GF502 IP PHONE
USER MANUAL

No.002-GF502-004
2008-011-24

CONTENTS

Preface

◎About this Manual

This guide provides the information you need to understand, install, configure, and manage the GF502 VOIP Phone on your network. It provides the required steps to get the GF502 VOIP Phone up and running on a SIP network. Not all features listed are available by default. Contact your network administrator or service provider to find out which features and services are available to you on your system.

◎Audience

Network engineers, system administrators and end users who need to understand how to install and use the IP phone on a SIP network should review this guide. Prior knowledge of IP telephony concepts is recommended.

◎Related Documentation

- ***Dial Plan Guide For VoIP Device.pdf***

The user guide of local dial plan for GF502 VoIP phone.

- ***Provisioning Guide for VoIP Device.pdf***

The user guide of provisioning for GF502 VoIP phone.

◎Document Conventions

This document uses the following conventions:

---

5 Means ***reader be careful.*** In this situation, you might do something that could result in equipment damage or loss of data.

---

1 Means ***reader take note***. Notes contain helpful suggestions or references to material not covered in the publication.

Chapter 1: An Overview of the GF502 VOIP Phone

◎Introduction

The GF502 IP Phone provides voice communication over an IP network using the SIP protocol. It functions much like a traditional analog telephone, allowing you to place and receive phone calls and to access features such as mute, hold, forward, transfer, and speed dial. In addition, because the phone is connected to network, it offers enhanced IP telephony features, including access to network information and customizable features. A GF502 VOIP Phone, like other network devices, must be configured and managed.

◎Main Features

- Support three sip accounts at the same time.
- Support multi-line.
- Support Static, DHCP and PPPoE mode.
- DHCP server and address pool.
- Support Bridge or NAT.
- Support QoS, ToS.
- Support SNTP.
- Support major G7.xxx CODEC.
- VAD, CNG, AEC, Jitter buffer.
- Support Local Dial Plan.
- Support Info, Inband and RFC2833 DTMF.
- Ring tones and tone set.
- Speed Dial, Phone Book, Call Log.
- Call Forward, Call Transfer.
- Three-Way and Conference.
- MWI
- Caller ID displays.
- DND, Mute, Hold.
- Provision function.
- Web UI management.
- Adjustable user password and super password.
- Adjustable volume of hands free and handset.
- Display logo.

◎Silkscreen

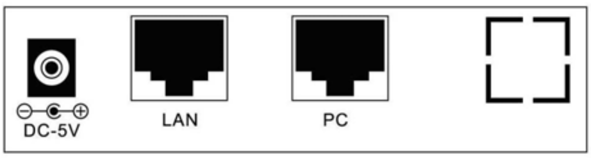


Figure1-1 GF502 SILKSCREEN

◎Description of Keypad

NO.	TITLE	DEFINE
1	OK	➢ Keep pace with the left word "OK", "Select" or "YES" at the screen bottom ➢ Enter the menu ➢ Confirm the modification
2	EXIT	➢ Keep pace with the right word "Cancel", "Back" , "Exit" , "Clear" or "NO" at the screen bottom ➢ Quit the menu ➢ Cancel the modification ➢ Delete input digit
3	<	➢ Move the pointer to the option above ➢ When editing, move the pointer to the left digit ➢ Enter the phone book list when Idle
	>	➢ Move the pointer to the option below ➢ When editing, move the pointer to the right digit ➢ Enter the phone book list when Idle
4	MIC + -	➢ When pick up the handset, adjust the handset volume ➢ When press hands free, adjust the speaker volume
	VOL + -	
5	HANDSFREE	➢ Get ready for dialing in idle mode ➢ Receive a call ➢ Dial out when selecting a phonebook or call log entry
6	REDIAL	➢ Dial out the last called number ➢ Switch the input mode between the number and the character when editing ➢ Enter the list of number dialed
7	HOLD	➢ During the call, come into the hold mode ➢ Press and hold to upgrade ➢ Check the phone's parameters of network
8	TRANSFER	➢ Transfer the call
9	FORWARD	➢ Configure forward type and number
10	DND	➢ Switch the DND mode on or off
11	LINE1	➢ Receive the new call during the call ➢ Make a new call during the call ➢ Switch between different conversations
	LINE2	

