User Manual

FV6020 Series
IP Phone
Version 2.0

Contents Catalogue

CHAPTER 1

1 Introduction

About This Manual

This Manual provides basic information on how to install and connect FV6020 IP Phone to the network. It also includes features and functions of FV6020 IP 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 FV6020 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.

FV6020 IP Phone

FV6020 IP phone is a stand-alone device, which requires no PC to make Internet calls. FV6020 IP 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 FV6020 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 our website timely, thanks for your understanding and continuous support.

CHAPTER
2

2 Installation

2.1 Appearance

FV6020 series IP phone are designed to look like the conventional telephones. The following photo illustrates the appearance of FV6020 IP phone.

2.2 Package list

1) **One FV6020 IP phone (Main body + Handset + Cord)**
2) **One Straight Ethernet cable**
3) **One universal power adapter**
4) **One User Manual**

Warning: Although the adapter of FV6020 series IP Phone is compliant with UL standard, please do not attempt to use other difference power adapter or cut off power supply during configuration or updating phone. Using other power adapter may damage FV6020 series phone and will void the manufacturer warranty.

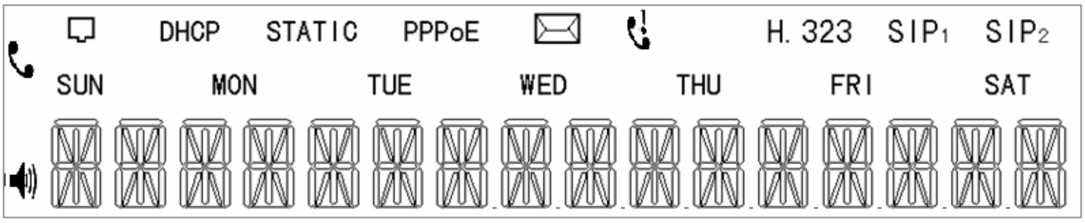
Caution: All operations of our product must abide by the instructions provided by IP Phone manufacturer. Any Changes or modifications to this product without formal authorization by manufacturer, or operation of FV6020 series phone in any way other than the instructions stated on this user manual will void the manufacturer warranty.

2.3 Installation

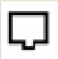
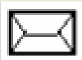
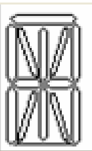
1) **Insert handset cord into the handset jack and left jack of IP Phone**
2) **Insert the power adapter's plug into the phone front Power jack (DC 5V) and the 2-prong plug end of which into grounded power outlet**
3) **Start up the IP phone by turning the front switch stated 'ON' & 'OFF' to 'ON'**
4) **Remove the LAN cable for Internet connection from your PC and connect it to WAN port of FV6020, then follow below installation checking way**
5) **If need to set up small LAN network, find the LAN cable in the box and connect between LAN port and your PC (PC is not required to set up for making a call)**

2.4 Check Installation by ICON of LCD

FV6020 IP Phone has a 74mmx28mm LCD that can display three lines of below characters each. 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. The definitions for each character displayed on LCD described as below table.

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 mode and FLASH when DHCP client is not registered successfully. OFF when Phone is work on another mode.**
STATIC	**Network Status Icon: ON when Phone work on Static mode and FLASH when IP address is disable. OFF when Phone is work on another mode.**
PPPoE	**Network Status Icon: ON when Phone work on PPPoE mode and FLASH when PPPoE is not registered successfully. OFF when Phone is work on another mode.**
	**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 successfully, ON when enable register and register successfully, OFF when disable register**
	**Handset Status Icon: ON when off-hook OFF when on-hook**
	**Hand-free Status Icon: ON when phone work on hand-free mode OFF when IDLE or work on handset mode**
SUN MON …	**Weekday Status Icon: Show the correct weekday according to the phone current date**
	**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**

3 Product Overview

CHAPTER 3

FV6020 IP Phone is a next generation IP network telephone based on industry open standard SIP (Session Initiation Protocol).FV6020 series IP Phone offer customer superb sound quality and rich functionalities at mass-affordable price.

3.1 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**
- **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 hand 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 Specifications

The below table describes the hardware specifications of FV6020 Phone

Item		FV6020
Power Adapter		Input: 100-240VAC 50~60Hz Output: +5VDC, 1200mA
CPU		Infineon PSB21553 150MHz
Port	WAN	10/100Base T　RJ-45 for LAN
	LAN	10/100Base T　RJ-45 for PC
Power Consumption		Idle:1.4W / Active:1.8W
LCD size		3 inch (74×28mm)
Operating Temperature		0～40℃(32°～104℉)
Storage Temperature		-10°～60℃(14°～140℉)
Relative Humidity		10～65% (Non-condensing)
Dimension (W×H×D)		11.6× 8× 3 inch (29.5×20.5×7.5cm)
Weight (packaging included)		2.07 lb. (0.94kg)
Certification		CE / FCC Part 15 Class B

4 Basic Operations

CHAPTER 4

4.1 Get Familiar with Keypad

FV6020 phone has a 28-button keypad. Definitions of each state as below

Key Button	Mode	Definitions
0 -9	In the dial-up mode	Decimal digit number 0-9, star and pound keys are usually used to make phone calls
	In the keypad configure mode	Rapid first button press display the digit number 0-9,rapid second button press display the English character or others
*	In the dial-up mode	As one part of phone number when call out
	In the call hold mode	Ready to call a third party's number to make three-way (or conference) calling
	In the keypad configure mode	Equal to the dots notation when input IP address
#	In the dial-up mode	As one part of phone number if "#" is the first dialed number, otherwise as the ending symbol to end up dialing
SYSINFO	In the IDLE mode	Continuous thrice press display assigned IP address of WAN Port, gateway's IP address and the phone number registered on public server
ENTER	In the keypad configure mode	Confirm configuration or enter submenu mode
Exit	In the keypad configure mode	Cancel configuration or exit submenu mode
MENU	In the IDLE mode	Enter menu mode and display the tree menu system
HOLD	In the keypad configure ode	Temporarily hold the active call
Transfer	In the keypad configure mode	Transfer the active call to another party or Enter three-way (or conference) calling.
REDIAL/SEND	In the dial-up mode	Redial the number last dialed, or force a call to go out immediately before timeout

SPEAKER	In the IDLE mode	Enter hands-free mode
UP	In the keypad configure mode	Go back to previous menu item or increase handset/speakerphone volume
DOWN	In the keypad configure mode	Go down to lower menu items or reduce handset/speakerphone volume
DEL	In the dial-up mode	Delete a key entry, call log, voice mail and etc.
	In the keypad configure mode	Modify the current configuration parameter or delete input info
MUTE	In the IDLE mode	Mute an active call;
OUT	In the keypad configure mode	Browse the outgoing call records (maximum saving 100 records)
IN	In the keypad configure mode	Browse the incoming call records(maximum saving 100 records)
REC	In the IDLE mode	Enter voice mail submenu (maximum saving 5 records)
PBOOK	In the IDLE mode	Access to phone book.(maximum saving 100 records)

Note:
1 Outgoing , Incoming ,Missed call support maximum 100 records, but power off will lose the record.
2 Support 5 records: one for local message ,one for welcome message ,three for incoming call leaving message

4.2 Dialing and Making Calls

4.2.1 Dialing general PSTN Phone or extension.

There are three dialing modes:

1. **Pick up handset or press SPEAKER button, and then enter the phone numbers, IP phone will send out the numbers with the DTMF tones.**
2. **In the off hook mode or Hands-free mode, Press the REDIAL/SEND button directly to redial the number last called. Once pressed, the last dialed number will be displayed on the LCD with DTMF tones and an outgoing call is sent.**
3. **Make use of Speed dialing mode: enter PBOOK→SPEED DIAL→INPUT INDEX submenu to input the index which correspond to the phone number you want to dial, then the phone could automatically dial the number.**
   Process: PBOOKENTER—INPUT INDEX (1,2,3…) + #

4.2.2 IP to IP calling

Making IP to IP calling is nearly same as dialing general PSTN Phone, there are three ways to set IP phone number and domain (more details please refer to 5.2.4)

1. **"Peer to Peer" calling mode: direct make calls and no need to set phone number thru proxy server (user could refer to Dial peer setting on web configuration charter).The phone should be operated under following condition (satisfy one option)**
   - **Requirement 1 both two phones are assigned the public IP address individually**
   - **Requirement 2 both two phones using private IP address should be on the same LAN.**
2. **Dialing the IP number configured by public agency, both two phones should be already registered on the public server.**
3. **Dialing the IP number configured by private agency, both two phones should be already registered on the private server.**

Examples:

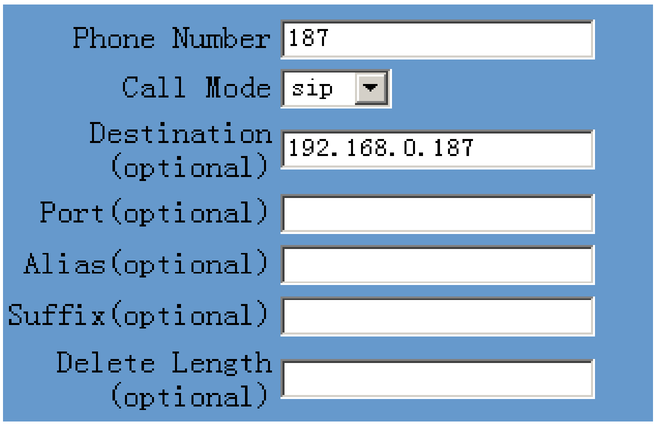
To dial a number on the proxy, such as 1001, simply pick up handset or press speakerphone, dial 1001 and then press the REDIAL/SEND button.

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 REDIAL/SEND button.

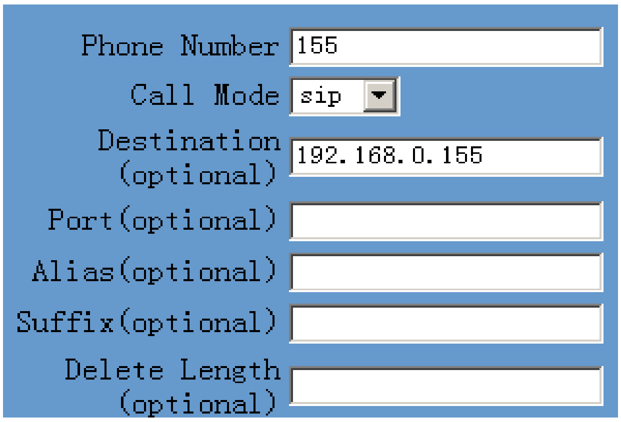
Example 1

If A dial number 187 , A can talk with B . If B dial number 155 , B can talk with A

A phone IP address 192.168.0.155
in dail peer config page



B phone IP address 192.168.0.187
in dail peer config page



4.3 Answering Calls

There are three ways to receive incoming calls:

- **Pick up handset to receive incoming calls**
- **Press the SPEAKER button to receive incoming calls**
- **Start-up the voice message function to record the incoming calls, afterwards listen to voice messages thru phone audio play .**

Operations: Enter REC→RECEIVED→NEW→UP OR DOWN to the LIST NO submenu (maximum is 3 message) 。

Precondition : must enable ☑ Enable Voice Record **in Advance /Call service**

Note: User can switch from a hands-free call to handset by picking up the handset. To switch from a handset call to a hands-free call, press hands-free button, and then hang up the handset.

4.4 Call hold

Pressing the HOLD button during current conversation enables you put an active calling on hold temporarily while a second call is answered or made, press this button again will go back to the previous call.

Precondition : must enable ☑ Enable Call Waiting **in Advance /Call service**

4.5 Call transfer

Press the TRANSFER button enables users could transfer an incoming call to the third party's number. When user A and user B both sides are on conversation, Users A press <TRANSFER > button ,and then dial the thid party user C . User A will hand up . User C ring , User B is on Hold state. User C pick up ,and talk with user B .

Precondition : must enable ☑ Enable Call Transfer **in Advance /Call service**

Process: Transfer button + the third party number

Example:A is talking with B , A press <Transfer> button and dial C number. A line break, then B will talk with C.

4.6 Three-Way Calling

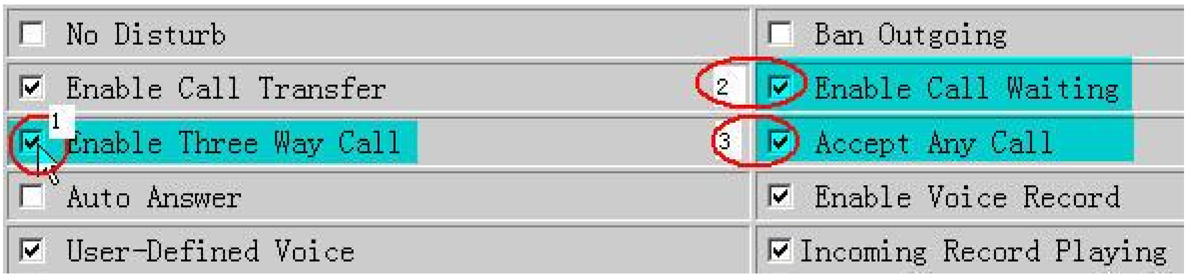
FV6020 IP phone support three-way (or conference) Calling. That is users could talk to more than one person (up to two) at the same time.

Process: press HOLD button →Dial third party's number->put through→ press HOLD button again →Press * button

Once the three-way initiator concludes the three-way calling, the other two sides can not continue the conversation call and hand up automatically.

Example:A are talking with B. A press HOLD button for holding B line , and dial the third party`s number , so A will talk with C . A press HOLD button again for holding C line , A dial * , make three-way (or conference) calling successfully.

Procondition: enable the three function as below picture.



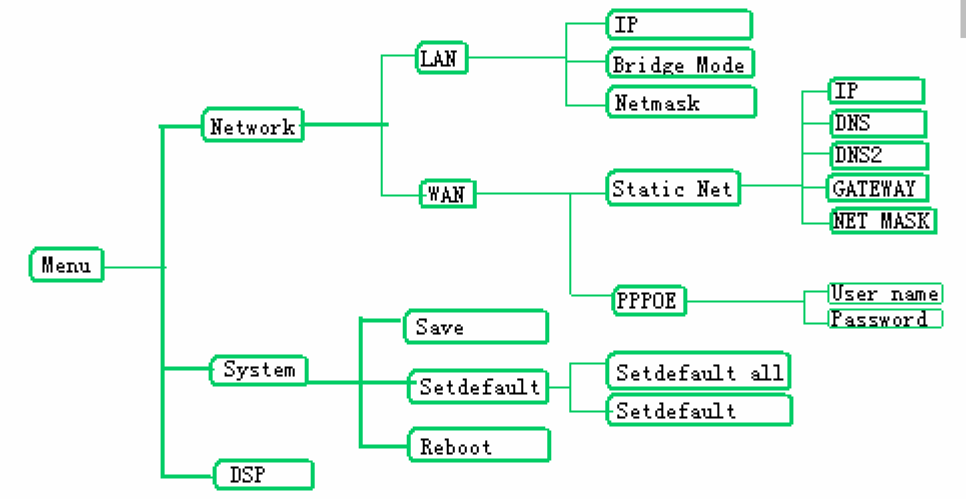
Note:
The function 4.4 & 4.5 & 4.6 could be started or closed thru system setting, so when you need to use above three functions please make sure of opening these functions.

