User Manual

TVP304

IP Phone

Version 1.0

index

1 Welcome

TVP304 series of SIP phones are innovative IP telephones that offer a rich set of functionality and superb sound quality. They are 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.

2 Installation

TVP304 series IP phones are designed to look and feel like standard telephones. The following photo illustrates the appearance of an TVP304 IP phone and the use of its key buttons.

2.1 Package List

The TVP304 phone package contains:
1) One TVP304 phone
2) One universal power adapter
3) One Straight Ethernet cable

2.2 Safety Compliances

The phone should only be operated with the universal power adapter provided with the package. Damages to the phone caused by using other unsupported power adapters would not be covered by the manufacturer's warranty.

3 Product Overview

TVP304 IP Phone is a next generation IP network telephone based on industry open standard SIP (Session Initiation Protocol) and H.323. Built on innovative technology, TVP304 IP Phone features market leading superb sound quality and rich functionalities at massaffordable price.

3.1 Key Features

- Support two models: 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 、G.729A/B 、G.726，and G.722
- Redundancy SIP server (or Gate Keeper): Can auto swap address between two servers address
- NAT transversal: Support STUN client, AVS and Citron 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
- Support standard voice features such as numeric Caller ID Display, Call Waiting, Hold, Transfer, Do-Not-disturb, Forward, in-band and out-of-band DTMF, Hotline (off hook autodial), auto answer,ban outgoing
- Full duplex hands-free speakerphone, redial, call log, volume control, voice record with indicator
- Support standard encryption and authentication (DIGEST using MD5,

MD5-sess)

- 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 operation (phone keypad 、 Web interface and Telenet) or automated centralized configuration file via TFTP or HTTP.
- Support firmware upgrade via TFTP/FTP and HTTP
- Support syslog, can send event of phone to syslog server.

3.2 Hardware Specification

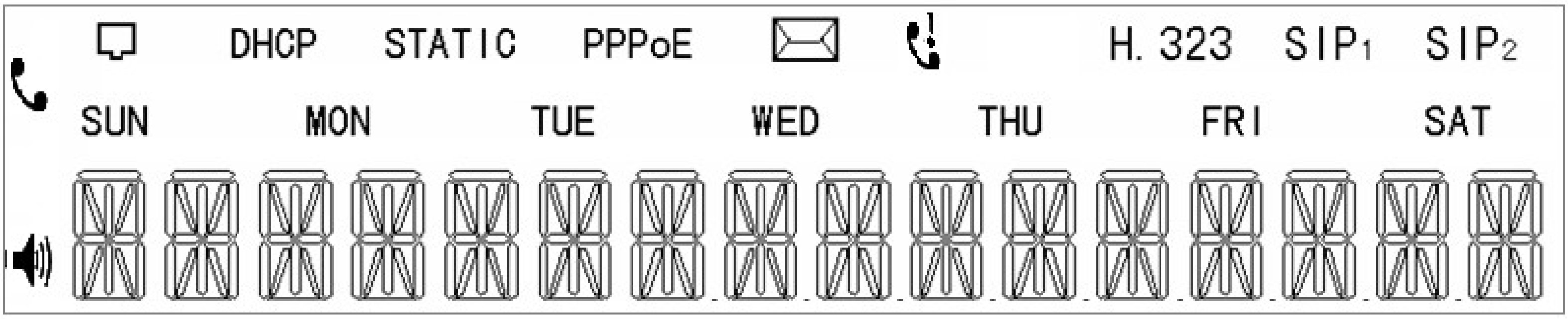
The table below describes the hardware specification.

Item		Specification
Power Adapter	Input	110-220V AC
	Output	5V DC 1A
Port	WAN	10/100Base T RJ-45
	LAN	10/100Base T RJ-45
Power Consumption		2.8W/1.9W
LCD		74mmx29mm
Operating Temperature		0～60℃
Relative Humidity		5～95%
Volume		
Weight		

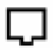
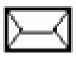
4 Basic Operations

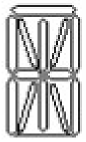
4.1 Get Familiar with LCD

TVP304 phone has a LCD of 74mmx29mm size and a backlight. Here is the display when all segments illuminate:



The LCD is equipped with a backlight. When the phone is in the normal idle state, the backlight is off. Whenever an event occurs, the backlight turns on automatically and brings the user's attention.

Icon	LCD Icon Definitions
	**Network Status Icon:** **FLASH in the case of Ethernet link failure or the phone is not registered properly.**
DHCP	**Network Status Icon:** **ON when Phone work on DHCP model and FLASH when DHCP client is not successful.** **OFF when Phone is work on another model**
STATIC	**Network Status Icon:** **ON when Phone work on Static model and FLASH when IP address is disable.** **OFF when Phone is work on another model**
PPPoE	**Network Status Icon:** **ON when Phone work on PPPoE model and FLASH when PPPoE is not successful.** **OFF when Phone is work on another model**
	**Message Status Icon:** **ON and Flash if Phone has new message include text message or voice record**
	**Missed call display** **ON and Flash if Phone has missed call and not be read.**
H. 323	**H323 register Status:** **FLASH when enable register and can not register successfully,** **ON when enable register and register successfully,** **OFF when disable register**
SIP1	**SIP1 (Public sip server) register Status:** **Flash when enable register and can not register successfully,** **ON when enable register and register successfully,** **OFF when disable register**
SIP2	**SIP2 (Private sip server) register Status Icon:** **Flash when enable register and can not register succedfully,** **ON when enable register and register succedfully,**

	OFF when disable register
	**Handset Status Icon:** **ON if off hook** **OFF if on hook**
	**Hand-free Status Icon:** **ON when phone work on hand-free model** **OFF when IDLE or work on handset model**
..SUN MON	**Weekday Status Icon:**
	**Numerical Numbers and Characters:** **0 - 9** *** # @** **A, B, C, D, E, F, G, H, I, J, K, L, M, N, O, P, Q, R, S, T, U, V, W, X, Y, Z**

4.2 Get Familiar with Keypad

TVP304 phone has a 28-button keypad.

Key Button	**Key Button Definitions**
0 - 9, *, #	Digit, star and pound keys are usually used to make phone calls
Sysinfo	Display basic information, including IP address and gateway address
ENTER	Enter key
Exit	Back key
Volume	Adjust volume by revolving
MENU	Enter MENU mode when phone is in IDLE mode.
UP	Previous menu item when phone is in IDLE mode Or increase handset/speakerphone volume
DOWN	Next menu item when phone is in IDLE mode Or reduce handset/speakerphone volume
HOLD	Temporarily hold the active call
Transfer	Transfer the active call to another party or Enter 3-way conferencing call.
Redial/Send	Dial a new number or Redial the number last dialed. After entering the phone number, pressing this key would force a call to go out immediately before timeout
SPEAKER	Enter hands-free mode
DEL	Delete a key entry, call log, voice mail and etc

MUTE	Mute an active call;
OUT	Dial call records
IN	Incoming call records
RECORD	Enter voice record menu
PBOOK	User can make a call directly by # button if choicing the proper person in phone number book.