	LINE3	
	LINE4	
12	CONF	➢ Diagnose network status. ➢ Conference or Three-way
13	PHONEBOOK	➢ Enter the phone book menu.
14	History	➢ Enter the call log menu.
15	MESSAGE	➢ Access the voice mail
17	SPEED DIAL	➢ Configure speed dial numbers
18	MUTE	➢ Turn off the volume of the microphone during the call
19	Dialing Pad	➢ Functions like a traditional telephone dialing pad. ➢ While inputting words, press number key successively, and the current characters will change in turn ➢ Press [#] after telephone number to ensure to dial out
20	LINK LED	➢ Turn on if no service is available. ➢ Blinking if not all accounts are registered.
	MESSAGE LED	➢ Turn on when there is MWI
21	LAN	➢ Interface to LAN or WAN
22	PC	➢ Interface to PC
23	PWR	➢ Interface of power supply

◎*Networking Protocol*

The GF502 VOIP Phones support several industry-standard networking protocols required for voice communication and data transmission.

- ➢ SIP: RFC3261
- ➢ DTMF: Inband, RFC2833, SIP Info

- RTP: RFC1889
- RTCP: RFC3605, RFC3611
- DNS: RFC1035
- FTP: RFC9590
- TFTP: RFC1350
- HTTP: RFC1945, RFC2616
- SNTP: RFC1361, RFC2030
- SNMP: RFC1157
- STUN: RFC3489
- DHCP: RFC2131, RFC2132
- PPPoE: RFC2516
- NAT: RFC1631, RFC2663

◎Support Codec

These phones support codec such as:

- G.711a
- G.711µ
- G.723.1 5.3/6.3
- G.729AB

◎Hardware Parameters

The GF502 VOIP Phone hardware features:

Standard

- CPU: Infineon PSB21553
- Data Memory: 8MB SDRAM
- Software Memory: 2MB Flash memory
- Ethernet Port: 2 RJ-45 plugs, IEEE802.3 10/100 Base-T
- Power Adaptor: Input 110-250V, Output 5V DC, 1A
- Keypad: 36keys
- LCD: 4x20 char-based, white backlight
- LED: 2 LEDs

Optional

- Data Memory: 16MB/32MB SDRAM
- Software Memory: 4MB/8MB Flash memory
- PoE: 802.3af
- LCD: 128x64 Pixel-base with blue/white backlight
- Headset Port: 2.5mm plug set

◎Environment

- Operating Temperature: 0℃ to 50℃（32℉ to 122℉）
- Storage Temperature: -10℃ to 60℃（14℉ to 140℉）

- Relative Humidity: 10%-90%, non-condensing

Chapter 2: Install the GF502 VOIP Phone

This section provides instructions for installing the GF502 VOIP Phone. Before you perform the installation, be sure you have met the following prerequisites:

- Planned the network and GF502 VOIP Phone configuration.
- Installed and configured the other network devices.
- 10BASE-T or 100BASE-T or better Ethernet cable. One cable is needed for each Ethernet connection.

1 After you install a phone, even if it is new, upgrade the phone to the current firmware image.

For information about upgrading, refer to Chapter 4.

◎The Connects of the Box

The GF502 IP Phone includes these components on the phone or as accessories for the phone:

Standard:

- Telephone base unit
- Bracket
- Handset
- Handset cord
- 5V Power adaptor

Optional:

- RJ-45 Ethernet cable
- Headset
- CD
- User's manual

◎Installation procedure

Step 1 Fix the bracket to the back of phone, connect the handset with phone base unit using the handset cord.

Step 2 Connect one end of a standard RJ45 Ethernet cable to the LAN port on the rear panel of the GF502 VOIP Phone. Connect the other end to your IP network device such as a hub, switch or directly to the Network. If your phone supports Power over Ethernet (PoE) , the GF502 VoIP IP phone can be powered from a switch via Ethernet cable, in which case the external power adaptor is not needed.

Step 3 (Optional) Connect one end of a standard RJ45 Ethernet cable to the PC port on the rear panel of the GF502 VOIP phone. Connect the other end to your PC.

1 If you want your PC connect to network through the voip phone or you want to configure the phone by PC port, perform this step.