CHAPTER 5

5 ConfigurationGuide

5.1 Config IP Phone through Keypad

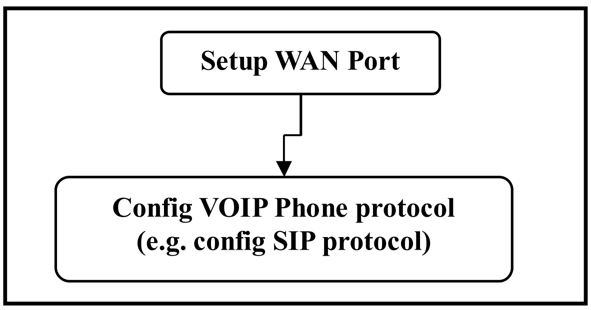
5.1.1Menu Structure:



5.1.2Keypad corresponding symbol table

Keypad	1 times	2 times	3 times	4 times
0	0	*	#	$
1	1	@	_	-
2	0	A	B	C
3	1	D	E	F
4	G	H	I	
5	J	K	L	
6	M	N	O	
7	P	Q	R	S
8	T	U	V	
9	W	X	Y	Z

5.2 configuration procedure for basic operations:



5.3 Minimum configuration

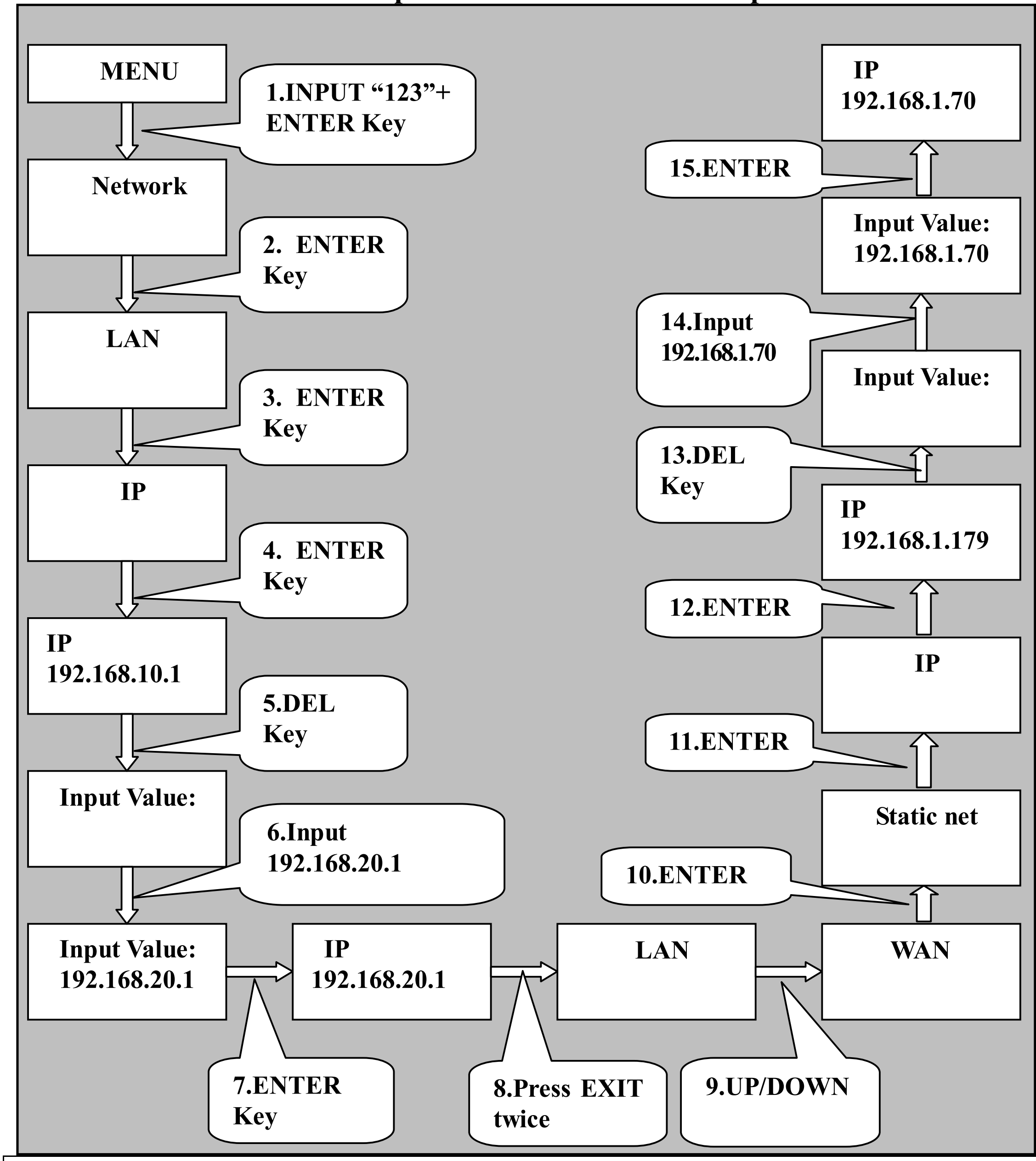
5.3.1Network configuration by keypad

Press MENU button→Input password "123" →Press ENTER button to confirm.
Configure WAN Port IP address: NETWORK→WAN→STATIC NET→IP
If using DHCP or PPPoE to get IP address dynamically, find user name on NETWORK→WAN→PPPoE→USER NAME
If using DHCP or PPPoE to get IP address dynamically, find password on NETWORK→WAN→PPPoE→PASSWORD
Configure WAN Netmask: Network→WAN→ STATIC NET →NETMASK
Configure WAN Gateway: Network→WAN→ STATIC NET →GATEWAY
Configure WAN DNS address: Network→WAN→ STATIC NET→DNS

Finished all above network configuration, Users can connect IP phone to internet successfully.

FV6020 IP Phone supports to modify IP address using keypad; Users could refer to the following flow chart to get a clear picture.

Configuration Example:
Set IP address of Phone WAN port to 192.168.1.70 and lan port to 192.168.20.1



Note:
IP Phone must config the correct WAN port IP address and Gateway IP address before connecting to internet. Due to FV6020 Phone Default mode is router ,(brideg mode is disable) so the WAN port IP could not be set to the same segment with LAN Port IP address when you modify WAN Port IP. Otherwise the,FV6020 can not get into internet. but if you had set wan port and lan port to same ip segment , you need to set fv6020 to factory default.
Process: power off ,and press # ,then power on ,input *#168 , and restart the fv6020
Default factory setting of WAN configuration is DHCP Client model
Default LAN Port IP=192.168.10.1 (Users could get them by pressing SYSINFO Key)

5.3.2 Common Shortcut Keys

- **Keep pressing 1 key for three seconds, phone transfer from internet connecting to Static mode**
- **Keep pressing 2 key for three seconds, phone transfer from internet connecting to DHCP mode**
- **Keep pressing 3 key for three seconds, phone transfer from internet connecting mode to PPPoE mode**
- **Keep pressing 5 key for three seconds, phone change default protocol to SIP**

5.4 Reboot IP Phone

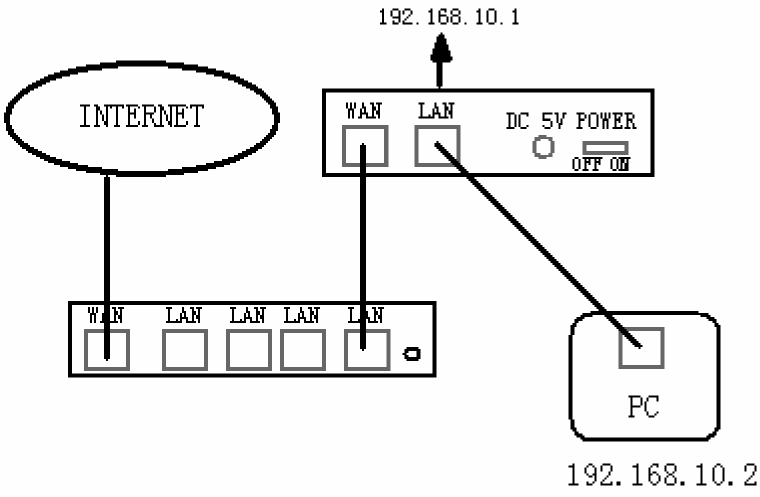
Enter MENU→SYSTEM→REBOOT submenu to reboot FV6020 IP Phone

Note: if no responding on phone, please cut off power supply to reboot phone.

CHAPTER 6

6.0 Web configuration

6.1 Physical connection

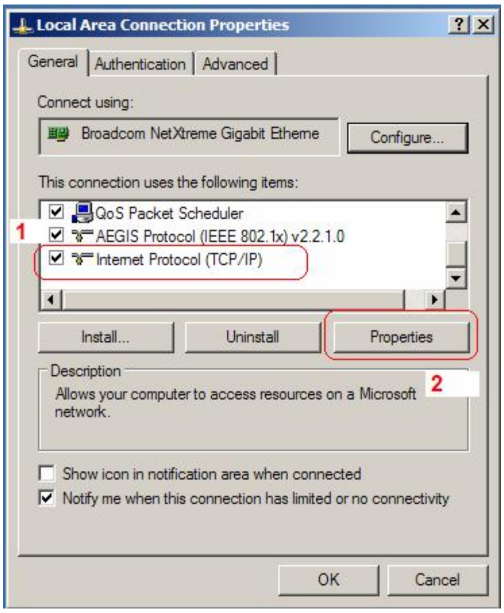


6.2 Preparation for Web configuration

The IP Phone Web Configuration Menu can be accessed by the following URI: http://Phone-IP-Address. The IP address can be set to either WAN IP address or LAN IP address, default factory setting of WAN configuration is DHCP Client model, default LAN IP address is "192.168.10.1".

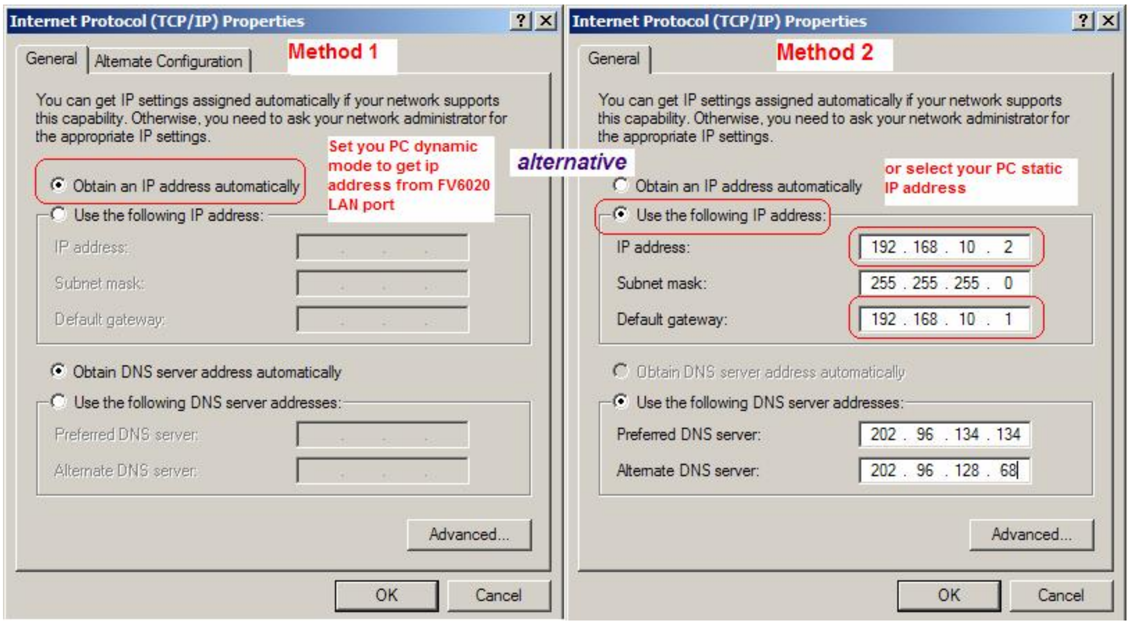
If connect PC with IP Phone LAN port and config to obtain IP address automatically, you could check the default gateway IP namely LAN IP address of IP Phone. The procedure as below

a) Access to "Property of local area connection" dialog box



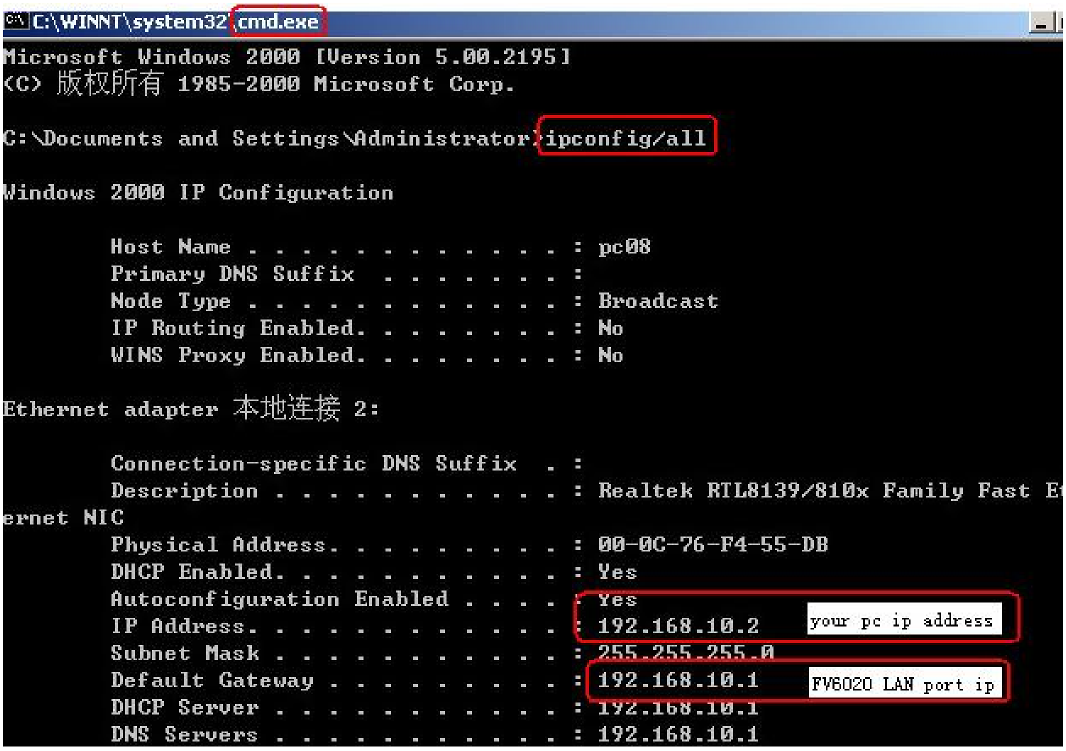
b) Select "Internet Protocol (TCP/IP)",click "Property" button

c) Setting refers to below dialog box, and then click "OK" button, PC will obtain IP address automatically.（or set your pc static ip 192.168.10.2）



Method 1 Method2

d) Input "cmd" command on the RUN submenu under PC START, key in "ipconfig/all on the command lines dialog box to find the default gateway IP address, which is LAN IP address of IP Phone.



