User Manual

WIP0020
DECT Phone
Version 2.1

Introduction

About This Manual

This Manual provides basic information on how to install and connect WIP0020 Dect Phone to the network. It also includes features and functions of WIP0020 Dect Phone components, and how to use them correctly. We sincerely hope you could enjoy the convenience and capabilities brought forward by our products.

Before Getting Started
Before you can connect WIP0020 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.

WIP0020 Dect Phone allows you to make and receive calls from both ordinary phone service and from IP telephony over the Internet.

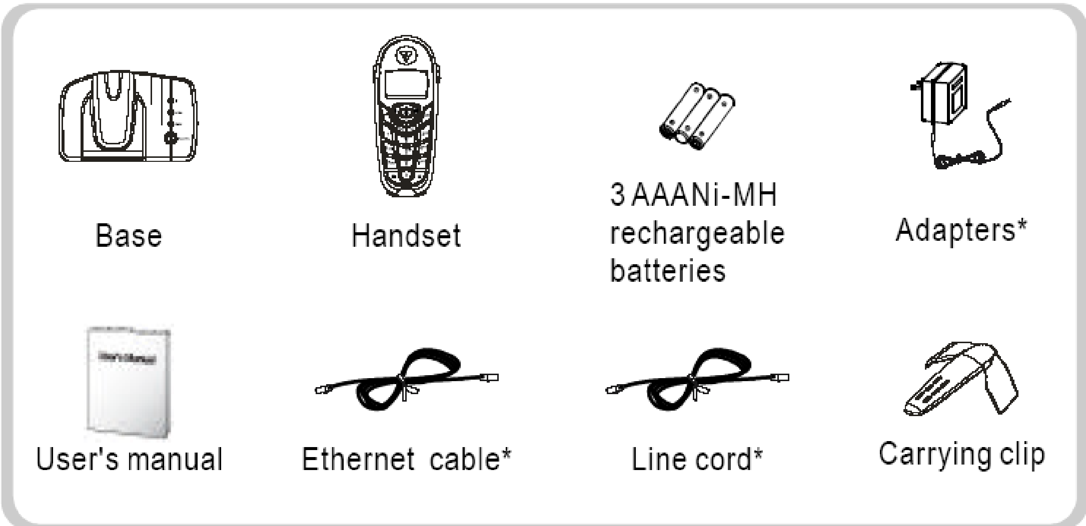
WIP0020 Dect Phone is a stand-alone device, which requires no PC to make Internet calls. WIP0020 Dect Phone supports both data and voice thru IP network, and also provides all the features and functionalities of conventional phone and more. Our IP phone guarantees clear and reliable voice quality on IP network, which is fully compatible with SIP and h.323 industry standard and able to interoperate with many other SIP or h.323 compliant devices and software on the market.

Notice
This publication describes the instruction for WIP0020 series IP phone functions only. We reserve the rights to do any changes or make enhancements of this publication without further notice. The most updated electronic revision of user manual can be downloaded from IPshop's website: www.ipshop.dk timely, thanks for your understanding and continuous support.

Before Using

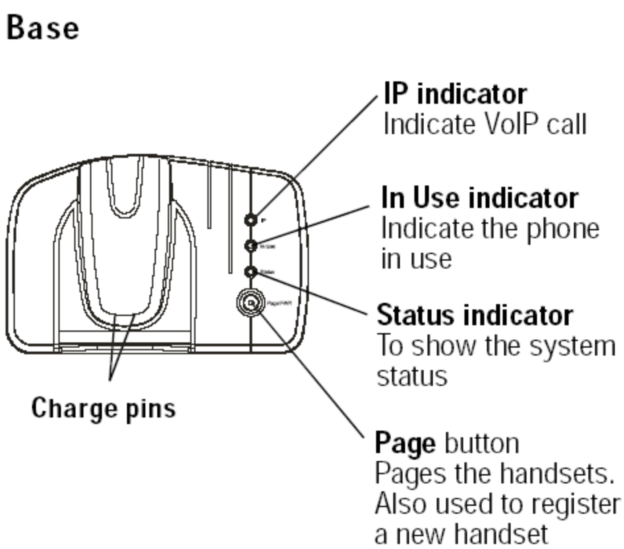
Package contents

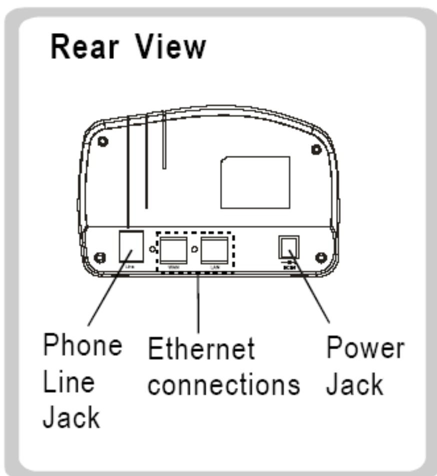
Once you have unpacked your phone, make sure that all the parts shown below are available. If any pieces are missing or broken, please promptly call your dealer.

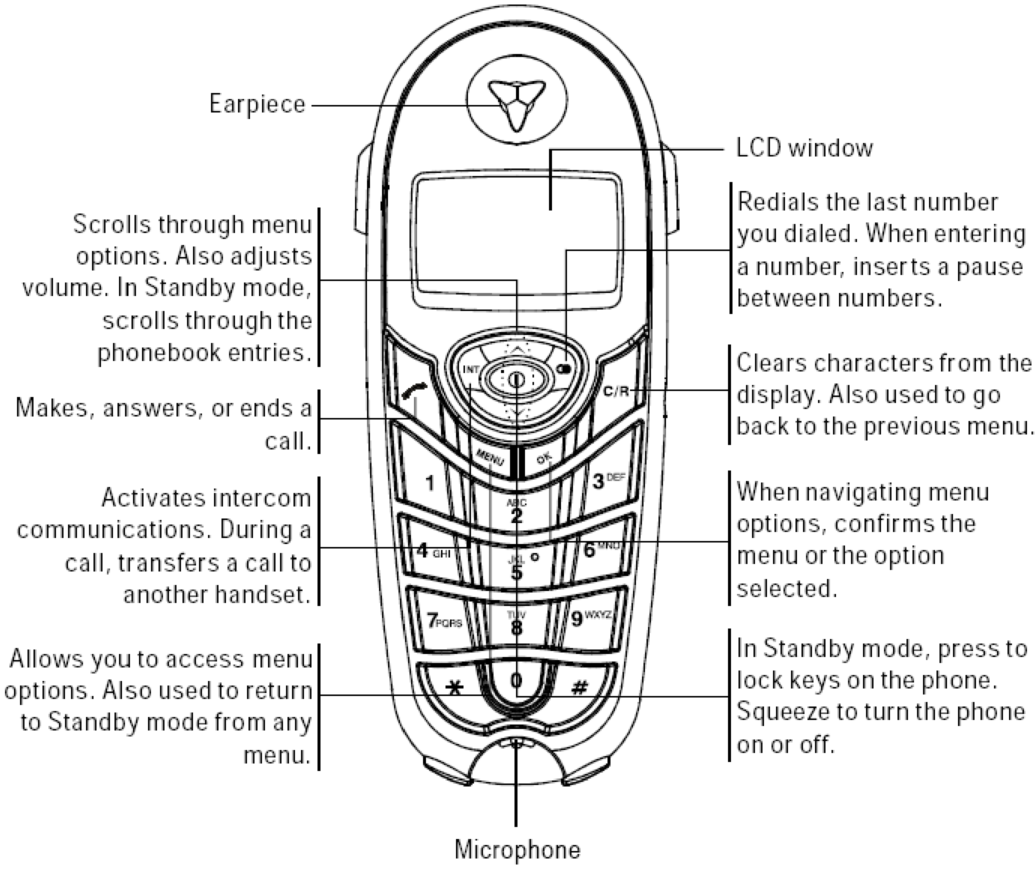


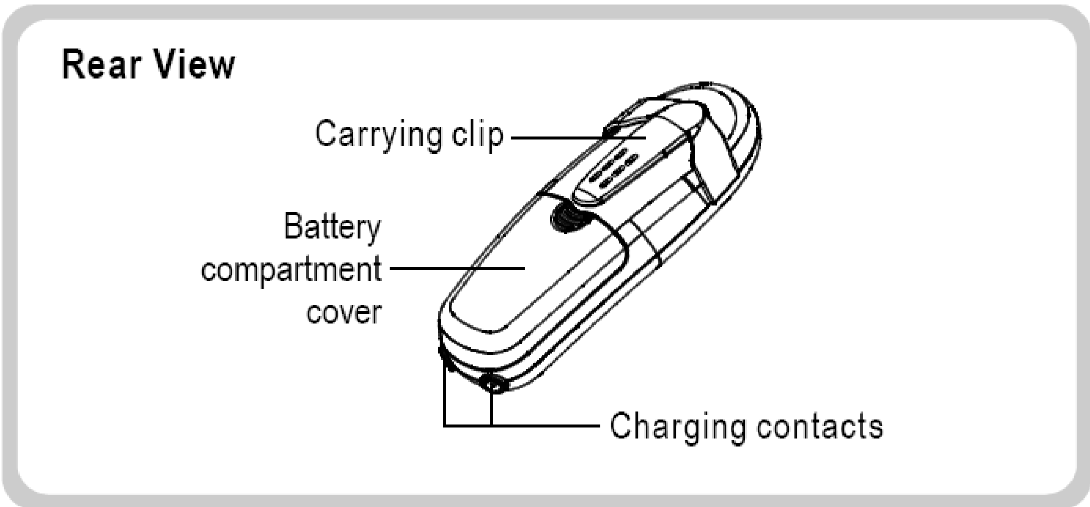
The shape of the plugs vary according to each country's specification.

Location and Function of Control

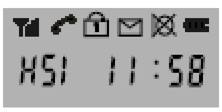








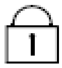
LCD Window Icon Descriptions



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, and the current time

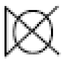
Signal Strength Icon This icon is always displayed when your phone is on, and shows the current signal strength. More bars indicate more signal strength.

Line Icon This icon indicates that the line is engaged.



Key Lock Icon This icon indicates the keys are locked.

New Call Icon This icon indicates that there is a new call. To view the caller, access the Call Log menu. See page

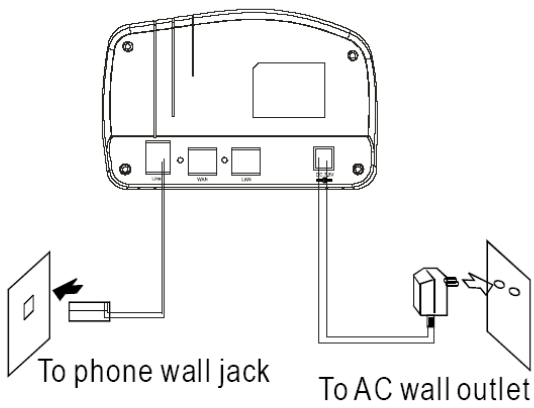


Mute Icon This icon indicates that your phone's microphone is off temporarily.

Battery Status Icon This icon is displayed at all times when your phone is on, and shows the level of your battery charge. The more bars, the greater the capacity.

Connecting Lines

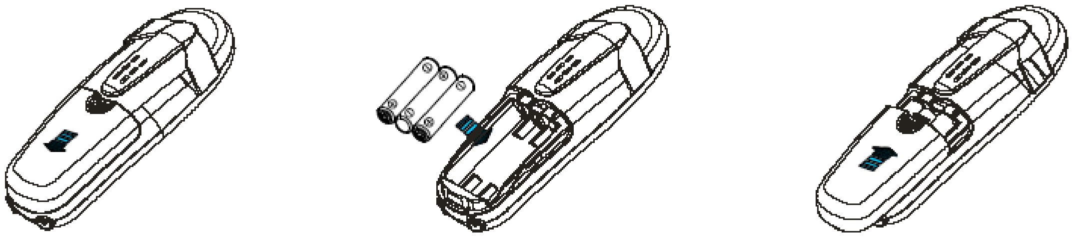
1. Connect one end of the phone line cord to the phone line jack on the bottom of the base, and the other end to a standard phone wall outlet.
2. Connect the modular end of the AC power Adapter to the power jack of the base, Then plug the AC adapter into a standard AC wall outlet.



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 oritentation may damage the handset.
3. To replace the battery cover, slide the cover up until it snaps shut.



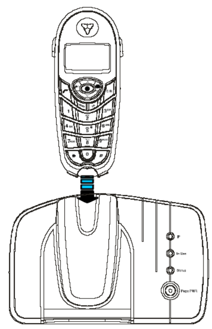
Notes:

- 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 or conventional alkaline batteries

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 charging LED on during the charge.



Using Handset Carrying Clip

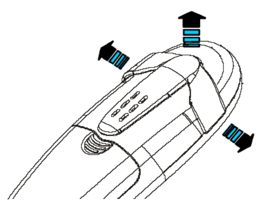
The supplied handset carrying clip allows you to conveniently carry the handset with you. It clips easily to your belt, waist hand, or shirt pocket.

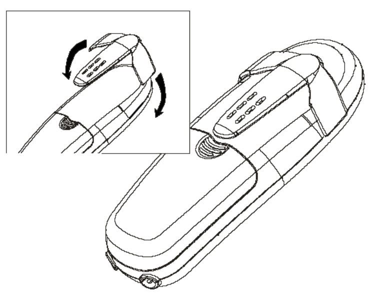
If you want to remove the carrying clip

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.

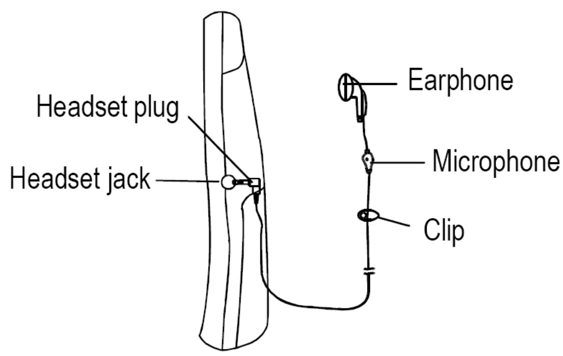




Using Headset (optional)

The headset jack is located in the middle right side of the handset and is 2.5mm standard plug. Simply plug the headset into the jack and the headset will be activated.

Note:



- When the headset is plugged into the headset jack, the microphone on the handset will be deactivated

Turning Handset On/ Off

When you place the handset on the charger, 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 button until you switch the display on.

Result: The first time you turn the phone on, the LCD window displays the handset number and the time. The phone is in Standby mode and ready for use.

If the time on the display is not correct, you can change the time. See " Setting Time" on page

2. To turn the handset off, press and hold the button until  **appears. Then the display turns off.**

Note:

- Nothing will appear in the LCD display when battery power is very low. **YOU SHOULD FULLY CHARGET THE HANDSET BEFORE USE.**

Part One: Cordless Phone

Your new VoIP (Voice over Internet Protocol) Phone can be used as a ordinary DECT (Digital Enhanced Cordless Telecommunication) Phones. 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

Basic Function

Choosing Dial Mode

In order to provide compatibility with other telephone systems, your phone can be set to either pulse dialing (same as rotary), or tone dialing (DTMF).