4.3 Make Phone Calls

4.3.1 Make Calls Using Regular Phone or Extension Numbers

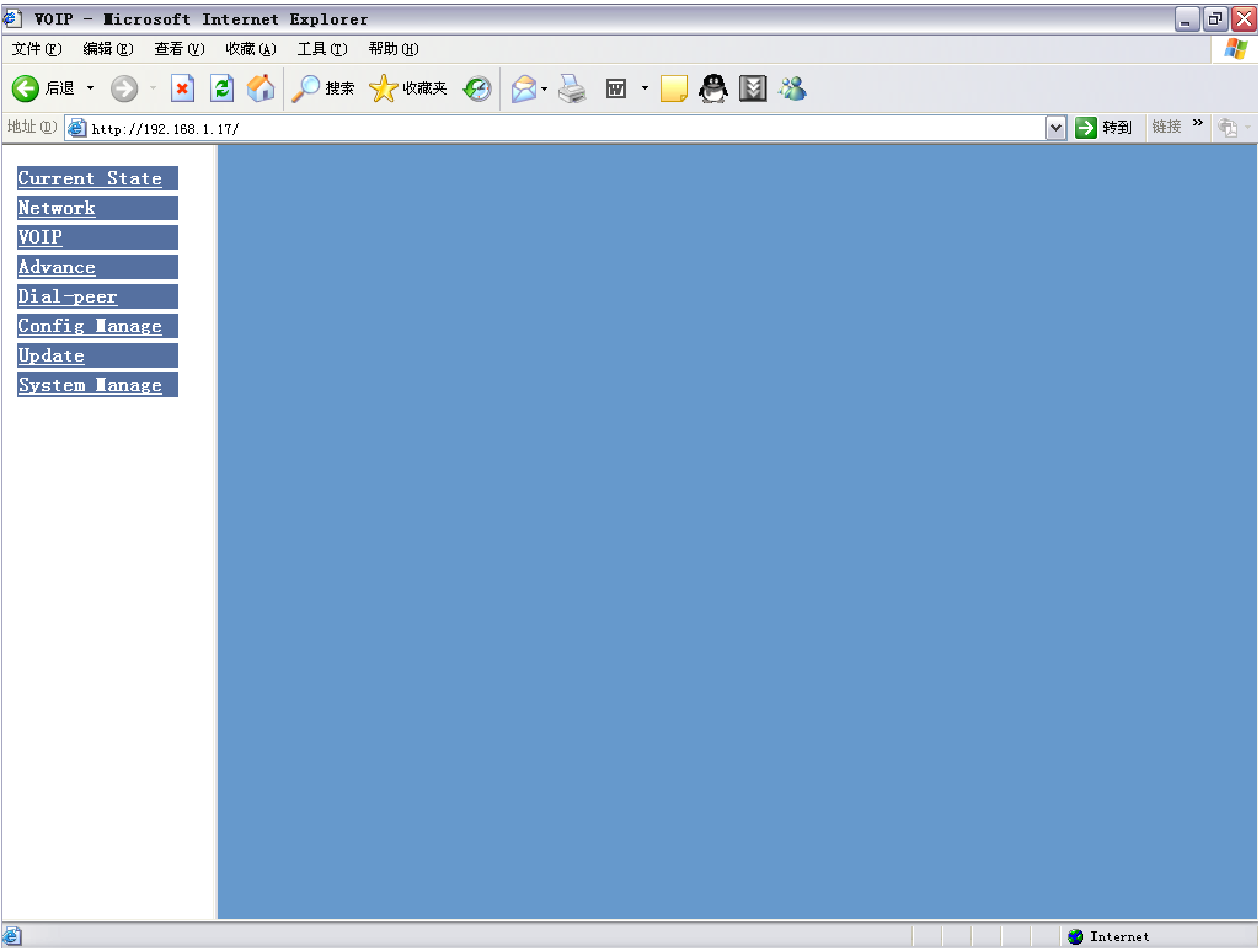
There are three ways to make phone calls:
1. Pick up handset or press SPEAKER button, and then enter the phone numbers
2. Press the SEND/REDIAL button directly to redial the number last called.Once pressed, the last dialed number will be displayed on the LCD as the corresponding DTMF tones are played out and an outgoing call is sent.
3. Browse the OUTGOING/INCOMING history and press the # button. Once pressed, the last dialed number will be displayed on the LCD as the corresponding DTMF tones are played out and an outgoing call is sent.

Examples:
To dial a number on the proxy, such as 1001, simply pick up handset or press speaker phone, dial 1001 and then press the “SEND/(Re)Dial” button.
To dial a PSTN number such as 62281486, you might need to enter in some prefix number followed by the phone number. Please check with your VoIP service provider to get the information. If your phone is assigned with a PSTN-like number such as 62281493, most likely you just follow the rule to dial 62281486 as if you were calling from a regular analog phone, followed by pressing the “SEND/(Re)Dial” button.

5 Configuration with WEB

The IP Phone Web Configuration Menu can be accessed by the following URI: http://Phone-IP-

Address. The default LAN IP address is “**192.168.10.1**” and WAN IP address is “**192.168.1.179**”. If the web login port of the phone is configured as non-80 standard port， then user need to input http://xxx.xxx.xxx.xxx：xxxx/， otherwise the web will show that no server has been found),it will be shown as follows:



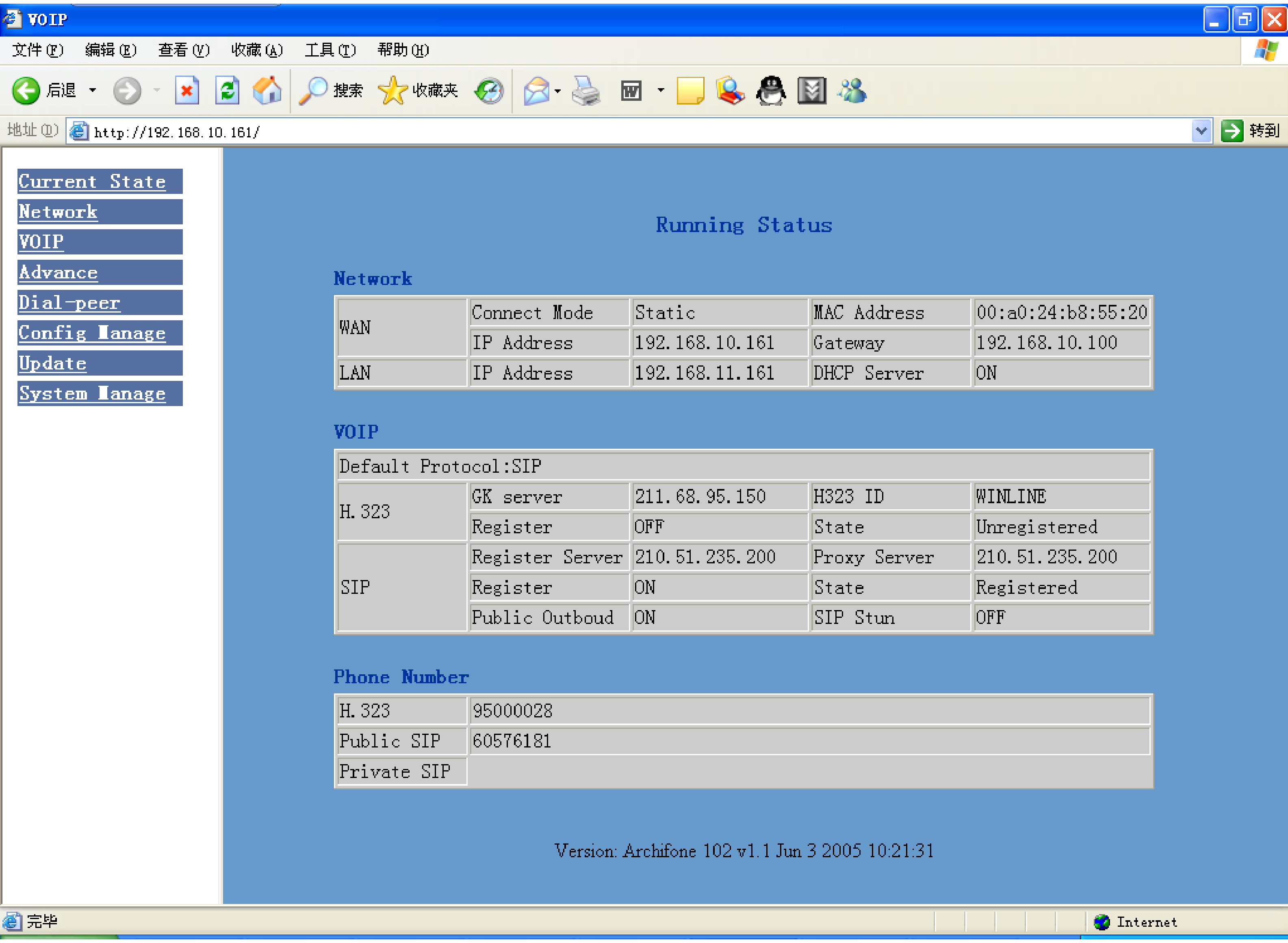
5.1 Current state

On this page user can gather information of each commonly-used parameter of the phone, it is shown as the following figure： the network section shows the current WAN, LAN configurations of the phone： including gaining way of WAN IP and IP （static state, DHCP, PPPoE)， MAC address， WAN IP address of the phone， LAN IP address of the phone， opening state of LAN DHCP server.

