

Contents

Prior To Use	1
Congratulations	1
Safety Precautions	1
Unpacking	2
Phone Layout and Keys Function	2
LCD Window Icons Description	4
Connecting Lines	5
Installing Batteries	5
Charging Handset	6
Using Handset Carrying Clip	6
Turning Handset On/Off	7

Part One: Cordless Phone

8

Basic Functions	9
Making a Call	9
Receiving a Call	9
Muting the Microphone	10
Adjusting Audio Volume	10
Ringer Setting	11
CID Book	12
Key Lock	14
Selecting a Language	14
HS Name	15
Auto Answer	15
Paging	16
Dial Mode	16
Handsfree	17
Flash Signal	17

Advanced Functions	18
Menu Navigation	18
Menu Map	18
Terminate HS	19
Flash Time	19
Modify PIN	20
BS Reset	20
Setting Alarm	21
Audio Setup	21
Key Tone	22
Setting Date & Time	22
Selecting a Base	23
HS Reset	23
Registration	24
Calling with more than one Handset	25
Making an Internal Call	25
Call Transfer between Handsets	25
Setting up a 3-way Conference Call	26
Caller ID on Call Waiting	27

Part Two: VoIP Phone **29**

Installation	30
Operations	30
Making an IP Call	30
Last Number Redial	31
Phone Book	32
Call Transfer	35
Three-way Conference Call	35
Configuration with Web	36
User Verification	37

Configuration Menu	38
Current State	39
Wizard Configuration	40
MMI Setting	42
Network Configuration	43
Wide Area Network (WAN)	43
Local Area Network (LAN)	44
QoS Settings	45
Service Port	46
DHCP Setting	47
Time Setting (SNTP)	48
VoIP Configuration	50
SIP Configuration	50
Advanced SIP Configuration	52
IAX2 Configuration	55
SIP Stun Configuration	57
Dial Peer	58
Phone Configuration	61
DSP Configuration	61
Call Service Configuration	63
Digital Map	65
Maintenance	67
Auto Provision	67
Syslog Configuration	68
Configuration	69
Update	69
Account Configuration	71
Reboot	72
Security	72
System Logout	78
Feature List	79
Troubleshooting	80

Troubleshooting

Symptom	Check & Remedy
No operation	<ul style="list-style-type: none"> ● Check that the power adapter is properly connected. ● Check that the telephone line cord is properly connected. ● Check that the handset is fully charged. ● Check that the handset batteries are installed properly.
No dial tone	<ul style="list-style-type: none"> ● Check that the telephone line cord is properly connected. ● Check that the power adapter is properly connected.
Nothing appears on the display	<ul style="list-style-type: none"> ● Check that the handset batteries are installed properly. ● Check that the handset batteries in full power. ● Check that the handset is on.
Handset seems to have very short battery life	<ul style="list-style-type: none"> ● Clean the charge contacts. ● Consistently short battery life may indicate that replacement of the batteries is necessary. ● Make sure the right batteries be used.
Caller's number is not displayed	<ul style="list-style-type: none"> ● Make sure you have subscribed to a Caller ID service via your network provider. ● The caller may have withheld their details. ● Let the phone ring a couple of times as there may be a delay in receiving the Caller ID information.
The static IP of the WAN and the LAN can't be configured	<ul style="list-style-type: none"> ● Check that when the phone in the NAT mode (not Bridge mode), the WAN Port IP is different from the LAN Port IP. For example, if the LAN Port IP Address is 192.168.1.X, then please don't configure the WAN Port IP Address to 192.168.1.X.
During a conversation, the phone sends busy tones suddenly	<ul style="list-style-type: none"> ● Check that the network and the connection of the phone and the cable. Because the phone can detect whether the current network is normal or not automatically. If the phone's network is disconnected during the conversation, so the phone will send busy tones to warn.

Prior To Use

Congratulations

on purchasing this VoIP (Voice over Internet Protocol) cordless phone. The phone enables carrier class residential and business IP Telephony services to be delivered over broadband or high-speed Internet connections. Moreover, it can be used as a ordinary DECT (Digital Enhanced Cordless Telecommunication) Phone. Similarly to GSM, this technology allows you to get the benefits of the digital wireless communication systems, which are better protected against interferences, tapping and intrusions.

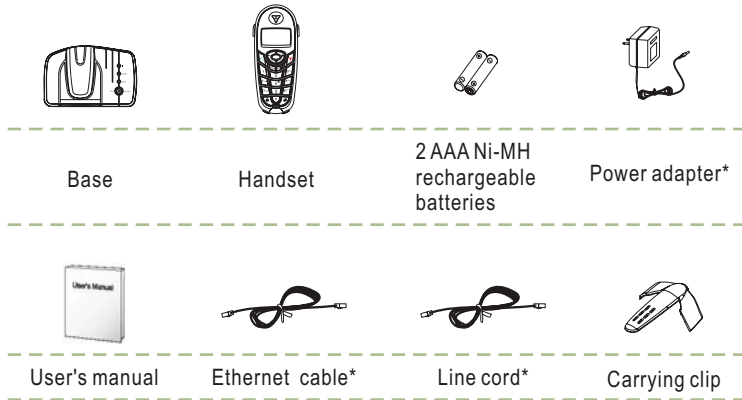
Safety Precautions

To reduce the risk of electrical shock and injury, please follow these basic safety precautions before using the phone.

1. Carefully read and observe the instructions in this manual.
2. Follow all warnings marked on the unit.
3. When cleaning, unplug the phone first, then use a damp cloth to wipe. Do not use liquid or aerosol cleaners.
4. Do not place objects on the line cord that may cause damage.
5. Do not use this phone in wet surroundings or environments where there is a risk of explosion.
6. Avoid spilling of any liquid on the phone.
7. Unplug this phone from the wall outlet and refer servicing to qualified service personnel only.
8. Pay attention to the polarity of the batteries, insert the rechargeable batteries in accordance with polarity symbols (this instruction is found in the installing batteries section.)
9. Use only **the batteries indicated in the User's Manual**. Never use other ordinary batteries or conventional alkaline batteries. Otherwise this may not only cause personal injuries but also damage to the unit.
10. Do not mix exhausted batteries with full batteries. Exhausted batteries shall not be disposed of with the usual household waste or in a fire.
11. If you are sure you will not be going to use the handset over a month, please take out the batteries from the battery compartment.
12. Use only **the power supply indicated in the User's Manual**.
13. Keep the phone out of the reach of children.
14. Use the phone only in the described manner.
15. Stop using the phone if it becomes damaged.
16. The phone is designed to work with in a temperature range from 0°C to 40°C.

■ Unpacking

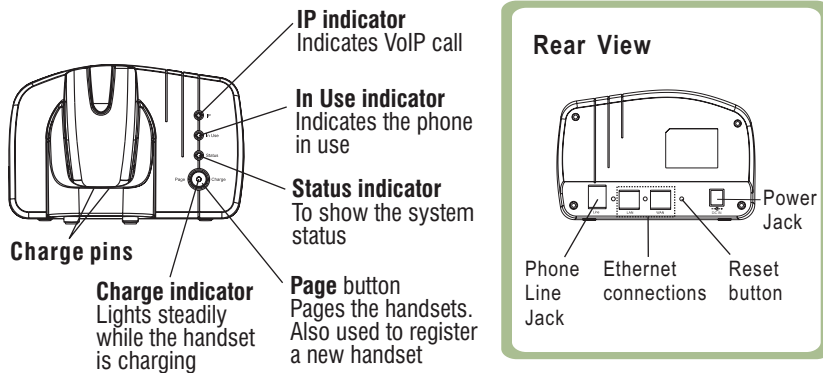
Once you have unpacked your phone, make sure that all the parts shown below are available. If any pieces are missing or broken, please promptly call your dealer.



*The shape of the plugs can vary according to each country's specification.

■ Phone Layout and Keys Function

Base



Feature List

DECT phone

1.8GHz DECT technology with GAP
 Up to 5 handsets for optional for one base
 Up to 4 bases for optional for one handset
 10 ring tones and volume selectable (5 levels+ off)
 Key tone ON/OFF selectable
 Full handsfree speakerphone
 Internal call
 Call transfer
 3-way conference call
 Receiver/Speaker volume adjustable (5 levels)
 Mute
 PIN protection
 Redial, Flash, Pause
 TONE/PULSE dialing mode selectable
 FLASH time selectable

Call duration display
 Date & Time setting
 Keypad lock
 9 languages selectable
 Auto answer ON/OFF selectable
 Alarm clock include snooze option
 Handset Name setting
 LCD backlight

Caller ID:
 FSK and DTMF dual system Caller ID
 Caller ID / Call Waiting
 Caller ID list with up to 40 entries
 Incoming calls number can be added to phonebook or deleted
 New call indicator

IP phone

SIP Features:

- Proxy and Register
- SIP domain
- DNS name of SIP server
- NAT transverse, STUN
- NAT transverse, SIP Express router
- Public Server/Private server.
- Dual public server
- Each password for each number
- SIP Call forward/transfer/holding/waiting
- Peer to peer SIP call

Networks Features:

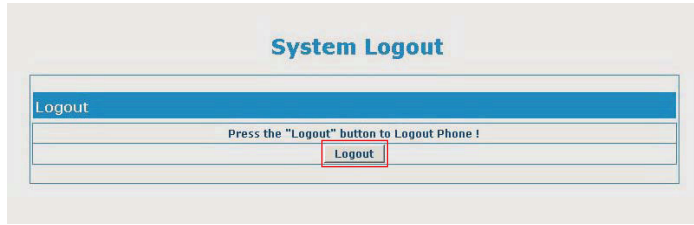
- WAN/LAN port with Router or Bridge Mode
- Under Bridge mode, Access internet by using NAT through PPPoE
- PPPoE for xDSL, automatically keep alive
- DHCP Client on WAN
- DHCP server on LAN
- DHCP Client with 2 servers IP

- DNS relay on LAN
- Auto configuration on LAN for IP and DHCP server

Call Control Features:

- Phone Book with up to 50 groups
- 9 VIP ring tones setting for phonebook entries
- Call routing table for Phone Book
- Multi numbers for same phone
- Public/Private number for phone
- HOTLINE Service, Pick up phone, dial immediately
- Redial book with up to 10 entries
- Black list for reject authenticated call
- Empty calling number reject service
- Limit dialing out number list
- No Disturb
- Caller ID display
- Call forward with no condition busy
- Call transfer
- 3-way talking service
- Dial out authentication

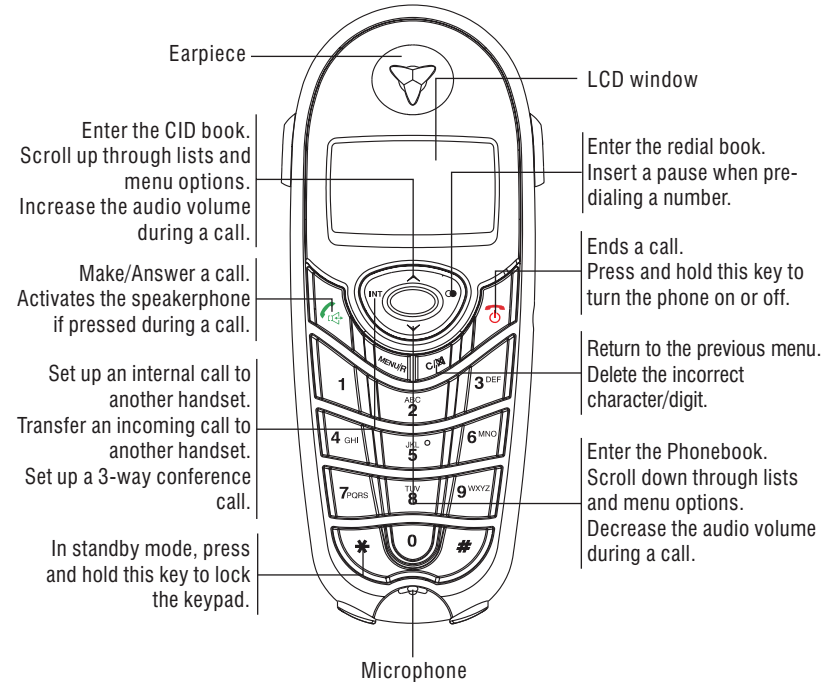
System Logout



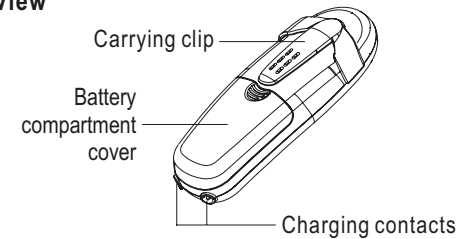
If you want to exit, click the **"Logout"** button.