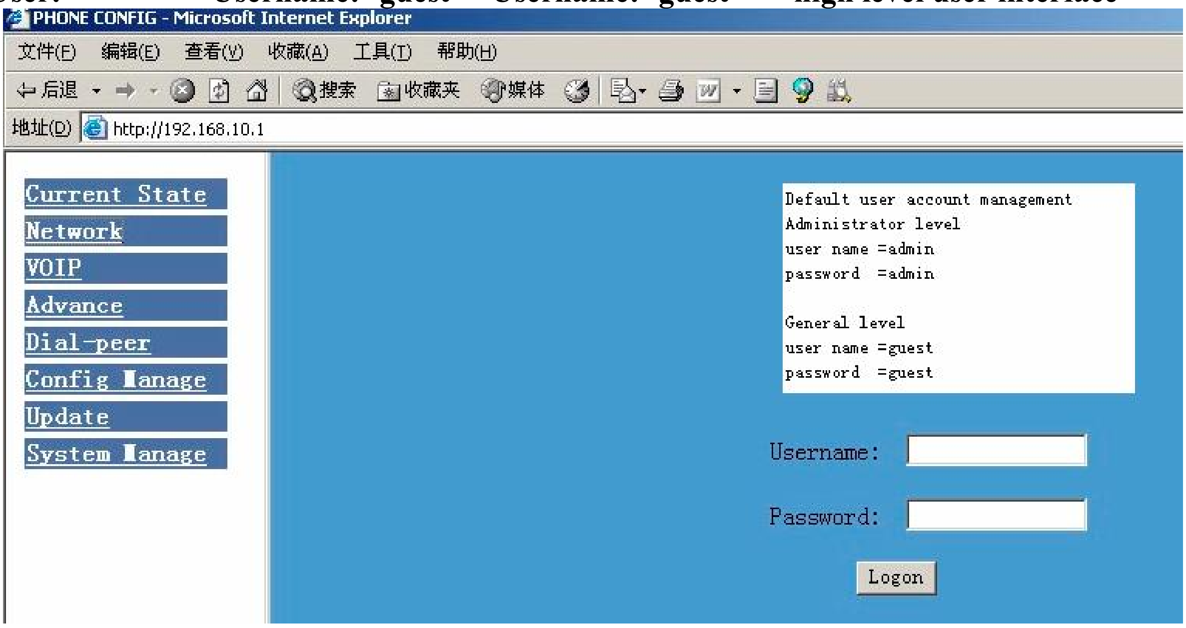
6.3 User verification

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 low level user interface

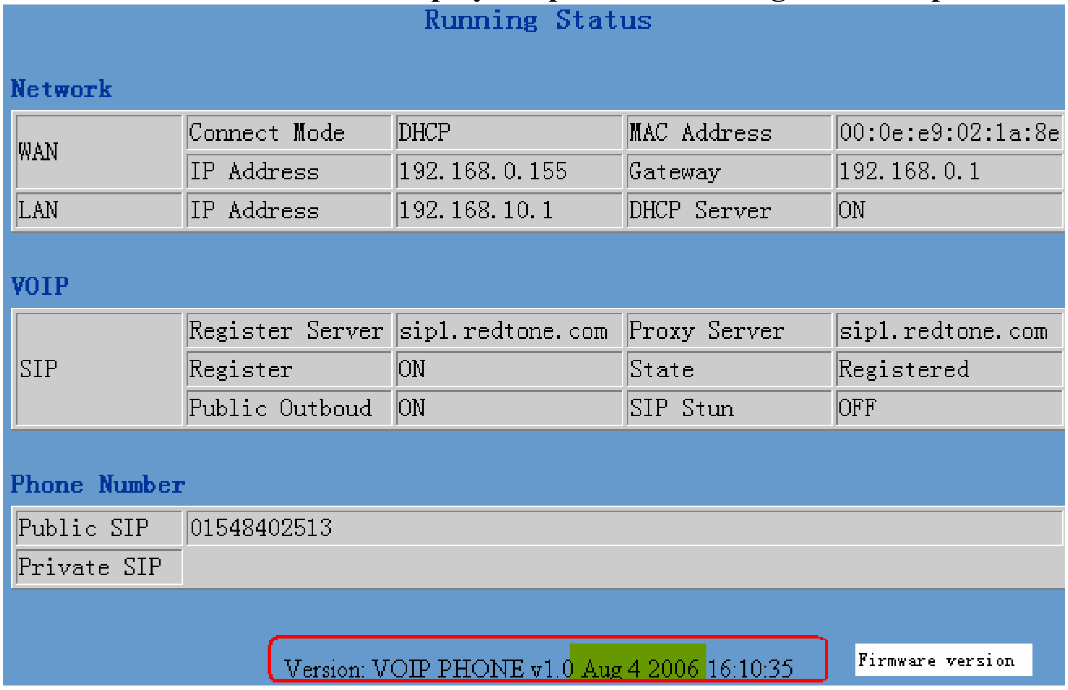
User: Username: guest Username: guest high level user interface



6.4 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**



Running Status

Network

WAN	Connect Mode	DHCP	MAC Address	00:0e:e9:02:1a:8e
	IP Address	192.168.0.155	Gateway	192.168.0.1
LAN	IP Address	192.168.10.1	DHCP Server	ON

VOIP

SIP	Register Server	sip1.redtone.com	Proxy Server	sip1.redtone.com
	Register	ON	State	Registered
	Public Outboud	ON	SIP Stun	OFF

Phone Number

Public SIP	01548402513
Private SIP	

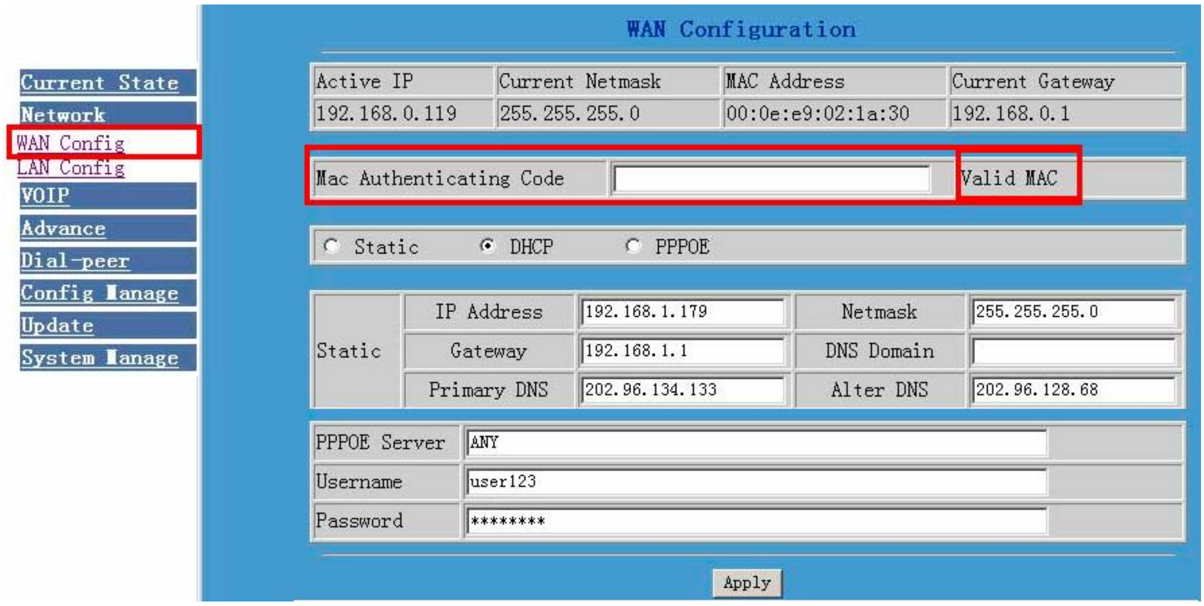
- **The version number and date of issue have been shown at the end of this page**

6.5 Network configuration

Network configuration includes WAN Config and LAN Config.

6.5.1 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 Favil or your provider)

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

Three models (Static /DHCP/PPPoE) are paratactic. Users can set the right model base on actual requirements.

- **Connect network to internet thru Static mode**

Default network config is DHCP model; Users need to set below parameters

Static	IP Address	192.168.10.71	Netmask	255.255.255.0
	Gateway	192.168.1.1	DNS Domain	voip.com
	Primary DNS	192.1.1.1	Alter DNS	192.1.1.1

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.

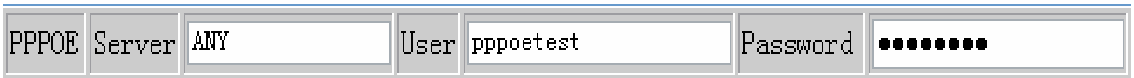
- **Connect network to internet thru DHCP mode**

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

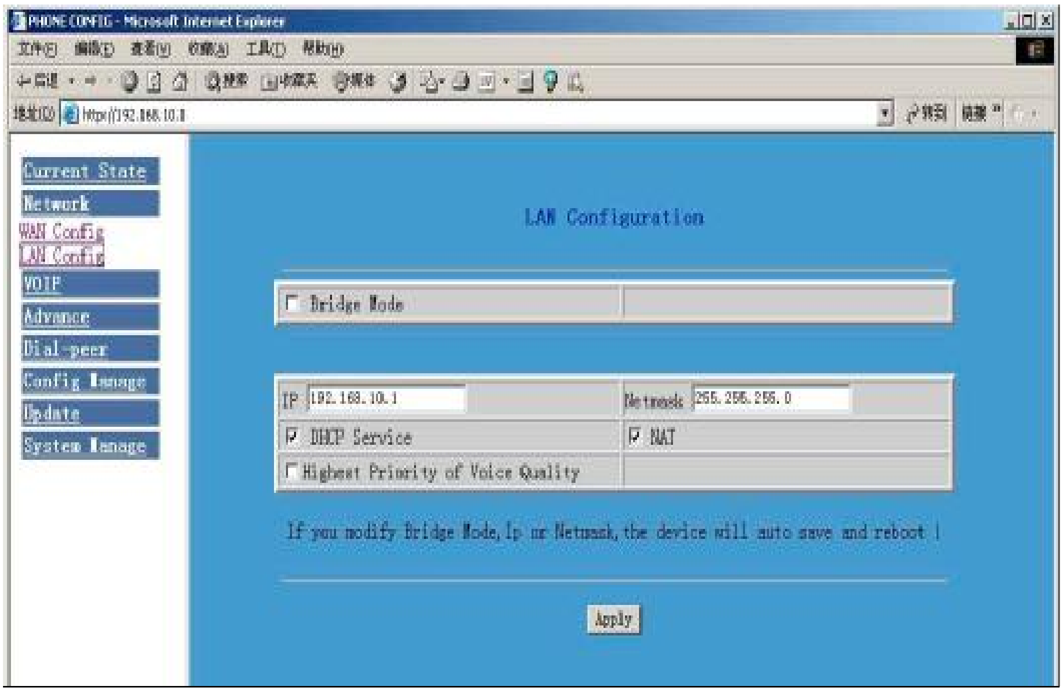


Server	**If ISP no special requirements, remains default setting**
User	Provided by ADSL ISP
Password	Provided by ADSL ISP

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.