Step 4 Insert the power adaptor cable into the 5V power adaptor cable receptacle on the phone. Ensure that the power adaptor jack is snugly attached to the GF502 VOIP Phone.

Chapter 3: Configure the GF502 VoIP Phone

Three different ways can be used to configure GF502 VoIP phone: web browser, phone keypad and provision function. Configuring by provision function refer to ***Provisioning Guide for VoIP Device.pdf.***

◎Configure via WEB UI

GF502 VoIP IP Phone has a built-in HTTP server for the user to do the configuration via web browser. Editing the parameters on the web, then press "Apply" at lower right corner and reboot the phone to make the change activated.

1 Do press the "Apply" after changes have been done in one page. Most parameters need to reboot the GF502 VoIP phone to take effect.

3.1.1 Access web setting page

You can access web page by the IP of LAN port or PC port. For a new GF502 VoIP phone the default network mode is set to DHCP mode, the NAT switch and DHCP server are set to on.

Access through the PC port: Use Ethernet cable to connect GF502 VoIP phone's PC port and your computer. The default gateway address and subnet mask are 192.168.0.1 and 255.255.255.0. Set your computer's IP to the network 192.168.0.x, subnet mask 255.255.255.0, gateway 192.168.0.1. Open your web browser and key in 192.168.0.1. Then you will see the logon page of the device.

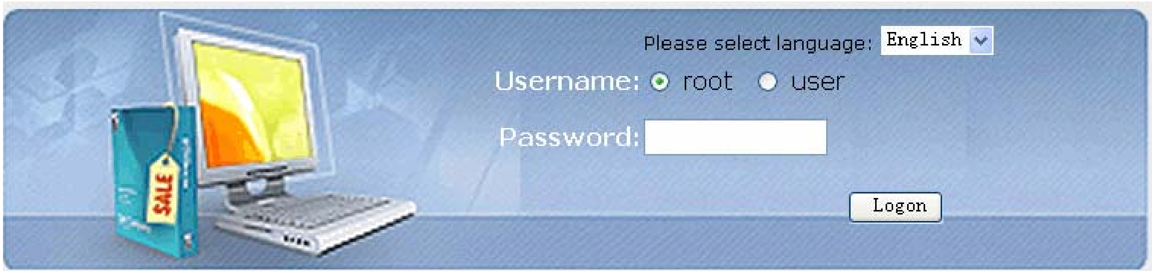
Access through the LAN port: Press **HOLD** of keypad in idle mode to check the LAN port IP. Open your web browser and key in the IP. Then you will see the logon page of the device.

1 The default username and password is root/111111 for administrator and user/000000 for user.

5 Do remember the password of root if you have changed it. You can't change it again if you forget it and do not setting provision.