Phone Layout and Keys Function (continued)

Handset



Rear View



LCD Window Icons Description



This area displays in-use information such as the other party's number, call duration, menus, etc. In standby mode, it displays the handset number, the signal strength icon, battery status icon and the current time.



New Call

Indicates you have missed one or more calls.



Off Hook

Indicates the line is engaged.



Hands-free

Indicates the handsfree function is enabled.



Battery Icon

Indicates battery charge level.



Signal Icon

Indicates the current signal strength. If the handset is too far from the base, this icon will blink on the LCD.



Keypad Lock

This icon appears when the keypad is locked.

4. VPN Tunnel

The screenshot shows the 'SECURITY' configuration page with the 'VPN' tab selected. The interface includes the following sections:

- VPN IP:** A text field containing '0.0.0.0'.
- VPN Mode:** Radio buttons for 'UDP Tunnel' (selected), 'L2TP', and 'Enable VPN'.
- UDP Tunnel:** A sub-section with fields for 'VPN Server Addr' (0.0.0.0), 'VPN Server Port' (80), 'Server Group ID' (VPN), and 'Server Area Code' (12345).
- L2TP:** A sub-section with fields for 'VPN Server Addr', 'VPN User Name', and 'VPN Password'.
- APPLY:** A button at the bottom right.

This function should use our private VPN server software.

VPN server addr: fill in VPN server public IP address

VPN server port: fill in 12000 if you do not modify the VPN server config

SIP 1 server register address : fill in VPN server VPN address 172.0.0.5 if you do not modify the VPN server config. Others sip parameter needn't change.

You can only use VPN function in the condition below:

- 1 It needs to use our VPN software.
- 2 Operation system should be linux, not windows2003.
- 2 SIP server software and VPN server software place in one server hardware.
- 3 SIP side must monitor the VPN tunnel packets.

Inside Port

LAN device port for port mapping.

Click to add to the list, click to delete it.

DMZ Setting

DMZ Table	
Outside IP	Inside IP
192.168.1.125	192.168.10.23

DMZ Table Option

Outside IP	<input type="text"/>
Inside IP	<input type="text"/>
Outside IP	192.168.1.125

Outside IP	Inside IP
192.168.1.125	192.168.10.23

Outside IP

DMZ outside IP address.

Inside IP

DMZ inside IP address.

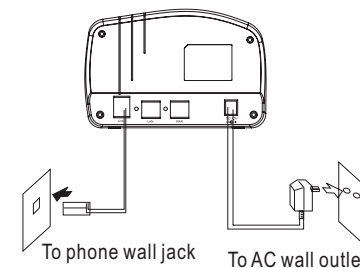
Outside IP

Select the IP you want to delete,

then click the to delete the selected one.

Connecting Lines

1. Connect one end of the phone line cord to the phone line jack on the rear of the base, and the other end to a standard phone wall outlet.
2. Connect the modular end of the AC power adapter to the power jack of the base, then plug the AC adapter into a standard AC wall outlet.



Installing Batteries

The rechargeable Ni-MH batteries (AAA size) come with your phone. Install the batteries before using your phone.

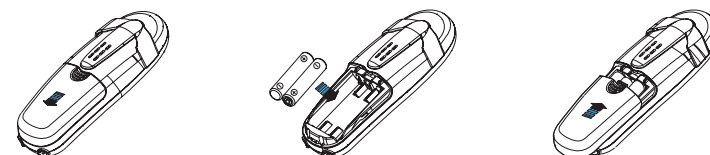
1. Slide the battery cover in the direction of the arrow and pull it out.

2. Insert new batteries as indicated, matching correct polarity (+, -).

Note:

- Reversing the orientation may damage the handset.

3. To replace the battery cover, slide the cover up until it snaps shut.



Notes:

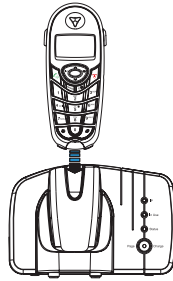
- The batteries need to be replaced if they do not recover their full storage capacities after recharging.
- When replacing the batteries, always use good quality Ni-MH re-chargeable batteries. Never use other batteries especially conventional alkaline batteries.

■ Charging Handset

Important Note: Before initial operation, **YOU SHOULD FULLY CHARGE THE HANDSET** for about **14-16** hours.

To charge the handset, you should place it on the base.

Result: When you place the handset on the base, the handset automatically turns on and the charge indicator on the base lights up.

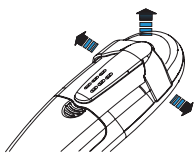


■ Using Handset Carrying Clip

The supplied handset carrying clip allows you to conveniently carry the handset with you. It clips easily to your belt, waist band, or shirt pocket.

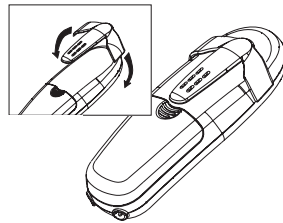
If you want to remove the carrying clip:

Insert a screw driver along the edge of one of its arms and release the clip. Then lift it off.



If you want to attach the carrying clip:

Attach the carrying clip to the back of the handset. Make sure that the carrying clip locks into place.



3. NAT Setting

SECURITY

MMI FILTER | FIREWALL | NAT | VPN

Protocol Set

<input checked="" type="checkbox"/> IPsec ALG	<input checked="" type="checkbox"/> FTP ALG	<input checked="" type="checkbox"/> PPTP ALG
<input type="button" value="APPLY"/>		

NAT Table

Inside IP	Inside TCP Port	Outside TCP Port
Inside IP	Inside UDP Port	Outside UDP Port

NAT Table Option

Transfer Type	Outside Port	
Inside IP	Inside Port	

IPsec ALG Enable /Disable IP Sec ALG, the default is enable.

FTP ALG Enable /Disable FTP ALG, the default is enable.

PPTP ALG Enable /Disable PPTP ALG, the default is enable.

Click to take effect.

NAT Table

Inside IP	Inside TCP Port	Outside TCP Port
Inside IP	Inside UDP Port	Outside UDP Port

The configurations of NAT table displays.

Transfer Type Transfer type using port mapping.

Outside Port WAN port for port mapping.

Inside IP LAN device IP for port mapping.

Specify current adding rule is deny rule or permit rule.

Source address, can be single IP address or network address.

Source address mask, indicates the source is dedicated IP if set to 255.255.255.255. Otherwise is network ID.


Destination address can be IP address or network address.

Destination address mask, indicates the source is dedicated IP if set to 255.255.255.255. Otherwise is network ID.

Then select , click **APPLY** to take effect.

■ Turning Handset On/Off

If the handset is in power off mode, when you place the handset on the base, it automatically turns on. To turn the handset on or off in standby mode, follow these steps:

1. To turn on the handset when it's off, press and hold the  key until you switch the display on. On power up, the handset will enter subscription mode and search for a registered base. A message as shown below will be displayed to the user, and the signal icon will flash.



If the handset is successful in finding a base, the handset will enter the standby mode, the display will show handset number or the handset name (if you have one saved as described on page 15), signal icon, battery icon and current time. If the handset is unsuccessful in finding a base, the handset will behave according to the Out of Range condition.

2. To turn the handset off, keep the  key pressed until the display turns off.

Note:

- Nothing will appear on the LCD when battery power is very low. YOU SHOULD FULLY CHARGE THE HANDSET BEFORE USING.

2. Firewall

Part One: Cordless Phone

Your new VoIP (Voice over Internet Protocol) phone can be used as an ordinary DECT (Digital Enhanced Cordless Telecommunication) phone. It is designed with advanced features. Similarly to GSM, this technology allows you to get the benefits of the digital wireless communication systems, which are better protected against interferences, tapping and intrusions.

The screenshot shows the 'SECURITY' configuration page with the 'FIREWALL' tab selected. It includes sections for 'Firewall Type' (with 'In_access Enable' and 'Out_access Enable' checkboxes), 'Firewall Input Rule Table', 'Firewall Output Rule Table', 'Firewall Set' (with dropdowns for 'Input/Output', 'Deny/Permit', and 'Protocol Type', and input fields for 'Src Addr', 'Des Addr', 'Src Mask', and 'Des Mask'), and 'Rule Delete'.

Firewall Setting Page. User may set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices to access the Internet.

Access list support two type limits: input_access limit or output_access limit.

This phone firewall filter is base on WAN port. So the source address or input destination address should be WAN port IP address.

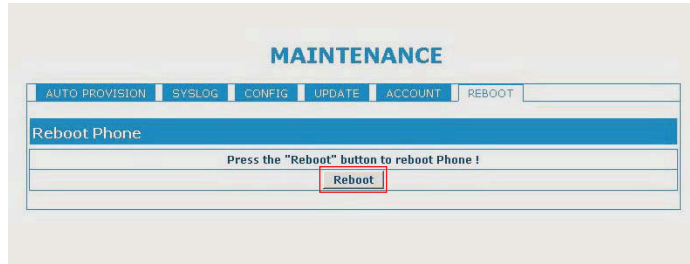
Configuration:

Out_access Enable Enable out_access rule.

The 'Firewall Set' section shows the following configuration: Input/Output is set to 'Input', Deny/Permit is set to 'Deny', Protocol Type is set to 'UDP', and Port Range is set to 'more than'. There are empty input fields for Src Addr, Des Addr, Src Mask, and Des Mask, and an 'Add' button.

Input/Output Specify current adding rule is input rule or output rule.

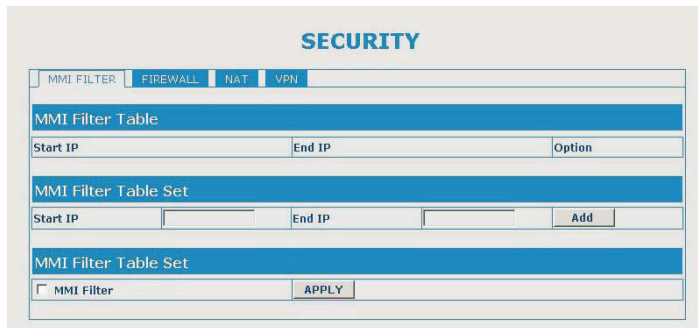
6. Reboot



If you want to restart the phone, click the "Reboot" button.

Security

1. MMI Filter




MMI filter is used to make access limit to this phone. When MMI filter is enable, only IP address within the start IP and end IP can access this phone.

Basic Functions

Making a Call

1. Pick up the handset and press the  key, after you hear the dial tone, press .

Result: The **In use** indicator blinks and the  icon appears on the LCD.

2. Dial a telephone number.


3. When you hear your called party, speaker with a normal voice.

4. To end the call, either press the  key or replace the handset on the base.

Receiving a Call

When a new call is received, the call information will appear.



If the caller can be identified, the caller's phone number is displayed.

