100-240V AC
	Output: +6V DC, 1.5A (with ferrite core) provided in the package
Speaker	32 Ohm/0.5 Watt speaker for speakerphone operation

5. Basic Installation

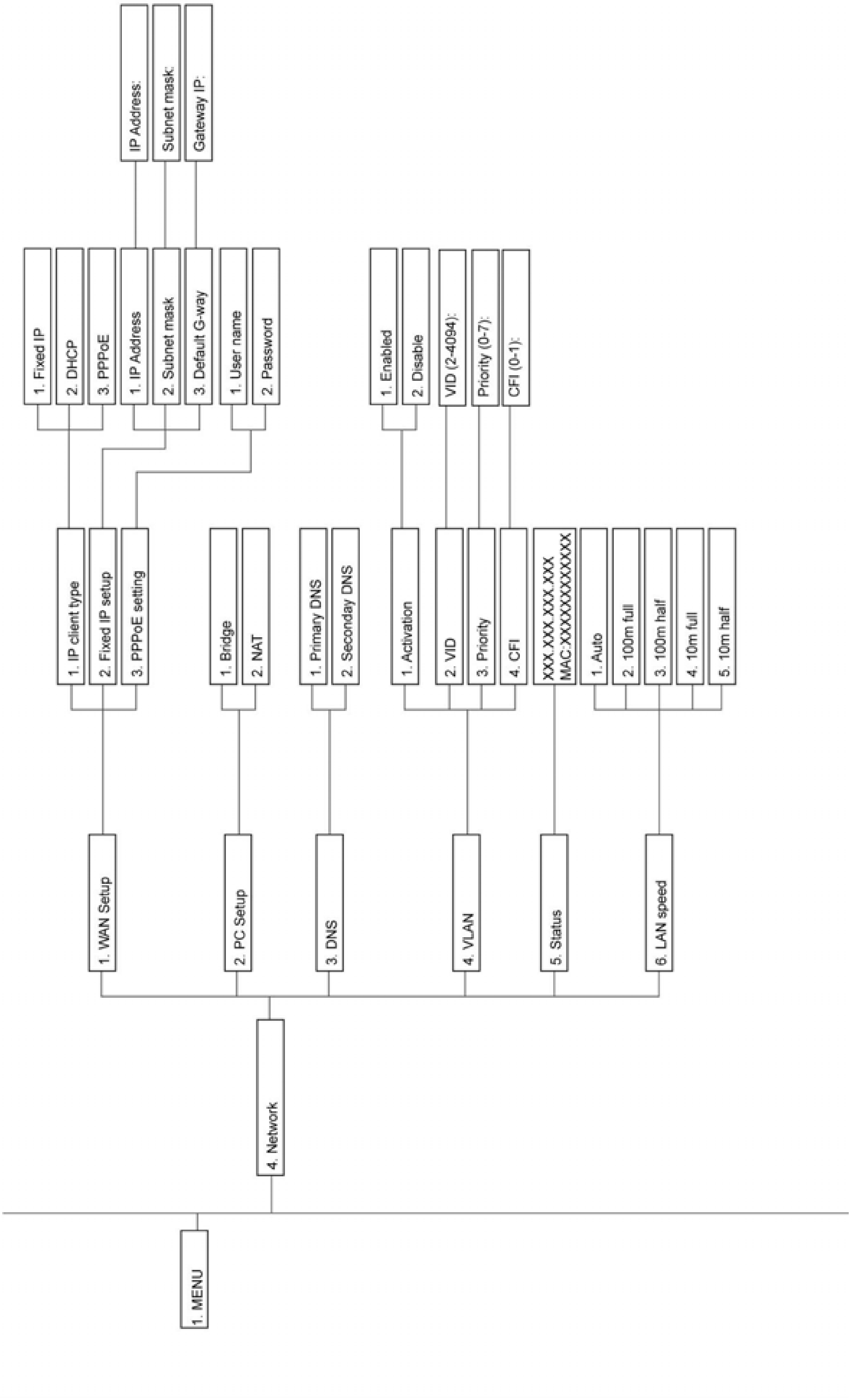
5.1. VoIP Phone Menu

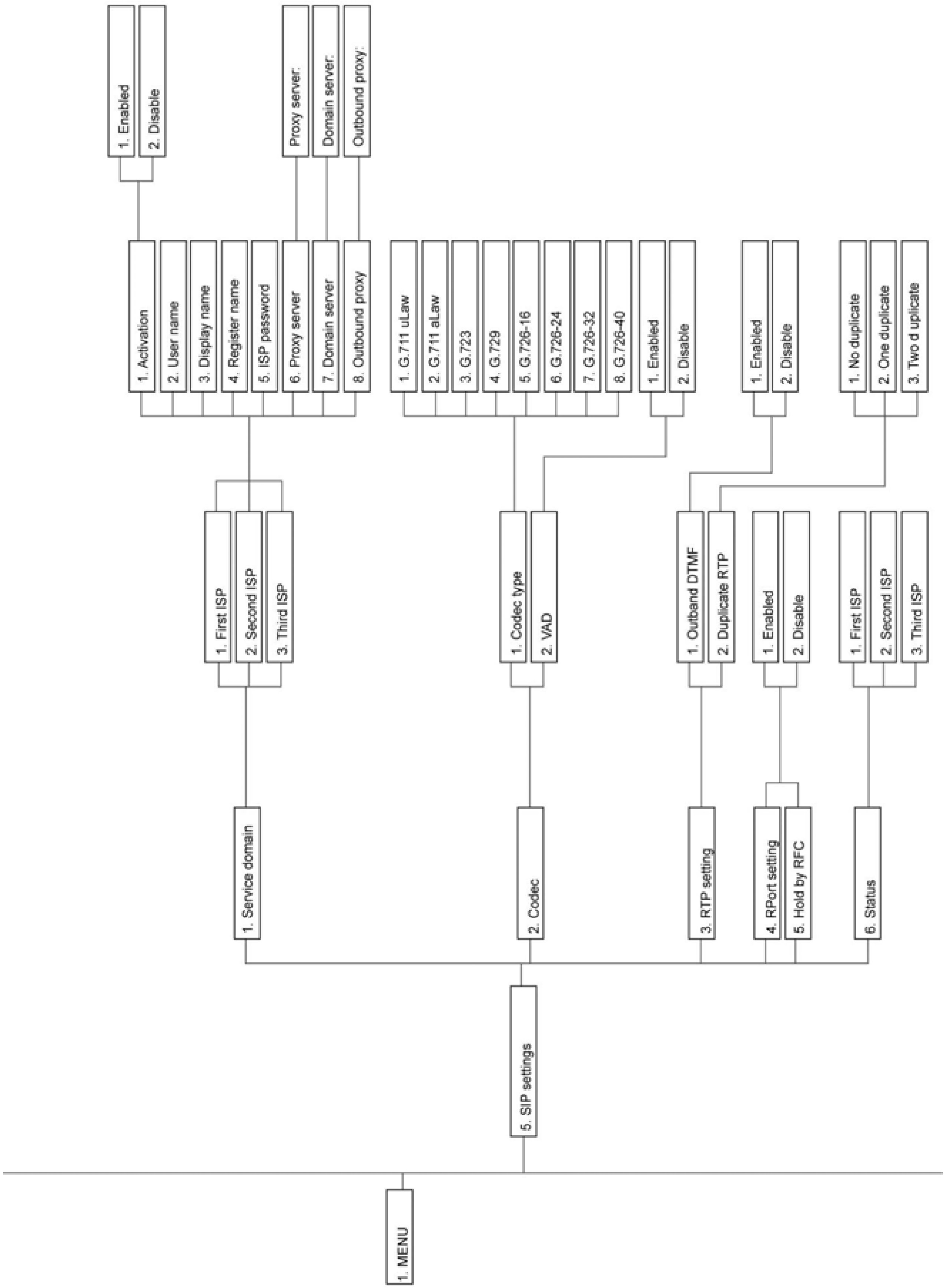
The following is the VoIP phone menu roadmap.