1. **Press the MENU button.**
2. **press the ▲ or ▼ button to choose BASE MENU, then press the OK button.**

Results: The LCD window prompts you to enter the PIN, The PIN is preset to “ 0000” at factory.

3. **Enter the PIN, then press the OK button**
4. **Press the ▼ button to choose DIAL MODE, then press the OK button.**

Result: The current setting is displayed.

5. **Press the ▲ or ▼ button to choose TONE or PULSE.**
6. **Press the OK button to save the selection.**

Setting Time

1. **Press the MENU button.**
2. **Press the ▲ or ▼ button to choose BASE MENU, then press the OK button.**

Result: The LCD window prompts you to enter the PIN, The PIN is preset to “ 0000”at the factory

3. **Enter the PIN, then press the OK button.**
4. **Press the ▲ or ▼ button to choose TIME SET, then press the OK button.**

Result: The current setting is displayed.

5. Enter the current time using the number keypad.

Notes:

- The time format is 24 hours. Select from 00 to 23.
- If you want to correct a digit in the middle while programming, use or button to move the cursor to the incorrect number, then enter a correct number.

6. Press the button to save the selection

Making a call

1. Pick up the handset and press the button.

Result: You hear a dial tone.

2. Dial a telephone number.

Note:

- You can store up to 10 telephone numbers in memory for automatic dialing. For details, **see page**

3. When the receiver answers, speak.

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

Note:

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

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

1. Enter a telephone number. Check the number in the LCD window.

Notes:

- If you make a mistake while entering a number, press the button to clear the last digit and correct the number.
- If you press the button for more than one second, all digits you have entered will be cleared and the phone returns to the Standby mode.
- When you enter the first digit(s) of the phone number, if the **Smart Dialing** feature is enabled and there are the matching numbers in the phone's memory, the number appears on the display. For details, see page

2. When the number appears correctly, press the button

Receiving a Call

When a call is received, the phone rings and the icon on the display blinks. If the caller can be identified, the caller's phone number is displayed. If the caller cannot be identified, only the icon blinks.

1. To answer the call, press any button. Or if the handset is on the base, just simply lift it up. You do not need to press any button.

Result: The phone is connected and the icon is steadily on. The LCD window displays the talk time.

2. You can speak. To end the call, either press the button or replace the handset on the base.

Note:

- After you lift the phone from the base to receive the call, the button does not work for 3 seconds to prevent the phone from being disengaged.

Switching the Microphone Off (Mute)

You can temporarily switch your phone's microphone off, so that the other party cannot hear you. Example: You wish to say something to another person in the room but do not want the other party to hear you.

To switch the microphone off temporarily, press the button during a conversation. The icon appears in the LCD window.

To switch the microphone back on, press the button again.

Adjusting Voice Volume

During a conversation, the or button adjusts the level of the earpiece volume. You can adjust the volume from level 1 to 3. The selected volume is displayed in the LCD window.

Adjusting Ring Volume

1. **Press the button.**
2. **Press the or button to choose , then press the button.**
3. **Press the or button to choose , then press the button.**

Result: The current ring level is displayed.

4. **Press the or button to choose the volume level you want.**

Result: You can adjust the volume from level 1 to 3. You can also turn the ringer OFF. Each time you press or button, the handset sounds its selected loudness.

5. Press the OK button to save the selection.

Note:

- If you set to turn the ringer OFF, When a call comes in, only the icon on the LCD blinks.

Last Number Redial

To redial the last number:

1. Press the button in Standby mode.

Result: The LCD window displays the last number you dialed.

2. Press the button to dial the number.

Your phone keeps the last 10 numbers you have dialed and allows you to retrieve the numbers.

To view and dial any of the last 10 numbers:

1. Press the button in Standby mode.

2. If you want to scroll through the memory, press the or button until you find out the number you want to call.

3. Press the button to dial the number.

Notes:

- When no numbers are found, EMPTY is displayed.
- When the redial memory storage is full, each time you dial a new number, the oldest number stored in the redial memory is automatically erased, and the redial memory is updated.
- You can modify the displayed number before pressing the button by using C/R button.

To delete a specific number in the Redial memory:

1. Press the button.

2. Scroll to the number you want to delete by using the or button.

3. When the number appears on the display, press the OK button.

Result: The display prompts you to confirm the deletion.

4. Press the OK button.

Result: You hear a beep and the phone returns to Standby mode after clearing the memory number.

To delete all numbers in the Redial memory:

1. Press the MENU button.

2. Press the or button to choose CLR REdIAL, then press the OK button.

Result: The LCD window prompts you to enter the PIN. The PIN is preset to "0000" at the factory.

3. Enter the PIN, then press the OK button.

Result: The phone prompts you to confirm the deletion.

4. Press the OK button.

Caller ID

When you receive a call, the caller's phone number is shown on the screen, if the caller's information is transmitted from the network on which the call was made. The last 10 received calls are stored in the Caller ID memory, and you can use the list to make a call to any of the numbers.

The Call ID icon appears when there are new calls in the memory. Once you view all the new calls using the Call Log menu, the icon will disappear.

To review and dial any of the received numbers:

1. **Press the MENU button.**
2. **Press the or button to choose CALL LOG, then press the OK button.**

Result: The LCD window shows the last caller's number (up to 20 digits). The time when the call was received is also displayed. If there is no caller ID received, EMPTY is displayed.

3. **Scroll the Caller ID numbers by using the or button and choose the desired number.**
4. **To dial the selected number, press the button.**

To delete a specific number from the Call Log:

1. **Scroll to the number you want to delete from the Caller ID memory, then press the OK button.**
2. **Press the OK button to confirm the deletion.**

To delete all numbers in the Caller ID memory:

1. **Press the MENU button.**
2. **Press the or button to choose CLR CALL LOG, then press the OK button.**

Result: The LCD window prompts you to enter the PIN. The PIN is preset to "0000" at the factory.

3. **Enter the PIN, then press the OK button.**

Result: The phone prompts you to confirm the deletion.

4. **Press the OK button.**

Battery Level Indicator

The icon is continuously displayed at the top right corner of the LCD window. The icon shows the level of battery power. The more bars you see, the more power you have left.

When the battery is too low for the phone to operate, the handset will automatically turnoff. You should place the handset on the base to charge the handset battery.

Key Lock

If you switch this feature on, all buttons except for the button will be locked and will not function.

You can answer incoming calls or intercom calls by using any buttons on the phone. But when you hang up, the phone returns to the lock mode. This feature is useful to avoid pressing buttons by mistake.

1. To switch the feature on, press the button briefly and the icon appears in the LCD window.

2. To switch the feature off, press the button briefly and the icon disappears from the LCD window.

Notes:

- If a power failure occurs, the Key Lock feature is automatically cancelled.
- Do not hold the button for more than 3 seconds, or the handset will be turned off.

Smart Dialing

With the Smart Dial feature, your phone displays the full phone number when you enter the first digit(s) of the number provided that the number is currently stored in your phone's memory (for example, stored in the phonebook, received or dialed recently).

To place a call using the smart dial feature:

1. In Standby mode, enter the first digit(s) of the desired phone number.

Result: The phone searches for the matching phone number from memory (in the order of outgoing calls, Phonebook, incoming calls), then displays the first available number. Numbers blink except for the part of the number you entered.

2. If the number is not the one you want, enter the remaining digits until you see the desired number.

3. Press the button to dial the number displayed. Or press the button to exit this smart dial mode.

You can select to enable or disable the Smart Dialing feature. To change the setting:

1. Press the button.

2. Press the or button to choose SMART dIAL, then press the button.

Result: The current setting is displayed.

3. Press the or button to choose On or OFF. To enable this feature, select On.

4. Press the button to save the selection.

Out of Range Indication

If the handset is too far from the registered base, the handset cannot properly engage the telephone line. The icon at the top left of the LCD window blinks to warn you and the signal strength indication bars next to the icon disappear.

If you carry the handset too far from the base unit during a call, the telephone line might be disconnected and the handset returns to Standby mode. Check if there is noise or static. If so, move the handset closer to the base station.

Registered Recall

During a conversation, the C/R button is used to end the call for making a new call or to local transfer a call to another parallel if under PABX.

To flash, simply press the C/R button while the line is engaged.

Result: The display shows F.

Tone Dialing Switchover

To access certain services such as voice mail or interactive telephone system, it is necessary to use tone dialing. When your phone is set to the pulse mode, DTMF dialing is available temporarily.

Press the * button while the phone is in the pulse dial mode. Your phone is converted to the tone dial mode temporarily.
When you hang up the call, the mode returns to the pulse mode.

Paging

You can page the handset from the base unit. Using this feature, you can locate the lost handset. Press the PAGE button on the base unit. All handsets registered to the base will ring for about 30 seconds.
To stop paging, press the PAGE button on the base or any button on the handset.
Note:

- If a call comes in while the handset is being paged, the handset stops paging and the bell will ring.

Advanced Menu Functions

Menu Navigation

To access a menu option:

1. **To display menu items, press MENU the button.**
2. **To scan through menu options, press the ⌃ or ⌄ button repeatedly.**
3. **To select an option, press the OK button when the option you want appears in the LCD window.**
4. **Repeat if necessary.**

To return to Standby mode from any menu:

If you press the C/R button from any menu (not in the number or text input mode) **the phone returns to**

the previous screen.

To return to Standby mode from any menu, press the button or the MENU button, or press and hold the C/R button.

Also, the phone will automatically return to Standby mode from any menu if no button is pressed for about 30 seconds.

Menu Map

1. Store Phone (see page 18)
2. Call Log (see page 13)
3. Call by Call (see page 19)
4. Key Beep (see page 20)
5. Ring Ring Type (see page 21)
   Ring Level (see page 11)
   VIP Ring (see page 22)
6. PIN Change * (see page 22)
7. Smar t Dial (see page 15)
8. Access Code (see page 22)
9. Register * (see page 23)
10. HS Reset * (see page 24)
11. Clr Redial* (see page 12) *
12. Clr Call Log * (see page 13) *
13. Select Base (see page 25)
14. Base Menu *
   Call Bar (see page 25)
   Dial Mode (see page 8)
   Time Set (see page 8)
   Release (see page 26)

* PIN is required to access these menus.

Note:

If there are several handsets registered to the base unit and if there is anyone who is using the Base Menu, you cannot access the Base Menu until the other handset has finished.

Phonebook

You can store frequently used phone numbers (up to 10) in your phone's internal Phonebook so that you can easily make a call without having to remember or enter the phone number.
Storing a Phone Number in Phonebook:

1. **Press the MENU button.**
2. **Press the OK button to access STORE PHONE.**
Result: The first phonebook entry appears on display.
3. **Press the or button to find the entry you want.**
4. **Enter the desired number (up to 24 digits), then press the OK button.**
5. **Repeat if necessary.**

Note:

- If you make a mistake while entering a number, use the C/R button to correct the mistake. Each time you press the button, the last digit is cleared. To clear all digits, press and hold the C/R button. Then enter the correct number.

Using a Pause

A pause allows to have adequate time for the phone number to register with the telephone company's system and complete the call. A pause provides a delay of 3 seconds.

If you want to insert a dialing pause between numbers when you store a number in memory, press the button until P appears at the pause entry.

Viewing the Phonebook Entries:

1. **In Standby mode, press the or button.**
2. **Press the or button until the phone number you want is displayed.**

Dialing a Number from Phonebook:

Find out the number you want to dial. Refer to "View the phonebook entries". Then press the button to dial.

Or in Standby mode, press and hold the entry number (0-9) until the stored number appears.

Result: The number is automatically dialed.

Editing the Phonebook Entries:

1. **Press the MENU button.**
2. **Press the OK button to access STORE PHONE .**
3. **Press the or button until the phone number you want to edit displays.**
4. **If necessary, press the C/R button to clear the digit(s) then enter the desired number.**
5. **Press the OK button to save the number.**

Call by Call

This feature lets you add a special service number such as special network service number. You can recall the number easily, then attach the phone number you want to call by entering the phone number manually or accessing Phonebook.

The special Call by Call number will be dialed, followed by the phone number.

You can store up to 3 Call by Call numbers.

Storing Call by Call Numbers:

1. **Press the MENU button.**

2. Press the or button to choose CALL by CALL , then press the OK button.

3. Press the or button to choose the desired memory cell.

4. Enter the desired number (up to 10 digits), then press the OK button.

Dialing Call by Call Number:

When you dial phone number manually:

1. Press the OK button in Standby mode to recall the call by call numbers.

Result: The currently stored call by call numbers are displayed.

2. Press the or button to choose the desired Call by Call number, then press the OK button.

3. Enter the phone number you want to call.

Result: The phone number displays after the selected Call by Call number.

4. Press the button to dial the number.

When you dial from Phonebook:

1. Press the OK button in Standby mode to recall the call by call numbers.

Result: The currently stored Call by Call numbers are displayed.

2. Press the or button to choose the desired number, then press the OK button.

3. Press the or button to recall Phonebook entries.

4. Find the number you want using the or button.

5. When you find the desired entry, press the OK button.

Result: The selected phone number displays after the preselected Call by Call number.

6. Press the button to dial the number.

Updating Your Call by Call Numbers:

You can delete or change the numbers.

1. Press the MENU button.

2. Press the or button to choose CALL by CALL , then press the OK button.

3. Press the or button repeatedly to choose the memory cell you want to delete or change.

Result: The stored number displays.

4. Using the C/R button, erase or change the number, then press the OK button.

Key Beep

Every time you press a key, your handset acknowledges it with a key tone. You can set your handset with a key tone or to disable the key tone for a silent use.

1. **Press the MENU button.**

2. **Press the ▲ or ▼ button to choose KEY BEEP, then press the OK button.**

Result: The display shows the current setting.

3. **Press the ▲ or ▼ button to scroll through the key tone options.**

Result: You can choose from:
- TYPE1 - Standard key tone.
- TYPE2 - Two frequency tone.
- OFF - The key tone does not sound.

4. **Press the OK button to save the selection.**

Selecting Ring Type

You can select your own ringing sound. 6 ring types are available.

1. **Press the MENU button.**

2. **Press the ▲ or ▼ button to choose RING , then press the OK button.**

3. **Press the OK button to access RINGTYPE.**

4. **Press the ▲ or ▼ button to choose the ring type you want.**

Result: Each time you press the ▲ button or ▼ button, the handset sounds the ring you have chosen.

5. **Press the OK button to save the selection.**

Setting VIP Ring

This option allows you to instantly identify callers you've assigned to the phone's memory by generating a distinctive ring. You can use this feature only when the caller's service network transmits the caller's information.
You can specify 2 callers and select a VIP ring tone for each caller among 6 different tones.