1. To answer the call, press the  key. If you have activated the Auto Answer function, when the handset is on the base, simply lift it up to answer.

2. To end the call, either press the  key or replace the handset on the base.

Result: After you hang up the call, the LCD displays the call duration.


Notes:

- You can select the ringer volume by using  or  key when the phone rings with an incoming call.
- If the polyphonic ring tone is set, the ring tone may continue for 4 seconds after the caller's hang-up or after the parallel phone had answered the call.

Muting the Microphone



You can temporarily switch your phone's microphone off, so that the other party cannot hear you. During the period you can still hear him/her voice.

Example: You want to say something to another person in the room but do not want the other party to hear you.

To mute your microphone, press the  key during the conversation. The "**CALL MUTED**" appears on the LCD.

To unmute the microphone, press the  key again. You will restore the normal condition.

Adjusting Audio Volume

During a conversation, you can use the  or  key to adjust the level of the earpiece (or the handsfree speakerphone one, if activated) volume from **VOL 1** to **VOL 5**. The selected volume will be displayed on the LCD.



You can also select the earpiece volume or the speaker volume by the **AUDIO SETUP** in Menu under **HS SETTING**, as described in page 21.

5. Account Configuration

Users can edit users account and modify existing users' authority on this web page.

User Name	User Level
admin	Root
guest	General

Add User

User Name:

User Level:

Password:

Confirm:

Account Option

User Name	User Level
admin	Root
guest	General

Current users' list of the phone:

List existing phone user account name.

Show existing user account level: Root and General. Root level users have the right to modify config. General level users have the right to modify some config only.

Configure the new account's password.

Enter the password again to confirm.

a. Web Update

On this page, user can select the upgrade document (firmware or config file) from hard disk of the computer directly to run the system upgrade. After upgrade completed, reset the phone and it will be usable immediately. Firmware format is *.z as suffix.

b. FTP Update















Users can download upgrade documents or lead in configuration files thru FTP mode. Please make sure export and import rights are authorized by FTP server before using FTP update way.

Definition of each parameter described as below:



Server	Set IP address for upload or download FTP/ TFTP server
Username	Set username of the upload or download FTP server. If user select TFTP mode, no need to input username and password
Password	Set upload or download of FTP server password
File name	Set file name for system upgrade documents or system configuration files. system file take .z as suffix, configuration files take .cfg as suffix
Type	Config export/import/upgrade file type [three options]: "Application update" is system documents upgrade "Config file export" is export configuration files to server "Config file import" is import configuration files to gateway
Protocol	Set transport protocol type [two options]: FTP and TFTP

■ Ringer Setting

You can select your own external (from the telephone Network) or internal (from other handsets registered to the same base) ring tone and adjust the volume.

1. Press the  key.
2. Press  or  key to choose **HS SETTING** menu, then press the  key.
3. Press  or  key to choose **RING SETUP**, and press the  key.
4. Use  or  key to choose **INT RING/EXT RING**, press the  key.
5. Press  or  key to choose **MELODY** or **VOLUME**, press the  key.
Result: You can select the ring tone from Melody 1 to 10 or adjust the volume from Volume 1 to 5, or off. When you adjust the ring tone or volume, the phone plays the selected melody or sounds the selected loudness.
6. Press the  key to confirm.






CID Book

When you receive a call, if the caller's information is transmitted from the network on which the call was made (and the caller doesn't hide it), the caller's phone number is displayed. Moreover, if you missed one or more calls, the  icon will appear on the LCD. If the CID memory is full, the  icon will flash on the LCD.










OUT OF AREA - This message will display when someone calls from an area where the telephone company is not offering the caller identification services or is not yet providing number delivery to your area.

PRIVATE - If the caller has exercised the option to prevent his name and number from being sent, the message will show on the LCD.

View any of the CID numbers

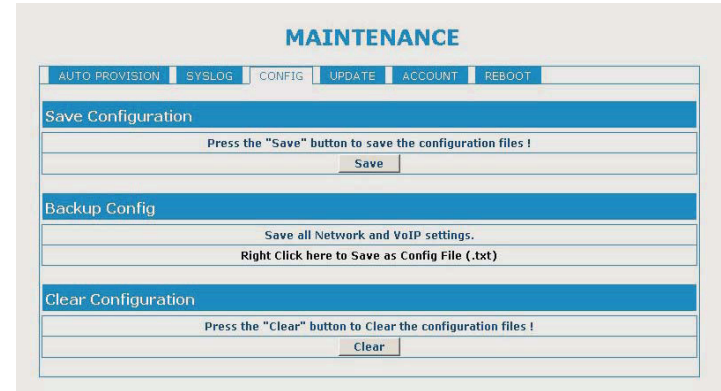
1. Press the  key.
2. Press the  key again to choose the **CID BOOK**, and if available, the numbers are displayed. (You can also access the CID book directly by pressing the  key in standby mode.)
3. Use  or  key to view the numbers.

Add CID Number to Phone Book

1. Press the  key.
2. Press the  key again to choose the **CID BOOK**, and if available, the numbers are displayed.
3. Use  or  key to view the numbers. When the desired number displays on the LCD, press the  key to enter **ADD TO PB** menu.
4. Press the  key, you are prompted to enter the name.
5. Enter the name and press the  key, then you can modify the number you want to store.
6. Press the  key, you are able to select the ring tone from Melody 1 to Melody 10, press the  key to confirm and exit, you will hear a confirm beep.

3. Configuration

User can save, backup and clear the current configuration on this page.

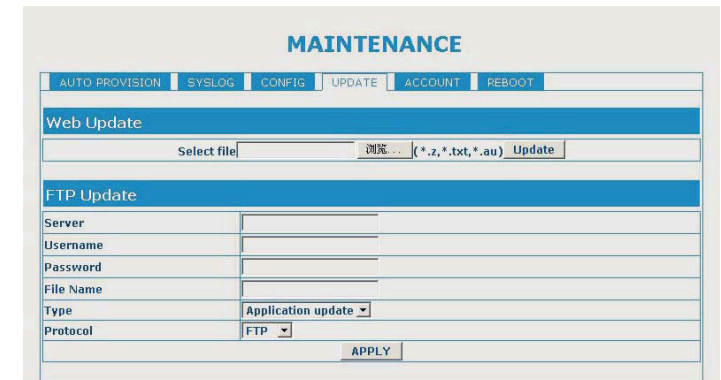


The screenshot shows the MAINTENANCE menu with a navigation bar at the top containing: AUTO PROVISION, SYSLOG, CONFIG, UPDATE, ACCOUNT, and REBOOT. The main content area has three sections:

- Save Configuration**: A blue header bar. Below it, a text prompt says "Press the 'Save' button to save the configuration files!". At the bottom of this section is a "Save" button.
- Backup Config**: A blue header bar. Below it, a text prompt says "Save all Network and VoIP settings. Right Click here to Save as Config File (.txt)".
- Clear Configuration**: A blue header bar. Below it, a text prompt says "Press the 'Clear' button to Clear the configuration files!". At the bottom of this section is a "Clear" button.

The system configuration can be set as factory default configuration. Click the "**Clear**" button to clear config page and the phone will restart automatically.

4. Update



The screenshot shows the MAINTENANCE menu with a navigation bar at the top containing: AUTO PROVISION, SYSLOG, CONFIG, UPDATE, ACCOUNT, and REBOOT. The main content area has two sections:

- Web Update**: A blue header bar. Below it, there is a "Select file" input field, a "浏览..." (Browse...) button, and a file type filter "(*.z,*.txt,*.au)". To the right is an "Update" button.
- FTP Update**: A blue header bar. Below it, there are several input fields: "Server", "Username", "Password", and "File Name". There are also two dropdown menus: "Type" (set to "Application update") and "Protocol" (set to "FTP"). At the bottom right of this section is an "APPLY" button.

2. Syslog Configuration

The screenshot shows the MAINTENANCE menu with the following options: AUTO PROVISION, SYSLOG, CONFIG, UPDATE, ACCOUNT, and REBOOT. The Syslog Set screen is displayed with the following fields:

Syslog Set	
Server IP	0.0.0.0
Server Port	514
MGR Log Level	None
SIP Log Level	None
IAX2 Log Level	None
Enable Syslog	<input type="checkbox"/>
<input type="button" value="APPLY"/>	

Server IP Configure syslog server IP address.

Server Port Configure syslog server port, click **APPLY** button after inputting server IP & server port to take effect.

MGR Log Level Configure MGR log level.

SIP Log Level Configure SIP log level.

IAX2 Log Level Configure IAX2 log level.

Enable Syslog Select to enable Syslog.

Delete a number in the CID Book


1. Press the key.
2. Press the key again to choose the **CID BOOK**, and if available, the numbers are displayed.
3. Scroll to the number you want to delete by using or key.
4. When the desired number appears on the display, press the key.
5. Use or key to choose **DELETE**, press the key. The display will show "**CONFIRM ?**". Press the key to confirm, you will hear a beep and the LCD shows the next number.



Delete all numbers in the CID Book



1. Press the key.
2. Press the key again to choose the **CID BOOK**, and if available, the numbers are displayed.
3. Press the key and use or key to choose **DELETE ALL** menu, then press the key.
4. The display shows "**CONFIRM ?**", press the key to confirm.

Result: You hear a beep and the phone returns to the standby mode after clearing all the CID numbers.

■ Key Lock

If you turn on this feature, all keys will be locked. You can answer calls by using the  key. But when you hang up, the phone returns to the locked mode. This feature is useful to avoid pressing keys by mistake.

To lock the keys, in the standby mode just keep the  key pressed till the display shows "HS LOCKED", then the  icon appears.











To unlock the keys, keep the  key pressed again till the key lock icon  disappears from the display.

Note:

- In locked mode, if you press any key, the phone will generate a warning tone, and the "HS LOCKED" message will be redisplayed.

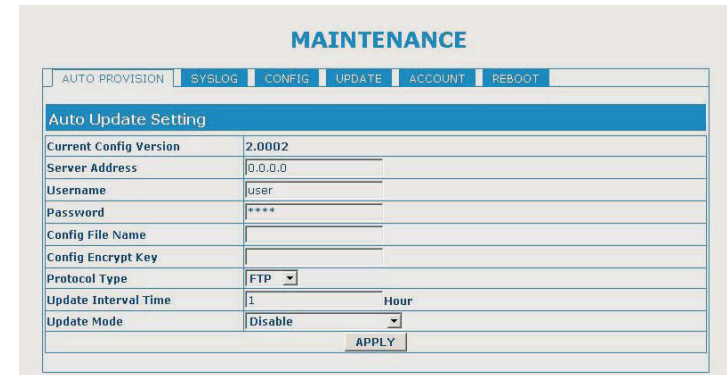
■ Selecting a Language

The handset supports up to 9 predefined languages: English, German, French, Italian, Spanish, Dutch, Russian, Portuguese and Turkish. You can change the language used to display the menu messages.

1. Press the  key.
2. Press  or  key repeatedly to choose **HS SETTING**, press the  key.
3. Use  or  key to choose **LANGUAGE**, then press the  key.
4. The languages will be displayed. Use  or  key to scroll through the options. Each language will be displayed in its own native translation.
5. Press the  key to select the required language and you will hear a confirm beep.