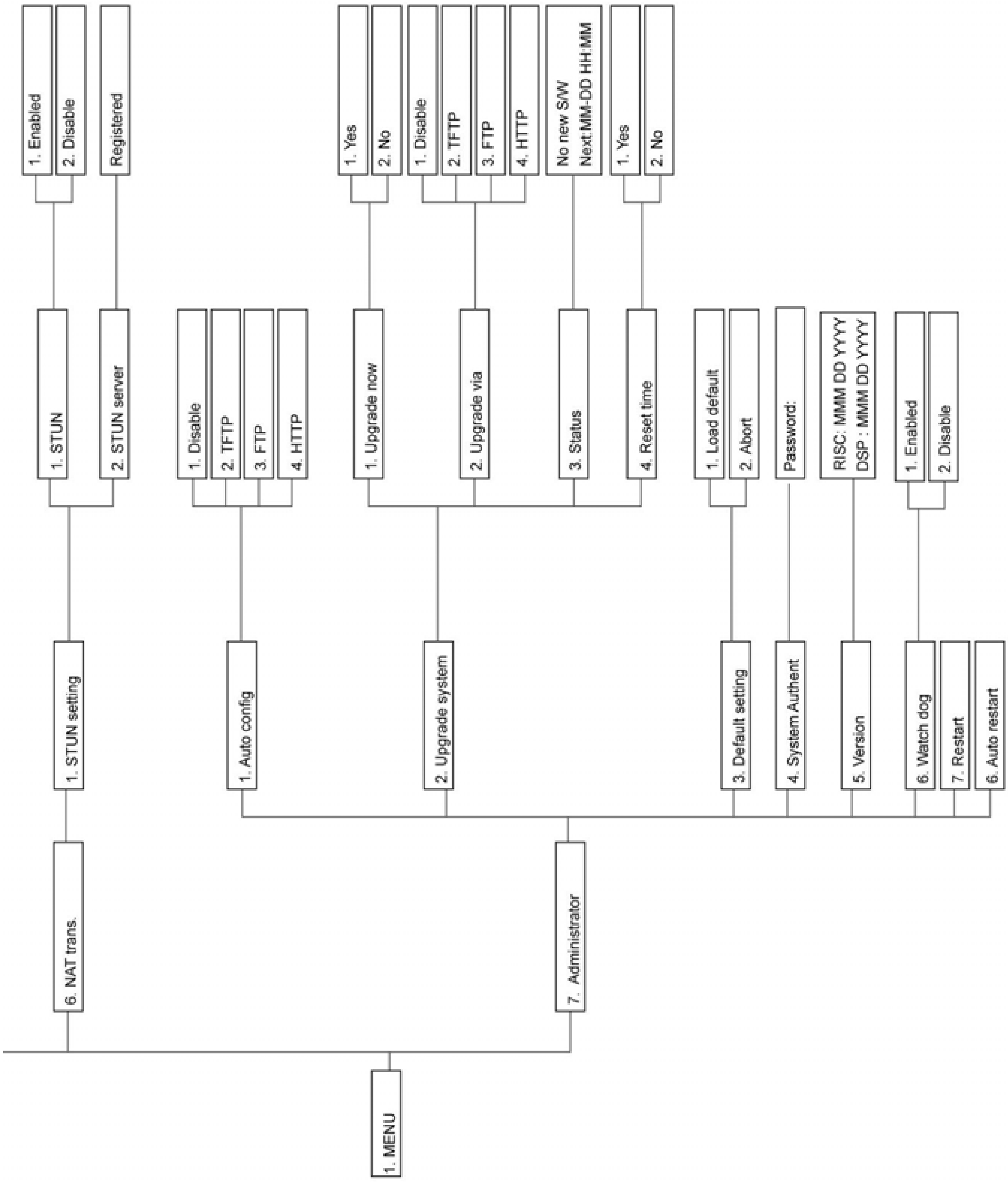


1. MENU
3. Alarm setting
1. Activation
2. Alarm time
1. Enabled
2. Disable
Alarm set at:
00:00
4. Date/Time
1. Date & Time
2. SNTP setting
SWTP must be disabled first
Date & Time:
YYYY-MM-DD HH:MM
1. SNTP
2. Primary SNTP
3. Secondary SNTP
4. GMT time zone
5. Adjustment time
1. Enabled
2. Disable
SNTP sever IP:
- GMT time use
1..24
5. Volume & Gain
1. Handset volume
2. Speaker volume
3. Handset gain
4. Speaker gain
Handset volume: (12=Max)
Speaker volume: (12=Max)
Handset gain: (12=Max)
Speaker gain: (12=Max)
6. Ringer
1. Ringer volume
2. Ringer type
Ringer volume: (10=Max)
Ringer type: (4=Max)
7. Auto dial
Auto dial: (3-9 seconds)
8. Pick up
Pick up code:
9. Voice mail
Voice mail code:


1. MENU
4. Network
1. WAN Setup
2. PC Setup
3. DNS
4. VLAN
5. Status
6. LAN speed
1. IP client type
2. Fixed IP setup
3. PPPoE setting
1. Bridge
2. NAT
1. Primary DNS
2. Seconday DNS
1. Activation
2. VID
3. Priority
4. CFI
XXX.XXX.XXX.XXX
MAC:XXXXXXXXXXXX
1. Auto
2. 100m full
3. 100m half
4. 10m full
5. 10m half
1. Fixed IP
2. DHCP
3. PPPoE
1. IP Address
2. Subnet mask
3. Default G-way
1. User name
2. Password
1. Enabled
2. Disable
VID (2-4094):
Priority (0-7):
CFI (0-1):
IP Address:
Subnet mask:
Gateway IP:



1. MENU
5. SIP settings
1. Service domain
1. First ISP
2. Second ISP
3. Third ISP
1. Activation
1. Enabled
2. Disable
2. User name
3. Display name
4. Register name
5. ISP password
6. Proxy server
Proxy server:
7. Domain server
Domain server:
8. Outbound proxy
Outbound proxy:
2. Codec
1. Codec type
1. G.711 uLaw
2. G.711 aLaw
3. G.723
4. G.729
5. G.726-16
6. G.726-24
7. G.726-32
8. G.726-40
2. VAD
1. Enabled
2. Disable
3. RTP setting
1. Outband DTMF
1. Enabled
2. Disable
2. Duplicate RTP
1. No duplicate
2. One duplicate
3. Two d uplicate
4. RPort setting
5. Hold by RFC
1. Enabled
2. Disable
6. Status
1. First ISP
2. Second ISP
3. Third ISP


1. MENU
6. NAT trans.
1. STUN setting
1. STUN
1. Enabled
2. Disable
2. STUN server
Registered
7. Administrator
1. Auto config
1. Disable
2. TFTP
3. FTP
4. HTTP
2. Upgrade system
1. Upgrade now
1. Yes
2. No
2. Upgrade via
1. Disable
2. TFTP
3. FTP
4. HTTP
3. Status
No new S/W
Next:MM-DD HH:MM
4. Reset time
1. Yes
2. No
3. Default setting
1. Load default
2. Abort
4. System Authent
Password:
5. Version
RISC: MMM DD YYYY
DSP : MMM DD YYYY
6. Watch dog
1. Enabled
2. Disable
7. Restart
6. Auto restart

5.2. Installation Procedure

Step 1：

Please take out your VoIP Phone and its handset. Then connect the handset and handset cord to VoIP Phone by plugging the left jack of phone. If you use RJ-45 broadband connection, please connect your network cable from your broadband modem to WAN port of your VoIP Phone, such as NAT, HUB, ADSL and CABLE.

Step 2：

If you would like to have PC online at the same time, please connect PC port to your PC.

Step 3：

Please plug in your power adaptor to your VoIP Phone and power source. LCD of your VoIP Phone will display "Loading Program!……." . Wait approximately 6 seconds the unit is ready to configure the phone.

6. Configuration by Keypad

The VoIP Phone can be configured by keypad easily. Almost all the configurations can be done through the keypad and LCD screen display on the phone set in a few seconds. In order to make a VoIP call, please do the configurations through keypad as described in the following sections.

Notice:

When need to input an English character in any menu item, please press that key button quickly to switch between different characters to set the correct one.

6.1. Network Configurations

The first thing in using the VoIP Phone is to set the network configuration to let the VoIP Phone connect to Internet. Depends on your network environment and VoIP phone, please use the proper method to configure the VoIP Phone to connect Internet.

6.1.1. Dynamic IP Method (DHCP)

Most of the network environment at office or at hotel room or at home is under a NAT (Network address translation) IP sharing device/router device. Under this environment, the easiest way to connect to Internet is using the DHCP (Dynamic Host Configuration Protocol) method. The VoIP Phone when configured using the following method, it will get the IP parameters dynamically, and connect to Internet automatically.

Please press "ENTER" key to set Dynamic IP when using DHCP method. Most of the cable modem connections also use this DHCP method.

6.1.2. PPPoE Method

Most of the broadband network environment provided by ISP (Internet service provider) is the ADSL (Asymmetric Digital Subscriber Line) connection, under this environment, the VoIP Phone can directly connect to the ADSL modem by setting PPPoE account (user name and password) provided by ADSL service provider, please follow below.

Please press "ENTER" key to set PPPoE (Point-to-Point Protocol over Ethernet) when using PPPoE method. And then key in the account from the ISP vendor.

Username – please input the user name of the account given by ADSL ISP.
Password – please input the password of the account given by ADSL ISP.

6.1.3. Fixed IP Method

For the other network environment, the users will need to set the static IP provided by ISP or from the IT administrator at office.

Under the "Fixed IP" setting submenu, please key in the IP host, network mask Gateway IP and DNS Server settings provided by your ISP or private IP address.

Once all the network settings were completed, please restart your VoIP Phone. Then you are able to check whether your Internet connection is working properly or not by going to network status.

6.2. Registration to Proxy Server

After the network environment were set and connected to Internet, you can register the VoIP Phone to the SIP Proxy server by the account from your VoIP vendor/operator. Please choose one of the following methods to register to a gatekeeper or proxy server.

6.3. "Forward" Configurations

The VoIP Phone supports three different kinds of call forward functions.

6.3.1. ALL Forward

Under "Forward mode" submenu, users can forward all the calls to designated number immediately by enable the all forward activation. Then, input the designated number to be forwarded to. For example, when A calls B, and B's phone has enable all forward with number 555-5555, then, under any circumstance, this phone call from A will be forwarded to number 555-5555.

When the "ALL forward" is enabled, the "busy forward" and "no answer forward" cannot be enabled at the same time.

6.3.2. Busy Forward

Under "Forward mode" submenu, users can forward the calls to designated number when the phone is busy (active) by enable the busy forward activation. Then, input the designated number to be forwarded to. For example, when A calls B, and B's phone is busy and enabled the Busy forward number with number 555-5555, then, this phone call from A will be forwarded to number 555-5555.

6.3.3. No Answer Forward

Under "Forward mode" submenu, users can forward the calls to designated number when no one answered during the specific period by enable the no answer forward activation. Then, input the designated number to be forwarded to. For example, when A calls B, and B's phone is not answered and enable the no answer forward with number 555-5555 and no-answer time equals 10, then, this phone call from A will be forwarded to number 555-5555 after ringing for 2-8 RMB no answered.

7. Call Functions

7.1. Making Calls

To make a call, users can pick up the handset, dial the party's number that wish to call. An alternative way to make a call is to press the number, then pick up the handset or press HANDSFREE/HEADSET key at the down left corner to make the call.

7.2. Receiving Calls

When a call is incoming and the VoIP Phone is ringing, to receive this call, just pick up the handset or press HANDSFREE/HEADSET key to answer.

7.3. Check call history (Incoming calls/ Dialed numbers)

The call history can be displayed on the LCD screen by pressing the submenu and then "▲" key or "▼" key when the VoIP Phone is in idle state.

There are two kinds of call history:
The call history will show in <Call Number> MM-DD HH:MM (V/M) format. The last digit 'V' represents the answered incoming call while 'M' represents the unanswered incoming calls.

"Incoming calls" is the record of numbers and time of last 64 incoming calls when answered or unanswered.

"Dialed numbers" is the record of numbers and time of last 64 successful dialed numbers.

When viewing the number of any call history, you can press "MENU" key to do one of the following two actions:

1. Dial numbers – to call out to that number by directly pressing the "HANDSFREE/HEADSET" key again.
2. Cancel – to delete the item in the call history.

7.4. Auto Redial

Users are able to call out with last dialed number (redial) by pressing "REDIAL" key after off-hooked or in Handsfree/Headset mode.

7.5. Call Forward

You can set the call forward functions through keypad or web configurations.

7.6. Phone Book and Speed Dial

By simply pressing Phone Book button, users are able to restore totally 140 phone numbers with this function. When users press Phone Book button, two options shown in LCD display are Phone Book and Speed Dial.

After selecting Phone Book option, users will user Search, Add entry, Erase all function to manage the phone book. To add person's name and numbers, users may select Add function. To edit or delete phone book information, users may simply select edit or delete function in this category. Users may delete all information in the entire phone book by selecting erase All.