1. **Press the MENU button.**

2. **Press the ▲ or ▼ button to choose RING, then press the OK button.**

3. **Press the ▲ or ▼ button to choose VIP RING, then press the OK button.**

4. Press the or button to choose the ring address you want, then prese the button.

5. Enter the phone number you want to designate, then press the button.

6. Press the or button to scan through the ring types available, then press the button to save the selection.

Changing PIN

The PIN is required for the following options: Registering a new handset, Resetting a handset and the items under bASE MENU . The PIN is preset to "0000" at factory. To change the PIN code:

1. Press the button.

2. Press the or button to choose PIN CHANGE , then press the button.

3. Enter the current PIN, then press the button.

Result: The LCD window does not display the PIN you entered in order to maintain secrecy. If you enter a wrong PIN, your phone returns to Standby mode.

4. Enter a four-digit PIN you want to use, then press the button.

5. Enter the new PIN again to confirm the number, then press the button.

Setting Access Code

Some telephone system requires an access code (9, for example) and listen to a second dial tone before dialing an outside number. You can set your phone to insert one-digit access code automatically when you use a phone number from your Call ID list.

1. Press the button.

2. Press the or button to choose ACCESS CODE, then press the button.

3. Enter the one-digit access code using the number keypad.

Note:

- When you dial a phone number in the Call Log list, the number will be dialed following the access code and a preset pause time.

4. Press the button to save the selection.

Registering a New Handset

The handset which comes with the base unit was already registered as handset 1. Each additional handset you purchase must be registered to the base unit.

Note: A handset can be registered to up to 4 different base units. And a base can be used with up to 6 handsets.

To register a handset:

1. **Press the MENU button.**

2. **Press the or button to choose REGISTER, then press the OK button.**

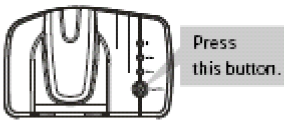
3. **Enter the PIN, then press the OK button.**

Note:

- The PIN is preset to "0000" at the factory. You can change the PIN. For details, see page 22. If you want to register other handsets from different manufacturers to this base, you must enter their PIN codes in this step.

4. **Press the or button to choose the base number you want, then press the OK button.**

5. **Press and hold the PAGE button on the base for more than 5 seconds, then the IN USE indicator starts to blinks. Then release the button.**



6. **Press the OK button.**

Result: The LCD window prompts you to enter AC.

7. **Enter your phone's AC (Authentication Code), and press the OK button. The AC is "0000".**

Result: When the registration is properly completed, the LCD window displays the handset number, and returns to Standby mode.

Note:

- AC remains the same even if you want to register other handsets from different manufacturers to this base.

Resetting Handset

If you reset the handset, all the user-selectable features including Phonebook, incoming and outgoing call numbers and other settings return to the initial setting at the factory.

To reset your handset:

1. **Press the MENU button.**
2. **Press the or button to choose HS RESET, then press the OK button.**
3. **Enter the PIN, then press the OK button.**
4. **Press OK button to confirm the reset.**

Selecting Base

Your handset can be used with up to 4 base units. This menu option allows you to select the base you want to use.

Note:

To use the handset with more than one base unit, you must register the handset to each base unit. See page 23.

To select a base unit:

1. **Press the MENU button.**
2. **Press the or button to choose SELECT bASE, then press the OK button.**
3. **Press the or button to scroll through the available bases.**

Result: If you select bEST bASE, the handset will automatically find the first available base unit when you lose contact while moving around.

4. **Press the OK button to save the selection.**

Call Barring

It is possible to set the phone to restrict numbers that can be dialed. Calls beginning with numbers you specified can not be dialed in this feature.

You can set up to 4 different restricted numbers containing up to 4 digits each.

If the feature is on, the message bARRING displays when the restricted number is dialed.

To activate a call barring number:

1. **Press the MENU button.**
2. **Press the or button to choose bASE MENU, then press the OK button.**
3. **Enter the PIN, then press the OK button.**
4. **Press the OK button to access CALL bAR.**
5. **Enter the number you want to restrict, up to 4 digits.**
6. **Press the OK button to save the number.**

To deactivate a call barring number:

1. **Press the MENU button.**
2. **Press the or button to choose bASE MENU, then press the OK button.**
3. **Enter the PIN, then press the OK button.**
4. **Press the OK button to access CALL bAR.**
5. **Press the or button to choose the cell number you want to deactivate, and clear the number.**

6. **Press the OK button to save the selection.**

Releasing Handset

You can remove the registered handset from the base if necessary.

1. **Press the MENU button.**
2. **Press the or button to choose bASE MENU, then press the OK button.**
3. **Enter the PIN, then press the OK button.**
4. **Press the or button to choose RELEASE, then press the OK button.**

Result: The LCD window displays all handsets currently registered to the base.

5. **If you want to select the handset to release individually, enter the number(s) of the handset(s).**

When the selected handset number(s) disappears in the LCD window, press the OK button.

Result: The selected handset(s) will be removed from the base. The LCD window displays UNREGISTER, If the removed handset is currently registered to another base unit and is within the range of the base unit, you can use the handset with the base.

Caller ID Type

There are two different Caller ID types which are DTMF and FSK. This feature allows you to change the Caller ID type.

1. **Press the MENU button.**
2. **Press the or button to choose bASE MENU, then press the OK button.**
3. **Enter the PIN, then [illegible]s the OK button. Then the LCD windowdisplays CALL bAR .**
4. **Press and hold the # button for about 3 seconds, then the current setting is displayed (FSK or DTMF).**
5. **If you want to change the Caller ID type, press the * button once. Then the phone exit to standby mode.**
6. **If not, press the MENU button to exit.**

Using Multi System

Up to 6 handsets can be used with a base. You can make an intercom call and transfer an external call between the handsets.

Intercom Between Handsets

If you have several handsets registered to the base, two handsets can talk to each other on an internal communication call.

1. **Press the button on your handset.**
2. **Enter the handset number (1~6) you want to page.**
3. **The paged handset rings. The LCD window on the paged handset displays your handset number (the paging handset).**

Notes:

- If you enter a handset number that does not exist, the handset sounds an error tone.
- To cancel the intercom call, press either the button or the button.

4. **To answer the call from you, the paged handset's user should press any button.**
5. **To end the call, press either the button or the button.**

Notes:

- If an external call comes in during an intercom conversation, you will hear beeps. When you hear the low beeps, finish the intercom call by pressing the button.
- Then the external line rings. Press the button to answer the call.

Call Transfer Between Handsets

You can transfer a call from one handset to another.

1. **During a telephone conversation, press the button. Your caller will be put on hold.**
2. **Enter the handset number (1~6) you want to transfer to.**
3. **The paged handset will ring. To answer the call from you, the paged handset should press any button.**
4. **You can speak to the handset (Intercom).**
5. **To transfer the external call to the paged handset, press the button.**

Result: The paged handset is connected to the outside party.

Or, to cancel the call transfer and talk with the outside party again, press the button.

Note:

- When you transfer a call, you may hang up before the transferred station answers your paging. The connection between the caller and the station is completed when you hang up. If the transferred station does not answer the call within a predetermined time, the call is transferred back to your station.

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 and H.323 industry standard and can interoperate with many other SIP or H.323 compliant devices and software on the market.

VoIP Software Features

- Support two modes: Bridge and Router (NAT&NAPT)
- Network Protocols: TCP/UDP/IP,ICMP,HTTP,DHCP Client (WAN Interface) ,DHCP Server (LAN Interface) ,DNS Client, DNS Relay, SNTP,

PPPoE, FTP,TFTP

- VOIP Protocols: Support H323 (V4)&SIP (RFC3261, RFC3262, RFC3264, RFC3265) synchronously
- Voice Codecs: G.711 (A-law/U-law) ,G.723.1(High/low),G.729
- NAT transversal: Support STUN client, etc. Can modify SIP register port, HTTP server port, Telnet server port and RTP port
- Support two SIP server synchronously：Can register two different SIP server, and can make a call by either proxy
- Black list and out –limint , Ban outgoing
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Provide easy configuration thru manual Web interface and Telenet) or automated centralized configuration file via TFTP or HTTP
- Support syslog, can send event of phone to syslog server

Installation

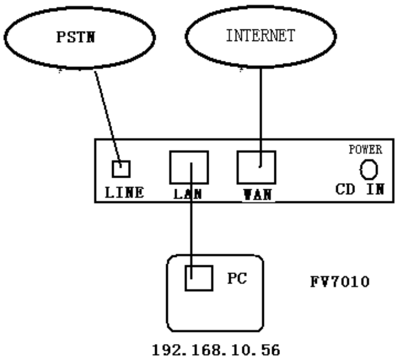
1. Remove the LAN cable for Internet connection from your PC and connect it to "WAN" port of the base.
2. Connect the power adapter in the box to "Power".
3. Find LAN cable in the box and connect between "Lan" port and your PC (PC is not required for set up or making a call.) .

Make Phone Calls

To dial a PSTN number or a number on the proxy, you might need to enter in some prefix number (you can change the lifeline prefix in the configuration web, for details, see page 48) followed by the phone number. We recommend to dial # after the number, this will improve the speed of getting through the line.

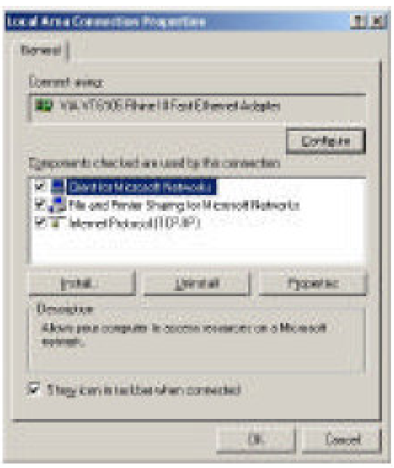
Connecting to the configuration WEB page

Physical Connection



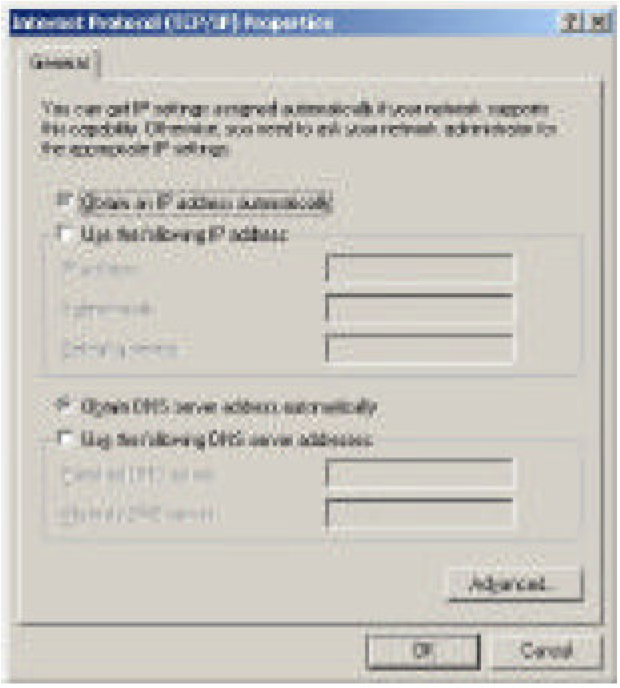
The IP Phone Web Configuration Menu can be accessed by the following default LAN IP address " 192.168.10.1". Before accessing the web, you must do the following steps:

1. Open the " **Local area connection properties** " window.



2. Select **"Internet protocol (**TCP/IP) ", then click the " ". **Internet Protocol(TCP/IP) Properties**
3. Select "**Obtain an IP address automatically** ", then click" ", the PC will obtain an IP

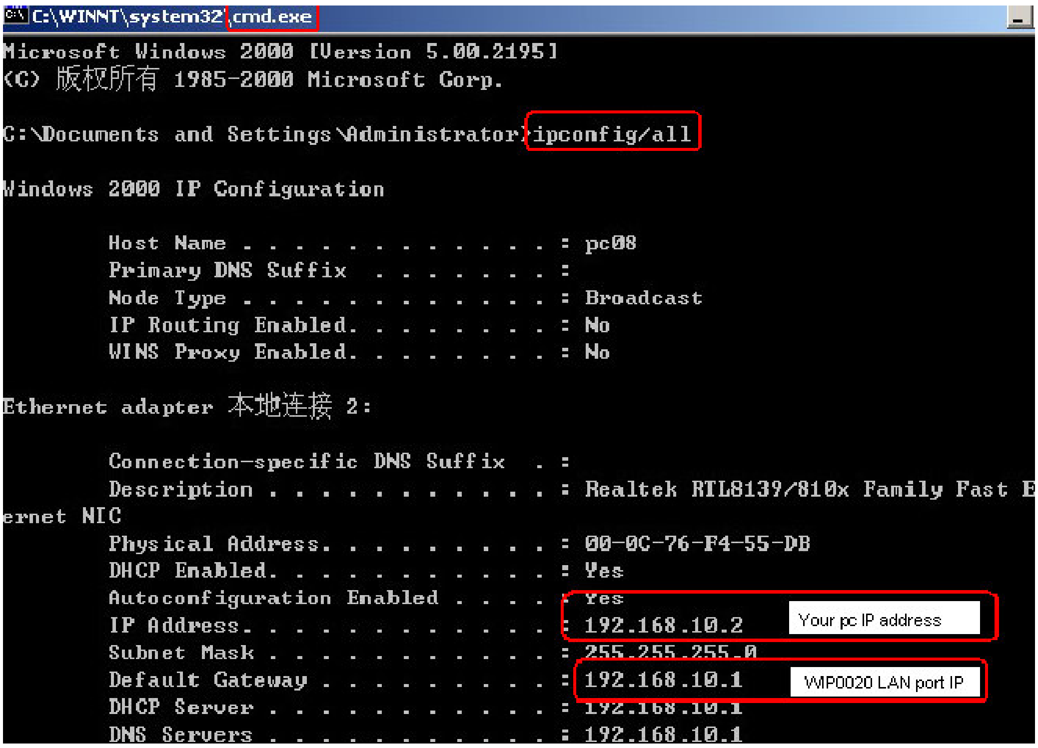
address automatically.



If your LAN IP was changed, do as follow to obtain your phone's LAN IP:

1. Perform steps 1 to 3 of the last paragraph.
2. Click"**Start** " to select"**RUN** ", after inputting" **cmd** " then click" ".
3. Then a command window will pop up. After inputting "**ipconfig/all** ", the window shows as below.

The " **Default Gateway** " is the LAN IP.