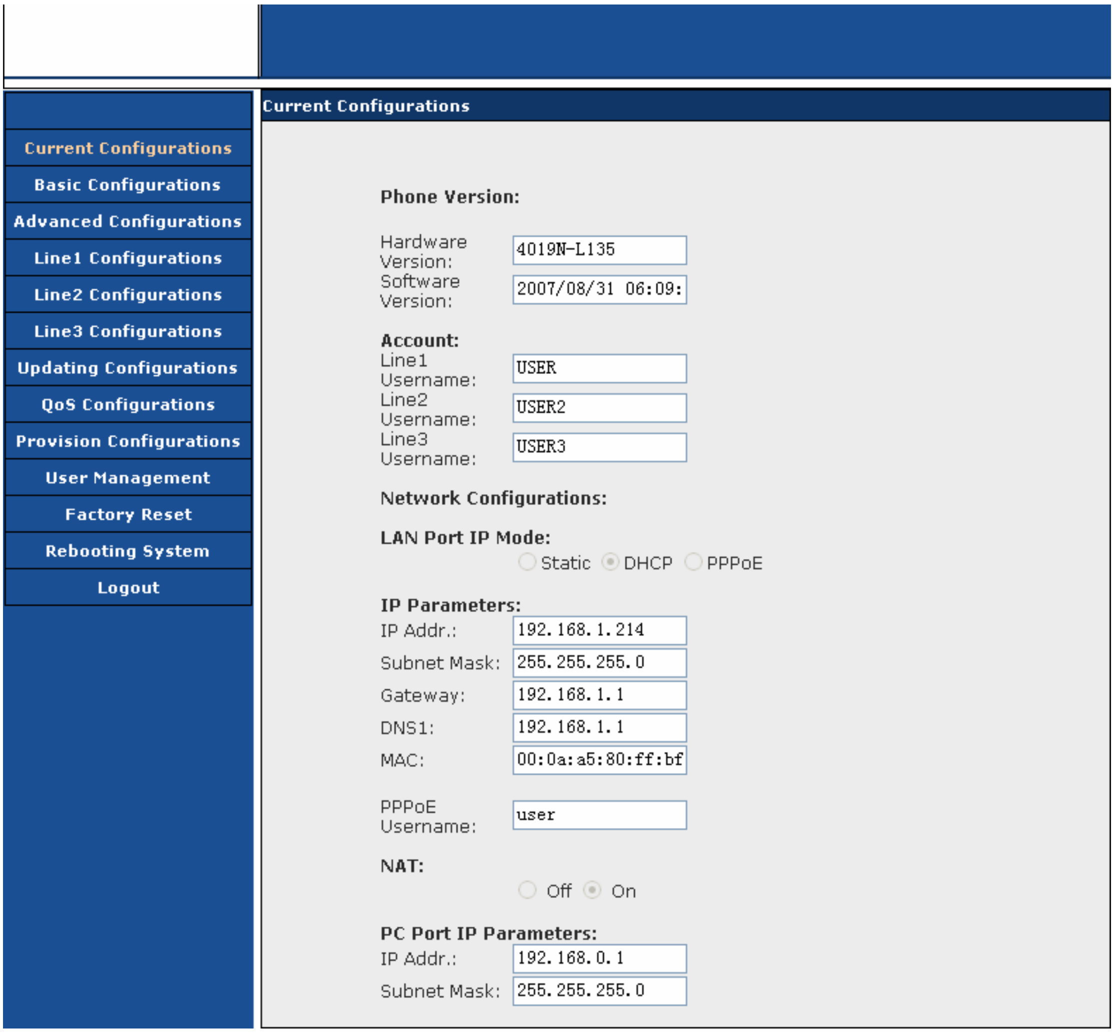
3.1.2 The logon page

Select the user name (user or root) and input password to logon. Root has no limit in view or configure via browser.



3.1.3 Current Configurations page

After logon, the page will show the GF502 VoIP phone running state about firmware version, accounts and network parameters.



Phone Version shows the hardware and the software version of this phone.

Account shows the account username which is in using.