To set up with speed dial function, users should select Speed Dial and M1- M10 will show in the menu. By selecting each slot, two options are Current Info and Change Setting. Current Info displays the phone number for this slot. By selecting Change Setting, users are able to change the person of the phone book list for this individual slot.

After the speed dial items M1-M10 were set, user can make speed dial call by directly pressing the M1-M10 key when the phone is off-hooked.

8. Setup the VoIP Phone by Web Browser

The default of the 9339 VoIP Phone was: NAT is enabled, WAN port is in DHCP Client Mode and PC Port is in DHCP Server Mode. You can connect the PC port, and then you will get an IP Address from the VoIP Phone.

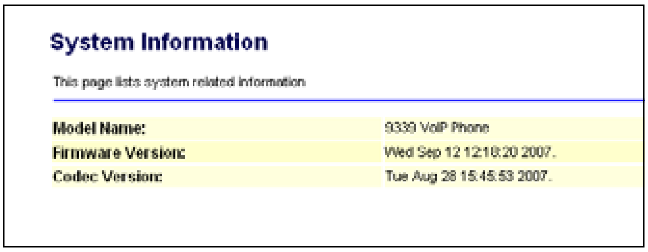
The VoIP Phone provides a built-in web server. You can use Web browser to configure the VoIP Phone. First please input the IP address http://192.168.123.1:9999 in the Web page. Please remember to add the port number ":9999".

8.1. Login

Step 1: Input the username and password into the blank field. The default setting is:

- ✧ For Administrator, the username is: setup; and the password is: test. If you use the account login, you can configure all the setting.
- ✧ For normal user, the username is: user; and the password is: 9339. If you use the account login, but you cannot configure the SIP setting.

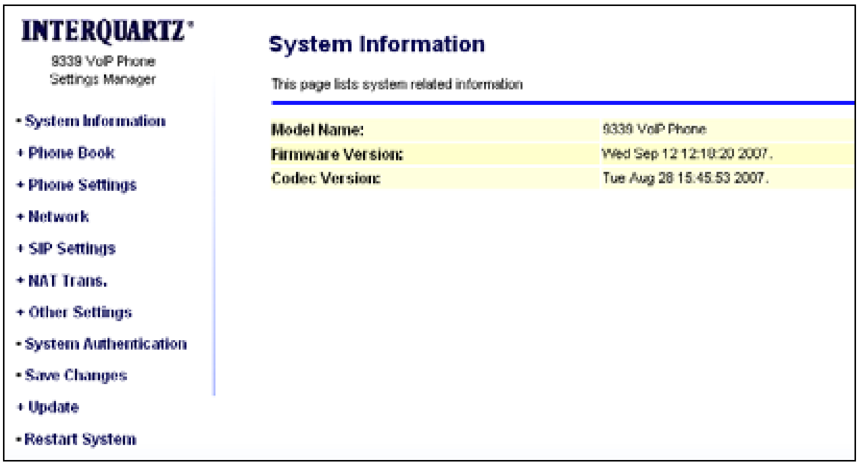
Step 2: Click the "Login" button will move into the VOIP PHONE web based management information page.



Step 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 settings, please click the Restart function in the left side, and click the Restart button in that page. After the system restart, all the settings were saved and becomes effective.

8.2. System Information

When you login the web page, you can see the VOIP PHONE current system information like Model Name, Firmware version and Codec Version. Besides, you can see the function lists in the left side and navigate for further set up.



8.3. Phone Book

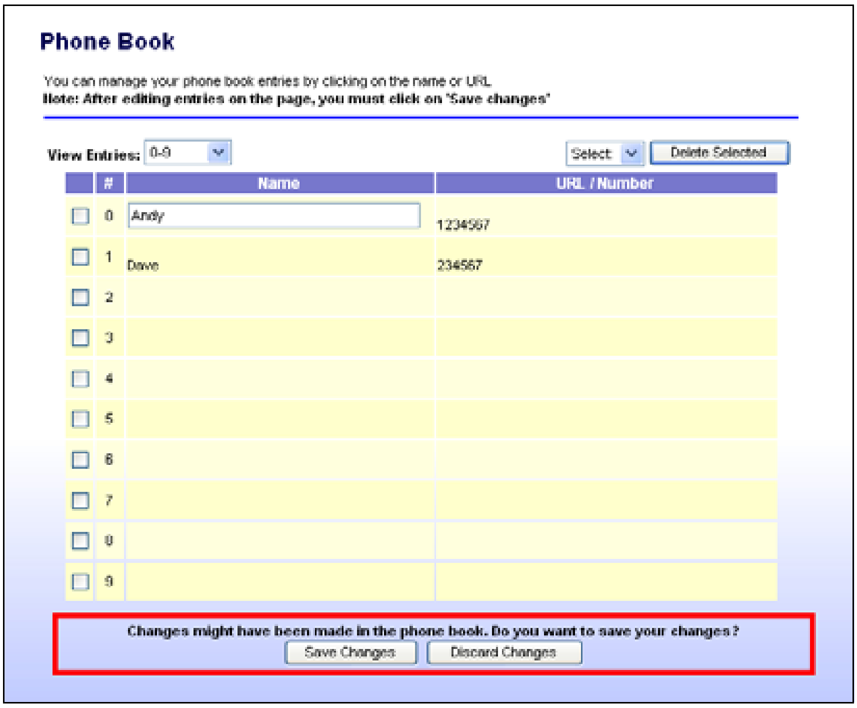
Phone Book contains Phone Book and Speed Dial Settings. The Phone Book can store 140 phone numbers and 10 Speed Dial phone numbers.

8.3.1. Phone Book

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

If you add a phone number into the phone book, input the position, the name and the URL/Number. When you finished, click the "Save Changes" button.

If you want to delete a phone number, you can select the phone number and click "Delete Selected" button. If you want to delete all phone numbers, you can click "Delete All" button.



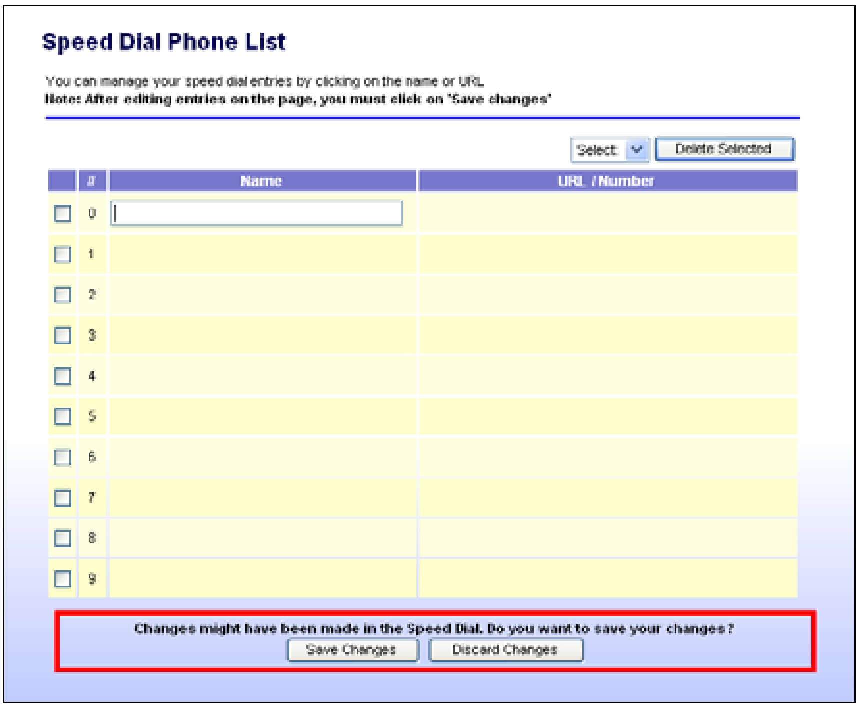
8.3.2.Speed Dial

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

If you need to add a phone number into the Speed Dial list, input the name and ULR/number to the designated position. When you finished, click the "Save Changes" button.

If you want to delete a phone number, you can select the phone number and click "Delete Selected" button. If you want to delete all phone numbers, you can click "Delete All" button.

If you want to use Speed Dial you just dial the speed dial number (from 0~9) then press "#".



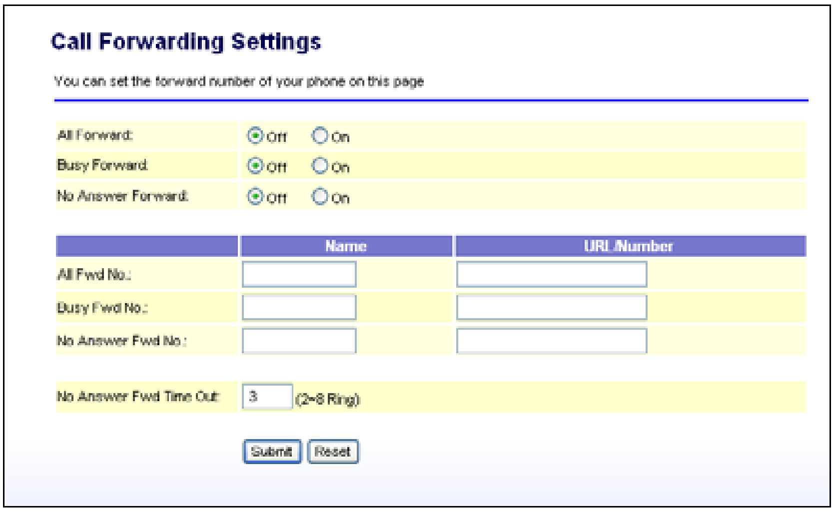
8.4. Phone Settings

In Phone Setting provides Call Forwarding Settings, SNTP Settings, Volume Settings, Ringers Settings, DND Settings, Dial Plan Settings, Call Waiting Settings, Soft-Key Settings, Hotline Settings and Alarm Settings.

8.4.1.Call forwarding

You can setup the phone number you want to forward in this page. There are three types of Forward modes; they are All Forward, Busy Forward and No Answer Forward. You can activate different modes by clicking the ON icon. When you finished the settings, please click the Submit button.

If there is nothing needs to change, please click 'Save Changes' in the left side. The change you made will save into the system and the system will restart automatically.



8.4.1.1.All Forward

All incoming calls will forward to the number you choose. You can input the name and the URL/Number in the 'All Fwd No' field. If you select this function, then all the incoming calls will direct forward to the number you choose.