The VoIP section shows the current default signaling protocol in use， and server parameter in use of each protocol： including GateKeeper IP of H323， H323ID， whether enables register，whether has registered on GK； Register server IP of SIP，proxy server IP，whether enables register， whether has registered on register server， whether enables outbound proxy，whether enables STUN server；

The Phone Number section shows corresponding phone number of each protocol；

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



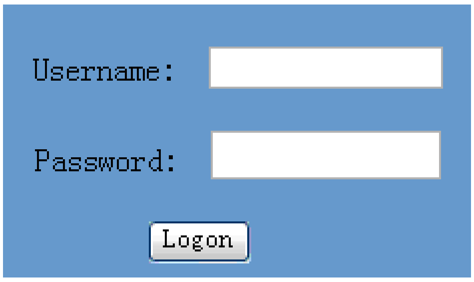
5.2 User verification

User should login before configurating dialogue machine.

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

Administrator account： the default username and password are all" admin"， this user can

configure the system.
Note： after inputting username and password， user press carriage return directly to enter the page.



5.3 Network configuration

5.3.1 Wide area network（WAN）

User 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.

Note： if IP address has been modified, the web page **will no longer respond owing to the modification， so new IP address should be input in the address field now.**

WAN Configuration

Active IP	Current Netmask	MAC Address	Current Gateway
192.168.1.97	255.255.255.0	00:01:02:03:04:06	192.168.1.68

◉ Static ○ DHCP ○ PPPOE

Static	IP Address	192.168.1.97	Netmask	255.255.255.0
	Gateway	192.168.1.68	DNS Domain	voip.com
	Primary DNS	192.168.1.68	Alter DNS	192.1.1.1

PPPOE	Server	ANY	User	user123	Password	●●●●●●●●●

Apply

Configuration Explanation：

Active IP	Current Netmask	MAC Address	Current Gateway
192.168.10.77	255.255.255.0	00:01:02:12:34:57	192.168.10.86

Current phone IP， subnet mask， mac address and current phone IP；

Static	IP Address	192.168.10.77	Netmask	255.255.255.0
	Gateway	192.168.10.86	DNS Domain	voip.com
	Primary DNS	192.168.10.86	Alter DNS	192.1.1.1

， Select acquisition way of IP for WAN； This is single option； Configure static IP parameter for WAN：

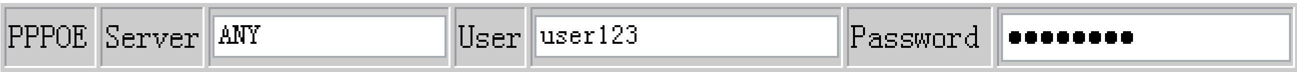
IP Address	192.168.10.77	Configure static IP address；
Netmask	255.255.255.0	Configure subnet mask；
Gateway	192.168.10.86	Configure IP address of the the phone；
DNS Domain	voip.com	Configure "dns

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；

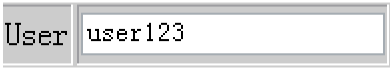
Main DNS server IP address；

The second DNS server IP address；

Configure PPPoE：

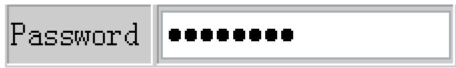


Service name， if PPPoE ISP has no special requirement for this name， generally is the default；



PPPoE account；

PPPoE

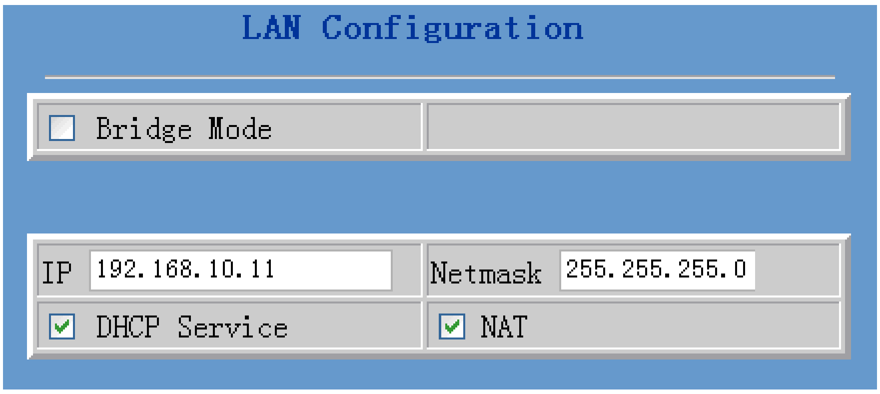


password；

Configure the parameter and then click "apply" to go into effect；

5.3.2 Local area network (LAN)

User can make local area network (LAN) configuration on this page, when bridging mode is selected, the local area network (LAN) configuration will no longer go into effect.



Configuration Explanation：

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；

Configure LAN static IP；

Configure LAN subnet mask；

Enable LAN port DHCP server；

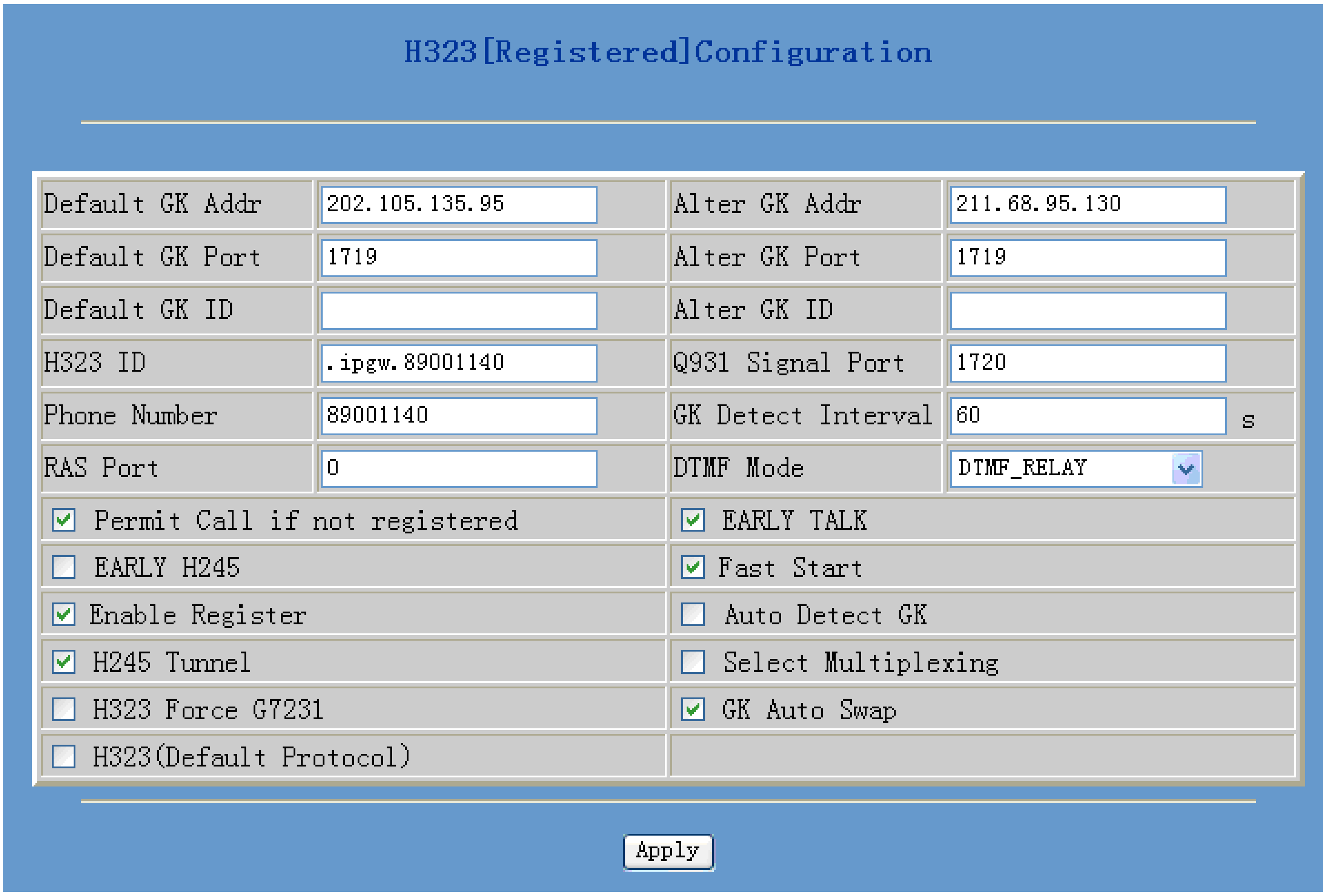
after user modify LAN IP， the phone will automatically modify the adjustment and save the configuration according to IP and subnet mask team DHCP Lease Table， user need to restart the phone to make DHCP server configuration go into effect;