■ Maintenance

1. Auto Provision



Auto Update Setting	
Current Config Version	2.0002
Server Address	0.0.0.0
Username	user
Password	****
Config File Name	
Config Encrypt Key	
Protocol Type	FTP
Update Interval Time	1 Hour
Update Mode	Disable
<input type="button" value="APPLY"/>	

- | | | |
|-------------------------------|---------|--|
| Current Config Version | 2.0002 | Current version, read only. |
| Server Address | 0.0.0.0 | Configure the IP address of FTP/TFTP server. |
| Username | user | Configure username of FTP. |
| Password | **** | Configure password of FTP. |
| Config File Name | | Configure the name of file which needs to update. The version of the file needs to be altered. |
| Config Encrypt Key | | Input the password here if the file is encrypted. |
| Protocol Type | FTP | Select the protocol type, FTP or TFTP. |
| Update Interval Time | 1 Hour | Configure the update interval time, the unit is hour. |
| Update Mode | Disable | Select the update mode. |










Example:

Digital Rule table	
	Rules:
	"[1-8]xxx"
	"9xxxxxxx"
	"911"
	"99T4"
	"9911x.T4"

- [1-8]xxx any 4 digits number between 1000 and 8999 sending out immediately
9xxxxxxx any 8 digits number starting with 9 sending out immediately
911 after finishing dialling 911, it will send out immediately
99T4 after finishing dial 99, it will send out in 4 seconds
9911x.T4 any more than 5 digits length starting with 9911, sending out within 4 seconds
End with "#", Fixed length, Time out these options are not repulsion.











■ HS Name

"HS N" is the default name of the handset after registration of handset. The handset number N= 1 to 5 reflects the handset is the Nth handset, which is registered to the base. The HS Label is displayed in the standby mode. You can modify it by the following steps:

1. Press the  key.
2. Press  or  key repeatedly to choose **HS SETTING**, press the  key.
3. Use  or  key to choose **HS LABEL**, then press the  key.
4. You can enter the handset name (up to 6 characters) by using the alphanumeric keys (use the  key to delete the previous character if necessary), then press  key to confirm and exit. The handset name will be modified.

■ Auto Answer

With this feature, you can answer a call by just picking up the handset from the base without pressing any key.

1. Press the  key.
2. Press  or  key repeatedly to choose **HS SETTING**, press the  key.
3. Use  or  key to choose **AUTO ANSWER**, then press the  key.
4. Use  or  key to choose **ON** or **OFF** to enable or disable the function, then press the  key, you will hear a confirm beep.

Paging

You can page the handset from the base unit making it ring with a special tone. It's useful to locate a lost handset.

Press the **Page** key on the bottom of the base (less than 5 seconds), all the handsets registered to the base will ring for about 60 seconds.










To stop paging, press the **Page** key on the base again or any key on the handset.

Note:

- On a long **Page** key press, more than or equal to 5 seconds, the base will enter subscription mode. For details, see page 24.

Dial Mode

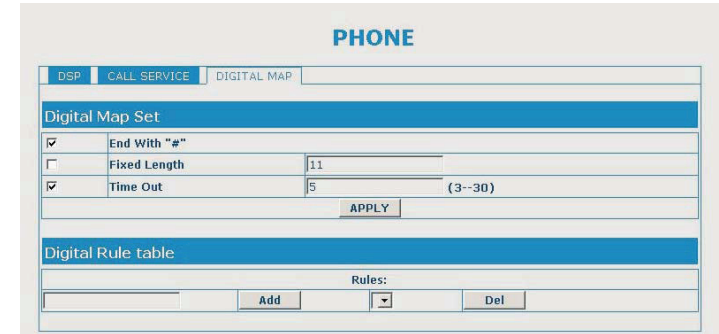
You can select to change the dial mode by the following steps:

- Press the  key.
- Press  or  key repeatedly to choose **BS SETTING**, and press the  key.
- Press the  key to choose **DIAL MODE**, then press the  key.
- You can press  or  key to select the required dial mode options (**TONE /PULSE**) and press the  key. A confirm beep will be heard.

Note:

- If you are not sure which dialing mode should be selected, please contact your local service provider.

3. Digital Map



The screenshot shows the 'PHONE' menu with three tabs: 'DSP', 'CALL SERVICE', and 'DIGITAL MAP'. The 'DIGITAL MAP' tab is selected. Below the tabs, there are two main sections: 'Digital Map Set' and 'Digital Rule table'. The 'Digital Map Set' section has three rows: 'End With "#"' with a checked checkbox, 'Fixed Length' with an unchecked checkbox and a value of '11', and 'Time Out' with a checked checkbox, a value of '5', and a range '(3--30)'. An 'APPLY' button is located below these rows. The 'Digital Rule table' section has a 'Rules:' label and an empty table with 'Add', 'Del', and a dropdown arrow button.

End With "#" Configure # for the end of dialling.

Fixed Length 11 When the length of the dialling match, the call will be sent.

Time Out 5 (3--30)

Specify the timeout of the last dial digit. The call will be sent after timeout. The unit is second.



The screenshot shows the 'Digital Rule table' section with a 'Rules:' label and an empty table with 'Add', 'Del', and a dropdown arrow button.

The dialling rules, as follows:

[] number location value range. It can be a number range (such as [1-4]), or number is separated by comma such as [1,3,5], or use a list such as[234].

x represents any one number between 0 and 9.

Tn represents the last digit timeout. Here [n] represents the time from 0~9 second, it is necessary. Tn must be the last two digit in the entry. If Tn is not included in the entry, we use T0 as default, it means system will sent the number immediately if the number matches the entry.

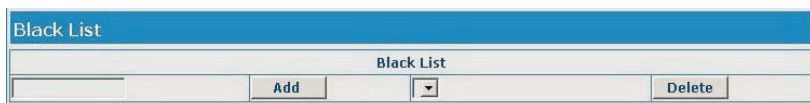
. (Dot) represents any number and no length limit.

Enable Call Transfer Configure to enable/disable call transfer (CT); after it is enabled, user accept calls, with hooking and dial directly, the phone will transfer the calls according to the above configurations of the port number IP images.

Enable Call Waiting Configure to enable/disable call waiting service; if it is selected, user can hold calls of the other party by hooking, with hooking again, the hold call can go on.

Enable Three Way Call Configure to enable/disable three way call; user can call the other part as the call origination, after talking, make hooking to hold this part and then press * key to hear the dialling tone, after call completion to the third party, hooking again to recover the talk with the second part, then the three way call concurrently.

Accept Any Call If it is selected, the phone can receive the calls that have wrong calling numbers but the calling IP is this phone.





Configure add/delete blacklist. If user don't want to answer a certain number, please add this number to the list, and then this number will be unable to get through the phone.





Configure out-limit list; for example, if user don't want the phone to dial a certain number, please add the number to this table, and the user will be unable to get through this number.

■ Handsfree

During a call, you may turn on the speakerphone built in the handset. In this mode you can simply put your handset on a surface (e.g. A desk) and enjoy the comfort of the handsfree conversation.


To turn the feature on, press the  key during a call (line engaged). The  icon appears on the LCD.


To turn the feature off, and keep on talking in normal mode, press the  key again. The  icon will disappear.

Notes:

- Before putting the handset near your ears, be sure you have turned the speakerphone off.
- To adjust the speakerphone volume, see page 10 or page 21.

■ Flash Signal

When the line is engaged, pressing the  key sends the Flash signal. This key is used in conjunction with special services (e.g. Call waiting) which can be available in your phone network. For further details please contact your network telephone company.

To use the said special services, just press the  key while the line is engaged: the display will show **R**.

Advanced Functions

Menu Navigation

To access a menu option:

1. To display menu items, press the **MENU** key.
2. To scroll through menu options, press **▲** or **▼** key repeatedly.
3. To select a menu, press the **MENU** key when the desired menu appears on the LCD.
4. Repeat if necessary.

To return to standby mode:

If you press the **C/M** key from any menu (not in the number or text input mode), the phone returns to the previous screen.

To return to standby mode, press the **END** key.

Also, the phone will automatically return to standby mode from any menu if no key is pressed in the next 40 seconds.

Menu Map

1. CID BOOK (see page 12)

2. PHONE BOOK

- ADD ENTRY (see page 32)
- CHANGE ENTRY (see page 33)
- DELETE ENTRY (see page 34)
- DELETE ALL (see page 34)
- PB STATUS (see page 34)

3. BS SETTING

- TERMINATE HS (see page 19)
- DIAL MODE (see page 16)
- FLASH TIME (see page 19)
- MODIFY PIN (see page 20)
- BS DEFAULT (see page 20)

4. HS SETTING

- ALARM (see page 21)
- AUDIO SETUP (see page 21)
- RING SETUP (see page 11)
- tone SETUP (see page 22)
- LANGUAGE (see page 14)
- HS LABEL (see page 15)
- AUTO ANSWER (see page 15)
- DATE & TIME (see page 22)
- SELECT BS (see page 23)
- HS DEFAULT (see page 23)

5. REGISTER (see page 24)

2. Call Service Configuration

On this page, user can set value added services such as hot-line, call forwarding, call transfer (CT), call-waiting service, three way call, blacklist, out-limit list and so on.

The screenshot shows the 'PHONE' menu with three sub-sections: 'Call Service Setting', 'Black List', and 'Limit List'. The 'Call Service Setting' section includes fields for 'Hot Line', 'No Answer Time' (set to 20 seconds), 'P2P IP Prefix', 'Do Not Disturb' (unchecked), 'Ban Outgoing' (unchecked), 'Enable Call Transfer' (checked), 'Enable Call Waiting' (checked), and 'Enable Three Way Call' (checked). There is an 'APPLY' button at the bottom of this section. The 'Black List' section has an 'Add' button, a dropdown menu, and a 'Delete' button. The 'Limit List' section also has an 'Add' button, a dropdown menu, and a 'Delete' button.

Configuration Explanation:

Hot Line Configure hot-line number of the port. With this number of the port, this hot-line number will be dialed automatically as soon as off-hook and user can't dial any other number.

No Answer Time 20 (seconds) Configure no answer time.

P2P IP Prefix Configure the prefix of Peer to Peer IP calling. For example, you set the prefix as "192.168.1." and the IP you want to call is "192.168.1.119", you only need to dial "#119" to call.

Do Not Disturb If selected, it enables the user to reject all incoming calls, and the callers will hear a busy tone. But the user can dial out any number.

Ban Outgoing If selected, it disables the user to dial out any numbers, but the user can answer incoming calls.

G729 Payload Length 20ms ▾ Configure the G729 payload length.

Signal Standard China ▾ Configure signal standard.

Handdown Time 1000 ms Configure handdown time, that is, if the hooking time is shorter than this time, then the gateway will not consider the user has handdown.










VAD Enable/disable voice activity detection.

G722 Timestamps 160/20ms ▾ G722 time stamps selection. You can select between 160/20ms and 320/20ms.

G723 Bit Rate 5.3kb/s ▾ G723 bit rate selection. You can select 5.3kb/s or 6.3kb/s.











■ Terminate HS

This function allows you to delete a handset registration from the base.

1. Press the  key.
2. Press  or  key repeatedly to choose **BS SETTING**, then press the  key.
3. Press the  key again to choose **TERMINATE HS**. The display shows "**PIN?**" to prompt you to enter the PIN Code (The default PIN Code is "**0000**"). Then press the  key.
4. Use  or  key to select a handset. Selecting either the current handset or a non-existent handset will sound a warning tone to the user.
5. Press the  key to confirm, and the selected handset will behave according to the unregistered condition.

■ Flash Time

You can select to change the Flash time by the following steps:

1. Press the  key.
2. Press  or  key repeatedly to choose **BS SETTING**, then press the  key.
3. Press  or  key to choose **FLASH TIME**, then press the  key.
4. The display shows the current setting. Use  or  key to select **SHORT/LONG**, press the  key.

■ Modify PIN

This function allows you to change the PIN number of the current active Base. The default PIN is 0000.

1. Press the **MENU** key.
2. Press **▲** or **▼** key repeatedly to choose **BS SETTING**, then press the **MENU** key.
3. Press **▲** or **▼** key to choose **MODIFY PIN**, then press the **MENU** key.
4. The display shows "PIN?" to prompt you to enter the old PIN. If the old PIN code is validated, you will be prompted to enter a new PIN code. If the old PIN is not validated, a warning beep will sound and you will be returned to the **MODIFY PIN** menu.
5. Enter the new PIN, press the **MENU** key. You will be requested to confirm the PIN code. Enter the new PIN again then press the **MENU** key to confirm and exit.