8.4.1.2.Busy Forward

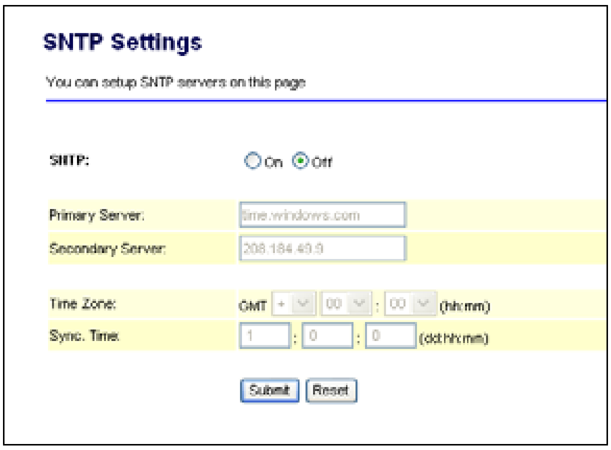
The new incoming calls will forward to the number you choose if the phone were in use. You can input the name and the URL/Number in the 'Busy Fwd No' field.

8.4.1.3.No Answer Forward

The incoming calls will forward to the number you choose if you cannot answer the phone. You can input the name and the URL/Number in the 'No Answer Fwd No' field. Also you have to set the numbers of ring of 'No Answer Fwd Time Out', then the incoming calls will start to forwarded the calls to the choose number as defined.

8.4.2.SNTP Settings

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 settings, please click the Submit button. If there is nothing needs to change, please click 'Save Changes' in the left side. The change you made will save into the system and the system will restart automatically.



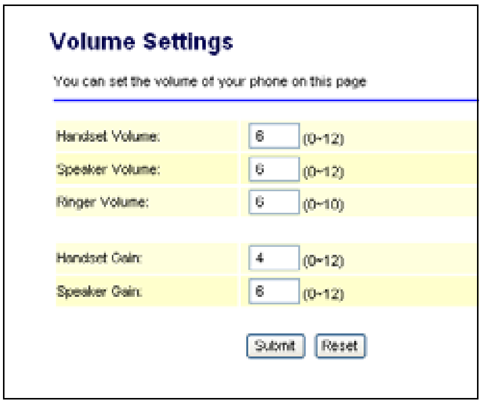
8.4.3.Volume Settings

You can setup the Handset Volume, Speaker Volume, Ringer Volume, Handset Gain and Speaker Gain.

- ✧ Handset Volume is to set the volume you hear from the handset.
- ✧ Speaker Volume is to set the volume you hear from the speakerphone.
- ✧ Ringer Volume is to set the ringer volume.
- ✧ Handset Gain is to set the volume send out from the handset.
- ✧ Speaker Gain is to set the volume send out from the microphone.

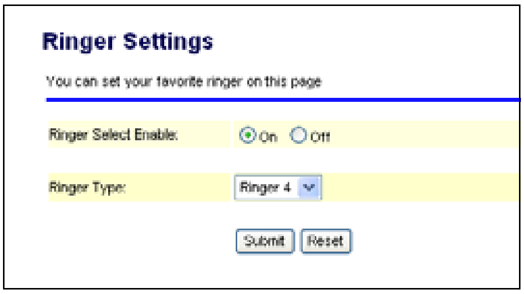
When you finished the settings, please click the Submit button.

If there is nothing needs to change, please click ‘Save Changes’ in the left side. The change you made will save into the system and the system will restart automatically.



8.4.4.Ringer Settings function

You can select the ringer melody for the incoming calls. When you finished the settings, please click the Submit button. If there is nothing needs to change, please click ‘Save Changes’ in the left side. The change you made will save into the system and the system will restart automatically.



8.4.5.DND Settings

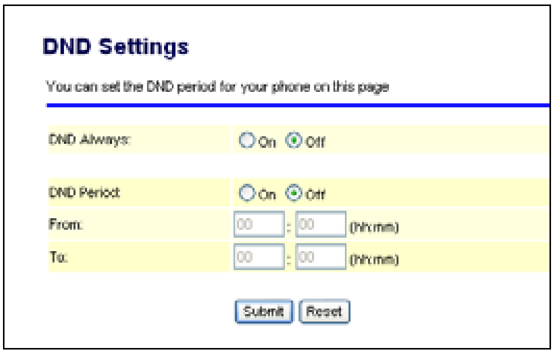
You can setup the DND Settings to keep the phone silence. You can choose Always DND or DND a period.

DND Always: All incoming calls 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.

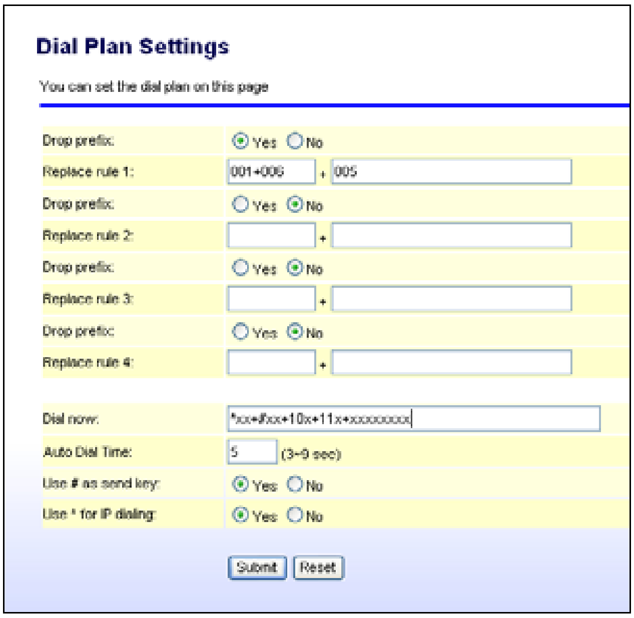
When you finished the settings, please click the Submit button.

If there is nothing needs to change, please click 'Save Changes' in the left side. The change you made will save into the system and the system will restart automatically.



8.4.6.Dial Plan Settings

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 explain:

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 now: When match with pattern then send dial number out but if first digit is '0' then dial plan will be ignored.

Example:

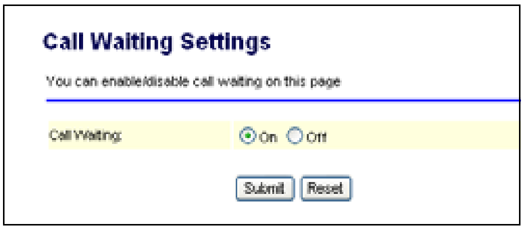
*xx	If matched with one of *00,*01....*99 then will send number out
#xx	If matched with one of #00,#01....#99 then will send number out
10x	If matched with one of 100,101....109 then will send number out
11x	If matched with one of 110,111....119 then will send number out
Xxxxxxxx	If dial with 8 digits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

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

If there is nothing needs to change, please click 'Save Changes' in the left side. The change you made will save into the system and the system will restart automatically.

8.4.7.Call Waiting Settings

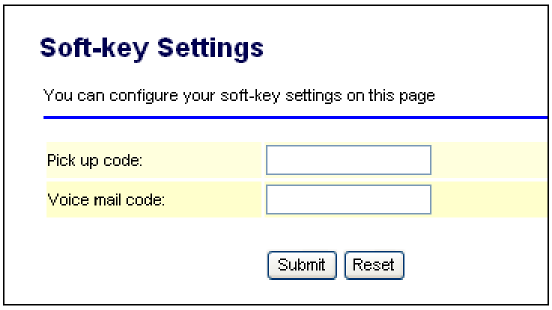
If user doesn't want to be informed by the new incoming calls, user can set the function off. When you finished the settings, please click the Submit button. If there is nothing needs to change, please click 'Save Changes' in the left side. The change you made will save into the system and the system will restart automatically.



8.4.8.Soft-key Settings

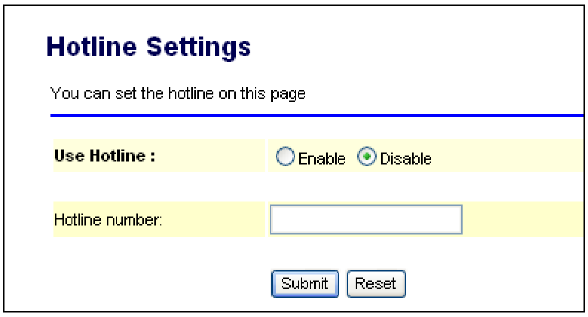
User can define the Pick Up and Voice mail function as a forty-key. This function needs to work with Proxy Server and the device also needs to have the Pick Up key and voice mail key. For example, if the Server define the pick up key is *7, then user have to input the *7 in Pick up key field. When user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

When you finished the settings, please click the Submit button. If there is nothing needs to change, please click 'Save Changes' in the left side. The change you made will save into the system and the system will restart automatically.



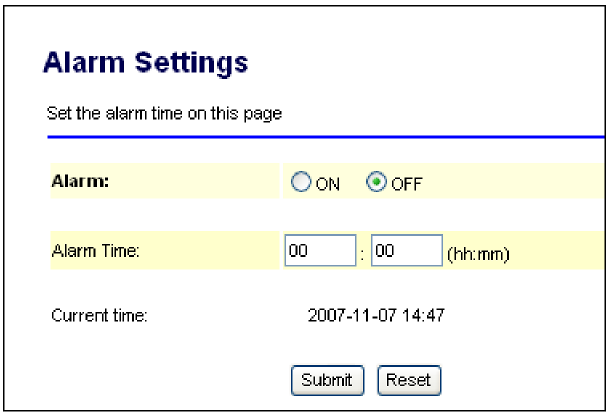
8.4.9.Hotline Settings

Hotline Settings allows dialing to a preset number automatically as long as pick up the phone.



8.4.10.Alarm Settings

Alarm Settings provides alarm function.

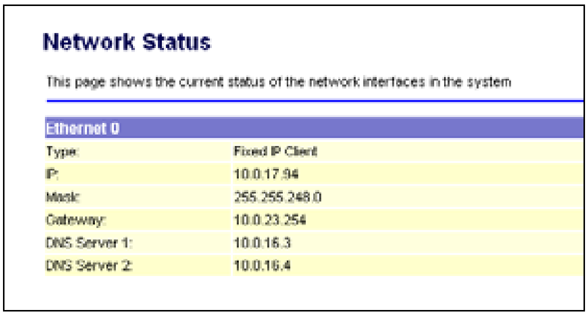


8.5. Network

In Network, you can check the Network status, configure the WAN Settings, PC Settings, DDNS settings, VLAN Settings, DMZ Settings, Virtual Server and PPTP Settings.