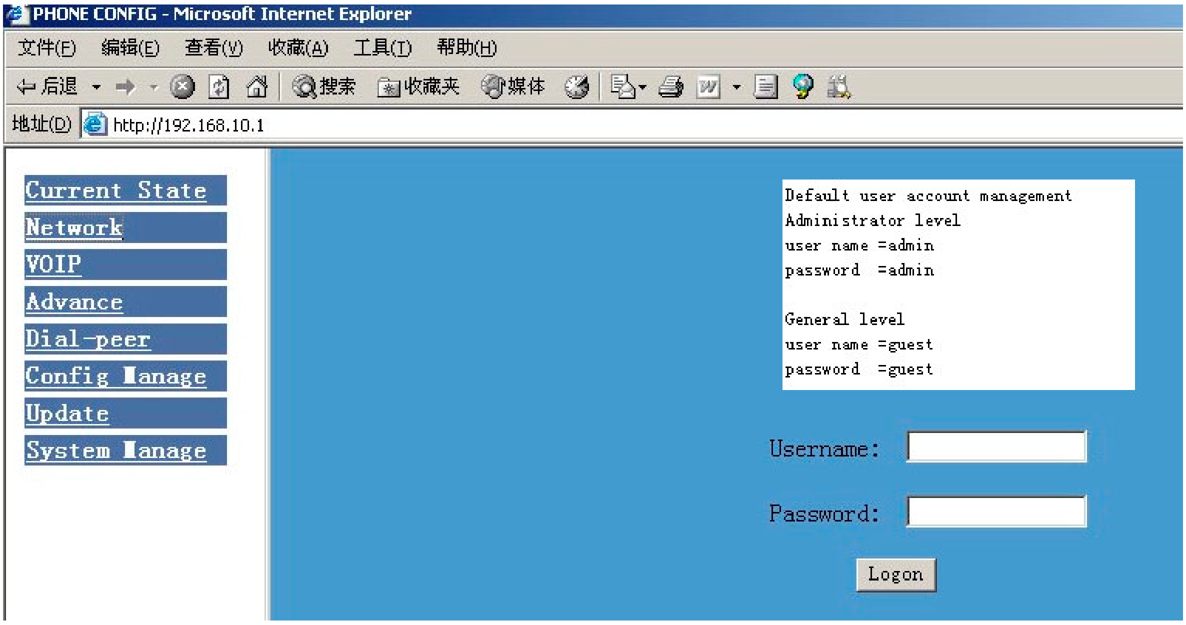
User verification

Users are requested to make verification when config or browse the IP phone thru web pages, users can direct login the config menu by inputting username and password as below:

Default username and password is:

Administrator: Username=admin password= admin high level user interface

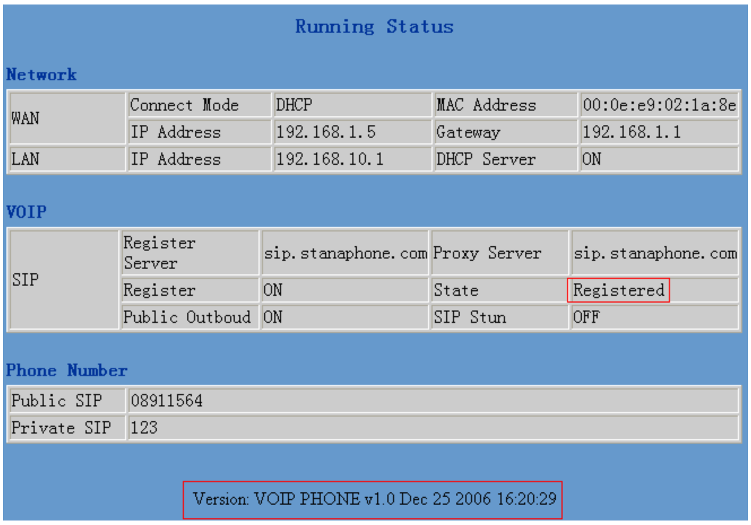
User: Username=guest password=guest low level user interface



Current State

On this page user can gather information of each commonly-used parameter of the phone, it is shown as the following figure:

- Network section: Display the current WAN, LAN configurations of the phone
- VoIP section: Display the current default signaling protocol in use，and server parameter in use of each protocol
- Phone Number section: Display the phone number against each protocol



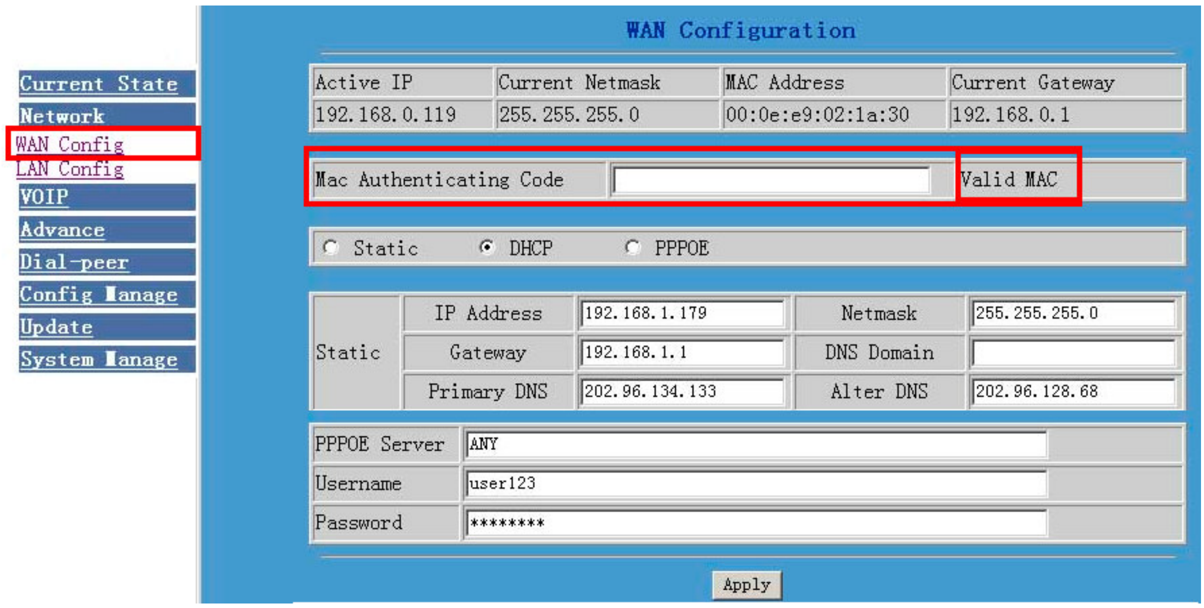
- The version number and date of issue have been shown at the end of this page

Network Configuration

Network configuration includes WAN Config and LAN Config.

WAN Configuration

This web page displays the WAN parameter configuration.



Display <valid MAC >, that means the phone had been certificated.

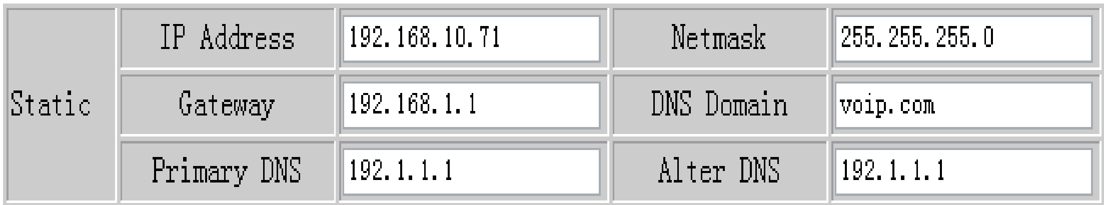
Display <invalid MAC>, that means the phone need a Mac Authenticating Code .(get it from your provider)

Display <invalid MAC, that means the phone can not work normally.

WAN port support Static /DHCP/PPPoE. Users can set the right model base on actual requirements.

- Connect network to internet thru Static mode

WAN default network config is DHCP model; So Users need to set below parameters



IP Address	WAN IP address
Netmask	Network mask
Gateway	Default gateway IP address
DNS Domain	Option configuration
Primary DNS	IP address for primary Domain Name Server
Alter DNS	Option configuration

Click "Apply" button after finished above setting, IP Phone will save the setting automatically with immediate effect.

If users visit IP Phone thru WAN, it need to input "ipconfig" command to get the new IP address and copy it to web browser bar to visit IP Phone.

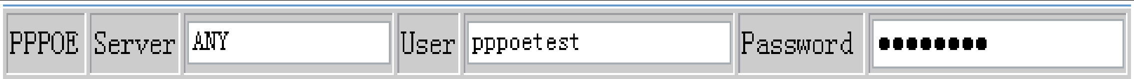
- Connect network to internet thru DHCP mode

Select "DHCP" on below single option, IP Phone will auto-config the WAN parameter with immediate effect.

- Connect network to internet thru PPPoE mode

Select "PPPOE" on below single option,

Set below parameter of PPPOE mode

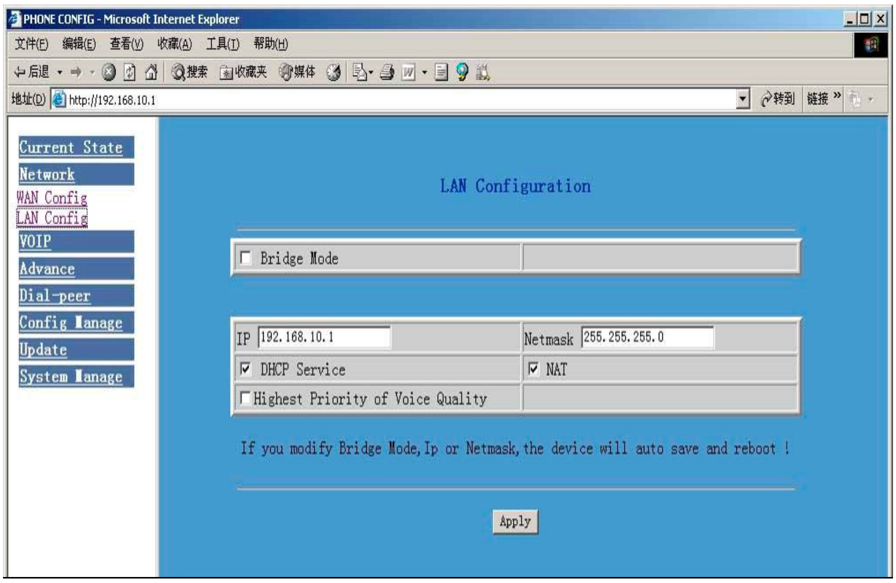


Server	If ISP no special requirements, remains default setting
User	Provided by ADSL ISP
Password	Provided by ADSL ISP

Click "Apply" button after finished above setting, IP Phone will auto-config the WAN parameter with immediate effect. The setting of WAN is still effective and enables IP Phone to connect to internet.

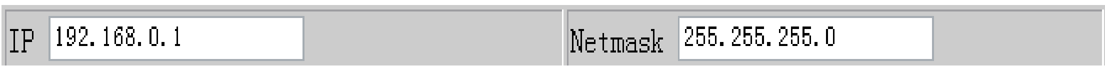
LAN Configuration

This web page displays the LAN parameter configuration. Please note once the bridging mode is selected, the LAN configuration will be no longer effective.



Configuration Example

- Config LAN: generally config one private IP address



IP	LAN IP address
Netmask	Network Mask

- Start LAN DHCP Service and NAT or not: default setting is start

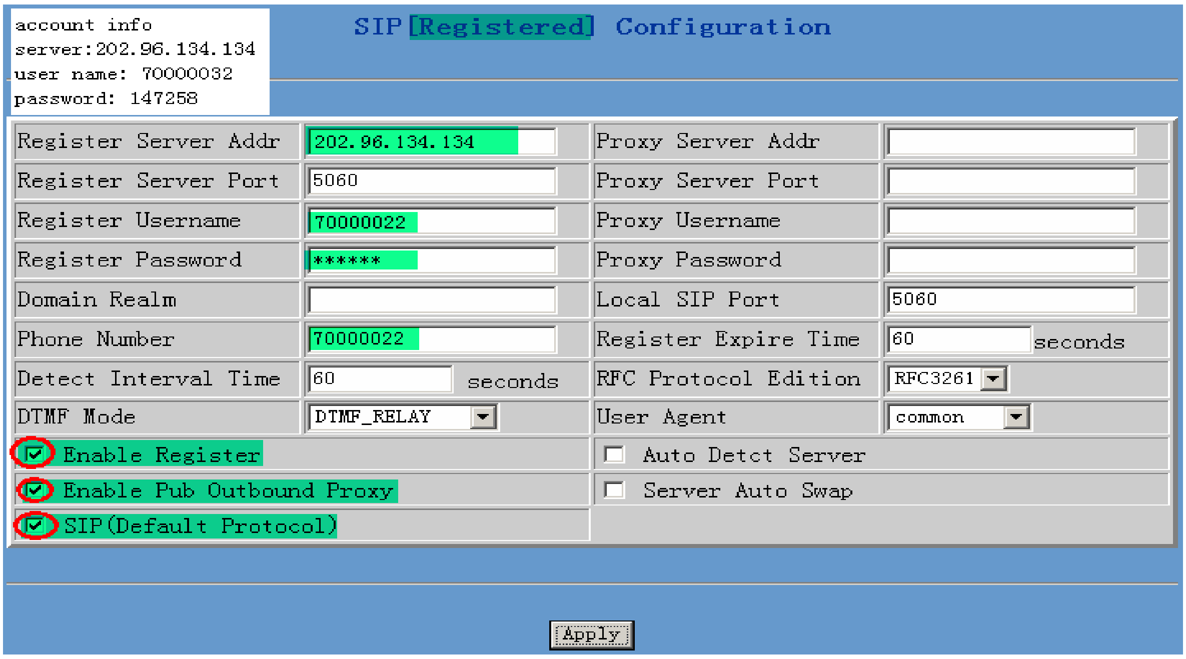
Start Bridge Mode or not(transparent mode): Once start Bridge Mode, some parts of LAN config will be disabled, and the phone will no longer set IP address for LAN physical port，LAN and WAN will join in the same network.

VOIP Configuration

This section is to config signaling protocol for the SIP Server and Client.

SIP 1 Configuration

User can configure specific parameter of SIP1 on this page:



Definition of each parameter described as below:

SIP[Unregistered] Configuration	SIP register state ; if register successfully, show “Registered” in the square bracket , otherwise show Unregistered
Register Server address	Set SIP register server IP address
Proxy Server addr	Set proxy server IP address(usually SIP will provide the same configuration of proxy server and register server, if different(such as different IP addresses), then each server's configuration should be modified separately)

Register Server Port	Set SIP register server signal port
Proxy Server Port	Set SIP proxy server signal port
Register Username	Set SIP register server account username (Usually it is the same with the config port number)
Proxy Username	Set the SIP proxy server account username
Register Password	Set password of SIP register server account
Proxy Password	Set password of SIP register account
Domain Realm	Enter the sip domain if any, otherwise WIP0020 will use the proxy server address as sip domain. (Usually it is same with registered server and proxy server IP address).
Local SIP Port	Set local signal port , the default is 5060
Phone Number	Set assigned phone number
Register Expire Time	Set expire time of SIP server register, default is 120 seconds
Detect Interval Time	Set detection interval time of server, default is 120 seconds
RFC Protocol Edition	Enable the phone to use protocol edition. When the phone need to communicate with phones using SIP1.0 such as CISCO5300 and so on, need to modify into RFC2543. the default is to RFC3261 ;
DTMF Mode	Set DTMF sending mode, support RFC2833, DTMF_RELAY (in-band audio) and SIP info
User Agent	Set the user agent if have, default is common
Enable Register	Configure enable/disable register
Auto Detct Server	Co-work with Server Auto Swap and Detect Interval Time. Enable this option, WIP0020 will periodically detect whether the public SIP server is available, if the server is unavailable, the WIP0020 will switch to the back-up SIP sever, and continue detecting the public sip server. WIP0020 will switch back to the primary SIP server if the server is available again.
Enable Pub Outbound Proxy	Configure to enable to use public outbound proxy ,if you have no stun server , advise to enable the option

Server Auto Swap	Configure main and backup auto-swap server ; if the phone enables main and backup server function , the automatic detection and auto-swap functions should both be chosen
SIP (Default Protocol)	Set SIP as the default signaling protocol

After finished the aforesaid network and VoIP configurations on the phone and network communication has been implemented , the user can make VoIP calls by the calling register and proxy server.