☑ NAT Enable NAT；

5.4 VOIP configuration

5.4.1 H.323 configuration

User can configure specific parameter of H323 signaling protocol on this page;



Configuration Explanation:

show H323 register H323[Unregistered]Configuration state ; if register successfully,there will show Registered in the square bracket, otherwise show Unregistered;

Configure default GateKeeper IP address; Default GK Addr 211.68.95.150

Configure default GK port; Default GK Port 1719

Configure default GK ID; if no special requirement of GK, user don't need to fill in anything; Default GK ID

The system initiates Q931 signal port, the default is 1720; Q931 Signal Port 1720

Configure the net gate RAS register port for the system; terminal user can logon to gatekeeper through RAS passage and make a request for allowing to initiate the call request. If the request has been allowed, then the gatekeeper will return a transport address (with IP address and port number) as the call signaling passage of the called party; RAS Port 0

DTMF Mode DTMF_RELAY

DTMF_RELAY
DTMF_RFC2833
DTMF_H245-STRING
DTMF_H245-SIGNAL

Fast Start

Auto Detect GK

Configure DTMF mode, RTP mode, RFC2833 mode, H245-string mode and H245-signal mode;

Permit Call if not registered

Configure permission for no-registered call, allow to initiate call without net gate register;

Early245 configuration, which means that when initiating a call,the 225 message transmission begins at the same time with 245 message transmission, the default is Disable; EARLY H245

Enable Register

Configure enable/cancel register

Configuration for transferring 245 message package to 225 message package; H245 Tunnel

Configure H323 to run the talking only by G.7231 encode, the default is Disable; H323 Force G7231

Configure the phone use H323 protocol as default call protocol; H323(Default Protocol)

Configure quick start mode to start H323 call;

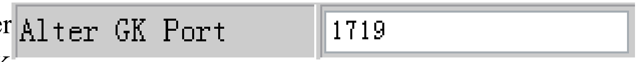
Configure multipexing of logical channel, the default is Disable;

Configure the phone can receive IVR, such as the voice prompt, dialing of PSTN color ring;

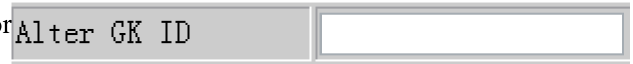
Configure GK backup and enable GK detecting and auto-swap functions, the phone will automatically swap to GK backup server when there is no response from default GK, and test the default GK; if the default GK recovers response, the phone will automatically swap to the default GK.

Configure GK backup server IP;

Configure server port for GK backup;



Configure ID for GK backup;



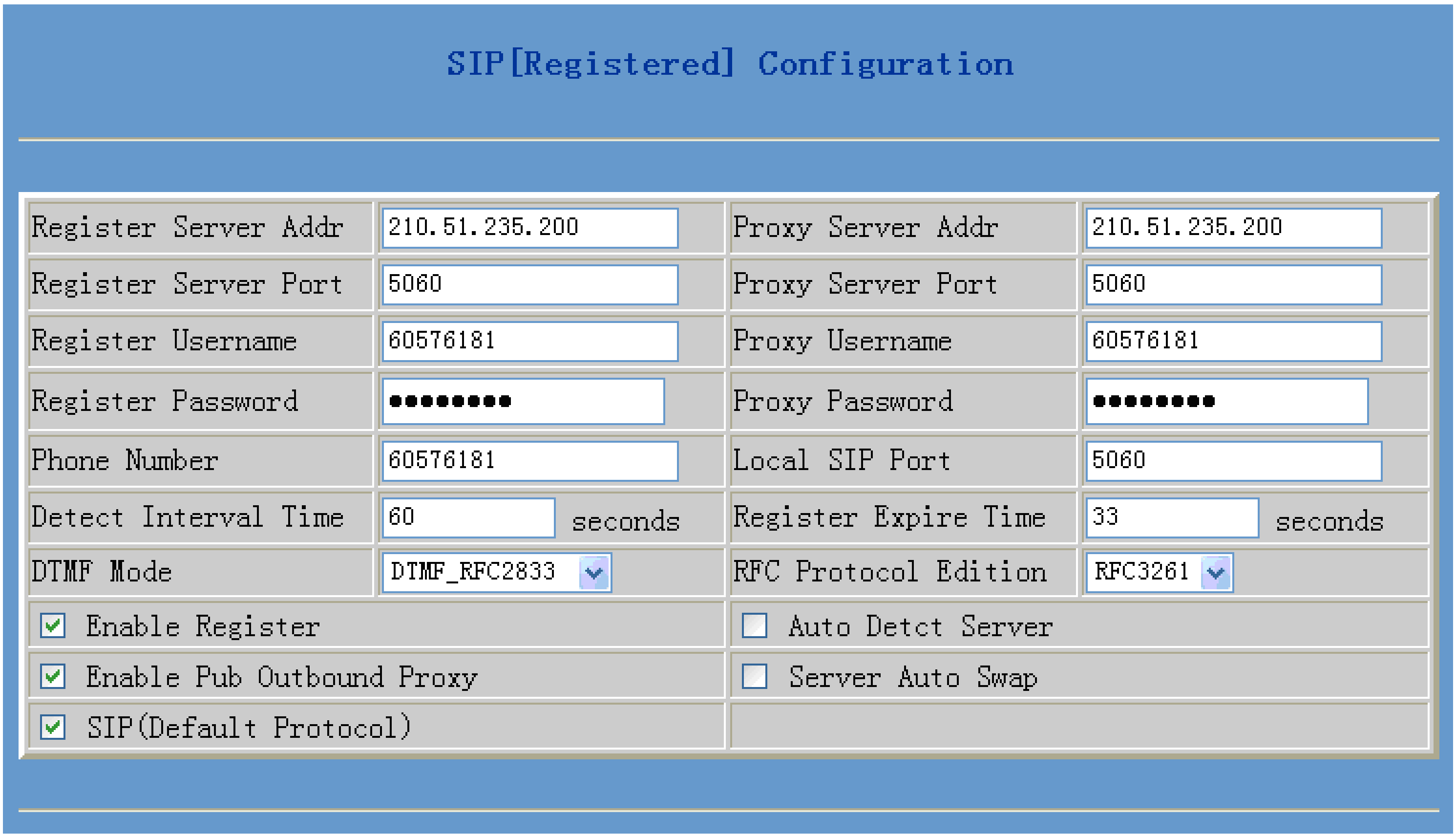
GK detection interval time configuration, the unit is second;