8.5.1.Status

You can check the current Network status in this page.



8.5.2.WAN Settings

In this page you can configure the VoIP Phone WAN port's settings. 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.

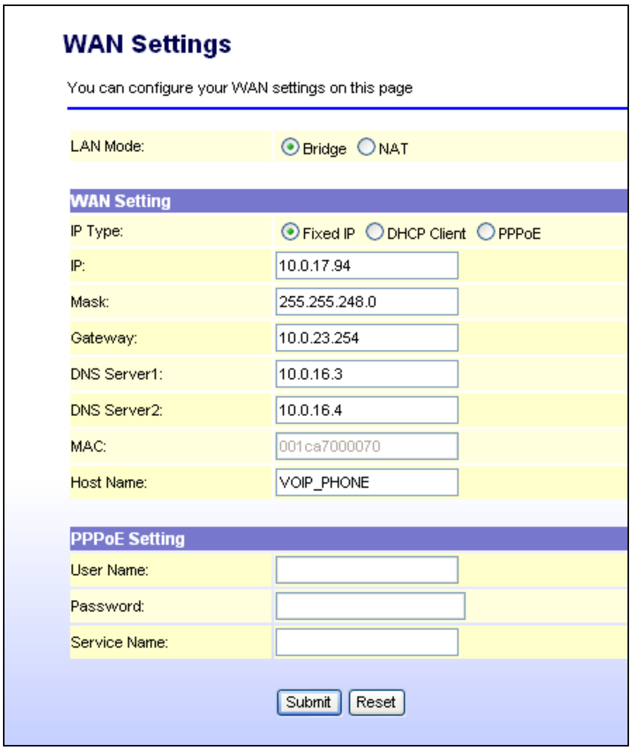
The VoIP Phone's default setting is NAT mode. If you don't need to use the NAT Mode, you can change to Bridge Mode. If you change the setting to Bridge Mode, then the PC setting will not effect and will be the same as WAN port.

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 settings, please click the Submit button. If there is nothing needs to change, please click 'Save Changes' in the left side. The change you made will save into the system and the system will restart automatically.



8.5.3.PC Settings

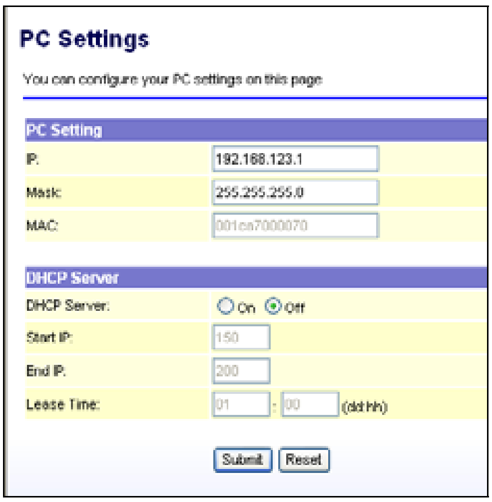
In this page you can configure the VoIP Phone PC port's setting.

The PC 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 address is 200. It is not necessary to change the PC settings.

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

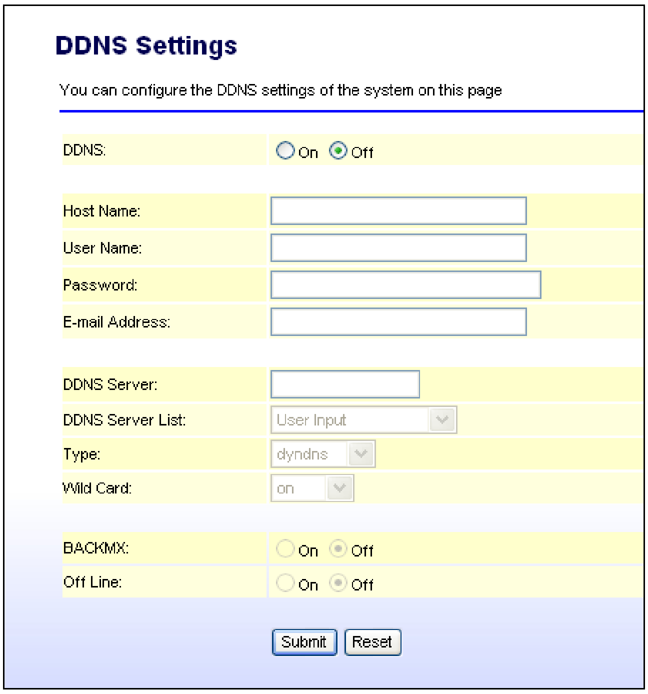
When you finished the settings, please click the Submit button.

If there is nothing needs to change, please click 'Save Changes' in the left side. The change you made will save into the system and the system will restart automatically.



8.5.4.DDNS Settings

You can configure the DDNS settings 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 settings, please click the Submit button. If there is nothing needs to change, please click ‘Save Changes’ in the left side. The change you made will save into the system and the system will restart automatically.



8.5.5.VLAN Settings

You can set the VLAN settings in this page. There are two parts in this page. First one is to set the packets related to the TA, and the second part is if you use the VLAN settings 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 PC 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 (0-7) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be defined by your service provider.

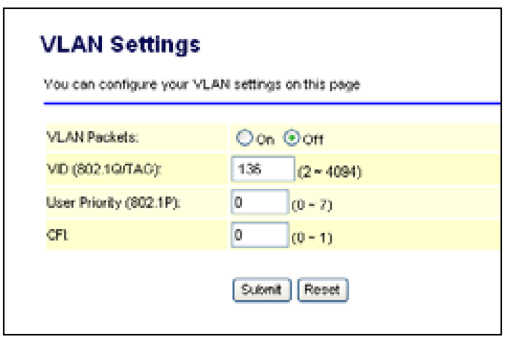
CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reason between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.

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 PC port to the PC.

NAT VLAN Settings

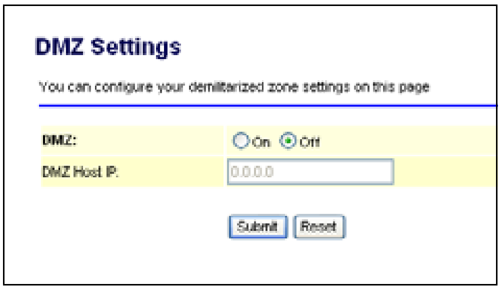
: When you set your device in NAT mode, the TA can help you to filter the wrong incoming packets. You can separate the other device connected behind the TA into 4 VLAN group. You can set different VID for these 4 groups. When the incoming packets go through the TA’s WAN port then the TA will check the 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 needs to change, please click ‘Save Changes’ in the left side. The change you made will save into the system and the system will restart automatically.



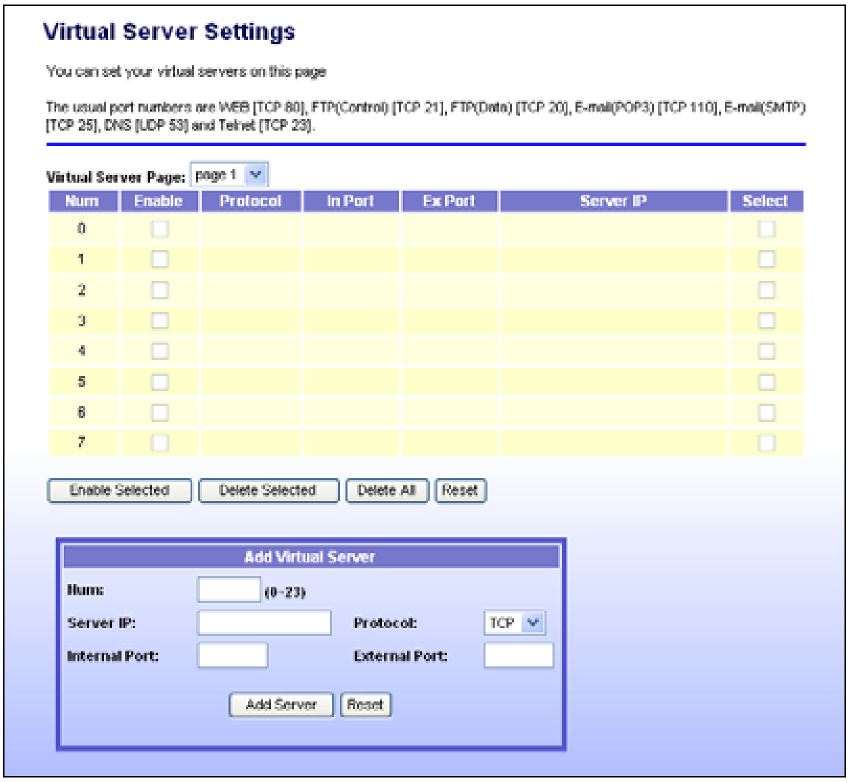
8.5.6.DMZ Settings

DMZ Settings provides DMZ data.



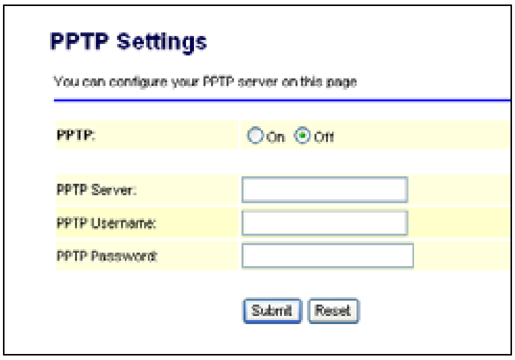
8.5.7.Virtual Server

Virtual Server Settings provides 24 sets of Virtual Server information.