> *Note:*
> *Some ISP internet may inhibit the phone to register and cancel the register in process, so user had better cancel apply or register soon and then submit registration repeatedly. Server may stop response of dialogue machine, then the phone receives no register/cancel login request and registration state will show incorrectness!*

Configuration Example

Firstly users should get the account info from VOIP Operator (Including Server IP address, port, username, password etc.) and follow below procedure.

- Config registered server and proxy server IP address and signaling port. (Support DNS for registered server and proxy server)

Register Server Addr	sip.stanaphone.com	Proxy Server Addr	
Register Server Port	5060	Proxy Server Port	

- Config the username and password for registered server and proxy server.

Register Username	08911564	Proxy Username	
Register Password	●●●●●●●●●●●	Proxy Password	

- Config the phone number (Usually phone number is same with SIP account) Remark: due to the above register username is “client”, so the phone number is different from SIP account)

Phone Number	62281493

- Config the domain realm (Usually it is same with registered server and proxy server IP address, Let it be blank)

Domain Realm	10.1.1.139

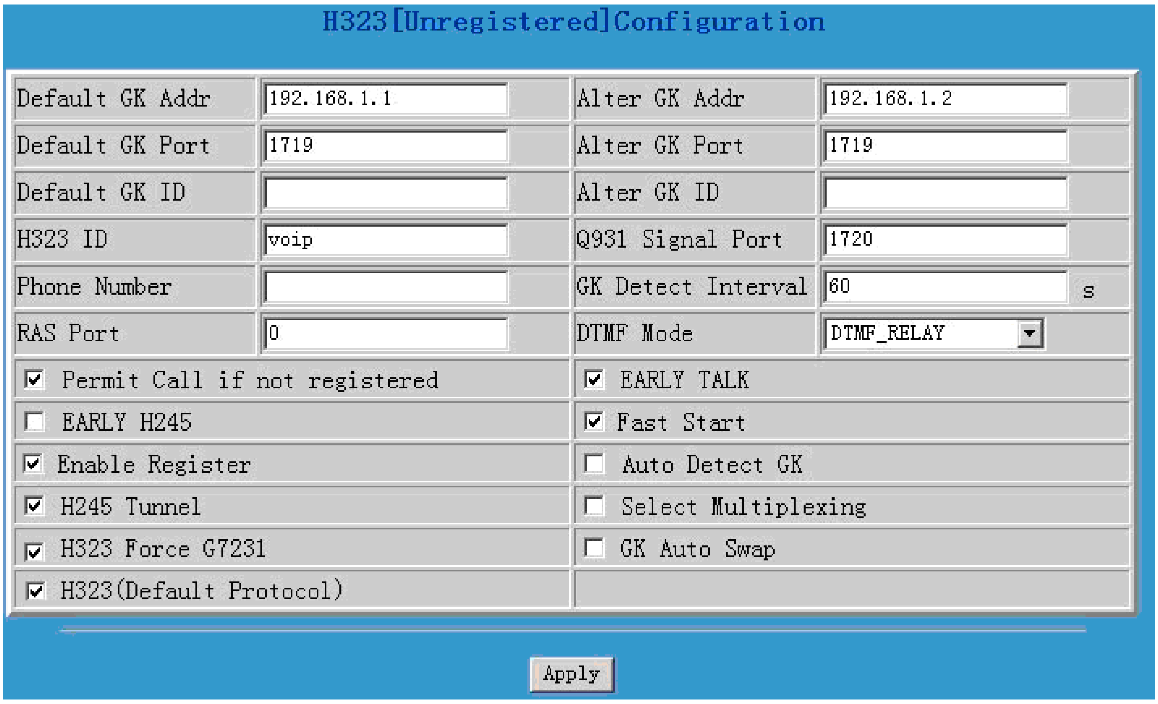
- Select below two option and registered in local outbound public proxy

☐ Enable Register & ☐ Enable Pub Outbound Proxy

Usually these two option need to be selected, when you want to use SIP1.

H.323 Configuration

User can configure specific parameter of H323 signaling protocol on this page；Definition of each parameter described as below



Definition of each parameter described as below

H323[Unregistered] configuration	Show H323 register state；if register successfully, show “Registered”, otherwise show “Unregistered” on bracket
Default GK Addr	Set default gatekeeper IP address
Alter GK Addr	Set backup gatekeeper server IP address
Default GK Port	Set default gatekeeper port
Alter GK Port	Set backup gatekeeper server port
Default GK ID	Set default gatekeeper ID, remains blank if no value

Alter GK ID	Set backup gatekeeper ID, remains blank if no value
H323 ID	Set H.323 ID, default is VOIP
Q931 Signal Port	Set system initial Q931 signal port, default value is 1720
Phone Number	Set assigned phone number
GK Detect Interval	Set GK detection interval time，the unit is second；
RAS Port	Set net gate RAS register port for the system
DTMF Mode	Set DTMF mode，RTP mode，RFC2833 mode，H245-string mode and H245-signal mode；
Permit call if not registered	Set permission for nor-registered call，allow to initiate call without net gate register；
EARLY TALK	Set receiving IVR ,such as the voice prompt, dialing of PSTN color ring；
EARLY H245	Early245 configuration，when initiating a call, the 225 message transmission begins at the same time with 245 message transmission，default value is disable
Fast Start	Set quick start mode to start H323 call
Enable Register	Set enable/disable register
Auto Detect GK	Set the phone enables to detect gatekeeper automatically
H245 Tunnel	Set enable/disable to start H245 Tunnel function
Select Multiplexing	Set multiplexing of logical channel，the default is Disable；
H323 Force G7231	Force to use codec G.723.1 when start H323 outgoing call
GK Auto Swap	Configure main and backup auto-swap server；if the phone enables main and backup server function，the automatic detection and auto-swap functions should both be chosen

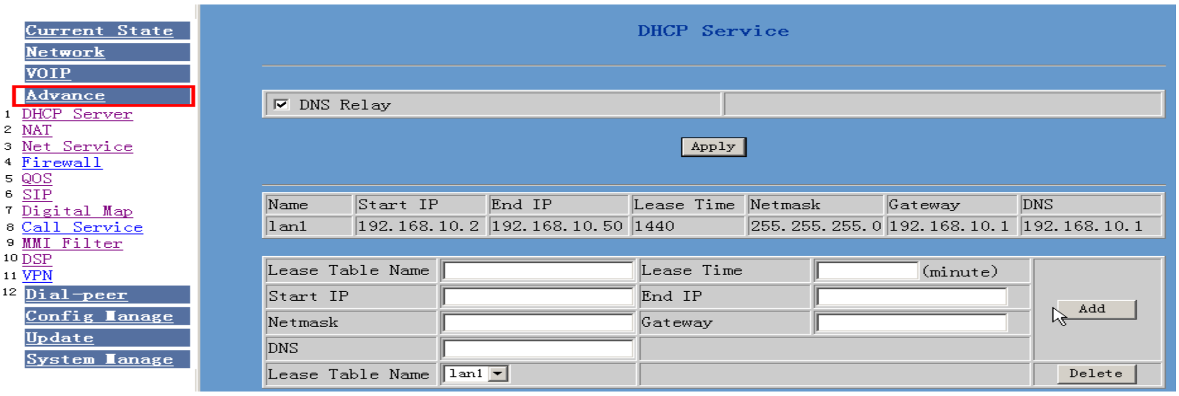
H323(Default Protocol)	Set H323 as the default signaling protocol

Advance

DHCP Server Configuration

When WIP0020 work as a router, this config is for WIP0020 LAN port network device

DNS Relay: DNS relay acts as a forwarder between the DNS Clients and the DNS Servers, DNS relay is designed for home/office networks where the users might want to dial into more than one Internet Service Provide (ISP)



DHCP server manage page.

User may trace and modify DHCP server information in this page.

DNS Relay: enable DNS relay function.

User may use below setting to add a new lease table.

Lease Table Name: Lease table name.

Lease Time: DHCP server lease time.

Start IP: Start IP of lease table.

End IP: End IP of lease table. Network device connecting to the WIP0020 LAN port can dynamic obtain the IP in the range between start IP and end IP.

Netmask: Netmask of lease table.

Gateway: Default gateway of lease table

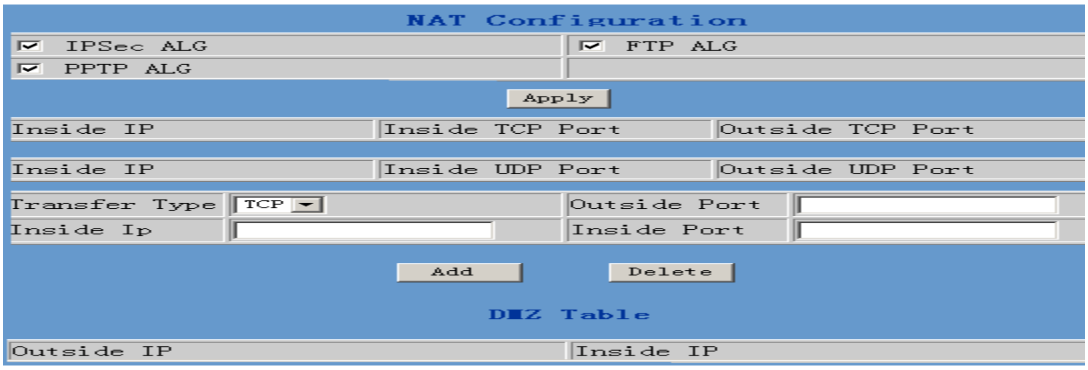
DNS: Default DNS server of lease table.

Notice: This setting won't take effect unless you save the config and reboot the device

NAT Configuration

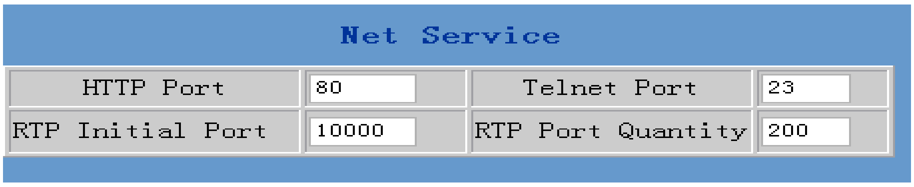
This page is for NAT configuration, such port forward, DMZ .

Network Address Translation (NAT) provides a mechanism for a privately addressed network to access registered networks, such as the Internet, without requiring a registered subnet address. This eliminates the need for host renumbering and allows the same IP address range to be used in multiple intranets. With NAT, the inside network continues to use its existing private or obsolete addresses. These addresses are converted into legal addresses before packets are forwarded onto the outside network.



Advance NAT setting: Maximum 10 items for TCP and UDP port mapping.
IP Sec ALG: Enable/Disable IP Sec ALG;
FTP ALG: Enable/Disable FTP ALG;
PPTP ALG: Enable/Disable PPTP ALG;
Transfer Type: Transfer type using port mapping.
Inside IP: LAN device IP for port mapping.
Inside Port: LAN device port for port mapping.
Outside Port: WAN port for port mapping.
Click Add to add new port mapping item and Delete to delete current port mapping item.

NAT Service Configuration



HTTP PORT

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 ;

Example:

The ip address is 192.168.1.70. you change the port value to 8090, the accessing address is http://192.168.1.70:8090

But if the value is 0, that imply it can not be configured by web browser.

TELNET PORT

Telnet Port	23

Configure telnet port ,the default is 23 port. You can change the value to others.

Example: The IP address is 192.168.1.70. you change the port value to 8023, the accessing address is telnet 192.168.1.70:8023

RTP 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；

DHCP SERVER lease

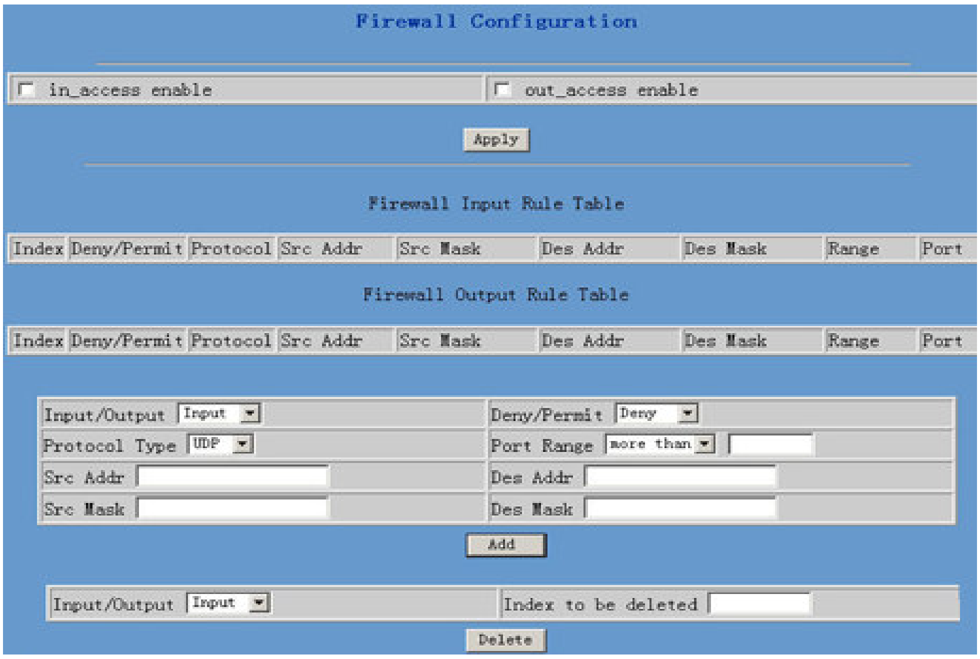
Leased IP Address	Client hardware Address

Leased IPMAC correspondence table of DHC. The table will display all device getting IP address from WIP0020 LAN port by DHCP.

Note *The configuration on this page needs to be saved after modified and will go into effect after restarting. If the Telnet, HTTP port will be modified, the port is better to be set as greater than 1024,because the 1024 port system will save ports.*

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

Firewall



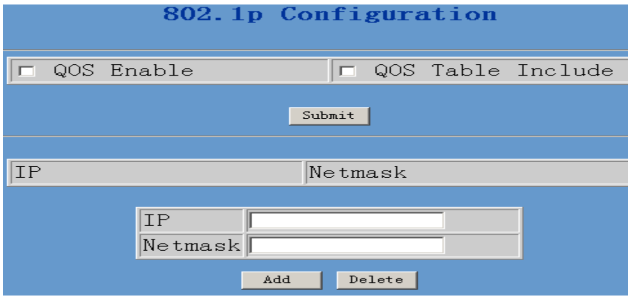
Firewall Setting Page. User may set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices to access the internet.