6.5.2 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	192.168.0.1	Netmask	255.255.255.0

IP	LAN IP address
Netmask	Network Mask

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

☑ DHCP Service	☑ NAT

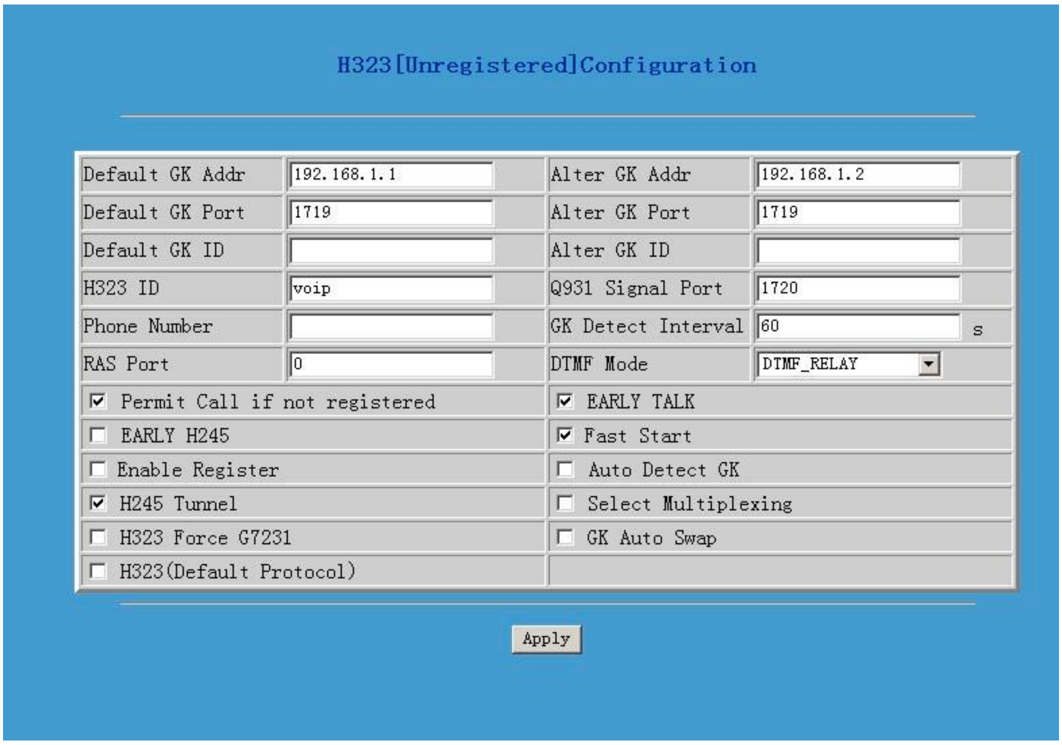
Start Bridge Mode or not（transparent mode）: Once start Bridge Mode, some parts of LAN config will be disabled, and the phone will no longer set IP address for LAN physical port， LAN and WAN will join in the same network;

6.6 VOIP Configuration

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

6.6.1 H.323 Configuration

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



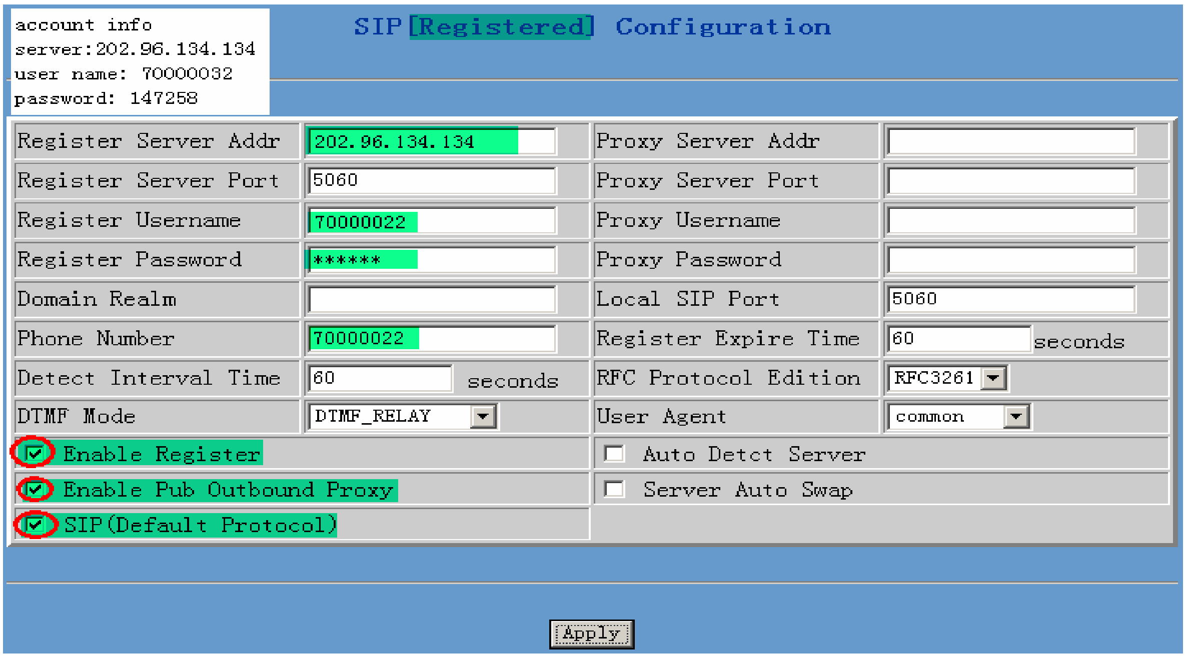
Definition of each parameter described as below

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

6.6.2 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 FV6020 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 (inband 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, FV6020 will periodically detect whether the public SIP server is available, if the server is unavailable, the FV6020 will switch to the back-up SIP sever, and continue detecting the public sip server. FV6020 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	10.1.1.139	Proxy Server Addr	192.1.1.139
Register Server Port	5060	Proxy Server Port	5060

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

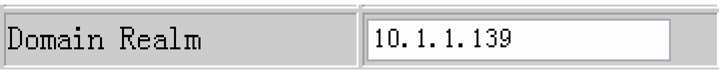
Register Username	client	Proxy Username	client
Register Password	••••	Proxy Password	••••

- **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)

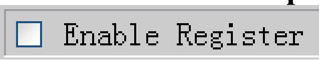
Phone Number	62281493

- **Config the domain realm (Usually it is same with registered server and proxy server**

IP address, Let it be blank)



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



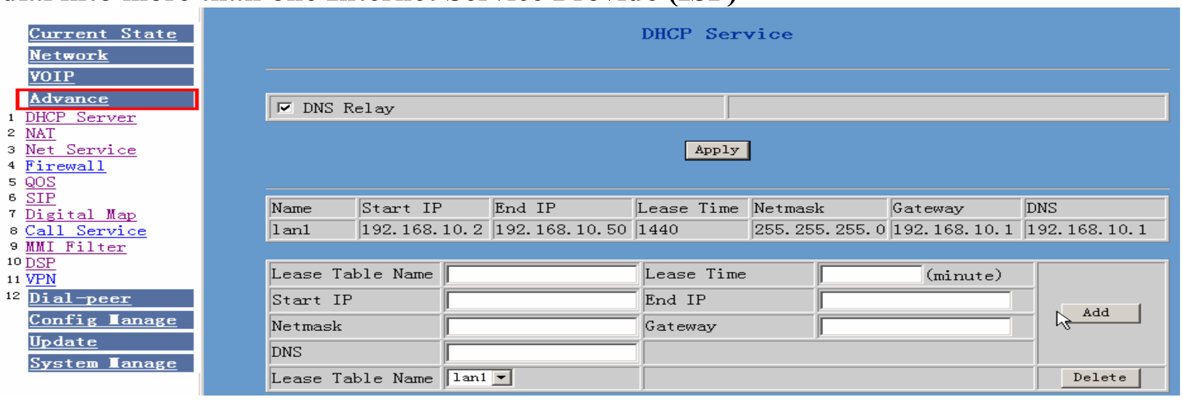
&


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

6.7 Advance

6.7.1 DHCP server configuration

when FV6020 work as a router, this config is for FV6020 LAN port network device DNS Relay : DNS relay acts as a forwarder between the DNS Clients and the DNS Servers,DNS relay is disigned 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.

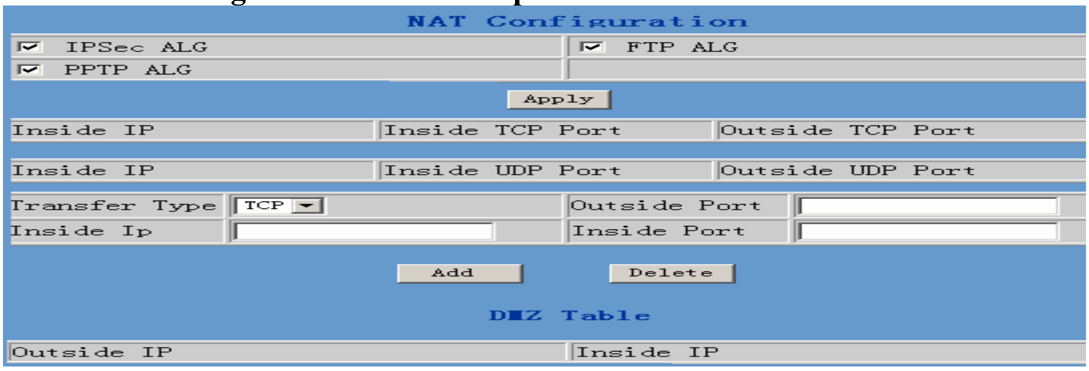
User may use below setting to add a new lease table.
Lease Table Name: Lease table name.
Lease Time: DHCP server lease time.
Start IP: Start IP of lease table.
End IP: End IP of lease table. Network device connecting to the FV6020 LAN port can dynamic obtain the IP in the range between start IP and end 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

6.7.2 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 theoutside network.



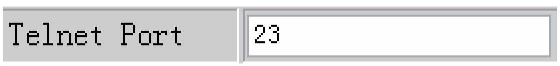
Advance NAT setting. Maximum 10 items for TCP and UDP port mapping.
IPSec ALG: Enable/Disable IPSec ALG;
FTP ALG: Enable/Disable FTP ALG;
PPTP ALG: Enable/Disable PPTP ALG;
Transfer Type: Transfer type using port mapping.
Inside IP: LAN device IP for port mapping.
Inside Port: LAN device port for port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

6.7.3 NAT service configuration

6.7.3.1

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.

6.7.3.2



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

6.7.3.3

RTP Initial Port	10000

Enable RTP initial port configuration. It is dynamic allocation；

6.7.3.4

RTP Port Quantity	200

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

Leased IP Address	Client hardware Address

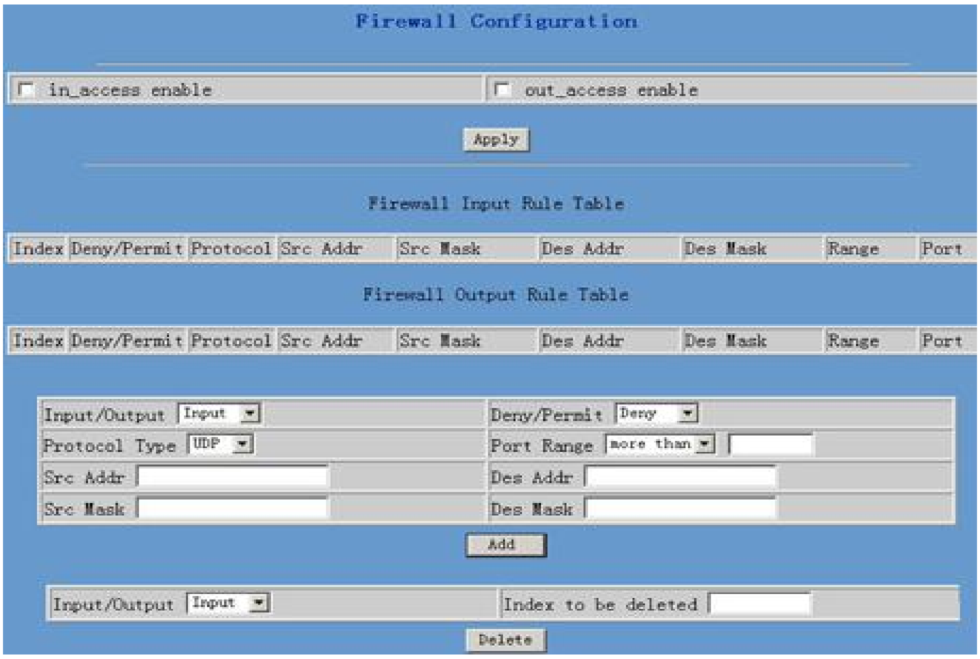
Leased IPMAC correspondence table of DHCP.The table will display all device getting ip address from FV6020 LAN port by DHCP .

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.

6.7.4 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.

FV6020 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 enable 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

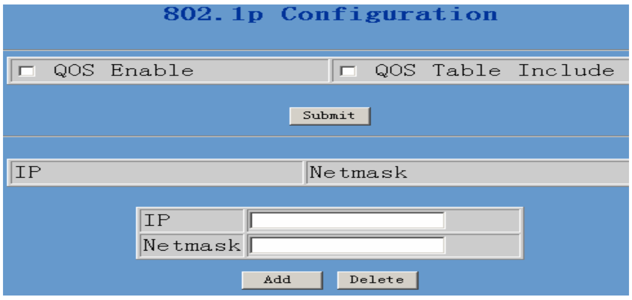
Src Addr: source address. can be single IP address or network address.

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

6.7.5 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. FV6020 will sorted the packets using the QoS and sends it to the destination.QoS provides service classes for accessing traffics in Internet.

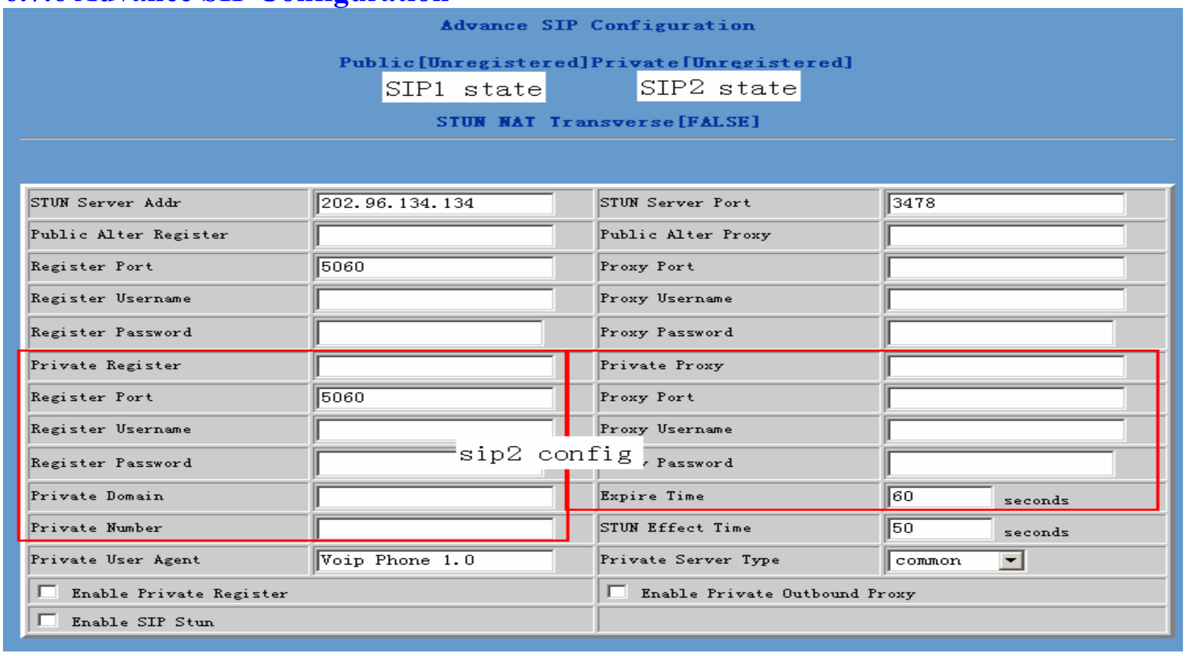
QoS Enable: Enable QoS service.

QoS Table : enable include QoS table, FV6020 will only provide QoS service to the network address included in the QoS table. Disable the option. FV6020 provides QoS service to the network address outside the QoS table.