8.5.8.PPTP Settings

PPTP Settings provide PPTP Server information. **Please use LAN to enter PPTP.**



8.6. SIP Settings

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

8.6.1. Service Domain

You need to input the account and the related information 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, and then you can input the following items:

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

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

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

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

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

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

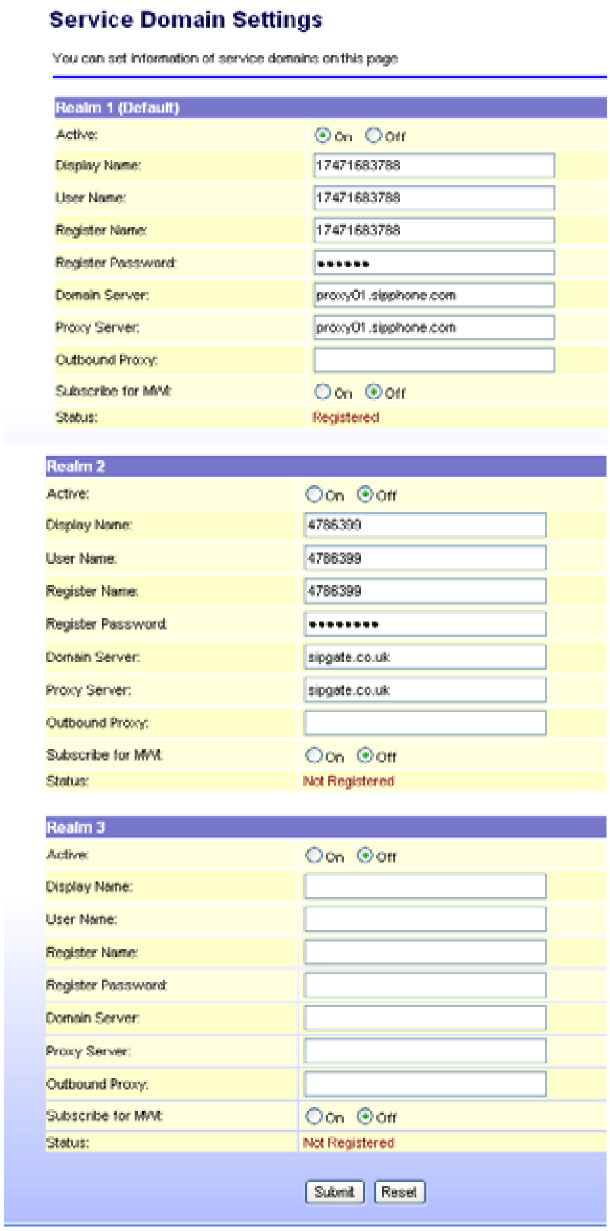
Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

Subscribe of MWI: subscribe for MWI function; your register SIP proxy server must support this function.

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 directly.

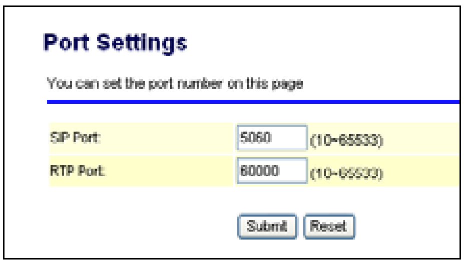
If you have more than one SIP account, you can follow the steps to register to the other ISP.

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



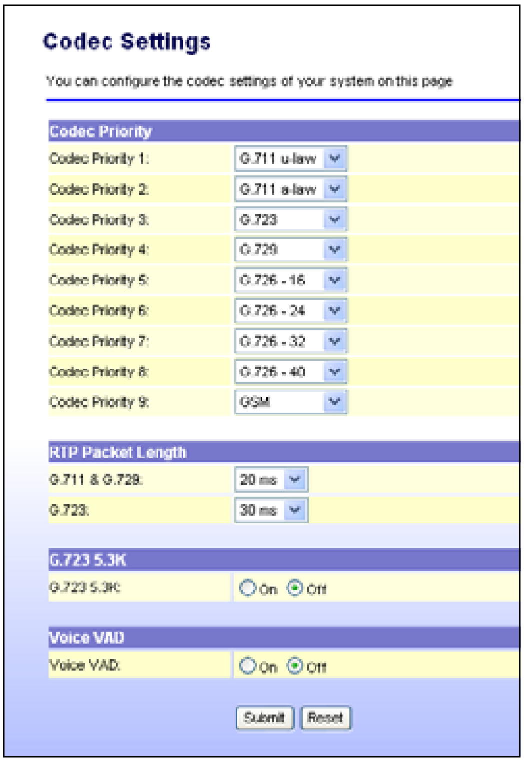
8.6.2.Port Settings

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



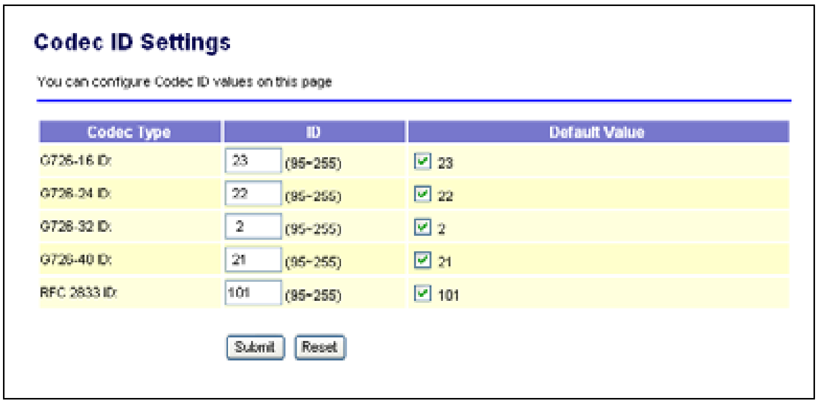
8.6.3.Codec Settings

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 settings, please click the Submit button.



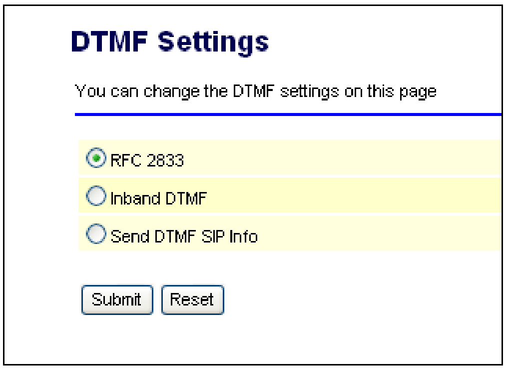
8.6.4.Codec ID Settings

You can set the Codec ID to meet the other device's requirement. When you finished the settings, please click the Submit button.



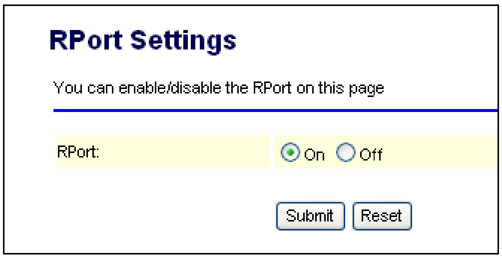
8.6.5.DTMF Settings

You can setup the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change these settings, please follow your ISP information. When you finished the settings, please click the Submit button.



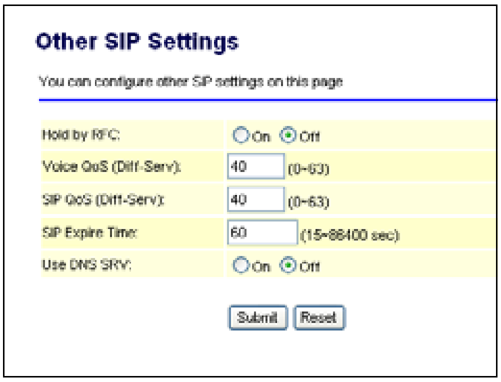
8.6.6.RPort Settings

You can setup the RPort Enable/Disable in this page. To change this settings, please follow your ISP information. When you finished the settings, please click the Submit button.



8.6.7.Other SIP Settings

You can setup the Hold by RFC, Voice/SIP QoS (Diff-Serv) and SIP expire time in this page. To change these settings please following your ISP information. When you finished the settings, 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.

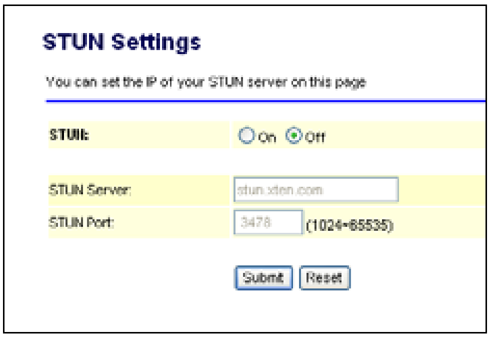


8.7. NAT Trans.

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

8.7.1.STUN Settings

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 settings, please click the Submit button.

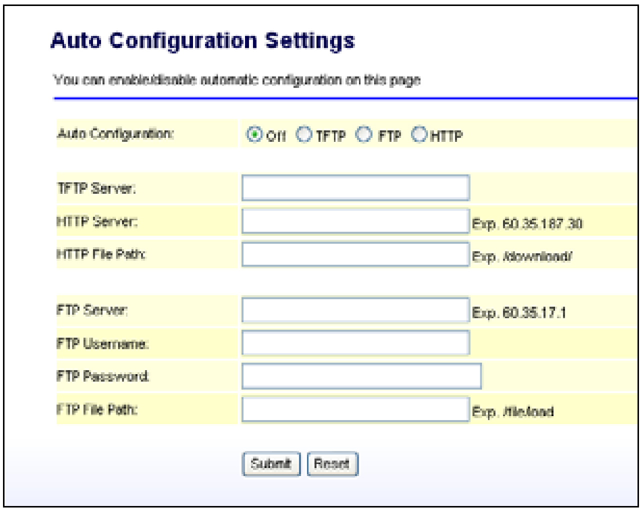


8.8. Other Settings

In Other settings, you can setup Auto Configuration Settings, MAC Clone Settings, Tone Settings and Advanced Settings function. The function can configure your VoIP Phone automatically.