Access list support two type limits: input_access limit or output_access limit. Each type support 10 items maximum.

WIP0020 firewall filter is base WAN port. So the source address or input destination address should be WAN port IP address.

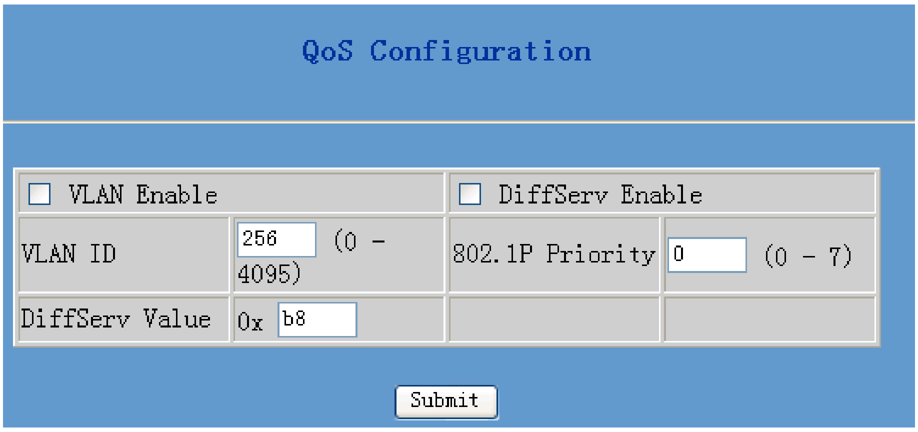
Configuration:
in_access enable: enable in _access rule
out_access enableA: enable out _access rule
Input/Output: specify current adding rule is input rule or output rule.
Deny/Permit: specify current adding rule is deny rule or permit rule.
Protocol Type: protocol using in this rule: TCP/IP/ICMP/UDP.
Port Range: port range if this rule
Src Addr: source address. can be single IP address or network address.
Dest Addr: Destination address can be IP address or network address.
Src Mask: source address mask. Indicate the source is dedicate IP if set to 255.255.255.255. Otherwise is network ID
Des Mask: Destination address mask. Indicate the source is dedicate IP if
set to 255.255.255.255. Otherwise is network ID

QOS 802.1p Configuration



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. WIP0020 will sort the packets using the QoS and sends it to the destination. QoS provides service classes for accessing traffics in Internet.

QOS 802.1p Configuration



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. WIP0020 will sorted 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 take IP service infomation that is necessary. It is strict three layer technology, it do not involve the low layer tranfer technology .

Advance SIP Configuration

Public[Registered]Private[Registered]

STUN NAT Transverse[FALSE]

STUN Server Addr		STUN Server Port	3478
Private Register	sip.stanaphone.com	Private Proxy	
Register Port	5060	Proxy Port	
Register Username	08911564	Proxy Username	
Register Password	●●●●●●●●●●●	Proxy Password	
Private Domain		Expire Time	60 Seconds
Private Number	08911564	STUN Effect Time	50 Seconds
Private Server Type	common	Private User Agent	Voip Phone 1.0
☑ Enable PRACK		☑ Enable Keep Authentication	
☐ Auto Detect Server		☐ Enable Session Timer	
☐ Signal Encode		☐ Rtp Encode	
☑ Enable Private Register		☐ Enable SIP Stun	

Public[Unregistered]Private[Unregistered]

To show the phone whether has been registered on public server or private server

SIP STUN Configuration:

STUN can support SIP terminal's penetration to NAT in the inner-net. In this way , as long as there is conventional SIP proxy and a STUN server placed in the public net, it will do; but STUN only supports three NAT modes : FULL CONE, restricted, port restricted.

STUN Server Addr	0.0.0.0

IF you have stun server .please input stun server address here.

STUN Server Port	3478

The STUN server default port is 3478

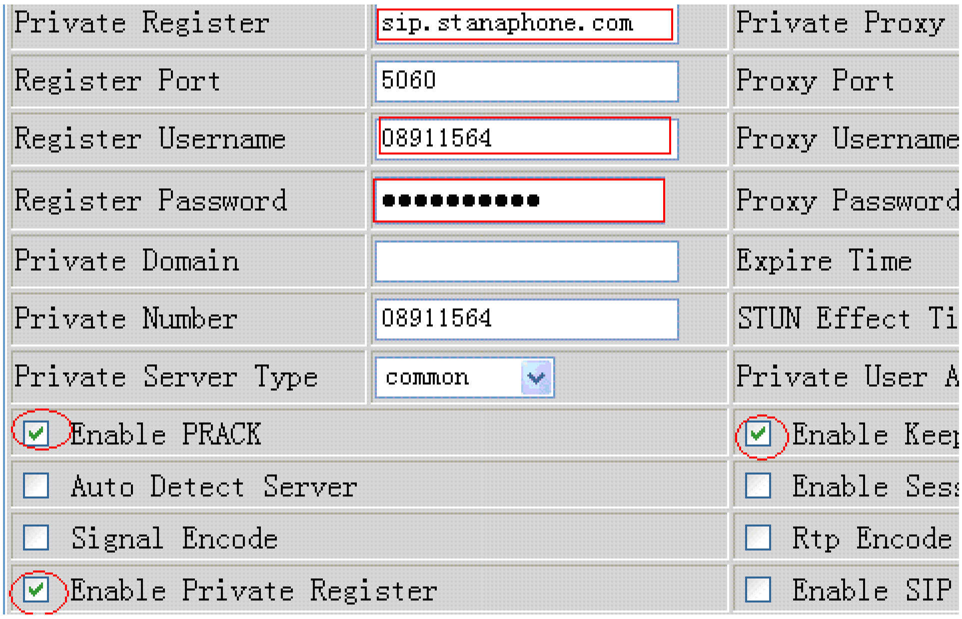
STUN Effect Time	minute

The unit is minute. if you have STUN server .please input interval time for STUN`S detection on NAT type.

☐ Enable SIP Stun

Configure enable/disable SIP STUN ; if you have stun server .please enabe the option.

Private Server(SIP2) Configuration.



Specific configuration parameter has the same meaning with public server.

Enable PRACK : Enable / Disable SIP prack function. Default is unchecked. Prack is same with ACK that is used for temporary response.

Auto Detect server :Enable/disable automatically detect server function. Enable the function when server forbid register time that is too short and server do not initiatively maintain to send terminal nat packets . Enable the function and set the sending the ip packets time is less than nat maintain time.

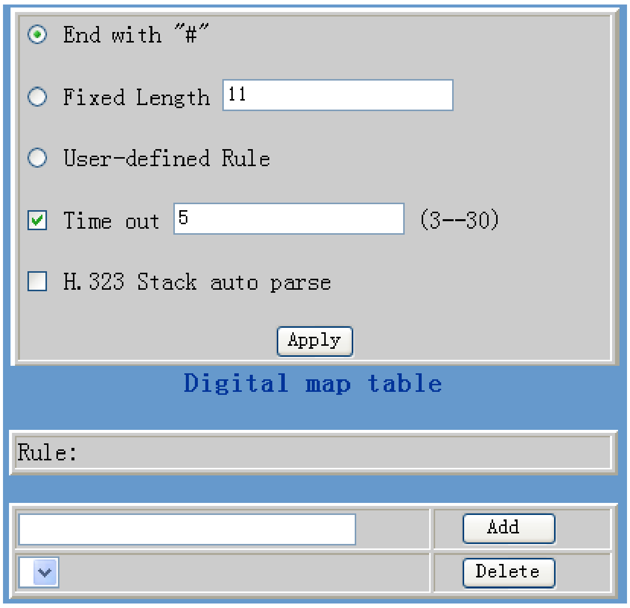
Singal Encode: Enable/disable singal encryption

Rtp Encode : Enable /disable Voice encryption

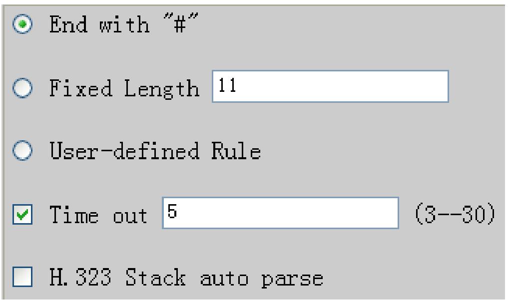
Enable Private Regitster : Configure permit/deny private server register

Enable Session Timer : Enable/disable support RFC4028

Digital Map Configuration



Fixed Digital Map



End With "#": Use # as the end of dialing.
Fixed Length: When the length of the dialing match, the call will be sent.
Timeout: Specify the timeout of the last dial digit. The call will be sent after timeout
User-defined rule User define the flexible ,more dial rule .digital map.

User Define Flexible Digital Map Table

Digit map is a set of rules to determine when the user has finished dialing. Digital Map is based on some rules to judge when user end their dialing and send the number to the server. With digital map, users don't have to press '#' key or "call"key after dialing. If the number dialed matches some item in the digital map table, or it doesn't match with any item, this number will be sent out immediately. It is not like using dial peer . Using digital map won't change the number dialed, the number sent is the same as the number dialed.

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.

[ ] 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]

Example:

[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 dialing 911 ,it will send out immediately

99T4 after finishing dial 911, it will send out in 4 second

9911x.T4 any more than 5 digits length starting with 9911, sending out within 4 seconds.

Digital map table	
Rules:	
[1-8]xxx	
9xxxxxxx	
911	
99T4	
9911x.T4	
	Add
[1-8]xxx	Del

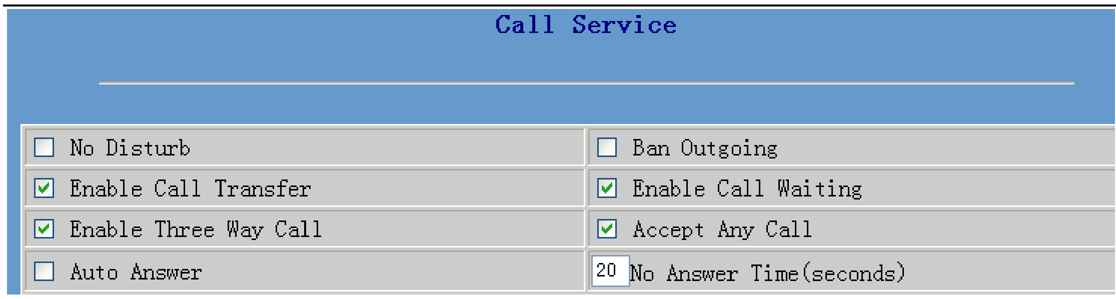
Using digital map can be combined with dial peer. First digital map will determine when the user finished dialing, then convert this number to the number actually sent according to "dial peer".

Dial-Peer

Number	Destination	Port	Alias	Suffix	Del length
2887	192.168.0.155	5060	no alias	no suffix	0
98765432	192.168.0.155	5060	no alias	no suffix	0
911	192.168.0.155	5060	no alias	no suffix	0
99	192.168.0.155	5060	no alias	no suffix	0
9911234	192.168.0.155	5060	no alias	no suffix	0

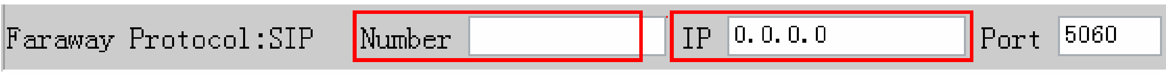
When user dial 2887 or 98765432、911、99 、9911234 , they will send out immediately.

Call service



Call forwarding

Call forward default is Disable. When Off is selected ,if the number dialed is engaged after the phone has received a call, then it will automatically transfer to the configured number according to the following picture (CF001 forward) configuration. when No Answer is selected，if the phone do not receive the incoming call .it will **automatically** in 10 No Answer Time(seconds) forward to the configured number according to the following picture (CF001 forward) configuration. When Always is selected，then the phone will directly transfer all incoming call to the number that had configured in advance like the picture showing.



Picture:CF001

Note:
1 Number can be sip server extension number or DID number (any PSTN number)
2 the function have no relationship with the option Enable Call Transfer *that enable or disable*

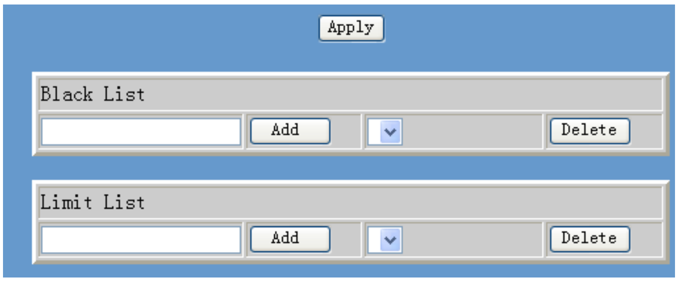
No Disturb

If it is enabled, the phone will not ring when there is a incoming call . DND, do not disturb, enable this option to refuse any calls.

Ban Outgoing

Enable this to forbid outgoing calls.

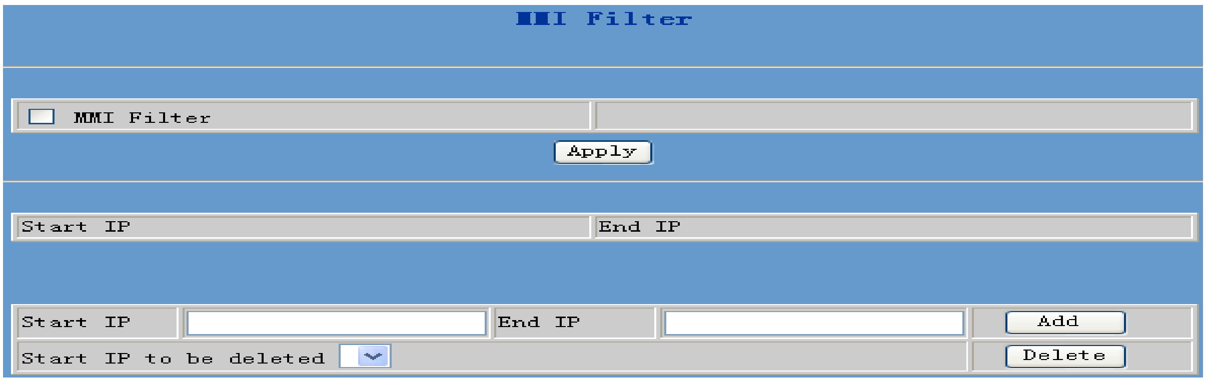
At the present : WIP0020 do not support call waiting ,call transfer ,three way conference



Black list Incoming call in these phone numbers will be refused. It is for precluding incoming communication like Call ID. If user doesn't want to answer a certain phone number, please add this phone number to the list, and then this number will be unable to get through the phone.