Enable the phone's auto-swap to GK;

Configure the phone to detect GK automatically

5.4.2 SIP configuration

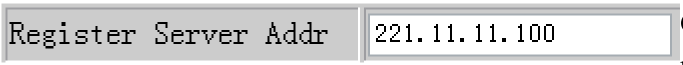
User can configure specific parameter of H323 signaling protocol on this page;



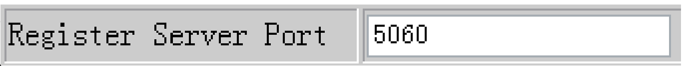
Configuration Explanation:

show SIP register 

state ; if register successfully,there will show Registered in the square bracket, otherwise show Unregistered;

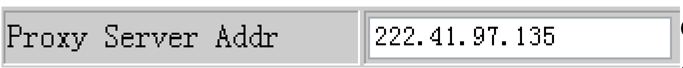


Configure SIP register server IP address;

Configure SIP register server signal port;


Configure SIP register account (usually it is the same with the port number that configured, some special SIP servers will have different port configurations,then the port configuration needs to be configured to be numbers, here the configuration account can be arbitrary character string);

Configure password of SIP register account;

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）；

Configure SIP proxy server signal port；

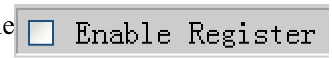
Configure proxy server account；

Configure proxy server password；

Configure local signal port，the default is 5060（this port will go into effect immediately, the SIP call will use the modified port for communication after modification）

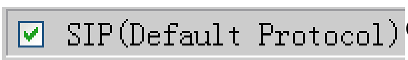
Configure expire time of SIP server register，the default is 600 seconds. If the expire time that server requires is more or less than that configured by the phone，the phone can automatically modify it to the recommended time limit and register；

Configure detection interval time of the server，if the phone enables SIP detection server function，the phone will detect once for whether the server has response every other detection interval time；

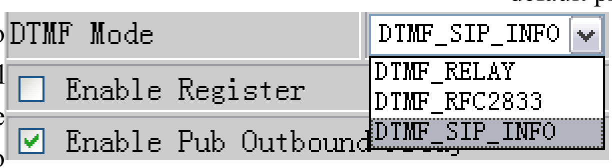


Configure enable/disable register；

Configure to enable public outbound proxy. If proxy server has been enabled，the phone will consider the user as using outbound proxy automatically. If the configuration has been disabled，the phone can still be registered to the server，but can't make SIP call；configuration of registered call by the phone will not have impacts on SIP point-to-point call；



Configure SIP of the phone as default protocol；



Enable the phone to use protocol edition.When the phone need to communicate with phones which is using SIP1.0 such as CISCO5300 and so on,then it should be configured into RFC2543 to communicate normally. the default is to enable RFC3261； DTMF sending mode configuration；three kinds：the above are basic configurations of SIP.

Note： if you want to register and call through server， you must configure corresponding numbers （which are usually SIP accounts）to local port， otherwise the phone will reject for sending out register message when it considers that there is no number.

☐ Auto Detct Server Configure automatic detection server of the phone;

Configure main and backup ☐ Server Auto Swap auto-swap server ; if the phone enables main and backup server function , the automatic detection and auto-swap functions should both be chosen;

After the aforesaid network and VoIP configurations have been configurated on the phone and internetwork communication **has been implemented**, the user can **make VoIP calls by the calling register and proxy.**

SOME ISP INTERNET MAY INHIBIT THE PHONE TO REGISTER AND CANCEL THE REGISTER IN SUCCESSION, SO USER HAD BETTER NOT APPLY OR REGISTER AND CANCEL SOON IN SUCCESSION AND SUBMIT REGISTRATION REPEATEDLY. SERVER MAY STOP RESPONSE OF DIALOGUE MACHINE, THEN THE PHONE RECEIVES NO CERTIFICATION OF REGISTER/CANCEL LOGIN REQUEST AND REGISTRATION STATE WILL SHOW AS INCORRECT!

5.5 Advance configuration

5.5.1 Net Service configuration

User can set up Telnet, HTTP, RTP port on this page and view DHCP table.

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

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;

Leased IP Address	Client hardware Address

Leased IP-MAC correspondence table of DHCP;

※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.

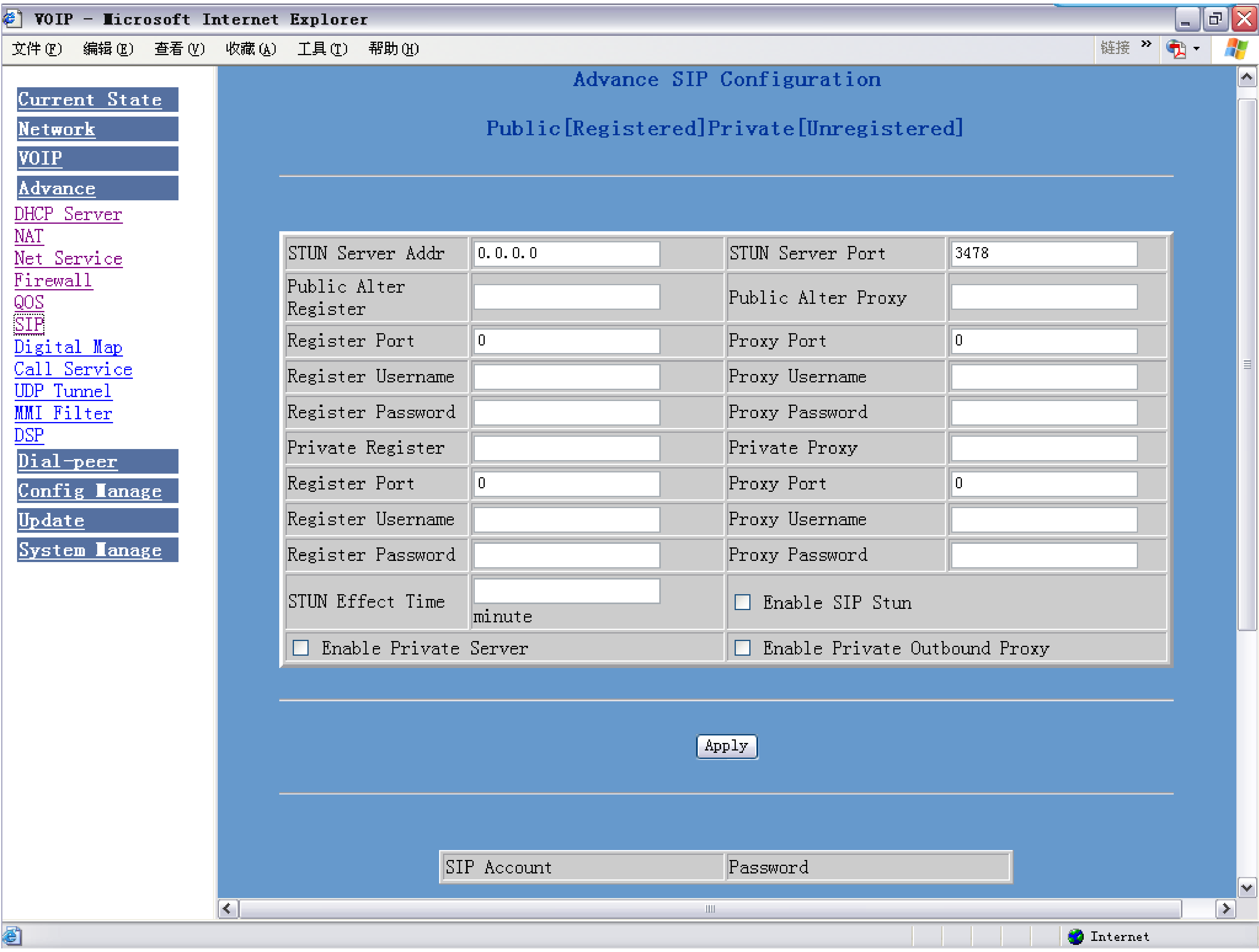
5.5.2 SIP advanced configuration

Set SIP STUN， private and backup server, user password and so on.