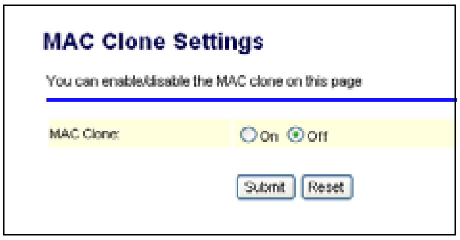
8.8.1.Auto Configuration Settings

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



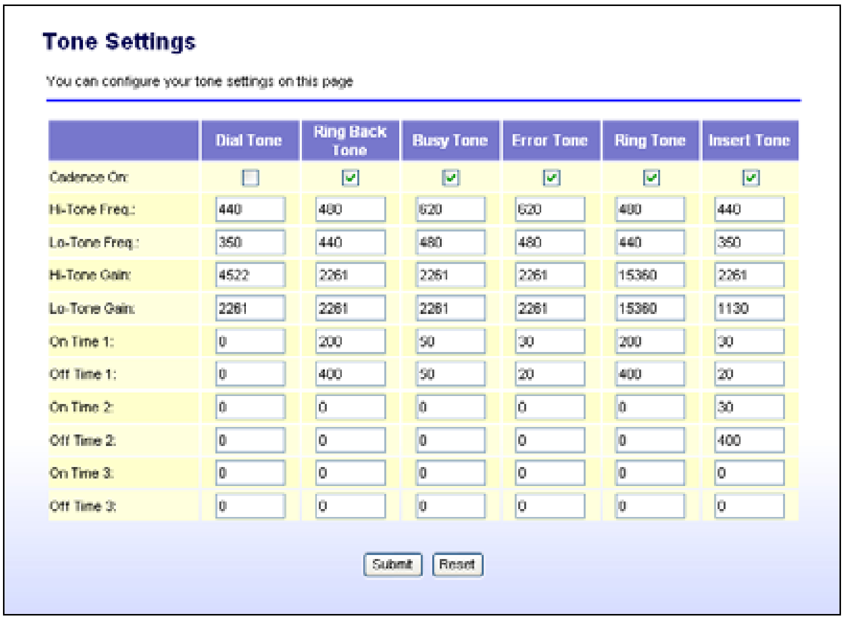
8.8.2.MAC Clone Settings

You can setup the MAC Clone Setting ON/OFF. When you finished the settings, please click the Submit button.



8.8.3.Tone Settings

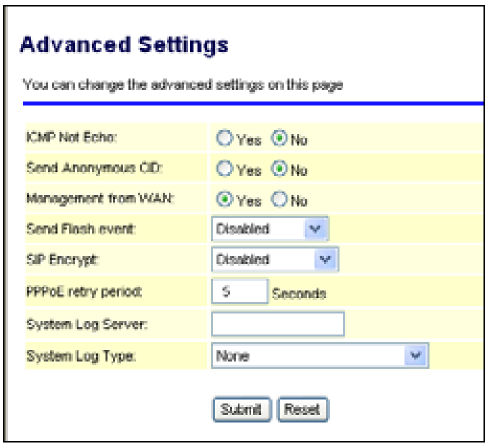
Tones Settings provide Dial Tone, Ring Back Tone, Busy Tone, Error Tone, Ring Tone, Insert Tone information. High Tone and Low Tone are available. When you finished the settings, please click the Submit button.



	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert 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

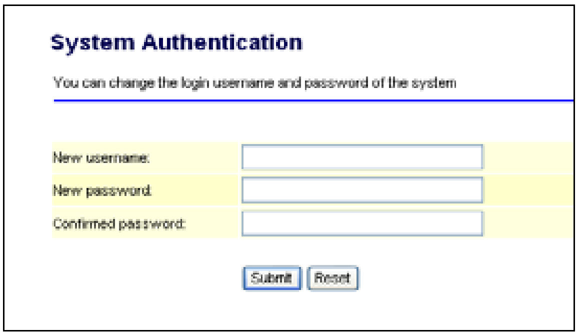
8.8.4.Advanced Settings

Advanced Settings provides ICMP not Echo, Send Anonymous CID, Management from WAN, Send Flash event, SIP Encrypt, PPPoE retry period, System Log Server and System Log Type functions.



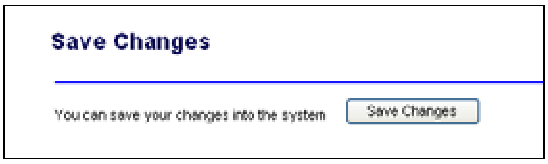
8.9. System Authentication

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



8.10.Save Changes

In Save Change you can save the changes you have done. If you want to use new settings 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.



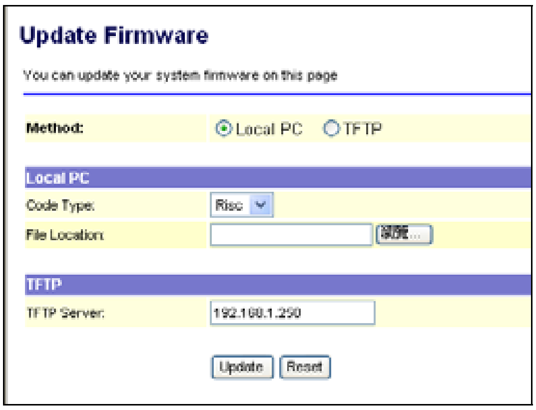
8.11.Update

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.

8.11.1.New Firmware

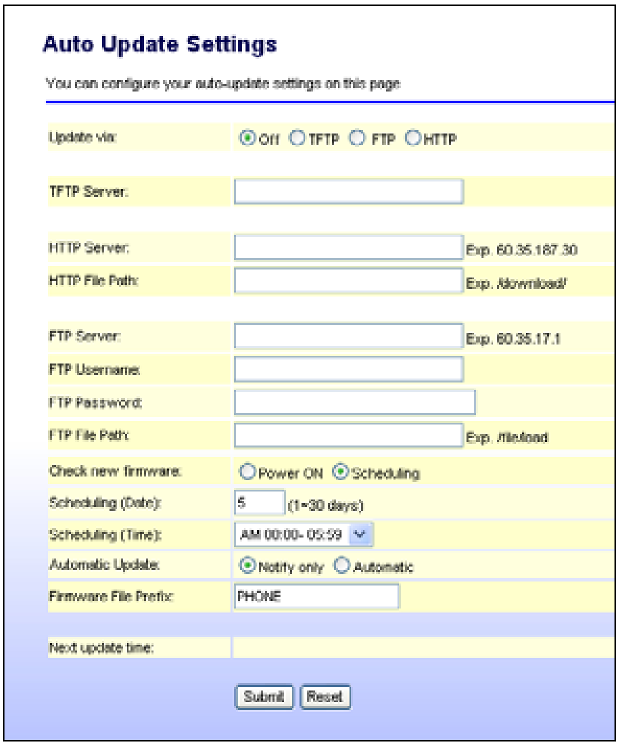
You can update new firmware via HTTP in this page. You can upgrade the firmware by the following steps:

Step1 : Select the firmware code type, Risc or DSP code.
Step 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.
Step 3: Select the correct file you want to download to the VoIP Phone then click the Update button.



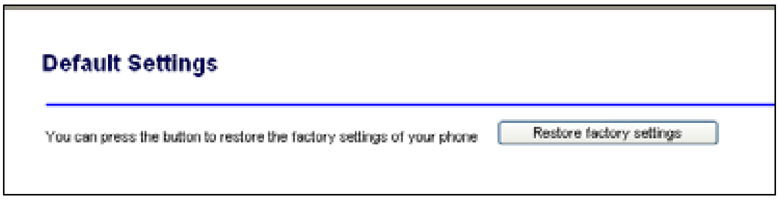
8.11.2.Auto Update Settings

You can auto update new firmware via HTTP in this page. Auto Update Settings page provide RISC (.gz) or DSP (.ds) format to update firmware, but not provide (.rom) format to update.



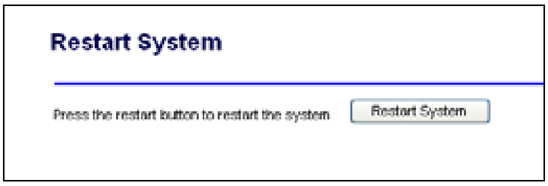
8.11.3. Default Settings

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



8.12.Restart System

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



9. Engineering webpage

Engineer usage webpage list. You have to login the system first then change the webpage to crystal.htm manually (http://ip address:9999/crystal.htm). In this webpage you will see the list about engineer webpage. You can change the webpage to what you want.

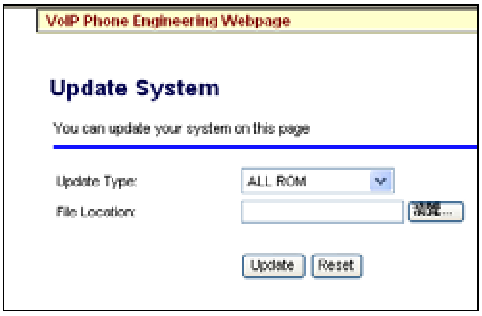
9.1. Update.htm

You have to login the system first then change the webpage to update.htm manually (http://ip address:9999/update.htm).

In this page you can update the system's ROM code, IC Test, Default setting and Logo.

Update ROM code. You can update the ROM code from this function. Please be noted that if you update the wrong file or during the update process the power is off, the system will be crashed.

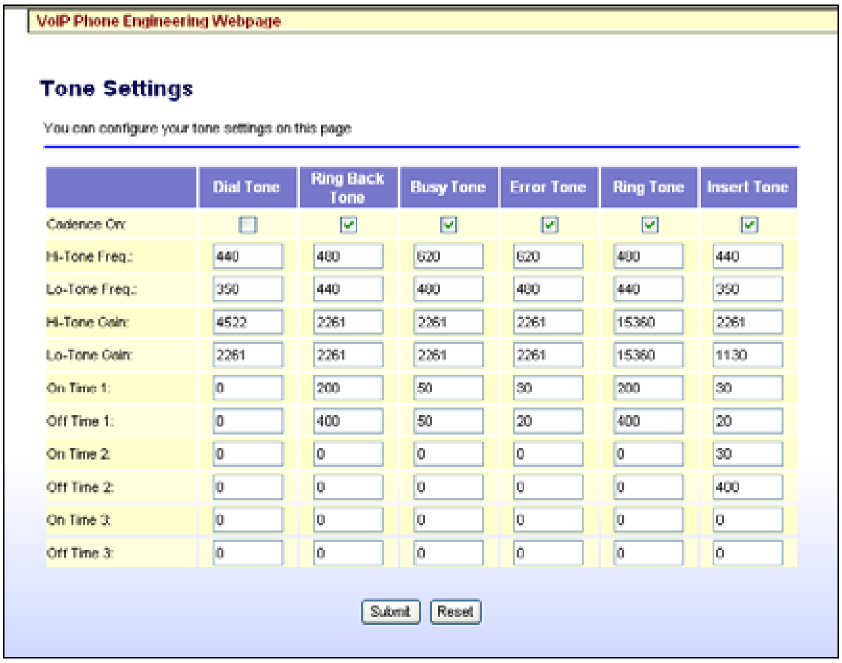
Update Logo. The Logo specification is 220x170 jpeg file.



9.2. Toneset.htm

You have to login the system first then change the webpage totoneset.htm manually (http://ip address:9999/toneset.htm).

In this page you can setup the Tone frequency and cadence to meet the requirement.