Limit list Outgoing calls with these phone numbers will be refused 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.

MMI Filter

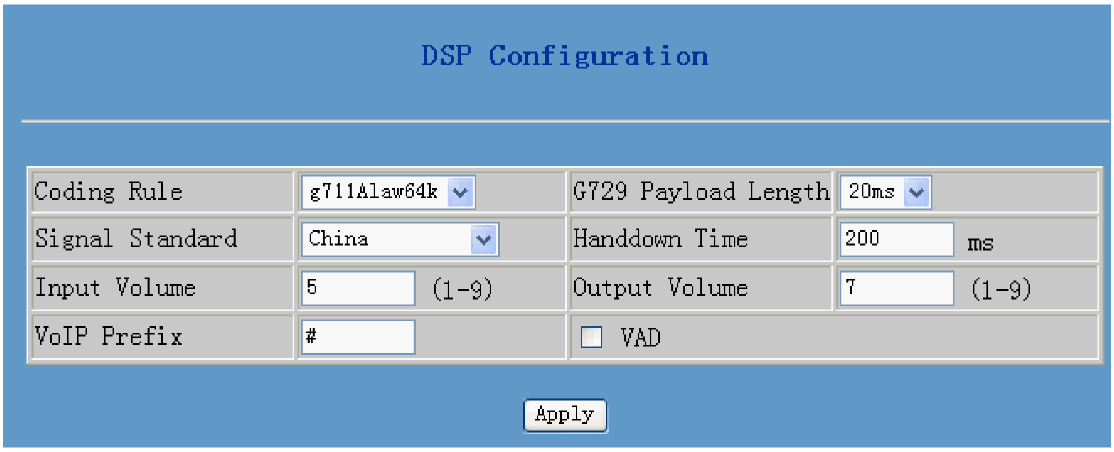


MMI filter is used to make access limit toWIP0020.

When MMI filter is enable. Only IP address within the start IP and end IP can access WIP0020.

DSP Configuration

On this page, user can set voip traffic prefix speech coding , IO volume control, cue tone standard, caller ID standard and so on.



Coding rule Configure Coding Rule: according to network bandwidth; support G.711a/u G.729 ,G7.23 -r53 /r63

G729 payload length Normally, G729 Payload Length don't need be changed into 10 ms;The voice payload size can be represented in terms of the codec samples. For example, a G.729 voice payload size of 20 ms (two 10 ms codec samples) represents a voice payload of 20 bytes [(20 bytes * 8) / (20 ms) = 8 Kbps]

Singal standard Configure Signal Standard: according to country's phone singal voice;

Handdown Time Configure hand-down time , that is, if the hooking time is shorter than this time, then the gateway will not consider the user has hand-down.

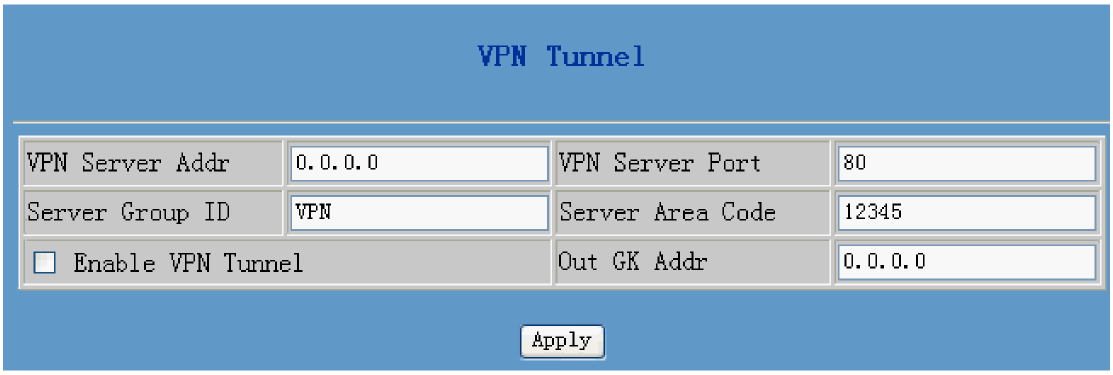
Input Volume Handset In Volume. The called party hearing volume

Output Volume Handset Out Volume. The caller party hearing volume.

VoIP Prefix User define the voip traffic prefix . support one digit , and the prefix will not send to voip server .It is just for separate the line traffic.
Default , the line traffic is PSTN line.

VAD: Enable/disable Voice Activity Detection.
selecting code G729 .and VAD is unchecked. That means g729a; selecting code G729 .and VAD is checked. That means g729b

VPN



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.

If you want to use VPN function ,It need the below condition.

1 It need 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.

More VPN detail introduction please refer to the VPN user manual .

Dial Peer

Number IP Table Configuration

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 configuration 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 number1234 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 manual swap. When making deletion or modification, select the number first and click load, then click Modify and complete the operation.

Display of calling number IP image list

Dial-Peer

Number	Destination	Port	Alias	Suffix	Del length
157	192.168.0.157	5060	no alias	no suffix	0
187	192.168.0.187	5060	no alias	no suffix	0
9T	255.255.255.255	5060	del	no suffix	1
8T	0.0.0.0	5060	all:0755	no suffix	0
010T	0.0.0.0	5060	rep:8610	no suffix	3
6T	192.168.0.187	5060	no alias	12345	0
741	192.168.0.187	5060	no alias	999	0

Add

Click Add, the following figure will be shown at the lower part of the page.

Phone Number
Destination (optional)
Port (optional)
Alias (optional)
Suffix (optional)
Delete Length (optional)

Phone Number 010T

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.

Destination 192.168.10.11

Configure destination address , if it is point-to-point call , then 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, then 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, this is optional configuration item : when nothing is input, the default of sip protocol is 5060 ; lifeline required no configuration of this item, shown as 0.

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.

add:	xxx	add xxx before number. in this way it can help user save the dialing length;
all:	xxx	the number is all replaced by xxx; speed dialing can be implemented, for example, user configure the dialing 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 can be decided by 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.

as no suffix ;

Example 1

T mean any length digit number.
Destination is 255.255.255.255 that mean calling out through SIP2 server.
Destination is 0.0.0.0 that mean calling out through SIP1server

Config page	Explanation	Example
Phone Number: 9T Destination (optional): 255.255.255.255 Port (optional): Alias (optional): del Suffix (optional): Delete Length (optional): 1	That means Any digits number starting with 9 pass through SIP2 server. Here alais is del Delete Length is 1 that means the phone will delete the first number before send number to server	User dial 93333 SIP2 server receive 3333
Phone Number: 2 Destination (optional): Port (optional): Alias (optional): all:33334444 Suffix (optional): Delete Length (optional):	It can be used for speed calling The number user dialed will be replaced fully by the number that is behind all) Here alias is all: (not all)	User dial 2 Sip1 server receive 33334444
Phone Number: 8T Destination (optional): Port (optional): Alias (optional): add:0755 Suffix (optional): Delete Length (optional):	It can be used to add local area or prefix. before sending out. It saves user dialing number. Here alias is add: (not add)	User dial 8309 SIP1 server receive07558309

Phone Number: 010T Destination (optional): Port (optional): Alias (optional): rep:8610 Suffix (optional): Delete Length (optional): 3	user want to dial PSTN (010 6228) by SIP1 , while actually the called number should be 86106228 , 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. Replace the number that user dialed before sending to SIP1 server. Here alias is **rep:**(not rep)	User dial 010 6228 SIP1 server receive8610 6228
Phone Number: 147 Destination (optional): Port (optional): Alias (optional): Suffix (optional): 0011 Delete Length (optional):	this is optional configuration item. It is to add number behind the number user had dialed. when no configuration has been made, shown as no suffix	User dial 147 Sip1 server receive 147 0011

Example 2

Dial-Peer

Number	Destination	Port	Alias	Suffix	Del length
20T	0. 0. 0. 0	5060	no alias	no suffix	0
200T	255. 255. 255. 255	5060	no alias	no suffix	0

The dial rule support exactitude matching priority.

When user dial 200 It will pass through SIP2
When user dial 2009 It will pass through SIP2
When user dial 20096 It will pass through SIP2
When user dial 201 It will pass through SIP1
When user dial 20 It will pass through SIP1

Config Manage (Save and Clear configuration)

Notice: clear config in admin mode, all settings restores to factory default; clear config in guest modem, all settings except sip, advance sip restore to factory default.

Save Config

Once change is made, Users should save the modified configuration to take effect, otherwise the IP Phone will go back to the last saved setting after phone reboot. The interface of "Save Config" as below, please follow the four steps below to config.

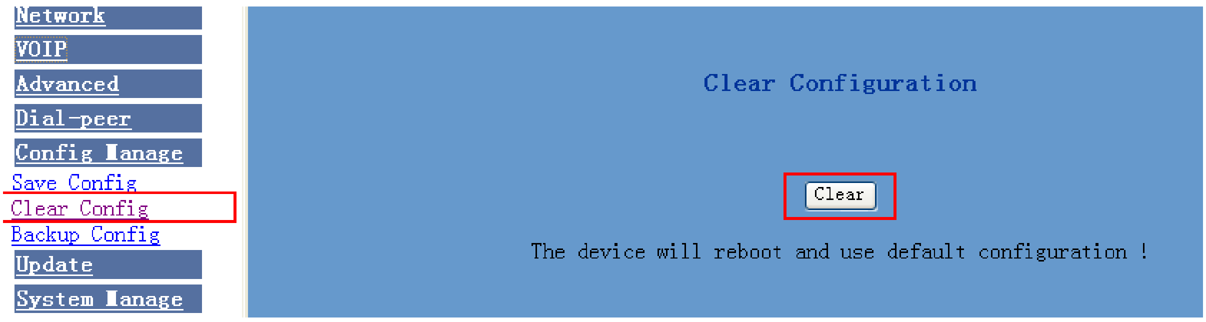
Enter "Config Manage" Menu →"Save Config" Submenu→ Click "Save" Button→ Return to "Current State" Web page

Clear Config

There are four method to clear config (set factory default), web 、telnet 、post mode、keypad. If the IP Phone doesn't work properly after modifying config, users can clear all modified config on "Clear Config" web page. The phone will clear all modified config and restore the default factory configuration. (Default network type for WAN is DHCP mode; default LAN IP address is 192.168.10.1)

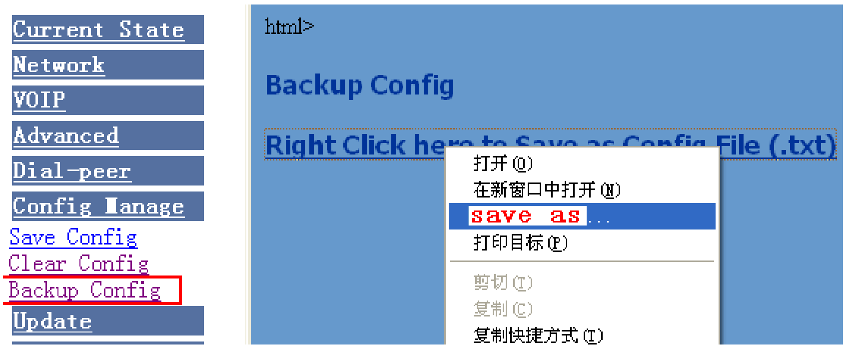
Process Please follow the below steps to clear config:

Enter "Config Manage" Menu →"Clear Config" →Click "Clear" Button →show "Submit Success" info on screen →Click "Return" button



Back up Config

Download phone config file by HTTP. Config file can be edit by WORDPAD.



Firmware Upgrade

Web Update

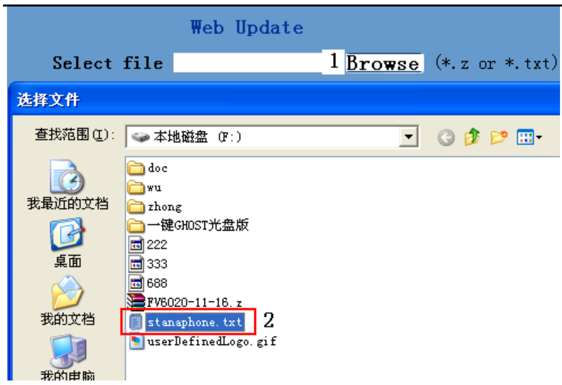
On this page, user can select the upgrade document (firmware or config file) from hard disk of the computer directly to run the system upgrade. After upgrade completed , reset the phone and it will be usable immediately. Firmware format is *.z as suffix

Firmware update

STEP:

Enter Update menu →WEB Update submenu→ click “browse” button→ download upgrade document from hard disk (firmware provided by manufacturer) → click “Update” button →reboot IP phone to go into effect

Config file download to phone



Commonly , set one phone all parameter needed .then download the phone config file to your FTP server (PC run a FTP software).

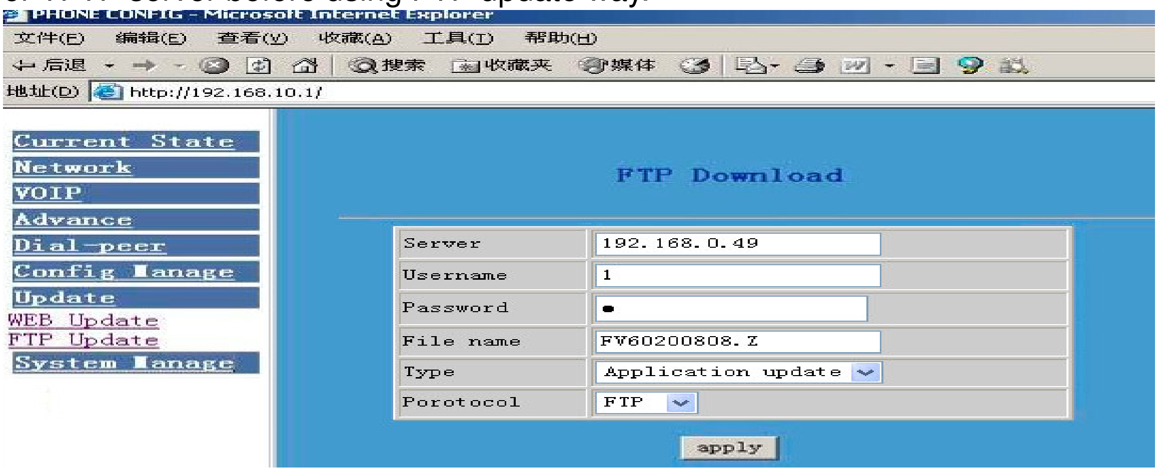
So when set another phone , download the config file to new phone for saving time .