Delete Enter the IP/MASK configure and select delete to delete corresponding item

Add: user can set the QoS Table using IP and Netmask. the IP can be network address (set netmask to 255.255.255.255)

6.7.6 Advance SIP Configuration



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.

Public backup server configuration

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	••••

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 server(SIP2) configuration.

Private Register	207.145.183.115	Private Proxy	
Register Port	5060	Proxy Port	
Register Username	2008	Proxy Username	
Register Password	••••	Proxy Password	
Private Domain		Expire Time	60 seconds
Private Number	2008	STUN Effect Time	50 seconds
Private User Agent	Voip Phone 1.0	Private Server Type	common
☑ Enable Private Register		☑ Enable Private Outbound Proxy	

Specific configuration parameter has the same meaning with public server.

☐ Enable Private Server Register

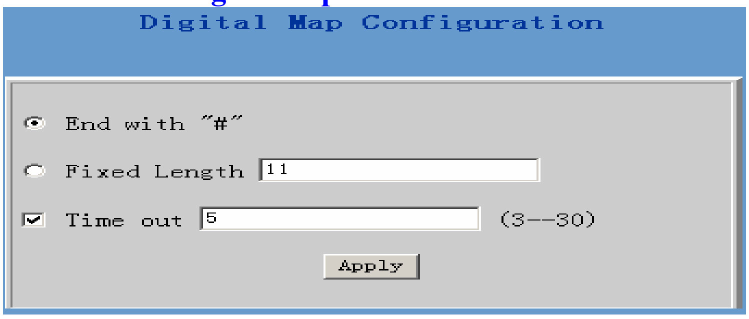
Configure permit/deny private server register;

Configure enable/disable private outbound proxy; if you have no stun server .advise to enable the option.

Note: about how to use SIP ,Please refer to the Dial peer chapter.

6.7.7 Digital map configuration

6.7.7.1 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

6.7.7.2 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.

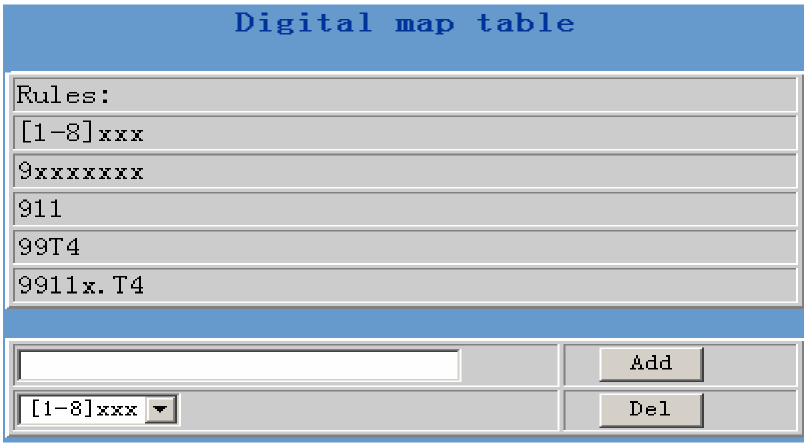
Tn represents the last digit timeout. here [n] represents the time from 0~9 second, it is necessary. Tn must be the last two digit in the entry. If Tn is not included in the entry, we use T0 as default, it means system will sent the number immediately if the number matches the entry.

. (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 in 4 second .



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

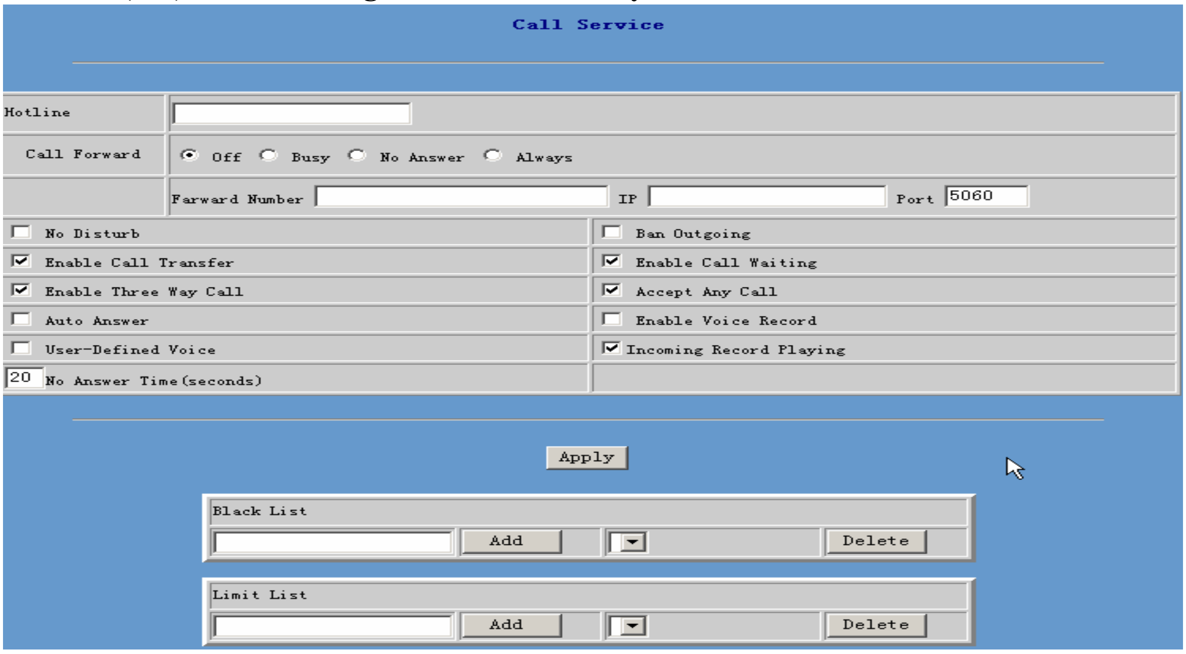
Number	Destination	Port	Alias	Suffix	Del length
2887	192.168.0.155	5060	no alias	no suffix	0
98765432	192.168.0.155	5060	no alias	no suffix	0
911	192.168.0.155	5060	no alias	no suffix	0
99	192.168.0.155	5060	no alias	no suffix	0
9911234	192.168.0.155	5060	no alias	no suffix	0

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

6.7.8 Call Service Configuration

Value added service configuration

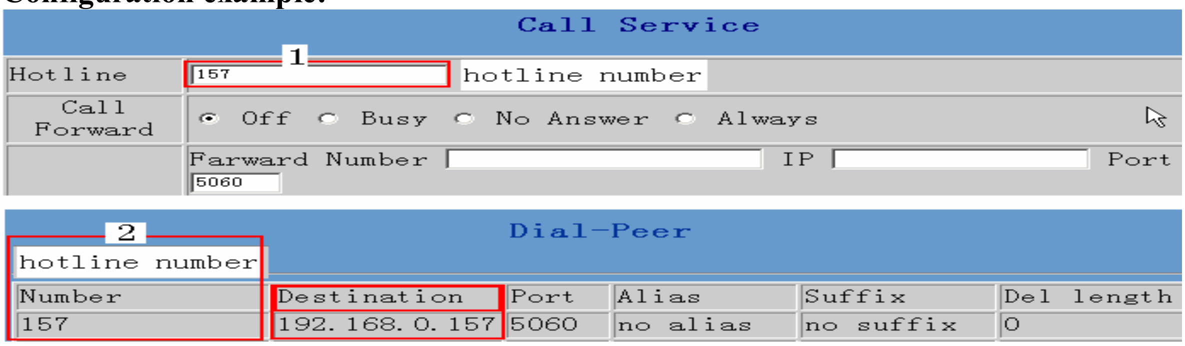
On this page, user can set value added services such as hot-line，call forwarding，call transfer (CT)，call-waiting service，three way call，blacklist，out-limit list and so on.



6.7.8.1 Hotline

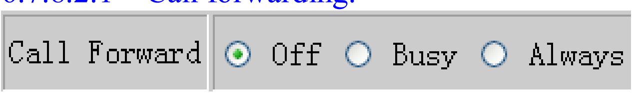
Configure hot-line number of the port. With this number of the port，this hot-line number will be dialed automatically as soon as off-hook and user can's dial any other number. If you do not use hotline , please let it be blank.

Configuration example:



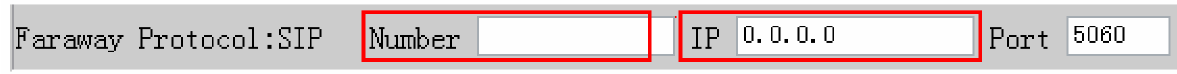
6.7. 8.2 Call feature

6.7.8.2.1 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.

Auto Answer

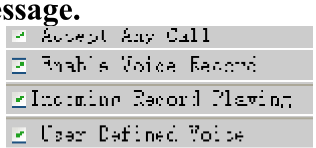
If it is enabled , when there is a incoming call .the phone will automatically press SPEAKER button.

Enable Voice Record

If it is enabled. The phone can record the incoming call leaving voice message. It support only 3 leaving message . it can not record leaving message when it had reach the maximum . after restarting the phone ,the message will lose . or you can delete the record message by manual(on phone operation).

If it is enabed, after playing the prompt message ,the phone will automatically press SPEAKER button and start to record leaving message.

Generally, these four option are selected toghter



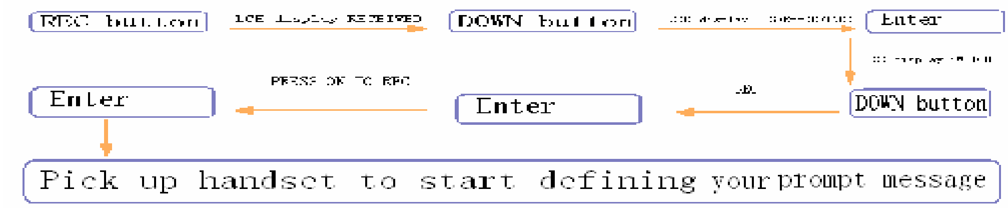
Incoming Record Playing

If it is enabled , the caller will hear FV6020 automatically play prompt message (welcome message.)

User-Defined Voice

User can difine only one prompt message (welcome message) that can be played when someone call the phone. The second defined voice will cover the first one .

Process: on phone operation



10 No Answer Time(seconds)

The unit is second. no answer call forward time setting.

6.7.8.2.2 Call Waiting configuration

☑ Enable Call Waiting

Configure enable/disable call waiting service; After it is enabled, user hold calls of the other party by <HOLD >button, by pressing <HOLD >button again, the call can go back to the previous call. If you want to use three way conference , this option must be enabled.

6.7.8.2.3 Call transfer configuration

☐ Enable Call Transfer

It is for enabling or disabling phone <TRANSFER> button transferring function.If it is enabled, when user A are talking with user B , A press <TRANSFER> button on the phone then dial the third party number (C user) directly. the phone will transfer the calls to C. The result is C phone will ring (user B LCD display HOLD ,A will hang up) . User C pick up phone then user B will talk with C .

If it is disable, On phone operating transfer call will fail.

6.7.8.2.4 Three way conference call

☐ Enable Three Way Call

Configure enable/disable three way call; When user A are talking with user B as the call origination, user A click HOLD button to hold user B line and then dial the third party user C. User A click HOLD button again to recover the talk with user B. At this time user A press * key to make C into the three way conference .

Operating process : HOLD button + third party number+ HOLD button+ *

Example:

A are talking with B .

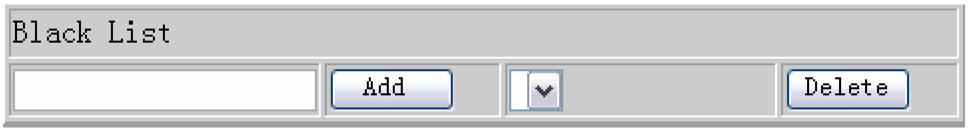
A click HOLD button (holding B line) .then dial C number .

A will talk with C .

A click HOLD button again (holding C line), A dial *

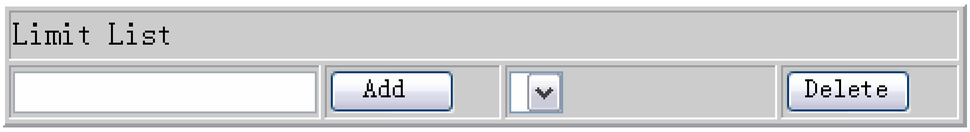
A 、B、 C make the three party conference sucessfully now

6.7.8.2.5 Black List



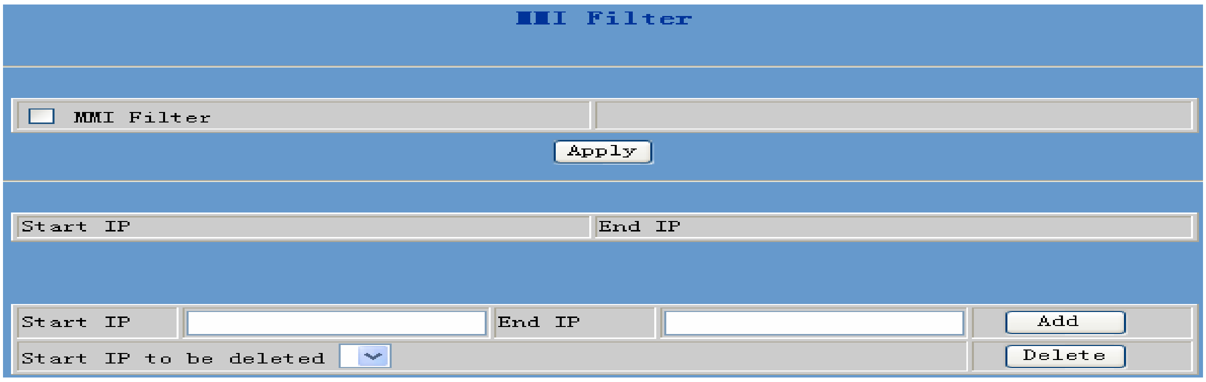
Incoming call in these phone numbers will be refused. It is for precluding incoming communication like Call ID. If user don't want to answer a certain phone number, please add this phone number to the list, and then this number will be unable to get through the phone.

6.7.8.2.6 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.