■ BS Reset

This function allows you to reset the Base setting to the factory default.

1. Press the **MENU** key.
2. Press **▲** or **▼** key repeatedly to choose **BS SETTING**, then press the **MENU** key.
3. Press **▲** or **▼** key to choose **BS DEFAULT**, then press the **MENU** key. You will be prompted to enter the PIN code.
4. Enter the PIN code (the default PIN is 0000), then press the **MENU** key. If the PIN code is validated all Base settings will be returned to the factory default, otherwise the settings will remain.

Result: After resetting, the handset will return to standby mode.

■ Phone Configuration

1. DSP Configuration

PHONE			
DSP CALL SERVICE DIGITAL MAP			
DSP Configuration			
First Codec	g711Ulaw64k	Second Codec	g723
Third Codec	g729	Fourth Codec	g711Alaw64k
Fifth Codec	g711Alaw64k	Handdown Time	1000 ms
Input Volume	3 (1-9)	Output Volume	7 (1-9)
G729 Payload Length	20ms	Signal Standard	China
G722 Timestamps	160/20ms	G723 Bit Rate	5.3kb/s
VoIP Prefix:	*	VoIP First:	<input checked="" type="checkbox"/>
VAD	<input type="checkbox"/>		
APPLY			

First Codec **g711Ulaw64k** Configure to select the first priority audio coding rule of DSP.

Second Codec **g723** Configure to select the second priority audio coding rule of DSP.

Third Codec **g729** Configure to select the third priority audio coding rule of DSP.

Fourth Codec **g711Alaw64k** Configure to select the forth priority audio coding rule of DSP.

Fifth Codec **g711Alaw64k** Configure to select the fifth priority audio coding rule of DSP.

Input Volume **3 (1-9)** Configure handset in volume.

Output Volume **7 (1-9)** Configure handset out volume.

Delete Length (optional)

Configure the replacing length, replace the number that user input according to this length; this is optional configuration item.

Of which the alias can be divided into four types, it should be combined with replacing length to make the setup:

Add: xxx, add xxx before number. User can save the dialling length in this way.

All: xxx, the number is all replaced by xxx; speed dialling can be implemented, for example, user configure the dialling number as 1, with the configuration "all", the actual calling number will be replaced.

Del: delete n bit in the front part of the number, n lies on the replacing length; this configuration can decide the protocol for appointed number.

Rep: xxx, n bit in the front part of the number will be replaced. N is decided by the replacing length. For example, user want to dial PSTN (010-62281493) by VoIP's voice over service, while actually the called number should be 8610-62281493, then we can configure called number as 010T, then rep: 8610, and then set the replacing length as 3. So that when user make a call with 010 prefix, the number will be replaced as 8610 plus the number and then sent out. It is a convenient thinking mode for user to make a call.

183 ▼













Delete

Modify

Delete or modify the selected phone number.

Setting Alarm

You are able to set the alarm on the handset and adjust the alarm settings.













1. Press the  key.
2. Press  or  key repeatedly to choose **HS SETTING**, then press the  key.
3. Press the  key again to choose **ALARM**.
4. The display shows the current set. Use  or  key to select **ON/OFF**, press the  key.
5. If you select **ON**, the display indicates you to enter the time in **HH:MM** format.
6. Press the  key, a beep sounds and the display shows **SNOOZE ON/OFF**, you can press  or  key to enable or disable the snooze function on the alarm and press the  key.
7. When the alarm sounds, press any key to switch it off.

Notes:

- At step 5 you need to enter the time in 24-Hour format.
- If you press any key at the right time the alarm to ring, the alarm will be silent.
- If you choose **SNOOZE ON**, the alarm will ring at regular intervals of 10 minutes.












Audio Setup

You can also use the menu to adjust the audio volume.

1. Press the  key.
2. Press  or  key repeatedly to choose **HS SETTING**, then press the  key.
3. Press the  key to choose **AUDIO SETUP**, then press the  key.
4. Use  or  key to choose **SPEAKER VOL** or **EARPIECE V**, press the  key.
5. The display show the current setting. Use  or  key to choose the volume level (VOLUME 1- VOLUME 5), press the  key.

■ Key Tone

Every time you press a key, your handset acknowledges it with a key tone. You can disable the key tones for a silent use. In certain error conditions, a warning tone will sound when an incorrect key is pressed.












1. Press the  key.
2. Press  or  key repeatedly to choose **HS SETTING**, then press the  key.
3. Use  or  key to choose **TONE SETUP**, then press the  key.
4. Press the  key again to choose **KEY TONE**.
5. You can use  or  key to choose **ON** or **OFF** to enable or disable the function.
6. Press the  key.

Note:





- For normal use, we recommend you leave the key tone enabled. This makes the phone easier to use.


■ Setting Date & Time

Date Format

1. Press the  key.
2. Press  or  key repeatedly to choose **HS SETTING**, then press the  key.
3. Press  or  key to choose **DATE & TIME**, then press the  key.
4. Press the  key again to choose **DATE FORM**.
5. You can use  or  key to choose the Date format (**DD-MM/MM-DD**), press the  key to save.

Setting date & time

1. Press the  key.
2. Press  or  key repeatedly to choose **HS SETTING**, then press the  key.

Add Dial Peer	
Phone Number	<input type="text"/>
Destination (optional)	<input type="text"/>
Port(optional)	<input type="text"/>
Alias (optional)	<input type="text"/>
Call Mode	SIP 
Suffix(optional)	<input type="text"/>
Delete Length (optional)	<input type="text"/>

Phone Number It is to add outgoing call number, there are two kinds of outgoing call number setup: One is exactitude matching, after this configuration has been done, when the number is totally the same with the user's calling number, the phone will make the call with this number's IP address image or configuration; Another is prefix matching (be equivalent to PSTN's district number prefix function), if the previous N bits of this number are the same with that of the user's calling number (the prefix number length), then the phone will use this number's IP address image or configuration to make the call. When configuring the prefix matching, letter "T" should be added behind the prefix number to be distinguished from the exactitude matching; the longest length is 30 bits.

Destination (optional) Configure destination address, if it is point-to-point call, input the opposite terminal's IP address, it can also be set as domain name and resolved the specific IP address by DNS server of the phone. If no configuration has been made, the IP will be considered as "0.0.0.0.". This is an optional configuration item.

Port(optional) Configure the other party's protocol signal port, it is an optional configuration item: if nothing is input, the default of h323 protocol is 4569, the default of sip protocol is 5060.

Alias(optional) Configure alias, this is optional configuration item: it is the number to be used when the other party's number has prefix. When no configuration has been made, shown as **no alias**.

Suffix(optional) Configure suffix, this is optional configuration item: it is the additive dial-out number behind the number; when no configuration has been made, shown as **no suffix**.

5. Dial Peer

Function of number IP table is one way to implement the phone's calling online, and the calling of the phone will be more flexible by configuring the number IP table. For example, user know the other party's number and IP and want to make direct call to the party by point-to-point mode: the other party's number is 1234, make a configuration of 1234 directly, then the phone will send the called number 1234 to the corresponding IP address; Or set numbers with prefix matching pattern, for example, user want to make a call to a number in a certain region (010), user can configure the corresponding number IP as 010T- protocol- IP, after that, whenever user dial numbers with 010 prefix (such as 010—62201234), the call will be made by this rule.

Bases on this configuration, we can also make the phone use different accounts and run speed calling without swap.

When making deletion or modification, select the number first, then click **Delete** or **Modify** to complete the operation.

The screenshot shows a web-based configuration interface for VOIP. At the top, there are tabs for SIP, TAXI, STUN, and DIAL PEER. Below the tabs is a table titled "Dial Peer Table" with columns: Number, Destination, Port, Mode, Alias, Suffix, and Del Length. Below the table is a form titled "Add Dial Peer" with fields for Phone Number, Destination (optional), Port (optional), Alias (optional), Call Mode (set to SIP), Suffix (optional), and Delete Length (optional). A Submit button is located below the form. At the bottom, there is a "Dial Peer Option" section with a dropdown menu and Delete and Modify buttons.

The following figure will be shown at the lower part of the page:

3. Press or key to choose **DATE & TIME**, then press the key.
4. Press or key to choose **SET DATE** or **SET TIME**, press the key.
5. Enter the current date or time by numeric keypad and press the key to save and exit.

Note:

- At step 5 you need to enter the time in 24-Hour format.

Selecting a Base

This function allows you to select a base from those already registered to the handset. You can access the options by scrolling the menu.

1. Press the key.
2. Press or key repeatedly to choose **HS SETTING**, then press the key.
3. Press or key to choose **SELECT BS**, press the key.
4. You can use or key to select the base you want, then press the key. If you select a non-existent base, the handset will sound a warning tone.
5. The display will show **SELECT BS X**. Press the key to confirm. A confirm tone will be heard.

HS Reset

This function allows you to reset the Handset settings to the factory default.

1. Press the key.
2. Press or key repeatedly to choose **HS SETTING**, then press the key.
3. Press or key to choose **HS DEFAULT**, press the key. You will be prompted to enter the PIN code.
4. Enter the PIN code (the default one is 0000), then press the key. If the PIN code is validated all handset settings will return to the factory default, otherwise a warning tone will be heard and the settings will remain.

Result: After resetting, the handset will return to standby mode.

Registration

The supplied handset is already registered with the number 1. You can register up to 5 handsets to your base unit in order to share the same line with other people.

Before registering a handset to the base, you should press and hold the **Page** key on the base unit for about 5 seconds until the In use indicator starts flashing rapidly.

1. Press the **MENU** key.
2. Press **▲** or **▼** key repeatedly to choose **REGISTER**, then press the **MENU** key.
3. Use **▲** or **▼** key to scroll through the Base names list, press the **MENU** key to choose a base to your necessary. The handset will search for the requested base and the relevant information will flash on the LCD.
4. If the base is found, and the handset is successfully registered, you will be prompted to enter the PIN code (the default one is 0000). On validation of the PIN code, the Registration tone will sound and the Base assigns a number to the handset. If the PIN code entered is invalid, a warning tone will sound, and the handset will return to the previous registration state.
5. If the base is not found, the handset will behave according to the Out of Range condition.

4. SIP Stun Configuration

The screenshot shows the VOIP menu with the following settings:

STUN Set	
STUN NAT Transverse	FALSE
STUN Server Addr	
STUN Server Port	3478
STUN Effect Time	50 Seconds
Local SIP Port	5060
APPLY	

Set Sip Line Enable Stun	
SIP 1	Load
Use Stun	<input type="checkbox"/>
APPLY	

STUN NAT Transverse **FALSE** Shows stun NAT transverse judgement. True means transverse, false means it cannot be transverse.

STUN Server Addr If you have stun server, please input stun server address here.

STUN Server Port Set the stun server port here, the default is 3478.

STUN Effect Time **Seconds** The unit is second. If you have stun server, please input interval time for stun's detection on NAT type.

Local SIP Port Configure local SIP port.

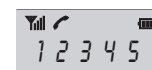
Use Stun Configure to enable/disable the phone to use STUN transverse.

- Voice Mail Text** Set the voice mail name.
- Echo Test Number** The number replace the Loop Back text number. User dial the test number to check the IAX2 voice quality.
- Echo Test Text** Loop back test text number.
- Refresh Time** **Seconds** Set the refresh time of IAX2 server register, default is 60 seconds. We recommend you make a choice from 60 to 3600.
- Enable Register** Configure enable/disable register.
- Enable G.729** Enable or disable the code G.729a/G729b. It is special for some PC software G729 compatibility.