VoIP Phone Engineering Webpage

Tone Settings

You can configure your tone settings on this page

	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert Tone
Cadence On:	[ ]	[x]	[x]	[x]	[x]	[x]
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

9.3. Speakerset.htm

You have to login the system first then change the webpage to speakerset.htm manually (http://ip address:9999/speakerset.htm).

In this page you can setup the Speaker function. Default we set the speaker is in half-duplex mode. If you want to set to full-duplex mode, it needs to check if your housing is suitable for this function. Or you have to set it as half-duplex mode.

VoIP Phone Engineering Webpage

Speaker Phone Settings

You can configure your speaker phone settings on this page

(•) Half-Duplex ( ) Full-Duplex

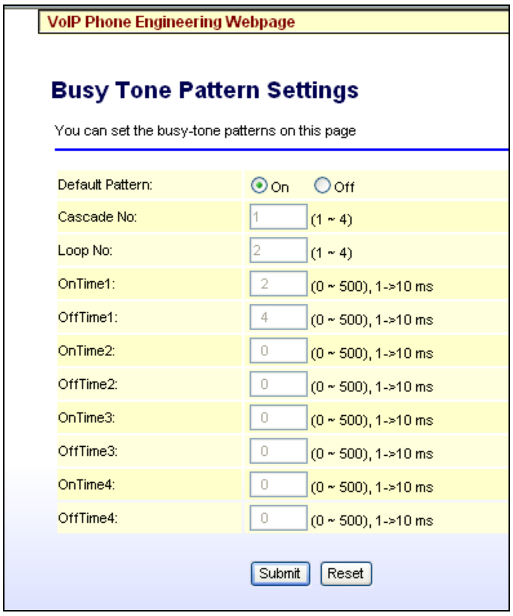
Cut-off Threshold:	0024
Cut-off Time Constant:	4000
Cut-off Hold Time:	0014

Submit Reset

9.4. BusyTonePTset.htm

You have to login the system first then change the webpage to busytoneptset.htm manually (http://ip address:9999/busytoneptsetset.htm).

In this page you can setup the Busy Tone Pattern.



9.5. Factory.htm

You have to login the system first then change the webpage to factory.htm manually (http://ip address:9999/factory.htm).

In this page, you can export the current system settings to a backup file before save in a new setting.

10. Safety and Compliance

WARNING: Changes or modification to this equipment not expressly approved by the party responsible for compliance could void the user authority to operate the equipment.

10.1. FCC Part 15 Rules

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

1. This device may not cause harmful interference, and
2. This device must accept any interference received, including interference that may cause undesired operation.

10.2.Class B Digital Device or Peripheral

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications.

However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

1. Reorient or relocate the receiving antenna.
2. Increase the separation between the equipment and receiver.
3. Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
4. Consult the dealer or an experienced radio/TV technician for help.

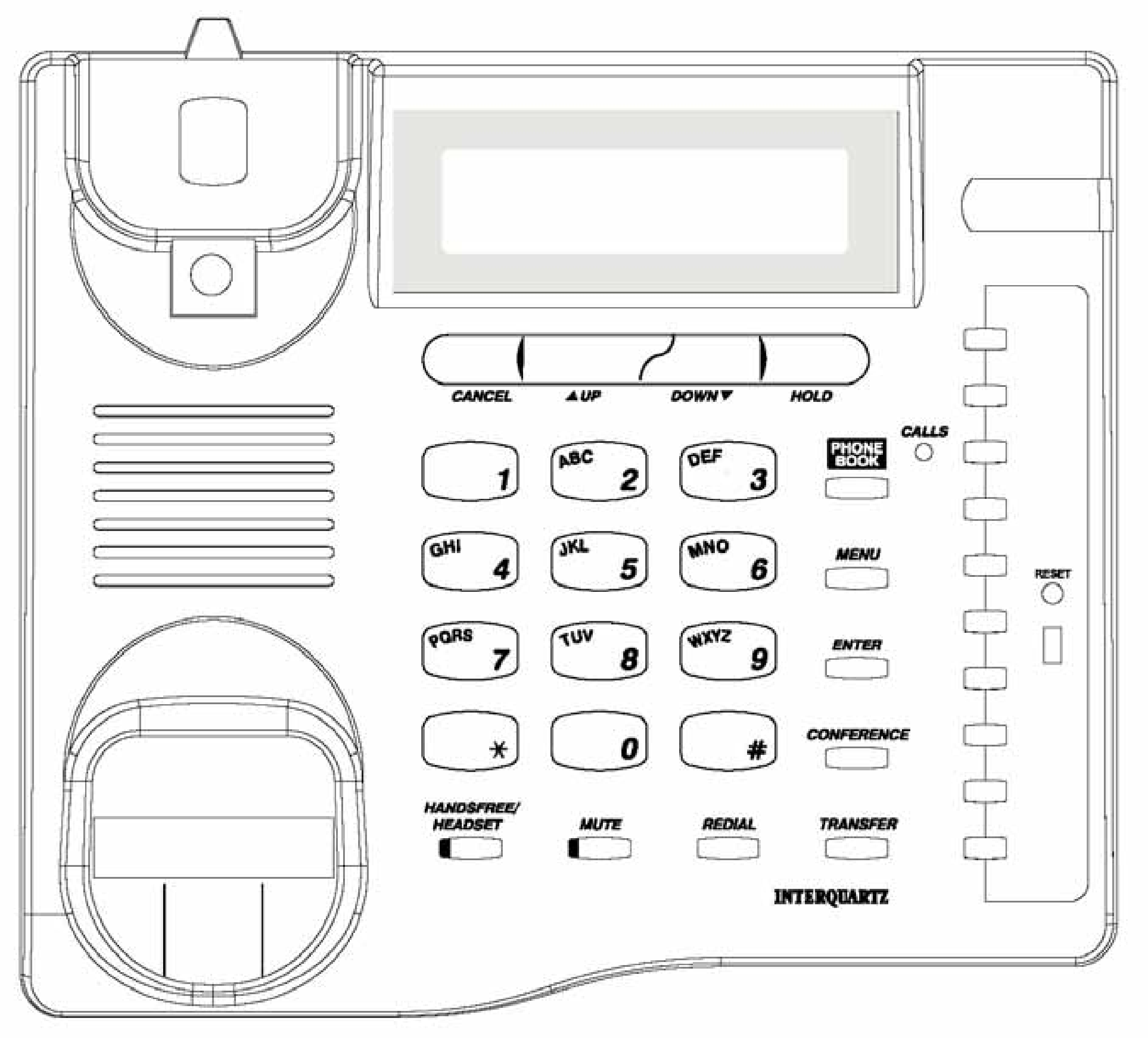


Figure 1 Line drawing of 9339HS VoIP Phone (Front View)

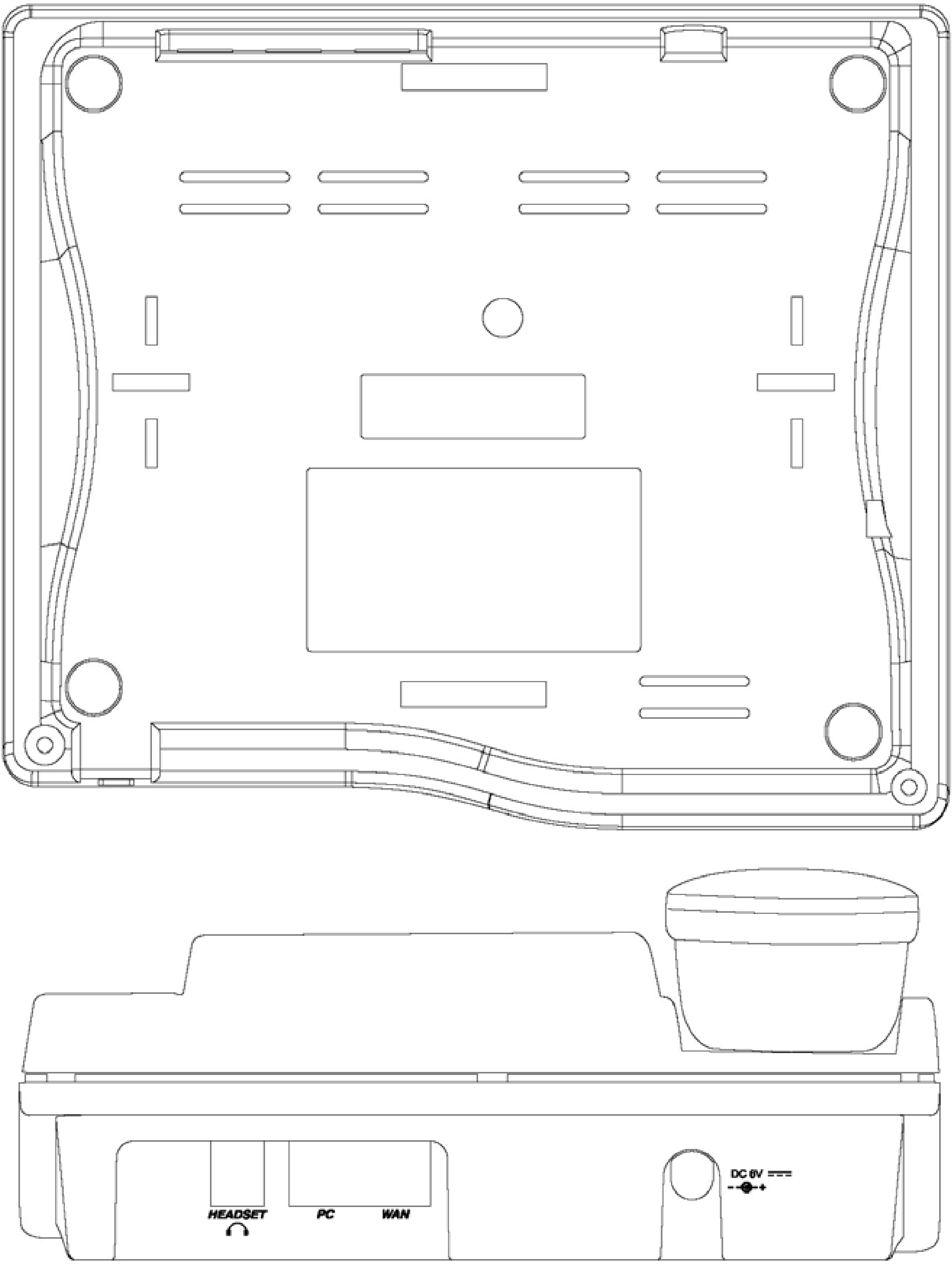


Figure 2 Line drawing of 9339 VoIP Phone (Bottom View & Rear View)