SIP STUN is a kind of server that used to realize the SIP's enablement of NAT， when the STUN server IP of the phone has been configured（generally the default is 3478）and Enable SIP Stun has been selected, conventional SIP server can be used to realize the phone's penetration of NAT.

Public backup server can implement the proxy of the dialogue machine through auto-swap function when no response to public server. When the phone detect response of public server， it will auto-swap to public server. Public backup server is redundancy backup of public server, it should have the same account with public server.

The phone's supports to two different kinds of SIP server concurrently can be implemented on private server. In this way user can register and use two different kinds of services concurrently.



Configure explanation of private server:

Public[Unregistered]Private[Unregistered]

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

Configure IP address of SIP STUN server; STUN Server Addr 0.0.0.0

Configure port of SIP STUN; STUN Server Port 3478

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;

Public Alter Register	10.1.1.11	Public Alter Proxy	0.0.0.0
Register Port	5060	Proxy Port	5060
Register Username	1234	Proxy Username	1234
Register Password	••••	Proxy Password	••••

Public backup server configuration； the specific configuration parameter has the same meaning with public server. It should be noted that the username and password should be the same with the public main server；

Private Register	210.25.132.124	Private Proxy	210.25.132.124
Register Port	5060	Proxy Port	5060
Register Username		Proxy Username	
Register Password		Proxy Password	

Private server configuration. specific configuration parameter has the same meaning with public server；

STUN Effect Time	minute

Interval time for STUN's detection on NAT type， the unit is minute；

☐ Enable SIP Stun

Configure enable/disable SIP STUN；

☐ Enable Private Server Register

Configure permit/deny private server register；

☐ Enable Private Outbound Proxy

Configure enable/disable private outbound proxy；

If user has accounts of a certain SIP server and each account has different password， then user should add each account and its corresponding password to the account& password table.

SIP Account	Password
1000	1000

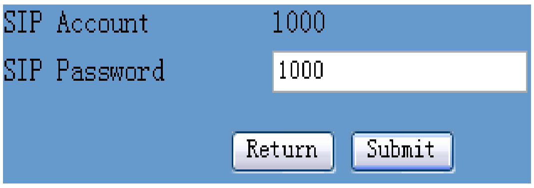
Configure display of account & password list；

Click Add to add account and password, it is shown as the following figure:

SIP Account	
SIP Password	
	Return Submit

Configure additive accounts
Configure additive passwords
Click submit to submit the configuration, click return to cancel the configuration and return;

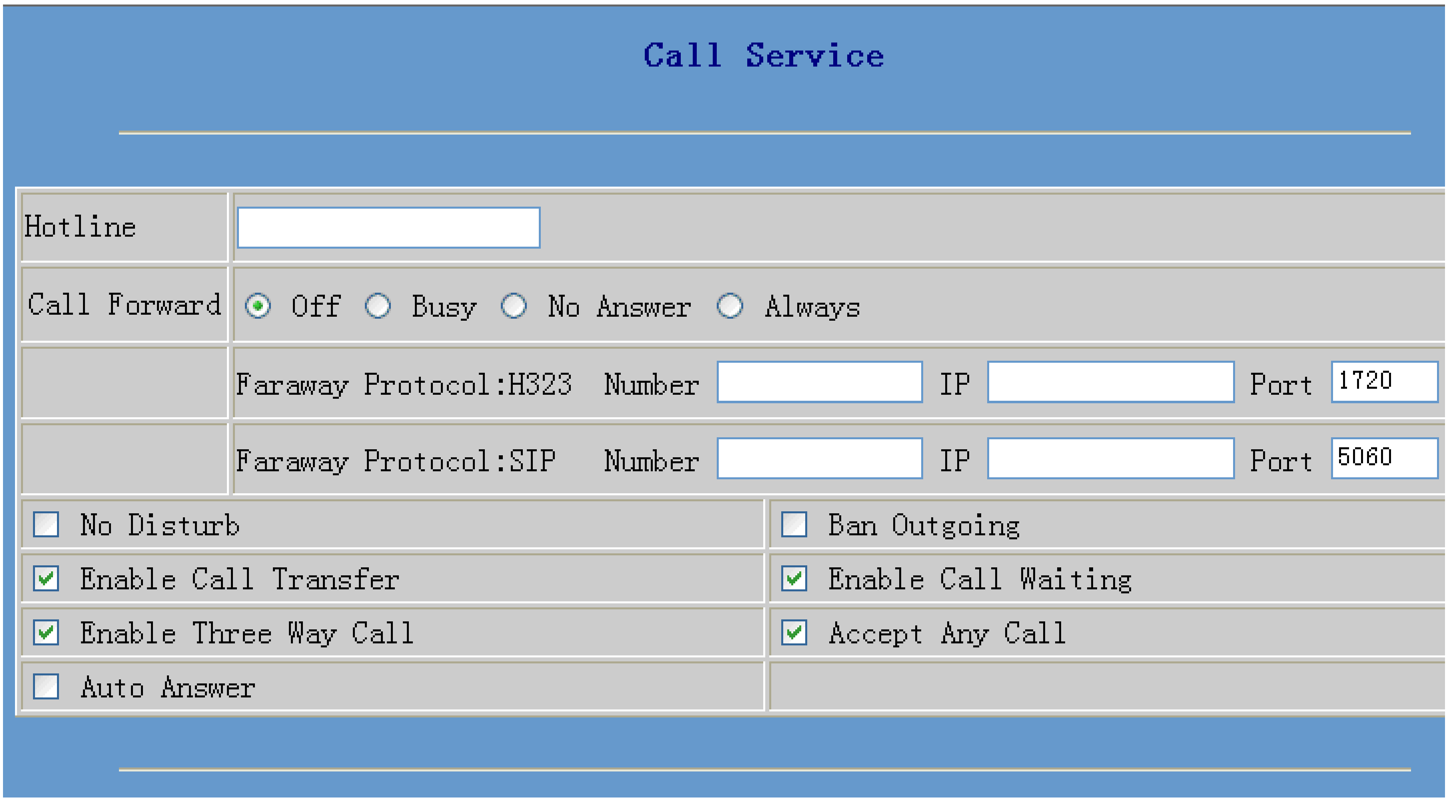
Select accounts that you want to delete from the drop-down menu , click delete. Select drop-down menu to select accounts that want to modify, click load to load the configuration and then click modify to modify:



Accounts to be modified, read-only;
Passwords to be modified;
Click submit to submit, click return to cancel the modification and then return;

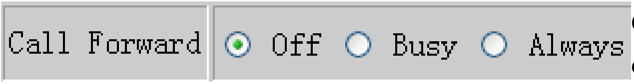
5.5.3 Value added service configuration

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.