Note:

> *Note:*
> *Under system upgrade progress, IP Phone may not be restarted normally due to some system reason (e.g. electricity shut off), users can re-download under post mode.*

FTP or TFTP Update

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

STEP:

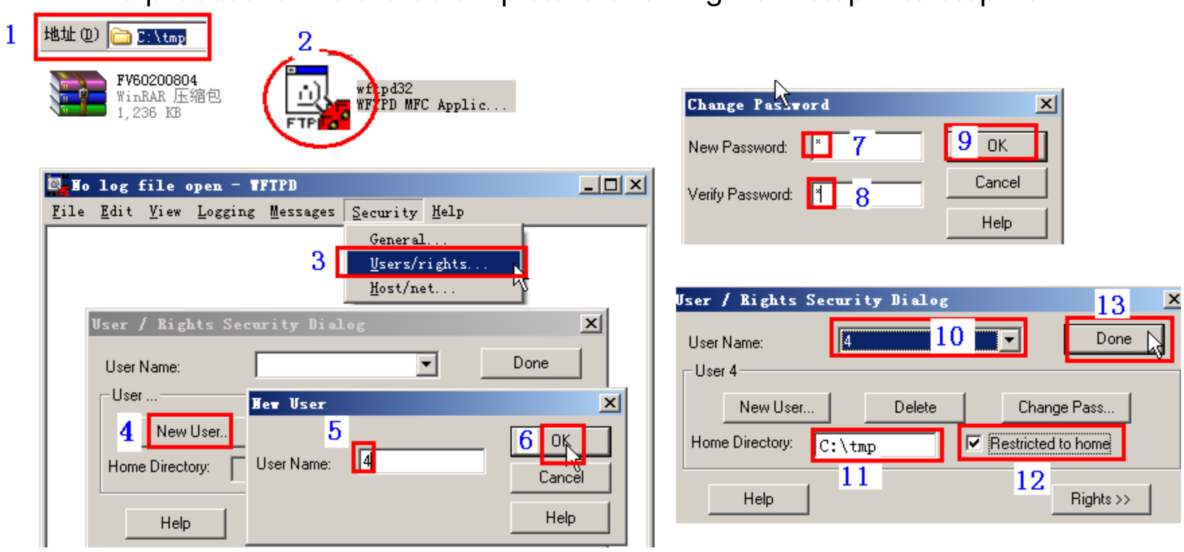
Enter Update menu →FTP Update submenu→ Config FTP/TFTP server → Config username and password of FTP server (if select TFTP mode, please skip this step) →key in file name → choosing file type from the dropdown menu→ choosing protocol type

Example: (export config file)

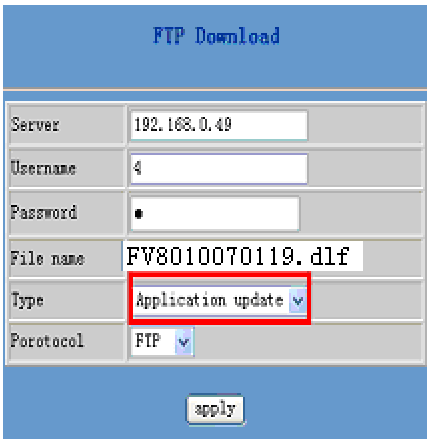
1 FTP

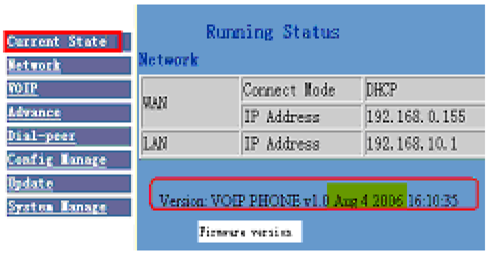
<1> Copy Wftpd32 software and WIP0020 Firmware into a new Folder (example c:/tmp)

<2> Run wftd32.exe. Set a user name and password for WIP0020 ftp updating The process is like the below picture showing from step 1 to step 13.



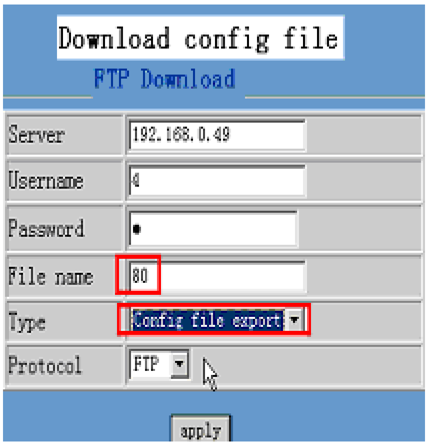
Update the firmware





After it update successfully. You will find the new time in Current State version.

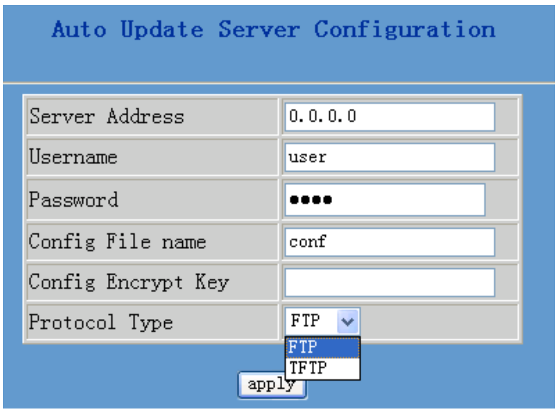
Download config file to you pc.



After you click apply , you can find the file that it had download to your pc <c:\temp>

Autoupdate

Auto update config file from server by FTP or TFTP. When the phone reboot ,it will automatically connect to your server to download the config file. After downloading successfully ,The phone will reboot.



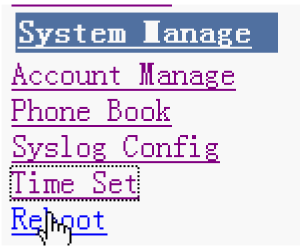
Server Addres Your FTP or TFTP server address.
Username FTP server login user name . If using TFTP , needn`t fill anything
Password FTP server login user password. If using TFTP , needn`t fill anything
Config File name The name of config file in FTP or TFTP server .
If "config file name " is unfilled means the phone will auto search

the right config file from FTP /TFTP server by MAC address matching.

Config Encrypt Key If the config file is encrypted. need to fill in the encrypt key. config file support aes (The Advanced Encryption Standard) 64 After download config file to phone , the phone will auto make decryption

Protocol Type update protocol selecting, Support FTP and TFTP protocol.

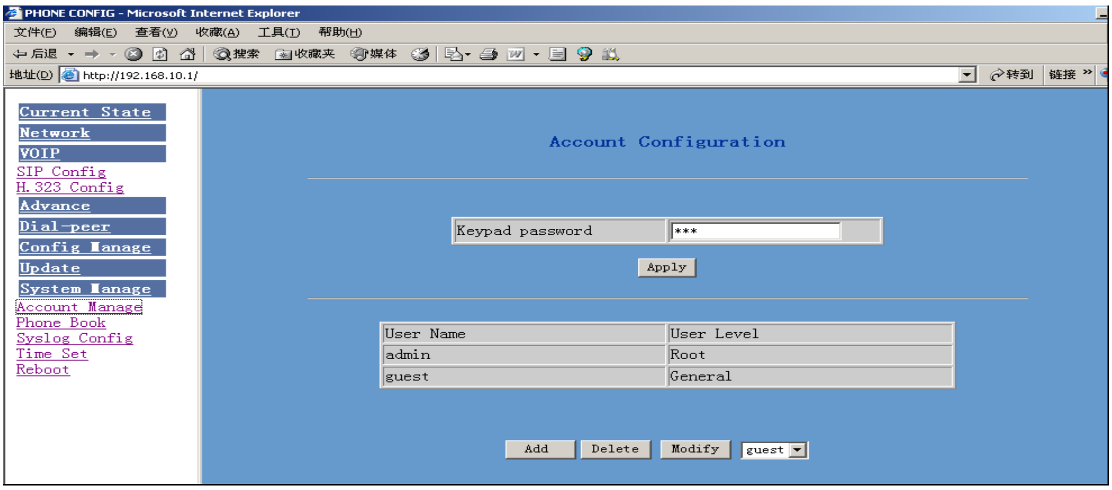
System Manage



Support account manage, syslog and reboot

Account Manage (maximum 5 accounts)

Users can edit users (add or delete) account and modify existing users' authority on this web page.



Definition of each parameter described as below

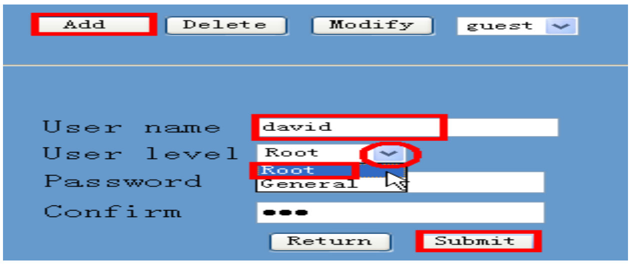
Keypad password	Set keypad operation config password, default is 123,users can input new password then click "apply" button ,"submit success" info will show on screen, reset password successfully
User Name	List existing phone user account name

User Level	Show existing user account level [two option]:Root and General: Root level users have the right to modify config; General level users have the right to read-only
Add	Add user account to IP phone
Delete	Delete increased user account
Modify	Modify increased user level and password

Operation Example

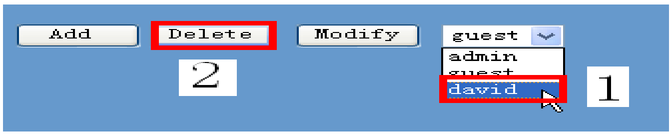
- Add one new account

Click “Add” button →input User name (No-Modify) →Choosing User level from dropdown menu →set new user password →confirm password →submit the new account info by clicking “submit” button →show “submit success” on screen →return to account configuration interface by clicking “Return” button



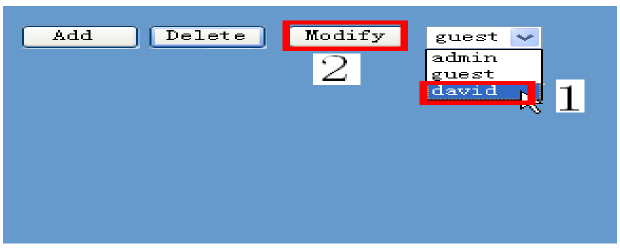
- Delete increased account

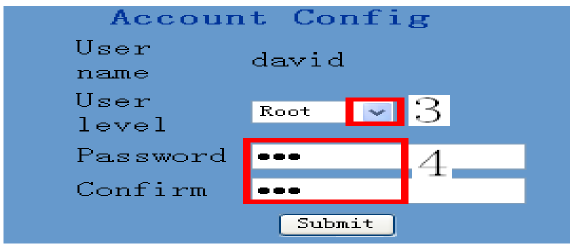
Choosing the account need to del. from dropdown menu→ Delete account by pressing “Delete button” →show “Submit Success” on screen



- Modify increased account (For Root-level user account only)

Choosing the modified account →enter below interface →modify user level or password →click “Submit” button to submit the modification



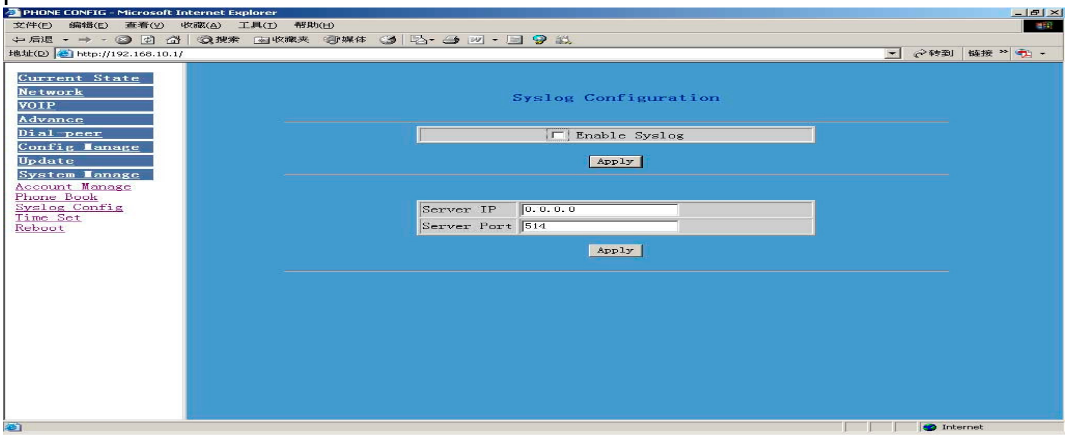


Owing to the phone's default account : accounts of the administrator level-admin account and the ordinary level - guest account are all weak account and weak password, the username and password will be easily to guess on public network, so the user had better modify the administrator and ordinary user.

Enter with manager level when making modification , create a administrator account and a browse account (you'd better not set the name as admin, administrator, guest, etc.) , set password and then save configuration , entering with new manager account, delete default manager and browse account and save configuration , security will be enhanced!

Syslog Configuration

Users can star or close syslog function and config syslog server IP address & port.

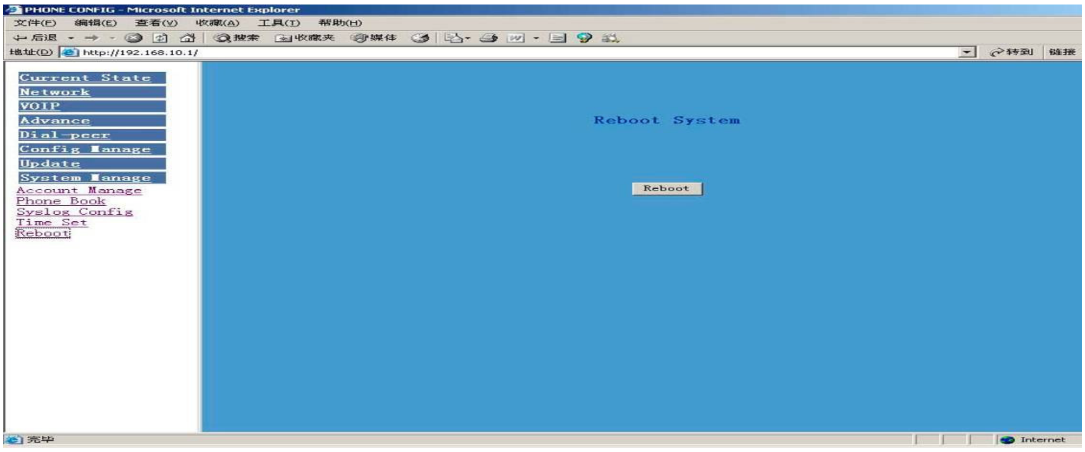


Definition of each parameter described as below

Enable Syslog	Config enable/disable syslog function, choose it and then click “Apply” button to go into effect
Server IP	Config syslog server IP address
Server Port	Config syslog server port, click “ Apply” button after inputting server IP & server port to take effect

System Reboot

Once any change of phone configuration is made, users need to reset IP phone to go into effect. Users should save the modified configuration before system reboot, otherwise the phone system configuration will go back to last saved setting. The system reboot interface as below


PHONE CONFIG - Microsoft Internet Explorer
http://192.168.10.1/
Current State
Network
VOIP
Advance
Dial-peer
Config Manage
Update
System Manage
Account Manage
Phone Book
Syslog Config
Time Set
Reboot
Reboot System
Reboot
Internet