6.7.9 MMI Filter

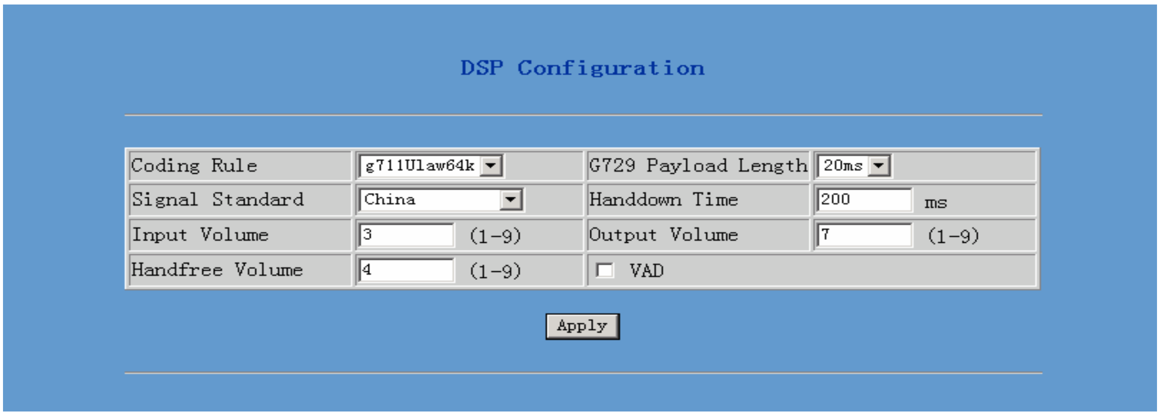


MMI filter is used to make access limit toFV6020.

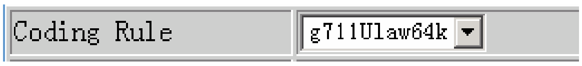
When MMI filter is enable. Only IP address within the start IP and end IP can access FV6020.

6.7.10 DSP configuration

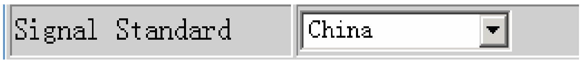
On this page, user can set speech coding，IO volume control, cue tone standard, caller ID standard and so on.



Configuration Explanation：



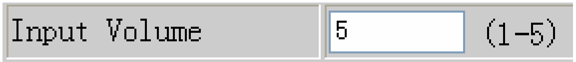
Configure Coding Rule according to network bandwidth; support G.711a/u G.729 ,G7.23



Congfigure Signal Standard according to country's phone voice;

Normally, G729 Payload Length don't need be changed into 10 ms;

Handset out volume.



Handset in 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.

VAD: **Enable/disable Voice Activity Detection**

6.8 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 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 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 configurating the prefix matching, letter "T" should be added behind the prefix number to be distinguished from the exactitude matching.

Call Mode sip

Configure the calling mode: H323 and 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.

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·

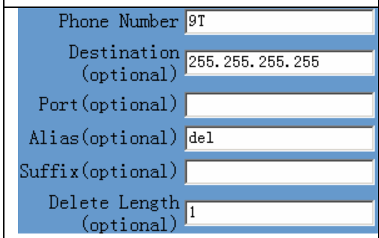
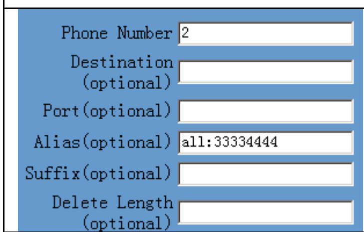
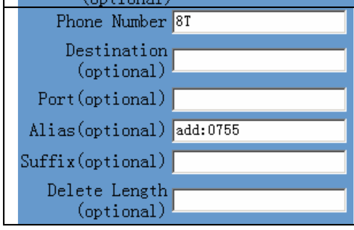
rep: xxx n bit in the front part of the number will be replaced. n is decided by the replacing length.

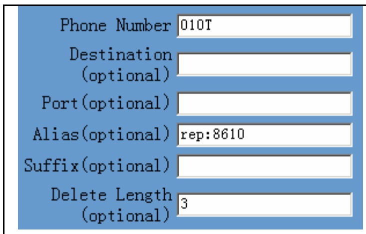
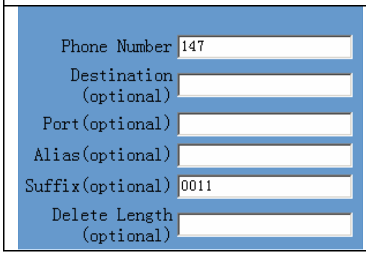
Suffix(optional)

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

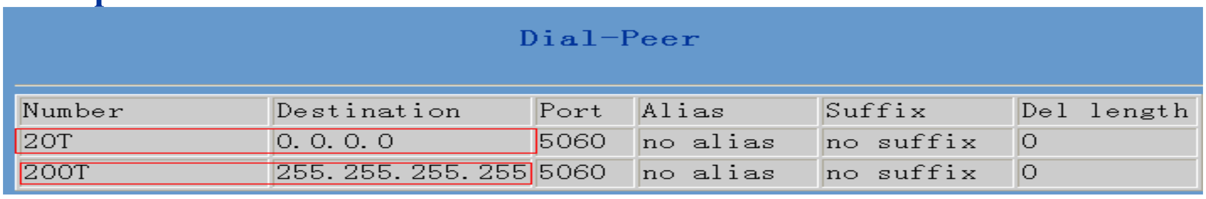
Example 1

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	Explaination	Example
	That means Any digits number starting with 9 pass throught 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
	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
	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

 	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. Relace the number that user dialed before sending to SIP1 server. Here alias is rep:(not rep)	User dial 010 6228 SIP1 server receive8610 6228
 	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

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

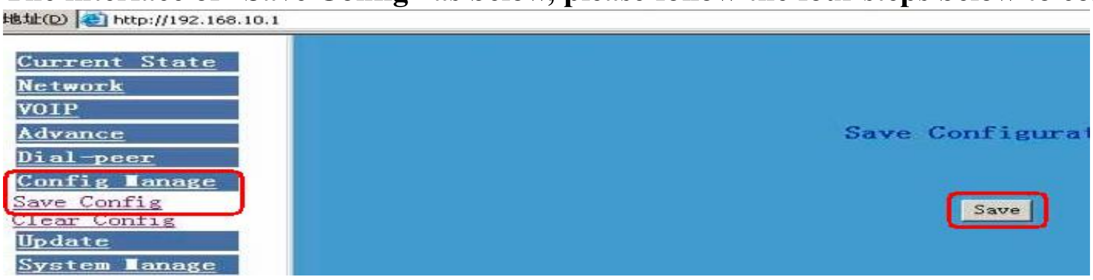
6.9 Config Manage (Save and Clear configuration)

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.

6.9.1 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.



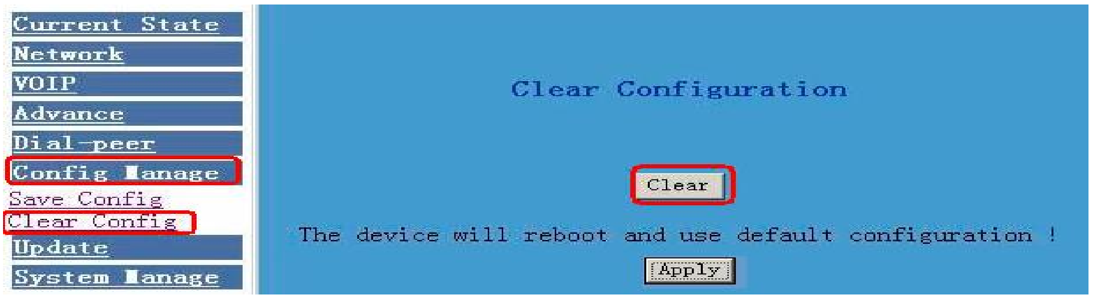
Enter "Config Manage" Menu →"Save Config" Submenu→ Click "Save" Button→ Return to "Current State" Web page

6.9.2 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:

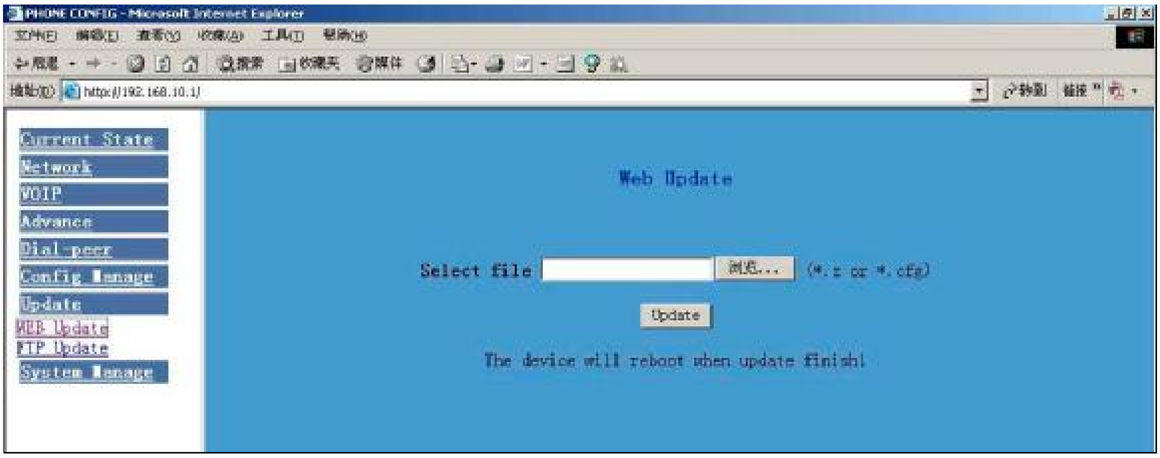
Enter "Config Manage" Menu →"Clear Config" →Click "Clear" Button →show "Submit Success" info on screen →Click "Return" button



6.10 Firmware Upgrade

6.10.1 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



STEP:

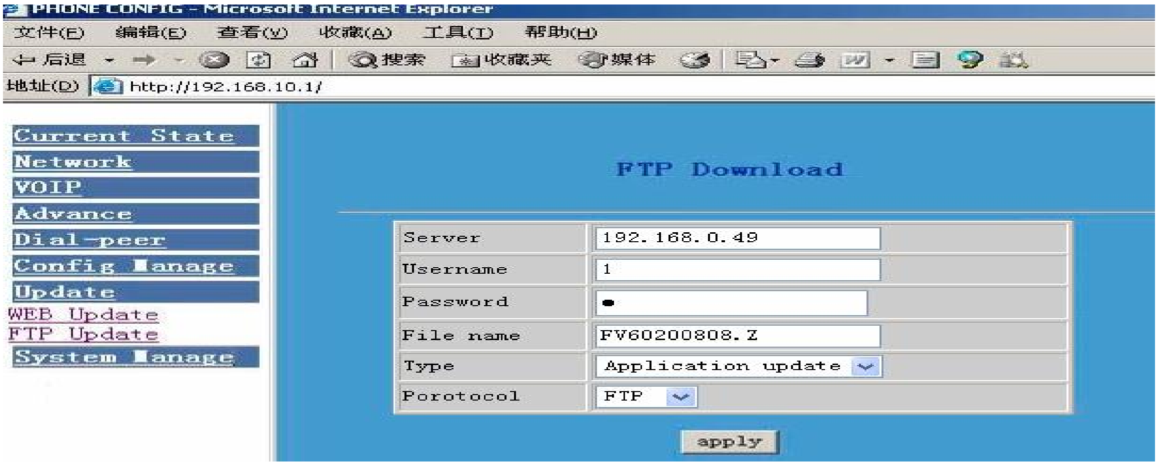
Enter Update menu →WEB Update submenu→ click "browse" button→ download upgrade document from hard disk (firmware or config file provided by manufacturer) → click "Update" button →reboot IP phone to go into effect

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.

6.10.2 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.

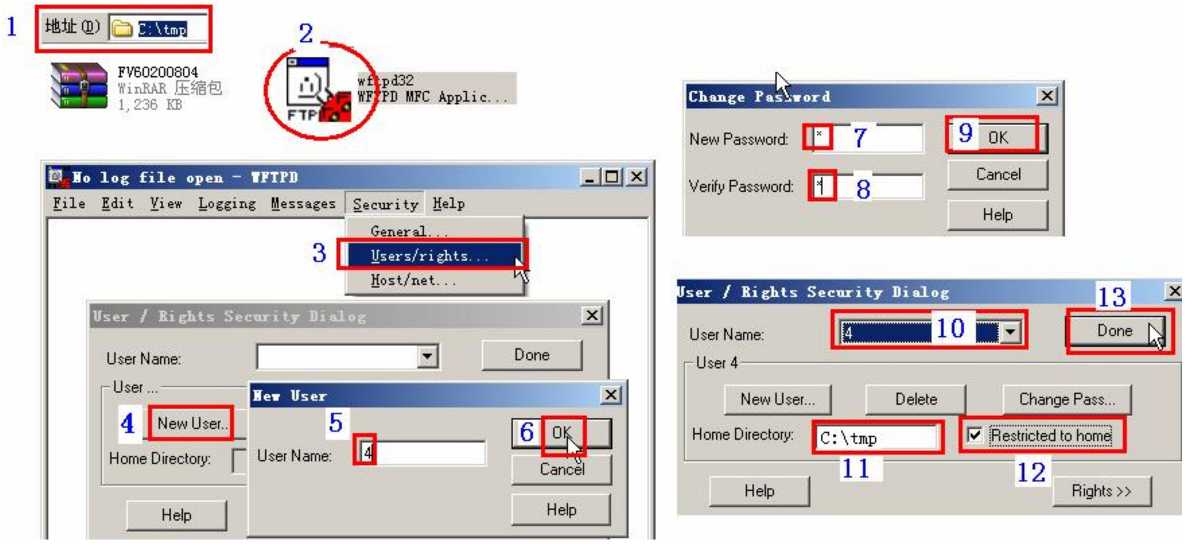


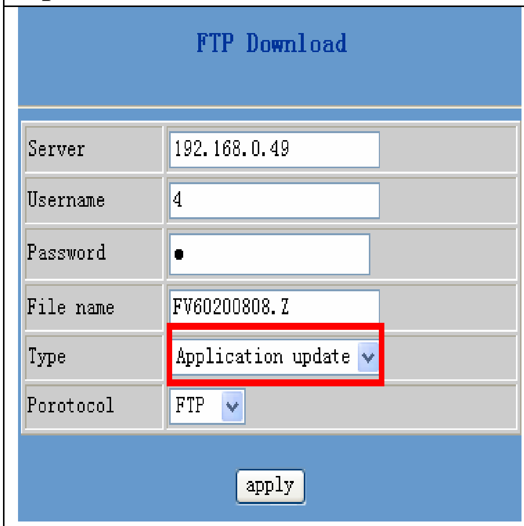
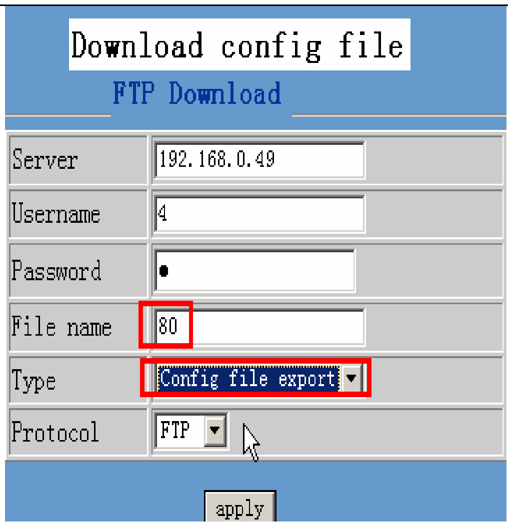
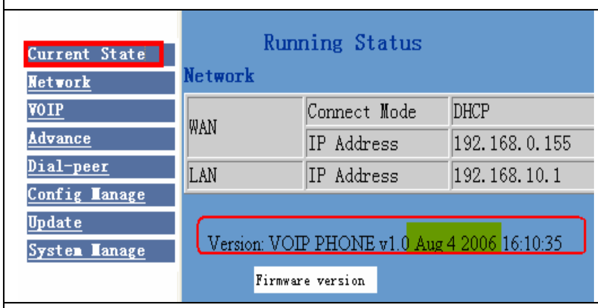
Definition of each parameter described as below

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 FV6020 Firmware into a new Folder (example c:/tmp)
<2> Run wftd32.exe. Set a user name and password for FV6020 ftp updateing
The process is like the below picture showing from step 1 to setp 13.