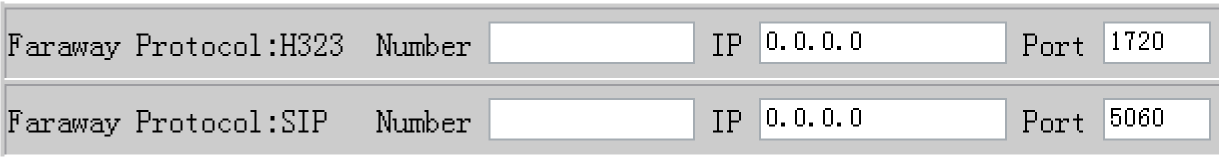


Configuration Explanation：

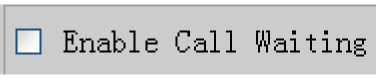
Configure hot-line number of the port. With this number of the port，this hot-line number will be dialed automatically as soon as off-hook and user can's dial any other number；



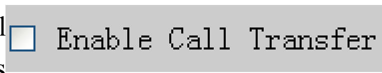
Call forwarding. The default is Disable；when busy is selected， if the number dialed is engaged after the phone has received a call, then it will automatically transfer to the configured number according to the following configuration；when always is selected， then the phone will directly transfer all the numbers that dial to this port to the configured numbers；



number IP configuration of call transfer (CT)；



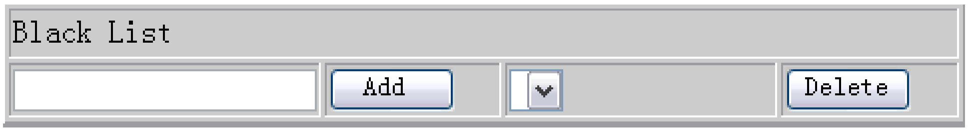
Configure enable/disable call waiting service ； After it is enabled, user can hold calls of the other party by hooking, with hooking again, the hold call can go on.



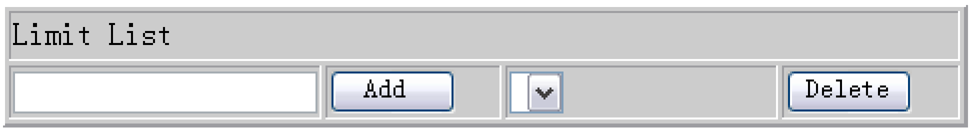
Configure enable/disable call transfer (CT) ； after it is enabled, user accept calls, with hooking and dial directly， the phone will transfer the calls according to the above configurations of the port number IP images；

Configure 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 dialing tone， after call completion to the third party, hooking again to recover the talk with the second part, then the three way call concurrently；

After the aforesaid configuration has been done, click apply to make them go into effect.



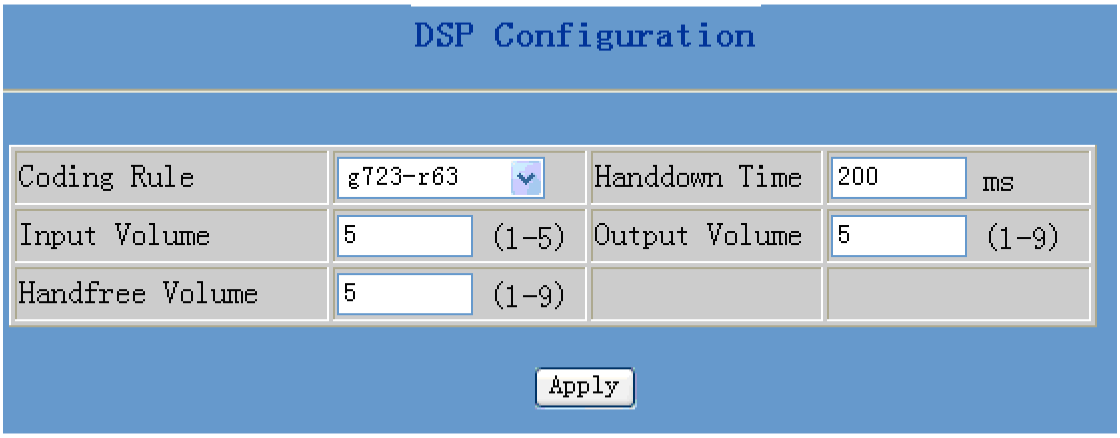
Configure add/delete blacklist. If user don't want to answer a certain number, pleasse add this number to the list, and then this number will be unable to get through the phone.



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.

5.5.4 DSP configuration

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

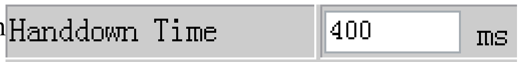


Configuration Explanation：

Configure output volume；

Configure input volume；

Configure handfree volume



Configure handdown time， that is, if the hooking time is shorter than this time, then the gateway will not consider the user has handdown；

5.6 Number binding configuration

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 configurating 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 swap.

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

Dial-Peer

Number	Call Mode	Destination	Port	Alias	Suffix	Del length
9T	h323	0.0.0.0	1720	del	no suffix	1
0T	sip	0.0.0.0	5060	del	no suffix	1

Add   Delete 0T   Modify 9T   Load

Configuration Explanation：

Number	Call Mode	Destination	Port	Alias	Suffix	Del length
0T	lifeline	0.0.0.0	0	no alias	no suffix	0
9T	sip	0.0.0.0	0	no alias	no suffix	0
1T	h323	0.0.0.0	1720	no alias	no suffix	0
8T	sip	255.255.255.255	5060	del	no suffix	1

Display of calling number IP image list；

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

Add

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 configurating 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.

Phone Number 010T

Configure the calling mode ： H323 and SIP；

Call Mode sip

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,then the default of h323 protocol is 1720, the default of sip protocol is 5060; lifeline required no configuration of this item, shown as 0;

Configure alias, this Alias(optional) 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;

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

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. 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. 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 leangth 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;

Delete selective number IP image; Delete 0T

Modify 0T Load If user want to modify a certain current number image, first select in the drop-down menu and then load the image parameter of the said number, click modify to make modification; of which:

this is the modified number. read-only; Phone Number 9T

To modify call mode; Call Mode sip

To modify destination address; this is IP or Domain (optional) 0.0.0.0

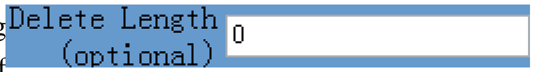
optional configuration item;

To modify destination phone port ; this is optional configuration item;

To modify alias ; this is optional configuration item;

To modify suffix ; this is optional configuration item;

To modify replacing length（if rep and del of alias have been configured）



Click submit to go into effect; click return to cancel configuration and return.

The basic application of the number IP table has been introduced, now let me introduce how to configure IP table of number to implement configuration of using multi-accounts concurrently:

For example, now user has a H323 account and two SIP accounts, then under the default condition, user can only make calls by the default protocol. Configure the number IP table to select the call protocol, then user don't need to select default protocol before making calls everytime.

The configuration process will not be repeated, now I will mainly introduce what kind of number IP image can implement this function.

By configuration, image table as follows will be gained:

Number	Call Mode	Destination	Port	Alias	Suffix	Del length
9T	sip	0.0.0.0	5060	del	no suffix	1
8T	sip	255.255.255.255	5060	del	no suffix	1
7T	h323	0.0.0.0	1720	del	no suffix	1

Image of 9T means when user configure public SIP server and register, then user just need to add a"9"before the calling number whenever making a call by public SIP;

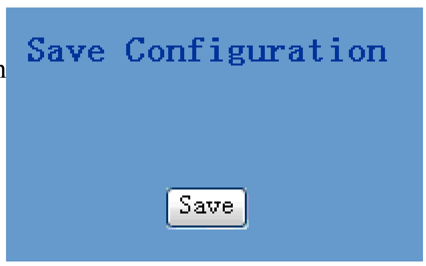
Image of 8T means when user configure private private server and register, then user just need to add a"8"before the calling number whenever making a call by private SIP;

Image of 7T means when user configure h323 server and register, then user just need to add a"7"before the calling number whenever making a call by H323 GK;

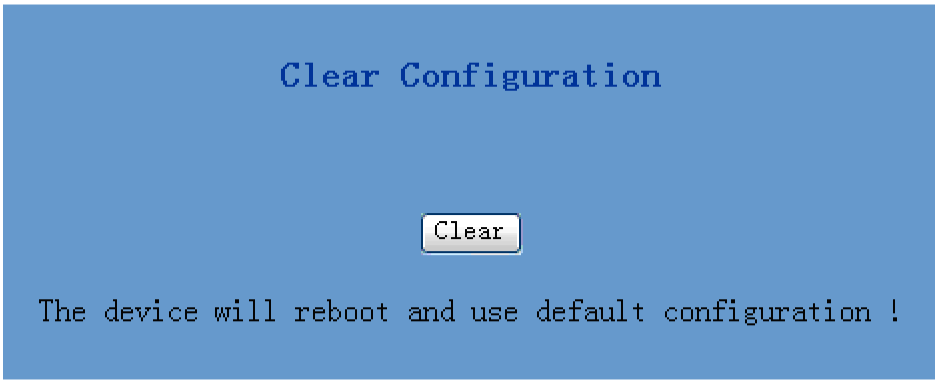
5.7 Save and Clear configuration

User can save the current configuration on this page.