Calling with more than one Handset

Making an Internal Call

1. Press the **INT** key, the LCD displays as follows, then enter the number of the handset you want to call.



2. If the destination handset is not available, the handset will return to standby mode. If the handset entered is valid, the called handset will ring and display **HSX CALLING**. The handset user can press the **ANS** key to answer the internal call.
3. At the end of your conversation, press the **END** key.

Note:

- During the internal call, if an external call comes in, the LCD will show the caller's number. You should press the **END** key to end the internal call first then press the **ANS** key to answer the call.

Call Transfer between Handsets

You can transfer an external call between the handsets those are registered to the same base.

1. During an external call, press the **INT** key then enter the number of the handset you want to transfer the call to.
2. If the called party can answer your call, he/she can press the **ANS** key to talk with you. At this time you can inform him/her of the incoming call.
3. When the called handset answers, press the **END** key or replace the handset to the base to complete the transfer.

■ Setting up a 3-way Conference Call

When you have both an external call and an intercom call in progress, you can set up a 3-way conference call, like this:

1. During an external call, press the **[INT]** key and then the number of the handset you want to join the 3-way call.
2. The called party press the **[📞]** key to answer.
3. The calling party can press the **[*]** key to start the 3-way call.
4. Either of the two internal handsets can press the **[📞]** key to leave the conference call at any time.

Note:

- If one of the handsets drops the conference call, the remaining handsets will continue the conference.

3. IAX2 Configuration

IAX2	
Register Status	Unregistered
IAX2 Server Addr	
IAX2 Server Port	4569
Account Name	
Account Password	
Phone Number	
Local Port	4569
Voice Mail Number	0
Voice Mail Text	mail
Echo Test Number	1
Echo Test Text	echo
Refresh Time	60 Seconds
Enable Register	<input type="checkbox"/>
Enable G.729	<input type="checkbox"/>
IAX2(Default Protocol)	<input type="checkbox"/>

IAX2 Server Addr Set IAX2 register server IP address or domain.

IAX2 Server Port Default port is 4569.

Account Name Set IAX2 register server account username.

Account Password Set password of IAX2 register server account.

Phone Number Set assigned phone number (usually it is same as account name).

Local Port Set local signal port, the default is 4569.

Voice Mail Number Configure the voice mail number. If IAX2 supports voice mail and the voice mail name is letter format, but you cannot input any letters via your phone, this number can be used to replace the voice mail name.

Enable DNS SRV Support the phone to search the domain name via DNS server by the manner of "_sip.udp".

Signal Encode Configure to enable/disable the phone to support signal encode.

Rtp Encode Configure to enable/disable the phone to support Rtp encode.

Enable Session Timer Configure to enable/disable the phone to support RFC4023.

Answer With Single Codec Configure the phone to answer with single codec.

Auto TCP If selected, when the main body of message exceeds 1300 byte, TCP protocol will be used automatically to transmit this message. Guarantee the availability of transmission.

Enable Strict Proxy Configure to make special server compatible.

Enable GRUU Configure to enable GRUU.

Enable Displayname Quote In order to make special server compatible. If selected, when sending signal, the display name will be enclosed by the sign " ", such as "display name".

Enable Subscribe If selected, you can subscribe information like others' status or voice memo after successful register.

Caller ID on Call Waiting

When you subscribe to Call Waiting service from your local telephone company, the telephone will display the name and number of the second caller while you are having a conversation.

Caller ID info displayed

Caller 1
4361234

Caller two's information is displayed

Caller 1
4
Caller 2
2915678

1. When you are on the line, the telephone will automatically display the name and number of the second caller.
2. Press the key to answer the second caller.
3. When you have finished, press the key to continue with your conversation with the first caller.

Note:

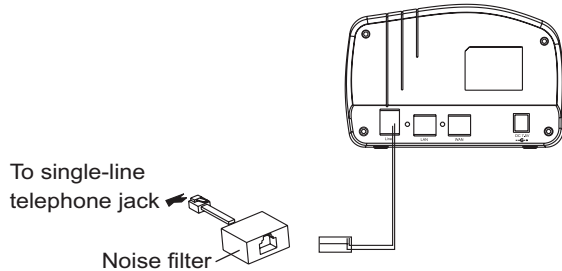
- If you have transferred an external call to another handset, you cannot use the call waiting function via the second handset.



If you subscribe to a DSL service

Please attach a noise filter (contact your DSL provider) to the telephone line between the base unit and the telephone line jack in the event of the following:

- Noise is heard during conversations.
- Caller ID features do not function properly.



RFC Protocol Edition Enable the phone to use protocol edition. When the phone needs to communicate with phones which are using SIP1.0 such as CISCO5300 and so on, then it should be configured into RFC2543 to communicate normally. The default is RFC3261.

Transport Protocol Select the transport protocol.

RFC Privacy Edition Configure the phone to use Anonymous to call out safely or not. Support the RFC3323 and RFC3325.

Transfer Expire Time seconds Configure the transfer expire time.

Subscribe Expire Time seconds Configure the subscribe expire time.

Hot Line Number Configure the hot line number.

Enable Keep Authentication Enable/disable the phone to send register request which keeps authentication. If you enable it, the server will send the confirm message directly after the register request has been received.

NAT Keep Alive Configure an auto-detect server.

Enable Via rport Configure to support RFC3581.

Enable PRACK Configure to enable/disable the phone to support Prack function of SIP. Recommend you to use the default setting.

Long Contact Configure to enable the field "**Contact**" to carry more parameters.

Enable URI Convert Enable the URI to convert "#" to "%23".

Dial Without Register Enable/disable to dial without register.

Ban Anonymous Call Enable/disable to ban anonymous calls.

2. Advanced SIP Configuration

Advanced SIP Setting			
Register Expire Time	60 seconds	Forward Type	Off
NAT Keep Alive Interval	60 seconds	Forward Phone Number	
User Agent	Voip Phone 1.0	Server Type	COMMON
Signal Key		DTMF Mode	DTMF_RELAY
Media Key		RFC Protocol Edition	RFC3261
Local Port	5060	Transport Protocol	UDP
RFC Privacy Edition	NONE	Transfer Expire Time	0 seconds
Hot Line Number		Subscribe Expire Time	300 seconds
Conference Number		Enable Conference Number	<input type="checkbox"/>
Enable Subscribe	<input type="checkbox"/>	Enable DNS SRV	<input type="checkbox"/>
Enable Keep Authentication	<input type="checkbox"/>	Signal Encode	<input type="checkbox"/>
NAT Keep Alive	<input type="checkbox"/>	Rtp Encode	<input type="checkbox"/>
Enable Via rport	<input checked="" type="checkbox"/>	Enable Session Timer	<input type="checkbox"/>
Enable PRACK	<input type="checkbox"/>	Answer With Single Codec	<input type="checkbox"/>
Long Contact	<input type="checkbox"/>	Auto TCP	<input type="checkbox"/>
Enable URI Convert	<input checked="" type="checkbox"/>	Enable Strict Proxy	<input type="checkbox"/>
Dial Without Register	<input type="checkbox"/>	Enable GRUU	<input type="checkbox"/>
Ban Anonymous Call	<input type="checkbox"/>	Enable Displayname Quote	<input type="checkbox"/>
<input type="button" value="APPLY"/>			

Register Expire Time 60 seconds Configure expire time of SIP server register, the default is 60 seconds.

NAT Keep Alive Interval 60 seconds Configure NAT keep alive interval time of server, default is 60 seconds.

User Agent Voip Phone 1.0 Set the user agent if have, default is Voip Phone 1.0.

Signal Key Set the signal key.

Media Key Set the media key.

Forward Type Off Select the forward type.

Forward Phone Number Set the forward number.

Server Type COMMON Select the type of signal encryption.

DTMF Mode DTMF_RELAY DTMF sending mode configuration, three kinds for selection. Different ISP may provide different modes.

Part Two: VoIP Phone

Your new VoIP phone is a stand-alone device, which requires no PC to make Internet calls. It supports both data and voice thru IP network, and also provides features of conventional phone. Your VoIP phone guarantees clear and reliable voice quality on IP network. It can be used thru Internet phone service to make basic Internet calls. It is fully compatible with SIP industry standard. It can interoperate with many other SIP or H.323 compliant devices and software in the market, too.

Before you can connect the phone to the network and use it, you must have a high-speed Internet connection installed. A high-speed connection includes environments such as DSL, cable modem and a leased line.

Installation

1. Remove the LAN cable for Internet connection from your PC and connect it to "WAN" port of the base.
2. Connect the power adapter in the box to the power port.
3. Find the ethernet cable in the box and connect between "LAN" port and your PC (PC is not required for set up or making a call) .

Operations

Making an IP Call

To dial an IP number, do as follows:

1. Pick up the handset and press the  key.

Result: The **In use** indicator blinks.

2. Dial a telephone number.

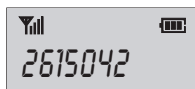
3. To end the call, either press the  key or replace the handset on the base.

Note:


- To make a call to the last number you dialed, use the **Redial** feature.

You can enter the desired phone number in standby mode, which allows you to make corrections before dialing. Follow these steps:

1. Enter an telephone number (up to 32 digits). Check it. You can also use the  key to insert a pause.



Note:

- If you make a mistake while entering a number, press the  key to clear the last digit and correct the number.

2. When the number appears correctly, press the  key.

Account Name Configure SIP register account.

Password Configure password of SIP register account.

Phone Number Set the assigned phone number.

Proxy Server Address Configure proxy server IP address (usually SIP will provide user with service of proxy server and register server which have the same configuration, so the configuration of proxy server is usually the same with that of register server, but if the configurations of them are different (such as different IP addresses), then each server's configuration should be modified separately).

Proxy Server Port Set SIP proxy server signal port.

Proxy Username Set the proxy server account user name.

Proxy Password Set the proxy server password.

Domain Realm Enter the sip domain if needed, otherwise the phone will use the proxy server address as sip domain. (Usually it is the same as registered server and proxy server IP address.)

Enable Register Usually this option needs to be selected when you want to use SIP1.

Advanced Set Start the advanced SIP configurations. Click this button to enter the advanced SIP configurations.

VoIP Configuration

1. SIP Configuration

The screenshot shows the VOIP configuration interface. At the top, there are tabs for SIP, FAX2, STUN, and DIAL PEER. The SIP tab is selected. Below the tabs is the 'SIP Line Select' section with a dropdown menu set to 'SIP 1' and a 'Load' button. Below that is the 'Basic Setting' section, which is a table with two columns. The first column contains fields for Register Status, Server Name, Server Address, Server Port, Account Name, Password, and Phone Number. The second column contains fields for Display Name, Proxy Server Address, Proxy Server Port, Proxy Username, Proxy Password, Domain Realm, and an 'Enable Register' checkbox. The Register Status field is highlighted in red and contains the text 'Unapplied'. At the bottom of the table is an 'APPLY' button, and below that is an 'Advanced Set' button.

This is a close-up of the 'SIP Line Select' section. It shows a dropdown menu with 'SIP 1' selected and a 'Load' button to its right.

Select the lines number of register SIP accounts, 1-2 for selection, click **Load** to take effect.

This is a close-up of the 'Register Status' field. The text 'Unapplied' is displayed in red.

SIP register state. When registering to the server, if register successfully, it shows "Registered" on the right. If the server refuse the request, it shows "Failed with xxx". If registration overtime, it shows "Time Out". If the server is trying registering, it shows "Trying". If you don't activate register, it shows "Unapplied". If you forget to configure the Phone number, it shows "System Error".

This is a close-up of the 'Display Name' input field. The field is empty.

Configure the display name, to display on the LCD of the other party's phone. Enables to input in English.

This is a close-up of the 'Server Address' input field. The field is empty.

Configure SIP register server IP address.