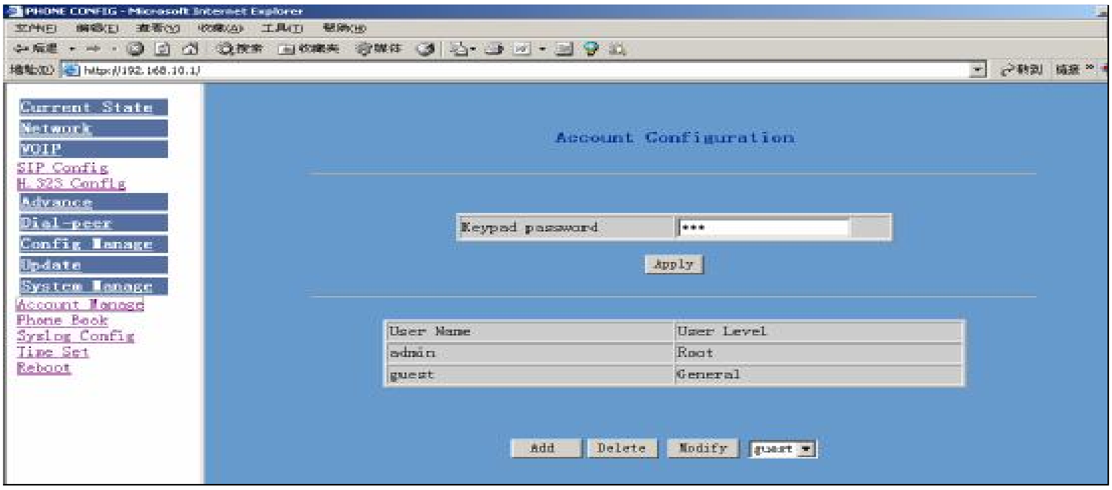


Update the firmware	Download config file to you pc.
	
	
After it update successfully. You will find the new version in Current State	After you click apply , you can find the file that it had download to your pc <c:\temp>

6.11 System Manage

6.11.1 Account Manage (maximum 5 account)

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



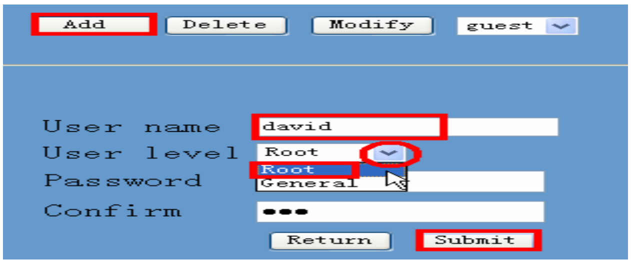
Definition of each parameter described as below

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

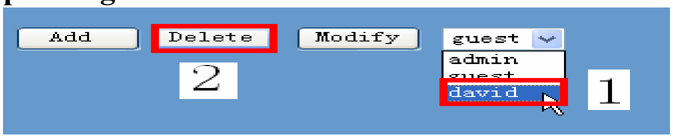
- **Add one new account**

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



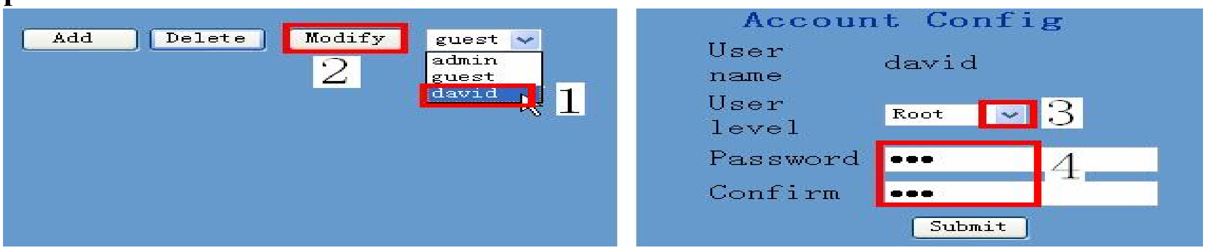
- **Delete increased account**

  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.

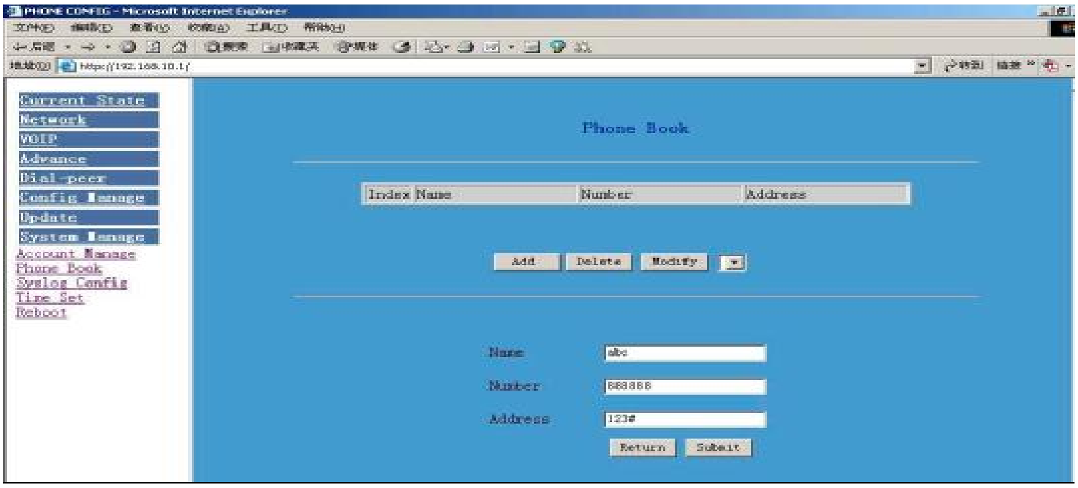
Enter with manager level when making modification，create a administrator account and a browse account (you'd better not set the name as admin, administrator, guest, etc.)，set password and then save configuration，entering with new manager account, delete default manager and browse account and save configuration，security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

6.11.2 Phone book configuration

On this page, user can save and configure telephone book.(maximum 100 record)

- **Add phone book record**

  Click "Add" button →Input Name name, number and address→ click "Submit" button→ show "Submit Success" on screen

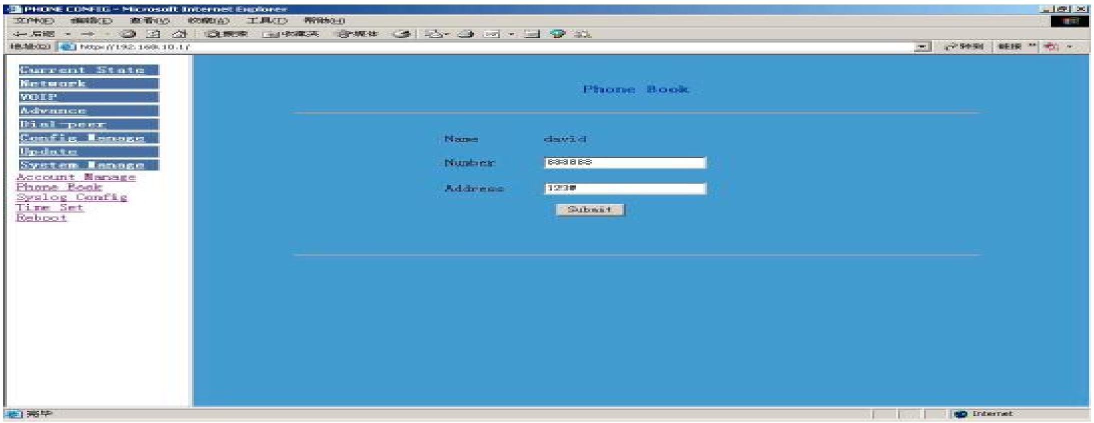


- **Delete phone book record**

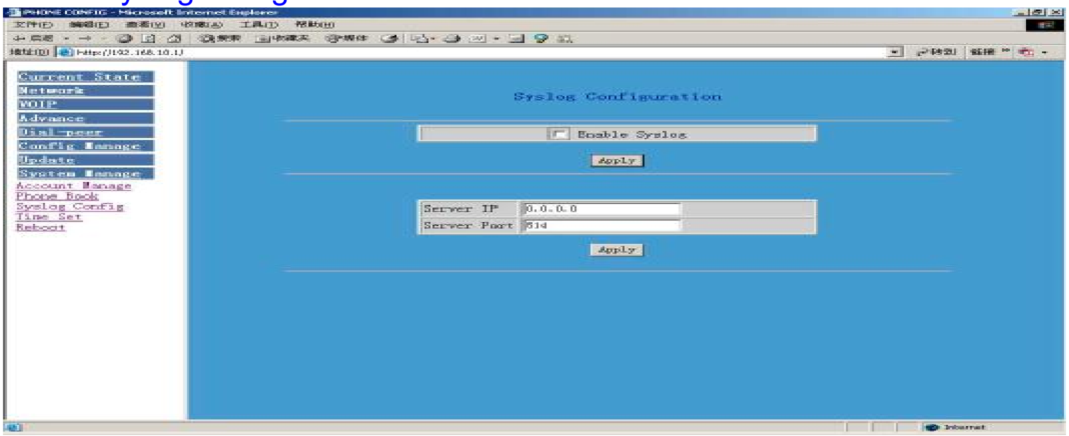
  Choosing the record need to del from dropdown menu →Del account by pressing "Delete button" → →show "Submit Success" on screen

- **Modify phone book record**

  Choosing the record need to modify from dropdown menu→ pressing the "Modify" button→ key in the new phone number and Address → click "Submit" button → Return to "Phone Book" interface



6.11.3 Syslog configuration

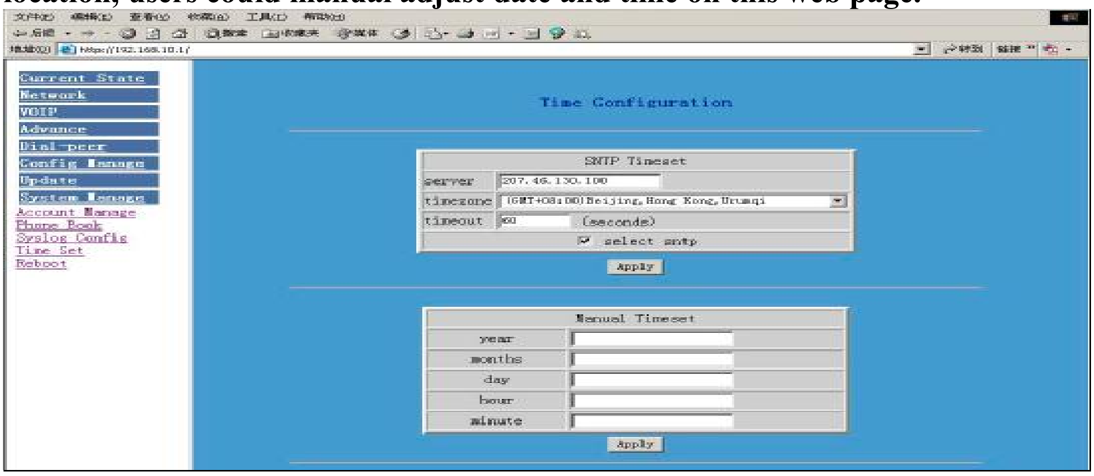


Definition of each parameter described as below

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

6.11.4 Time Set

Setting time zone and SNTP (Simple Network Time Protocol) server according to users location, users could manual adjust date and time on this web page.

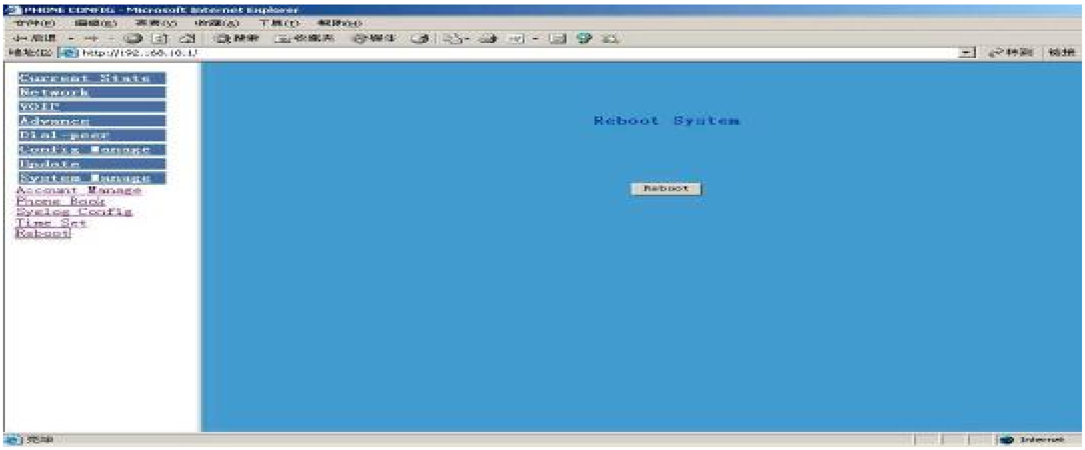


Definition of each parameter described as below

Server	**SNTP server IP address**
Time zone	Set time zone according to users location, Beijing time zone is GMT:+8:00
Timeout	use for socket calls which will block while doing the SNTP query, default value is 60 seconds
Manual Time set	Manual set date and time

6.11.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



7 Default Factory Setting

- Gain IP address thru DHCP mode, WAN Port static IP is 192.168.1.179, LAN port IP is 192.168.10.1.Default is to start the DHCP service and NAT function.
- Default communication protocol is to use SIP, SIP port is 5060
- Default HTTP port is 80, Telnet port is 23
- Default number end is "#" button
- Default user account is admin and guest
- Default phone time is to use SNTP protocol to get GMT

CHAPTER 8

8.0 Telnet configuration

8.1 Config Procedure

- **Input command "cmd" on Run submenu under PC START menu, and then key in "telnet phone-IP-address enables users to config IP phone thru telnet.**
- **Input username and password, both default username and password of Administrator account are "admin"**
- **Config IP Phone through command lines**

> *Note:*
> 1. *We suggest users to config IP Phone thru web browser instead of keypad or telnet.*
> 2. *After any change of configuration, please remember to make "write" command to save changes and then input "reload" command to reboot IP phone to take effect.*

8.2 Telnet basic Introduction

8.2.1 Basic structure

User may use telnet command to access and manage gateway.
FV6020 adopts tree structure for telnet. Every node contains its sub-nodes or local command. User can type "help" or "?" whenever to see sub-nodes and all local command under current node.
Besides local command, there are some global commands can be used in each node.