The system configuration can



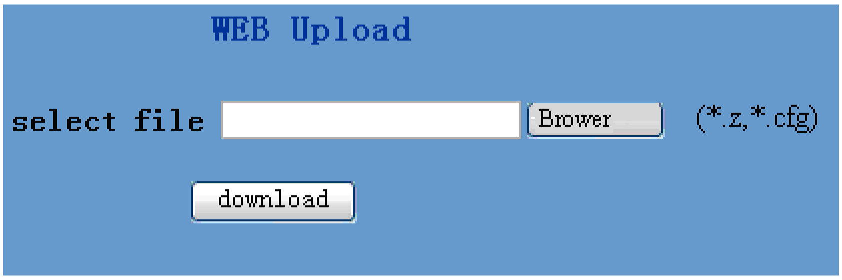
be set as factory default configuration on clear config page and the phone will restart automatically



5.8 Upgrade on-line

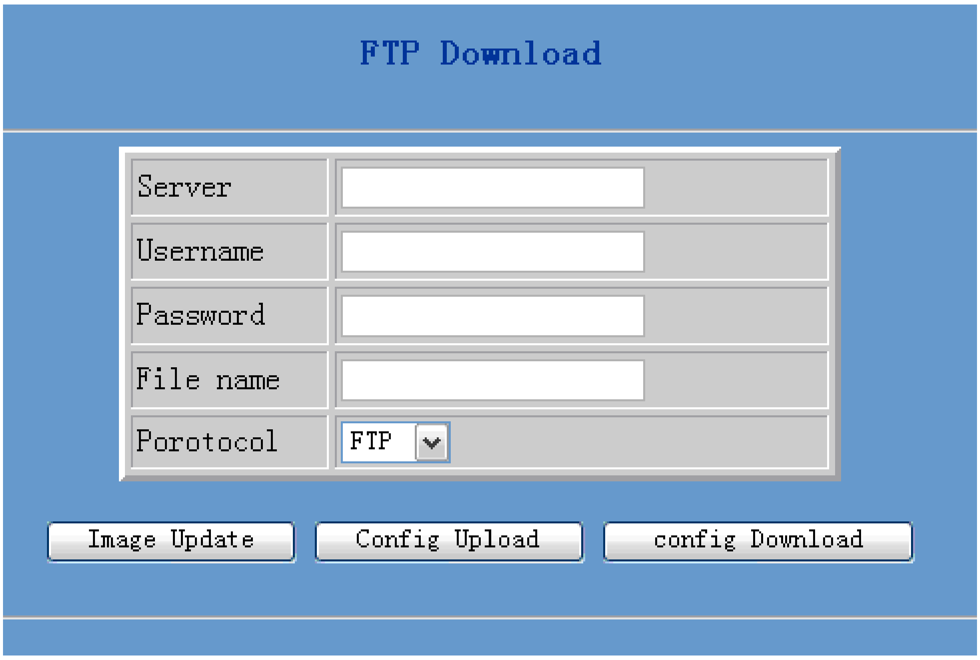
5.8.1 Upload WEB page

On this page, user can select the upgrade documents(**firmware or config file**) on hard disk of the computer directly to run the system upgrade. After the upgrade has been completed， restart the phone and it will be usable at once.



5.8.2 FTP download

On this page, user can upgrade system and configure files by FTP or TFTP mode.



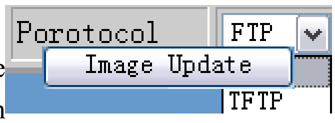
Configuration Explanation：

Configure upload or download FTP/ TFTP server IP address；

Configure username of the upload or download FTP server. If user select TFTP mode, username and password are not required to be configured；

Configure upload or download of FTP server password；

Configure upload or download system upgrade document or system layout file name.It should ne noted that system file take .dlf as suffix，configuration files take .cfg as suffix；



Select server type；

Click image update button， the phone will upgrade system file；

Click config upload button， the phone will upload its configuration files to FTP/TFTP server and save with names of user-defined configuration files；

Click config download button， the phone will download configuration files of FTP/TFTP server to the phone and the configuration will go into effect after restarting；

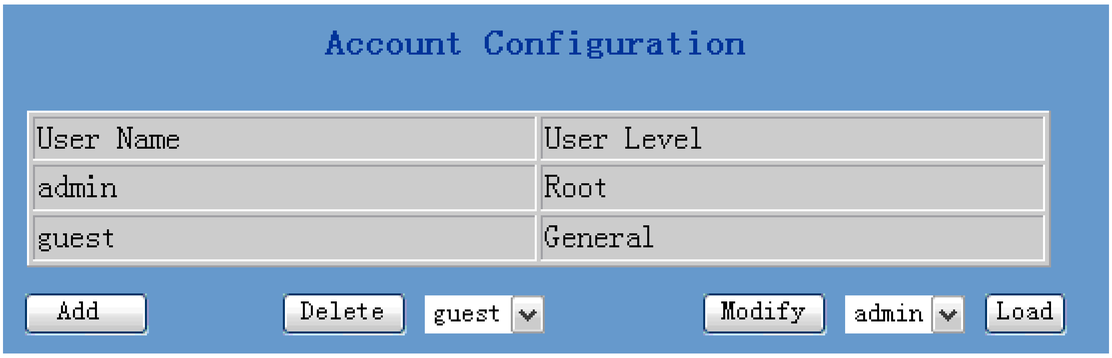
5.8.3 Configuration files WEB download

On this page, user can directly select the configuration files on the hard disk of the computer,and then make modification to the system configuration, after the download, restart the phone and the configuration will go into effect.

5.9 System management

5.9.1 Account management

On this page， user can add and delete users according to own needs and can modify user's authorities there have been.

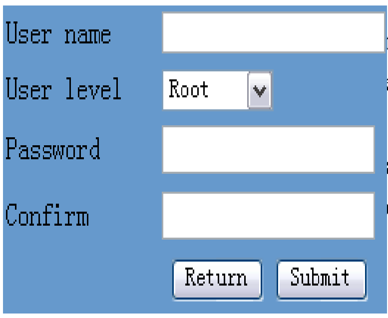


Configuration Explanation：

User Name	User Level
admin	Root
guest	General

display of phone user account list；

To add phone account； it will be shown at lower part of page as the following figure，of which：



ıts;

el； root possesses authorities to modify configuration，general

additive account；

ırd,to ensure correct setup of password；

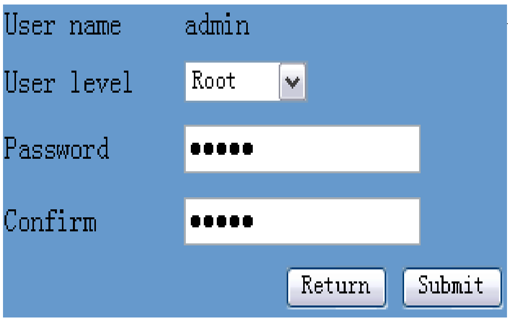
ck return to cancel configuration and return.

Select users that you want to delete in the drop-down menu， click

Delete.

To modify the chosen 

accounts , need to select account first, click load again and then click modify, it will be shown at lower part of page as the following figure, of which:



e;

s;

;

the modified user password;

> Owing to the phone's default account: accounts of the administrator level-admin and the ordinary level—guest are all weak account and weak password,the username and password will be easily to be guessed 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!

5.9.2 Phone book configuration

On this page ,user can save and configure telephone book.