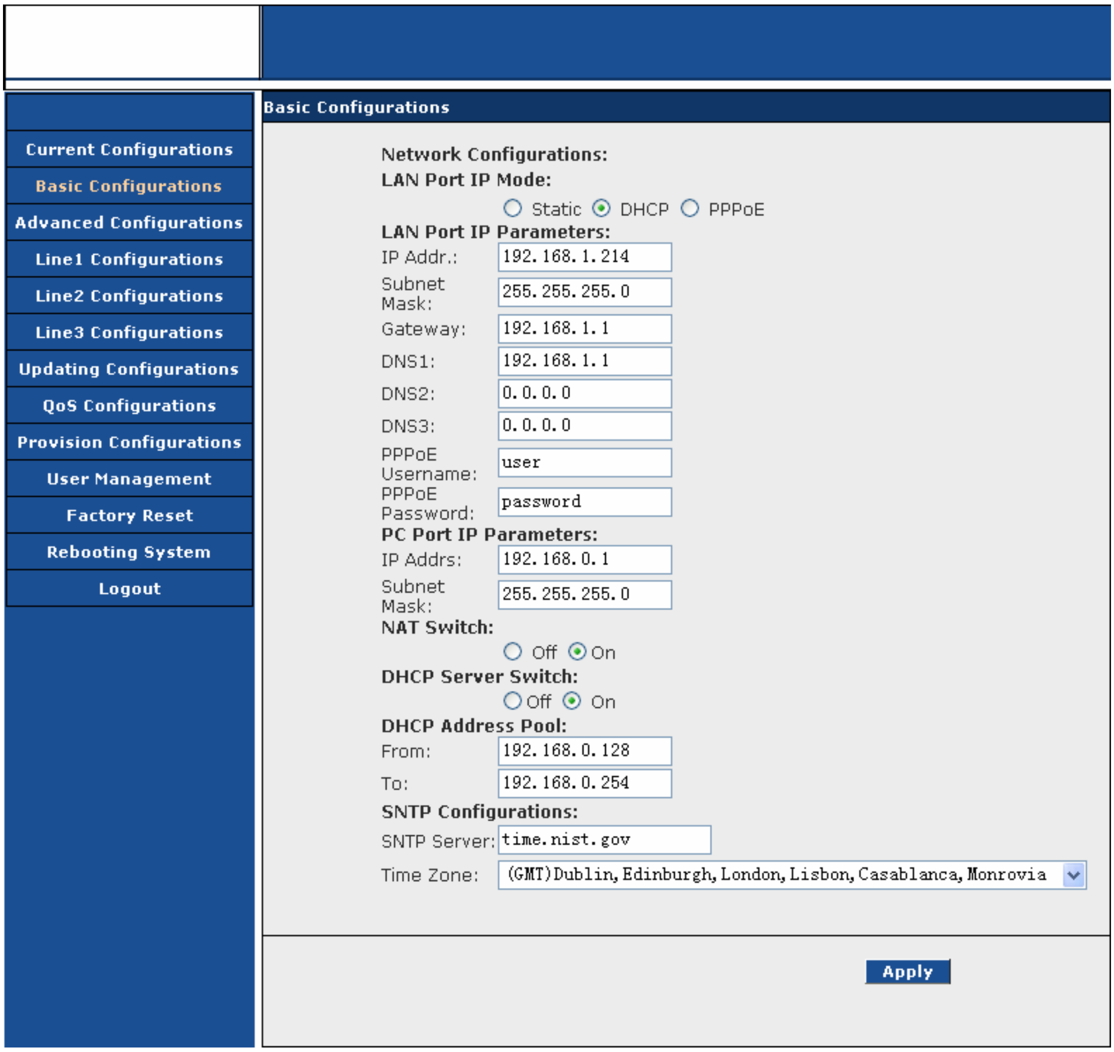
Network Configurations show the network mode, IP parameters and NAT.

3.1.4 Basic Configurations page

In the Basic Configurations page, parameters about network, time server and time zone can be set

1 The NAT switch and DHCP server are set to “on” as default, if you would not use NAT of phone, please set them to off. When the NAT switch is set to off, the network to PC will be set to bridge mode.

- If the network administrator assigns your phone a static IP or the DHCP function has been disabled (Generally company users using LAN network use this mode), please refer to Static Mode.
- If the network administrator has enabled the DHCP function, and the phone get IP address by DHCP, please refer to DHCP Mode.
- If you connect to the Internet directly by ADSL (Most home users and company users without LAN use this mode), please refer to PPPoE Mode.



- Configure Static IP:
  ----Enable *Static*;
  ----Set IP address in the *IP Addr.* field.
  ----Set subnet mask in the *Subnet Mask* field;
  ----Set gateway IP address in the *Gateway* field.
  ----Set DNS IP address in the *DNS1-3* field.
  ----Press *Apply.*

- Configure to dynamic obtain IP
  ----Enable *DHCP*.
  ----Press *Apply*

- Configure PPPoE:
  ----Enable *PPPoE*
  ----Enter PPPoE username and pin in the *PPPoE username* and *PPPoE password*.
  ----Press *Apply*

NAT Switch: Switch to Bridge mode (switch off) or NAT mode (switch on). The GF502 VoIP phone won't assign IP for its PC port in bridge mode and its PC and LAN port will be in the same network.

PC Port IP Parameters: Set the IP and subnet mask for the NAT of PC port.