This is a close-up of the 'Server Port' input field. The value '5060' is entered in the field.

Configure SIP register server signal port.

Last Number Redial

1. Press the **☎** key in standby mode.

Result: The LCD displays the last number you dialed.

2. Press the **☎** key to dial out the number.

Your phone allows you to retrieve the last 10 numbers you have dialed and recall them quickly.

Search for and dial a number in Redial Book

1. Press the **☎** key in standby mode.

2. If you want to scroll through the memory, press **▲** or **▼** key repeatedly until you find out the number you want to dial.

3. Press the **☎** key to dial out the number.

Notes:

- If no numbers are found, "EMPTY" is displayed.
- When the redial book is full, each time you dial a new number, the oldest number stored in the redial book is automatically erased.

Use Redial Book options

Using the redial book options, you can add a number to phone book or delete a number in the redial book.

1. Press the **☎** key.

2. Scroll to the desired number by using the **▲** or **▼** key.

3. When the number appears on the display, press the **MENU** key to choose the desired option:

Add to PB: allows you to add the number to the phone book. Complete storing the entry by starting from step 4 described in par."Adding the Phone Book Entries" (obviously, you will find the number already entered in the appropriate field).

Delete: allows you to delete the selected number.

Delete All: allows you to delete the whole redial book.

4. To exit, press the **C/M** key.

■ Phone Book










The phonebook built in your cordless phone allows you to store frequently used numbers so that you can easily make a call without having to remember or enter the phone number.

Character Map

To enter a specific alphanumeric character, press one or more times the relevant key for the required character according to the following table: once for the first character, twice for the second and so on.

Key	Characters in the displayed order
0	(spc) 0 & / .
1	- 1 @ _
2	A b C 2
3	d E F 3
4	G H I 4
5	J K L 5
6	M N O 6
7	P q R S 7
8	T U V 8
9	W X y Z 9

Adding the phone book Entries

1. Press the  key.
2. Press the  key to choose **PHONE BOOK**, press the  key.
3. Press  key again to choose **ADD ENTRY**.
4. The display show "**NAME ?**". Enter the name you want to use, then press  key to confirm. The display shows "**NUMBER ?**". Enter the desired number (up to 20 digits), press the  key.
- 5.*Use  or  key to select the ring you like to associate with the number, press the  key to confirm.
6. Repeat if necessary.

SNTP Time Set	
Server	209.81.9.7
Time Zone	{ (GMT+08:00)Beijing,Chongqing,Hong Kong,Urumqi
Time Out	60 (seconds)
SNTP	<input checked="" type="checkbox"/>

Server: The server IP address.

Time Zone: Set time zone according to your location, the default is Beijing time zone.

Time Out: Use for socket calls which will block while doing the SNTP query, default value is 60 seconds. Select **SNTP** then click **APPLY** to take effect.

Manual Timeset	
Year	<input type="text"/>
Months	<input type="text"/>
Day	<input type="text"/>
Hour	<input type="text"/>
Minute	<input type="text"/>

To set the time manually. All the options should be filled in to complete the setting.

Lease Time	<input type="text"/>	(minute) DHCP server lease time.
Netmask	<input type="text"/>	Netmask of lease table.
Gateway	<input type="text"/>	Default gateway of lease table.
DNS	<input type="text"/>	Default DNS server of lease table.

Click **Add** to submit to add DHCP lease table.

DHCP Lease Table Delete	
Lease Table Name	lan <input type="button" value="Delete"/>

Select the lease table you want to delete, click **Delete** to delete it from DHCP Lease Table.

DNS Relay <input checked="" type="checkbox"/>	Configure the DNS Relay way of the phone. The default is activate it. Select it, then click APPLY to take effect.
---	--

6. Time Setting (SNTP)

NETWORK	
WAN LAN QOS SERVICE PORT DHCP SERVER SNTP	
SNTP Time Set	
Server	209.81.9.7
Time Zone	(GMT+08:00)Beijing,Chongqing,Hong Kong,Urumqi
Time Out	60 (seconds)
SNTP	<input checked="" type="checkbox"/>
<input type="button" value="APPLY"/>	
Manual Timeset	
Year	<input type="text"/>
Months	<input type="text"/>
Day	<input type="text"/>
Hour	<input type="text"/>
Minute	<input type="text"/>
<input type="button" value="APPLY"/>	

* If you want to set a VIP number, you should select a special ring tone for it. The ring tone should be different from the external ring you have set. You can set up to 9 VIP ring tones associate with 9 VIP numbers.

Note:

- If you make a mistake while entering a number, use the key to correct it. Each time you press the key, the last digit is deleted. To clear all digits, press and hold the key.

Viewing the Phone book Entries

- In the Standby mode, press the key.
- Press or key repeatedly until the desired name is displayed.
- Press the key to show the relevant number.

Dialing a Number from Phone book

Find out the number you want to dial. Refer to "Viewing the Phone book Entries". Then press the key to dial out the number.

Editing the Phone book Entries

- Press the key.
- Press the key to choose **PHONE BOOK**, press the key.
- Press or key to choose **CHANGE ENTRY** menu. Press the key.
- Press or key repeatedly until the entry you want to edit displays, press the key to confirm.
- If necessary, press the key to clear the digit(s) then enter the desired name and number, press the key to save.
- Use or key to select the ring you like to associate with the number, press the key to confirm.

Deleting the Phone book Entries

1. Press the **☰** key.
2. Press the **☑** key to choose **PHONE BOOK**, press the **☰** key.
3. Press **⬆** or **☑** key to choose **DELETE ENTRY**. Press the **☰** key.
4. Press **⬆** or **☑** key repeatedly until the name you want to delete displays, then press the **☰** key. The display show "**CONFIRM ?**". You can press the **☰** key to delete it or press the **☒** key to exit.
5. At step 3, if you choose **DELETE ALL** menu and press the **☰** key, the display will show "**CONFIRM ?**". You can press the **☰** key to delete all the entries or press the **☒** key to exit.

PB Status

The PB Status shows the number of phone book entries already stored. When the **PB STATUS** shows on the LCD, press the **☰** key to enter it.

5. DHCP Setting

You can configure the DHCP service via this page, custom the assigning range of dynamic IP and other configurations.

The screenshot shows the NETWORK DHCP SERVER configuration page. At the top, there are tabs for WAN, LAN, QOS, SERVICE PORT, DHCP SERVER, and SNTP. The DHCP SERVER tab is selected. Below the tabs, there are four main sections:

- DHCP Leased Table:** A table with two columns: Leased IP Address and Client Hardware Address. The first row shows 192.168.10.2 and 00-50-ba-27-ce-b0.
- DHCP Lease Table:** A table with columns: Name, Start IP, End IP, Lease Time, Netmask, Gateway, and DNS. The first row shows lan, 192.168.10.1, 192.168.10.30, 1440, 255.255.255.0, 192.168.10.1, and 192.168.10.1.
- DHCP Lease Table Setting:** A form with input fields for Lease Table Name, Start IP, End IP, Lease Time (with a (minute) label), Netmask, Gateway, and DNS. There is an Add button at the bottom right.
- DHCP Lease Table Delete:** A table with columns: Lease Table Name and Delete. The first row shows lan and a Delete button.

DHCP Leased Table	
Leased IP Address	Client Hardware Address
192.168.10.2	00-50-ba-27-ce-b0

The IP-MAC contrast table of DHCP.

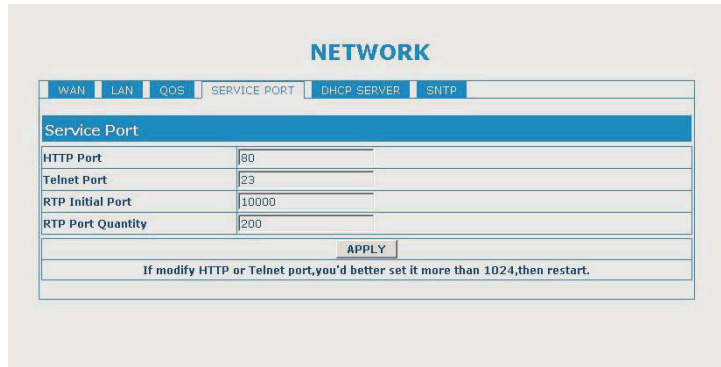
DHCP Lease Table						
Name	Start IP	End IP	Lease Time	Netmask	Gateway	DNS
lan	192.168.10.1	192.168.10.30	1440	255.255.255.0	192.168.10.1	192.168.10.1

Lease Table Name Add the lease table name.

Start IP The start IP of lease table.

End IP The end IP of lease table. Network device connecting to the phone LAN port can dynamically obtain the IP in the range between start IP and end IP.

4. Service Port



NETWORK

WAN LAN QOS SERVICE PORT DHCP SERVER SNTP

Service Port

HTTP Port	80
Telnet Port	23
RTP Initial Port	10000
RTP Port Quantity	200

APPLY

If modify HTTP or Telnet port, you'd better set it more than 1024, then restart.

Configuration Explanation:

HTTP Port 80 Configure web browse port, the default is 80 port, if you want to enhance system safety, you'd better change it into non-80 standard port.

Telnet Port 23 Configure Telnet port, the default is 23 port.

RTP Initial Port 10000 Enable RTP initial port configuration. It is dynamic allocation.

RTP Port Quantity 200 Configure the maximum quantity of RTP port. The default is 200.

※ The configuration on this page needs to be saved after modified and will take effect after restarting.

※ If the Telnet, HTTP port will be modified, you had better set the port greater than 1024, because the 1024 port system will save ports.

※ Set the HTTP port as 0, then the http service will be disabled.

Call Transfer

1. Blind Transfer

During a conversation, press the **MENU** button. After hearing the dial tone press the ***** button, then enter the number you want to transfer the call to, press the **#** button to confirm. After the transfer, your line will be disconnected automatically.

2. Attended Transfer

During a conversation, press the **MENU** button. After hearing the dial tone enter the number you want to transfer the call to, press the **#** button to confirm. After the line get through, press the **☎** button to complete the transfer. To use this feature, you should enable the "Call Waiting" and "Call Transfer" function.

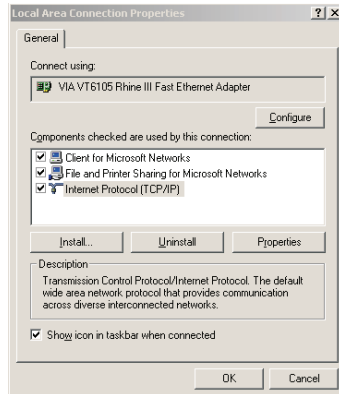
Three-way Conference Call

During a conversation, press the **MENU** button. After hearing the dial tone enter the number you want to transfer the call to, press the **#** button to confirm. After the line get through, press the ***** button to start the 3-way conference call.

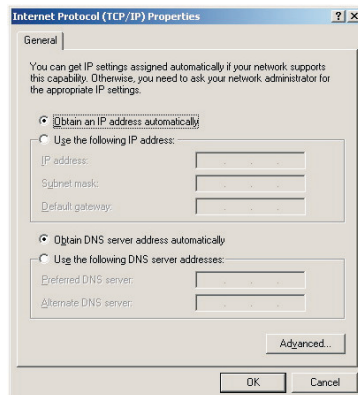
Configuration with Web

The IP Phone Web Configuration Menu can be accessed by the following default LAN IP address "192.168.10.1". Before accessing the web, you must do the following steps:

1. Open the "Local Area Connection Properties" window.



2. Select "Internet Protocol(TCP/IP)", then click the "Properties".
3. Select "Use the following IP address", enter the IP address, then click "OK".



LAN IP	192.168.10.1
--------	--------------

Configure LAN static IP.

Netmask	255.255.255.0
---------	---------------

Configure LAN subnet mask.