8.2.2 Basic command

Logout: exit telnet mode.
Write: save current settings.
Type sub-nodes name in current node to switch to sub-node.
Type "!" or "exit" in current node to return to parent-node.
Type "help" or "?" can see all sub-nodes and all local command under current node, every help item has comments such as <command> or <node> to distinguish sub-nodes and local command. Type "help" or "?" in command can see all parameters using in this command.
When typing node name or command, user no need to key the full name, use TAB button will make it more efficient.
There are two types in command parameters: optional and required. "required" parameter use "-" as prefix and "optional" use "_" as prefix. User may type "-" or "_" then press TAB button for complementarily.

8.3 Global Command

Global command is available under all nodes, FV6020 support following commands.

Command Name	Command function	Example
chinese	Set the language of help prompt info to Chinese	#chinese
clear	Clear screen	#clear
english	Set the language of help prompt info to English	#english
exit	Go back to upper level of node	#exit
help	1.Display help prompt info 2.Display all subnode under current node and local command	1． #help ping 2． #help
history	Display the history info for inputting command	#history
logout	Exit telnet config interface	#logout
ping	One test program using to check network or program availability	#ping www.google.com.
tree	Print out the tree structure of current node	#tree
who	Display current users login to PC	#who
Write	Save configuration to flash	#write
Reload	Reboot IP phone	#reload

Common Network command

Command Name	Command function	Example
ping	The same with above stated	#ping www.google.com
tracert	Print out the network path	#tracert www.google.com
show basic	Print current configuration status table	#show basic
show ip route	Print phone Router table	#show ip route
show ip arp	Print phone arp table	#show ip arp
show ip netstat	Run Netstat program	#show ip netstat
telnet	telnet another user host	#telnet 192.168.1.2
Set default	Clean phone configuration modification and restore default except network configuration	#setdefault
Set all default	Clean all configuration and restore default manufacturing setting	#setdefault all

8.4 Net configuration

8.4.1 LAN interface settings

Path: <config-interface-fastethernet-lan>#

Function	Command
[disable]enable bridge mode	[no]bridgemode
[disable]enable DHCP service	[no]dhcp-server
[disable]enable NAT	[no]nat
Show current DHCP rules	dhcpshow
Show LAN port IP address	ipshow
Show NAT info	natshow
Change LAN port IP address	ip –addr x.x.x.x –mask x.x.x.x

Example:

#config interfact fastethernet lan

<config-interface-fastethernet-lan>#ip –addr 192.168.1.10 –mask 255.255.255.0

8.4.2 WAN interface settings

path: <config-interface-fastethernet-wan>#

Function	Command
[disable]enable dhcp client	[no]dhcp
[disable]enable pppoe	[no]pppoe
[disable]enable QOS	[no]qos
Set default gateway IP	gateway x.x.x.x
Clear default gateway IP	no gateway
Set WAN port IP address	ip –addr x.x.x.x -mask x.x.x.x
Show WAN port settings	show

Example:

 # config interface fastethernet wan

<config-interface-fastethernet-wan>#ip –addr 202.112.241.100 –mask 255.255.255.0

You need to reconnect if the WAN port has been changed.

8.5 Netservice

path: <config-netservice>#

Function	Command
Set DNS address	dns -ip x.x.x.x _domain xxx
Set alternate DNS address	alterdns -ip x.x.x.x _domain xxx
Set hostname	hostname xxx
Set http access port	http-port xxx
Show http access setting	http-port
Set telnet access port	telnet-port xxx
Show telnet access port	telnet-port
Set RTP initial port and quantity	media-port –startport xxx –number xxxx
Add route rule	route –gateway x.x.x.x –addr x.x.x.x –mask x.x.x.x
Delete route rule	no route –gateway x.x.x.x –addr x.x.x.x –mask x.x.x.x
Show route info	Route
Show netservice info	show

Example:

#config netservice

<config-netservice>#dns –ip 202.112.10.36 _domain voip.com
<config-netservice>#media-port –startport 10000 –number 200
<config-netservice>#route –gateway 202.112.10.1 –addr 202.112.210.1 –mask 255.255.255.0

8.6 Port settings

path: <config-port>#

Function	Command
set callerid mode	callerid xxx
disable callerid	no callerid
Disable call forward	no callforward
[disable]enable call transfer	[no]calltransfer
[disable]enable call waiting	[no]callwaiting
Set DTMF gain	dtmfvolume xxx
[disable]enable3-way conference	[no]threetalk
Show port settings	show

8.7 SIP settings

path: <config-sip>#

Function	Command
[disable]enable registration	[no] register
[disable]enable auto detect server	[no] detect-server
Set sip domain	default-domain xxx
Set DTMF mode	dtmf-mode xxx
Set auto detect interval time	interval-time xxx
[disable]enable auto swap server	[no]swap-server
Set local SIP signal port	signalport xxx
Set proxy server	server proxy -ip x.x.x.x _port xxx _user xxx _password xxx
Set register server info	server register -ip x.x.x.x _port xxx –user xxx _password xxx
Set alter proxy info	alter-server proxy –ip x.x.x.x _port xxx _user xxx _password xxx
Set alter server info	alter-server register –ip x.x.x.x _port xxx _user xxx _password xxx
Show current sip info	show

Example:
#config sip server
<config-sip-server># server proxy ip 210.25.23.22 _port 5060 _user aaa _password 123456

8.8 User management

path: <config-user>#

Function	Command
Change user right.	access –user xxx –access xxx
Change user password	password –user xxx
Add new user	entry –user xxx –access 5 (or 12)
Delete user entry	no entry –user xxx
Show current sip info	show

Example:
#config user
<config-user>#entry –user abc –access 7
Example:<config-user>#access –user aaa –access 7

Note:
The command : **–user xxx –access digit**
Here if the digit is less 10 , then the user level is guest
If the digit is more that 10 ,then the user level is administrator

8.9 Debug (Level 0~7)

path: <debug>#

Function	Command
show debug setting	show
[disable]enable debug all modules	[no] all xxx
[disable]enable debug app module	[no] app xxx
[disable]enable debug cdr module	[no] cdr xxx
[disable]enable debug sip module	[no] sip xxx
[disable]enable debug h323 module	[no] h323 xxx
[disable]enable debug tel module	[no] tel xxx
[disable]enable debug dsp module	[no] dsp xxx

8.10 show system running info

path: <show>#

Function	Command
show: accesslist (firewall) settings	accesslist
show network status	basic
show current call info	call active
show CODEC capability	capability
show debug info	debugging
show LAN status and DHCP server info	dhcp-server
show digital-map info	dial-rule
show LAN info	interface fastethernet lan
show WAN info	interface fastethernet wan
show arp table info	ip arp
Show DNS gateway info	ip dns
Show netstate info	ip netstat
Show route info	ip route
Show icmp packets Stat.	ip icmp
Show igmp packets Stat.	ip igmp
Show ip packets Stat	ip ip
Show RTP packets Stat.	ip rtp
Show TCP packets Stat.	ip tcp
Show UDP packets Stat.	ip udp
show gateway memory	memory

show NAT information	nat
show caller-ID info	port callerID
show dsp info	port dsp
show hotline info	port hotline
show black list info	port in-limit
show outgoing limit info	port out-limit
show current phone number	port number
show current port status	port status
show PPPoE info	pppoe
show QoS table info	qos
show sip info	sip
show UDP tunnel info	udptunnel
show running time	uptime
show gateway version	version

8.11 Logout

Usage: #telnet –target -port

Login:xxx

Password:xxx #

#logout

8.12 tracert trace network path info

usage: #tracert –host

Example:#tracert www.google.com

8.13 update

usage:

 # update ftp –user xxx –password xxx –ip x.x.x.x –file xxx

 # update tftp –ip x.x.x.x –file xxx

Example:

 # update ftp –user abc –password 123 –ip 202.112.20.15 –file FV6020.z

8.14 upload configure file

usage:

 # upload ftp –user xxx –password xxx –ip x.x.x.x –file xxx

 # upload tftp –ip x.x.x.x –file xxx

8.15 download configure to flash

usage: #download tftp –ip x.x.x.x –file xxx

#download ftp –user xxx –password xxx –ip x.x.x.x –file xxx

Example:

#download ftp –user abc –password 123 –ip 202.112.20.15 –file FV6020.cfg

8.16 password

usage:

#password

Enter new password:xxx
Confirm new password:xxx

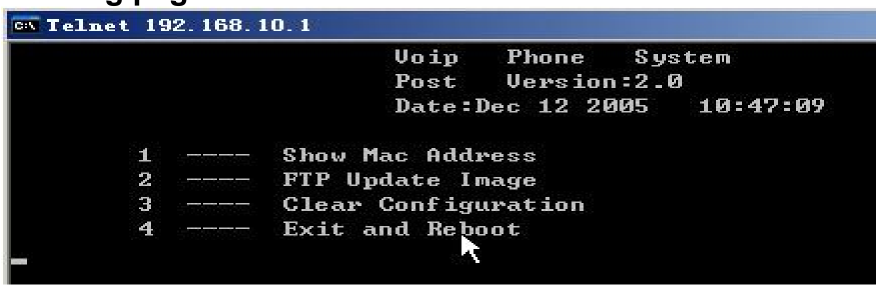
8.17 reload
usage: #reload
Reboot system

8.18 Network Diagnosis
There are some telnet commands for checking your network. Now Listing below for your information

Command	Function	Example
ping	Check if the destination is accessible	#ping www.google.com
tracert	Show network path info	#tracert www.google.com
show basic	Show network settings	#show basic
show ip route	Show route table	#show ip route
show ip arp	Show arp table	#show ip arp
show ip netstat	Netstat programe	#show ip netstat
telnet	Telnet to another device	#telnet 192.168.1.2

8.19 Restore to factory default
#setdefault **(clear gateway settings expect network part)**
#setdefault all **(clear all settings.)**

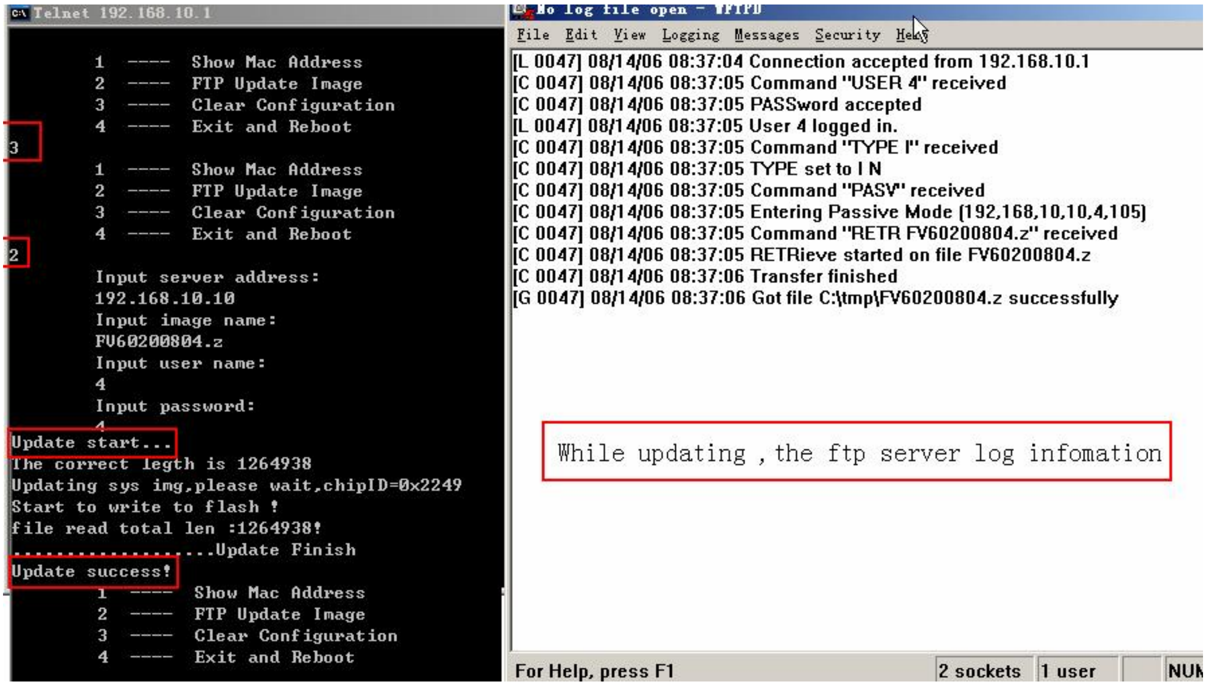
8.20 POST Mode(safe mode)
8.20.1 Access Post mode process
1 PC connect to FV6020 LAN port
2 power off
3 press # and power on till the LCD display POST MODE
4 telnet 192.168.10.1
5 config page is like below



8.20.2 Post mode clear
Input 3 and enter for clear configuration

8.20.3 post mode FTP update firmware

Input 2 for update firmware



FV6020 provide safe mode. When there is booting problem because of setting problem or firmware problem. User can restore the factory setting or upgrade to a new firmware to solve this problem.