DHCP Service	<input checked="" type="checkbox"/>
--------------	-------------------------------------

Enable LAN port DHCP server; after user modify LAN IP, the phone will automatically modify the adjustment and save the configuration according to IP and subnet mask team DHCP Lease Table, user need to restart the phone to make DHCP server configuration go into effect.

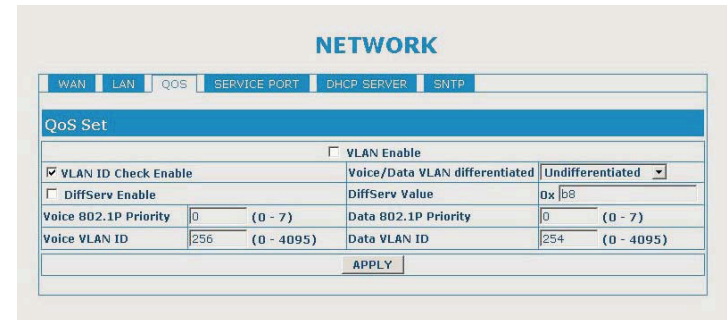
NAT	<input checked="" type="checkbox"/>
-----	-------------------------------------

Enable NAT.

Bridge Mode	<input type="checkbox"/>
-------------	--------------------------

Use bridge mode (transparent mode): bridge mode will make the phone no longer set IP address for LAN physical port, LAN and WAN will join in the same network.

3. QoS Settings



QoS Control based on 802.1p for different IP users. The QoS is used to mark the network communication priority in the data link/MAC sub-layer. The phone will sort the packets using the QoS and sends it to the destination. QoS provides service classes for accessing traffics in Internet.

DiffServ replace IP type of service, the field change to DS field. It takes IP service information that is necessary. It is strict three layer technology and does not involve the low layer transfer technology.

Gateway 192.168.1.1 Configure IP address of the gateway.

DNS Domain Configure "DNS domain" suffix. If user input "domain" and it can't be resolved, then the phone will add and resolve the "domain" after user has input.

Primary DNS 202.96.134.133 Main DNS server IP address.

Alter DNS 202.96.128.68 The second DNS server IP address.

PPPOE Server ANY Service name, if PPPoE ISP has no special requirement for this name, ANY is the default.

Username user123 PPPoE account.

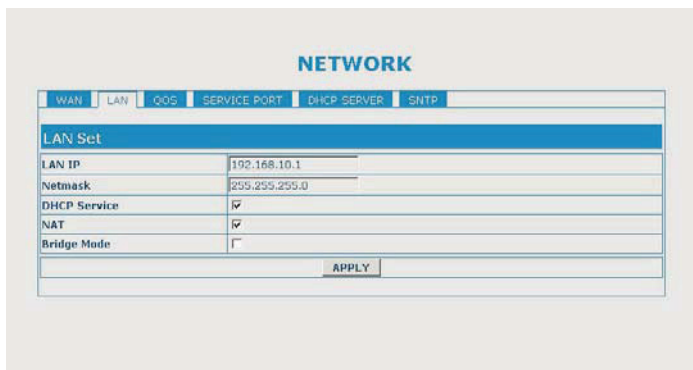
Password ***** PPPoE password.

Click "**APPLY**" button after finished above setting, the phone will save the setting with immediate effect.

Note:

- If you log on via WAN and alter the WAN IP, you must log on the new address again after submission.

2. Local Area Network (LAN)



User Verification

User should login before configuring dialogue machine.

Guest account: the default username and password are all "guest", user can only have a browse of system.

Administrator account: the default username and password are all "admin", this kind of user can configure the system.

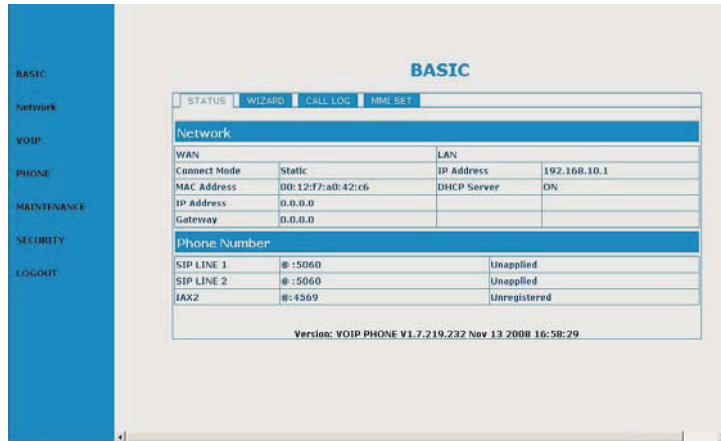
Note:

- After inputting username and password, click "**Logon**".



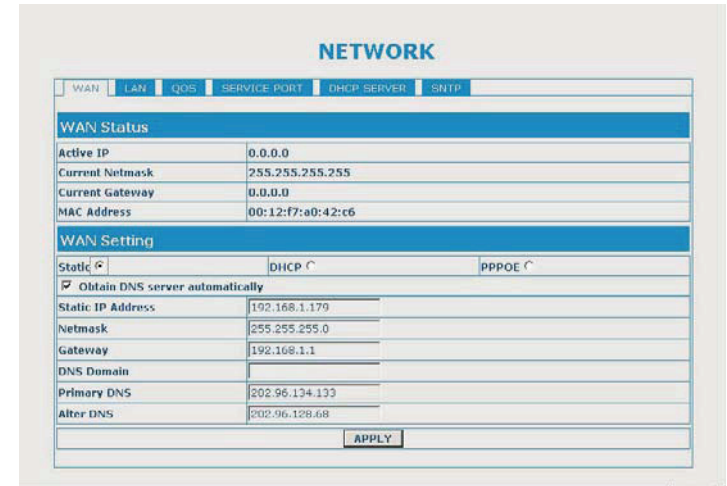
■ Configuration Menu

The web configure interface is composed of the configure menus and the corresponding configure interfaces at the right side.



■ Network Configuration

1. Wide Area Network (WAN)



You can view the current network IP linking mode of the system on this page.

User will be authorized to set the network IP, Gateway and DNS if the system adopts the static linking mode.

If the system selects DHCP service in the network which is using DHCP service, IP address will be gained dynamically.

If the system selects PPPOE service in the network which is using the PPPOE service, then the IP address will be gained by the set PPPOE ISP internet and password of the account.

Configuration Explanation:

Static DHCP PPPOE Select acquisition way of IP for WAN. This is single option, configure static IP parameter for WAN.

Static IP Address 192.168.1.179 Configure static IP address.

Netmask 255.255.255.0 Configure subnet mask.

3. PPPoE Mode

a. Select "PPPoE MODE", then click the "NEXT" button.

The screenshot shows the 'BASIC' configuration page with a navigation bar containing 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. Below the navigation bar is a blue header 'Network Mode Select'. Underneath, there are three rows of configuration options: 'Static IP MODE' with a right-pointing arrow, 'DHCP MODE' with a right-pointing arrow, and 'PPPoE MODE' with a right-pointing arrow. At the bottom of the form are two buttons: 'BACK' and 'NEXT'.

b. Configure the PPPoE parameters as the figure below, then click the "NEXT" button to complete the setting.

The screenshot shows the 'BASIC' configuration page with a navigation bar containing 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. Below the navigation bar is a blue header 'PPPOE Set'. Underneath, there are three rows of configuration options: 'PPPOE Server' with a dropdown menu set to 'ANY', 'Username' with a text field containing 'user123', and 'Password' with a text field containing '*****'. At the bottom of the form are two buttons: 'BACK' and 'NEXT'.

■ MMI Setting

You can edit the greeting message on this page.

The screenshot shows the 'BASIC' configuration page with a navigation bar containing 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. Below the navigation bar is a blue header 'LANGUAGE SELECTION'. Underneath, there is a row for 'Language Set:' with a dropdown menu set to 'English'. Below this is an 'APPLY' button. At the bottom of the page, the version information is displayed: 'Version: VOIP PHONE V1.7.219.232 Nov 13 2008 16:58:29'.

■ Current State

When you login in, you can see the current system information of the phone such as WAN settings, LAN settings, Phone numbers and Firmware version in this page.

The screenshot shows the 'BASIC' configuration page with a navigation bar containing 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. Below the navigation bar is a blue header 'Network'. Underneath, there are two sections: 'WAN' and 'LAN'. The 'WAN' section has a table with the following data:

Connect Mode	Static	IP Address	192.168.10.1
MAC Address	00:12:f7:a0:42:c6	DHCP Server	ON
IP Address	0.0.0.0		
Gateway	0.0.0.0		

The 'LAN' section has a table with the following data:

IP Address			
DHCP Server			

Below the 'WAN' and 'LAN' sections is a blue header 'Phone Number'. Underneath, there are three rows of configuration options: 'SIP LINE 1' with a dropdown menu set to '@:5060' and a status of 'Unapplied', 'SIP LINE 2' with a dropdown menu set to '@:5060' and a status of 'Unapplied', and 'IAX2' with a dropdown menu set to '@:4569' and a status of 'Unregistered'. At the bottom of the page, the version information is displayed: 'Version: VOIP PHONE V1.7.219.232 Nov 13 2008 16:58:29'.

■ Wizard Configuration

The setting contains Network Mode Select, Static IP and PPPoE Setting.

1. Static IP Mode

a. Select "**Static IP MODE**", then click the "**NEXT**" button.

The screenshot shows the 'BASIC' configuration page with a navigation bar containing 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. The main heading is 'Network Mode Select'. Below it, there are three radio button options: 'Static IP MODE' (which is selected), 'DHCP MODE', and 'PPPoE MODE'. At the bottom of the form, there are 'BACK' and 'NEXT' buttons.

b. Configure the Static IP parameters as the figure below, then click the "**NEXT**" button to complete the setting.

The screenshot shows the 'BASIC' configuration page with a navigation bar containing 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. The main heading is 'Static IP Set'. Below it, there are several input fields with the following values: 'Static IP Address' (192.168.1.179), 'Netmask' (255.255.255.0), 'Gateway' (192.168.1.1), 'DNS Domain' (empty), 'Primary DNS' (202.96.134.133), and 'Alter DNS' (202.96.128.68). At the bottom of the form, there are 'BACK' and 'NEXT' buttons.

2. DHCP Mode

a. Select "**DHCP MODE**", then click the "**NEXT**" button.

The screenshot shows the 'BASIC' configuration page with a navigation bar containing 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. The main heading is 'Network Mode Select'. Below it, there are three radio button options: 'Static IP MODE', 'DHCP MODE' (which is selected), and 'PPPoE MODE'. At the bottom of the form, there are 'BACK' and 'NEXT' buttons.

b. Configure the SIP parameters as the figure below, then click the "**NEXT**" button.

The screenshot shows the 'BASIC' configuration page with a navigation bar containing 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. The main heading is 'SIMPLE SIP SET'. Below it, there are several input fields with the following values: 'Display Name' (empty), 'Server Address' (empty), 'Server Port' (5060), 'User Name' (empty), 'Password' (empty), 'Phone Number' (empty), and 'Enable Register' (unchecked). At the bottom of the form, there are 'BACK' and 'NEXT' buttons.

c. The parameters you have configured will be shown on this page, click the "**Finish**" button to complete the setting.

The screenshot shows the 'BASIC' configuration page with a navigation bar containing 'STATUS', 'WIZARD', 'CALL LOG', and 'MMI SET'. The main heading is 'WAN'. Below it, there are several input fields with the following values: 'Connect Mode' (DHCP). Below the 'WAN' section, there is a section for 'SIP' with the following values: 'Register Server' (empty), 'Account/User Name' (empty), 'PhoneNumber' (empty), and 'Register' (OFF). At the bottom of the form, there are 'BACK' and 'Finish' buttons.