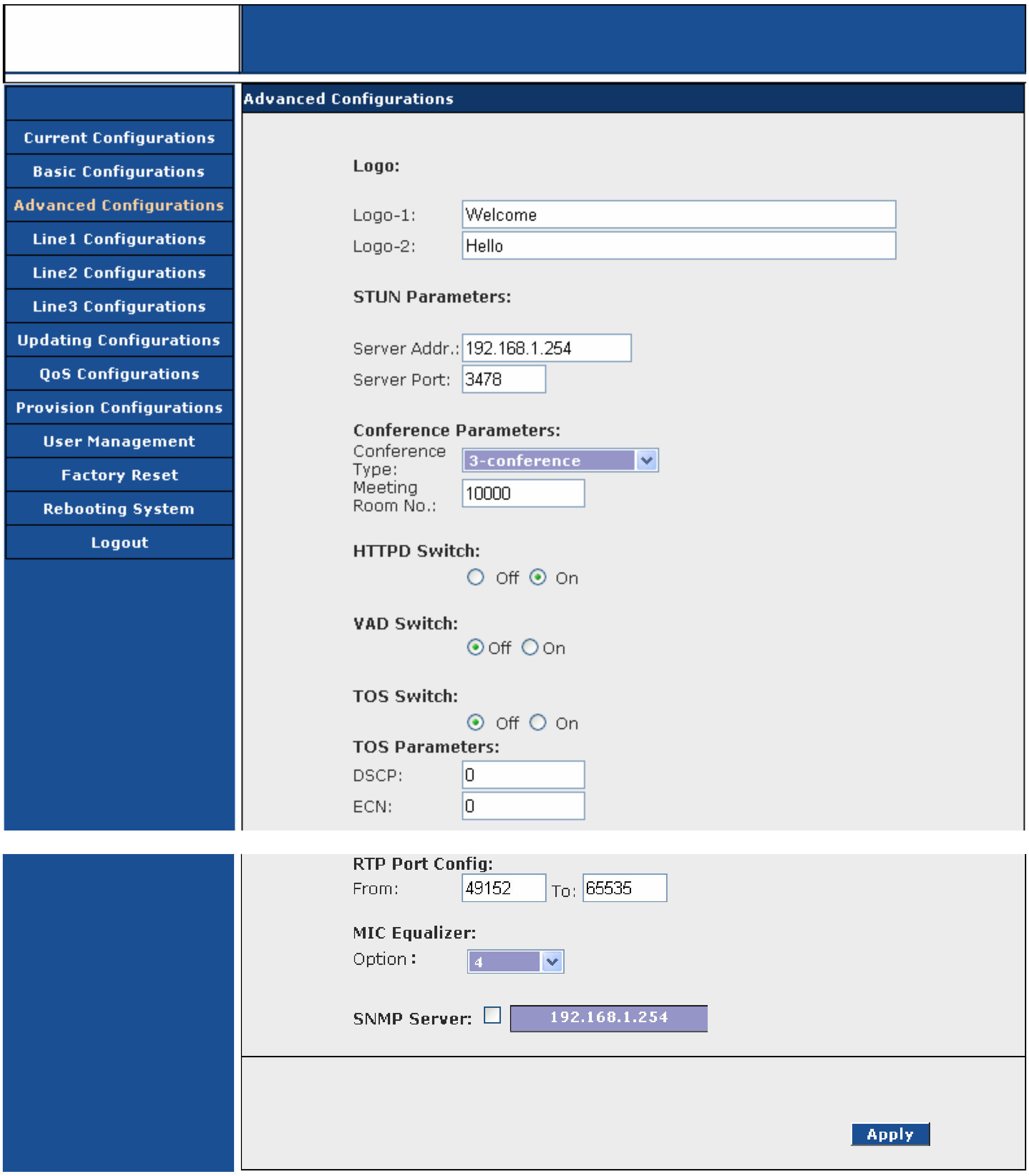
DHCP Server Switch: Enable the DHCP service in PC port.

DHCP Address Pool: Set the DHCP address for the DHCP service.

SNTP Configurations: Set the SNTP server and time zone. The default SNTP server is "time.nist.gov" and the default time zone is GMT+0.

3.1.5 Advanced Configurations page

In this page, some common parameters can be configured.



Logo: These will be display in the second line and third line of LCD when idle.

STUN Parameters: Set STUN server address and port if necessary.

Conference Parameters: Set the conference type and the conference room number.

HTTPD Switch: Enable or disable the WEB access. If you can't access the web, you can turn on the switch via keypad.

VAD Switch: Enable or disable Voice Activity Detection.

ToS Switch/ToS Parameters: Set the parameters about ToS.

RTP Port Config: Set the RTP port range.

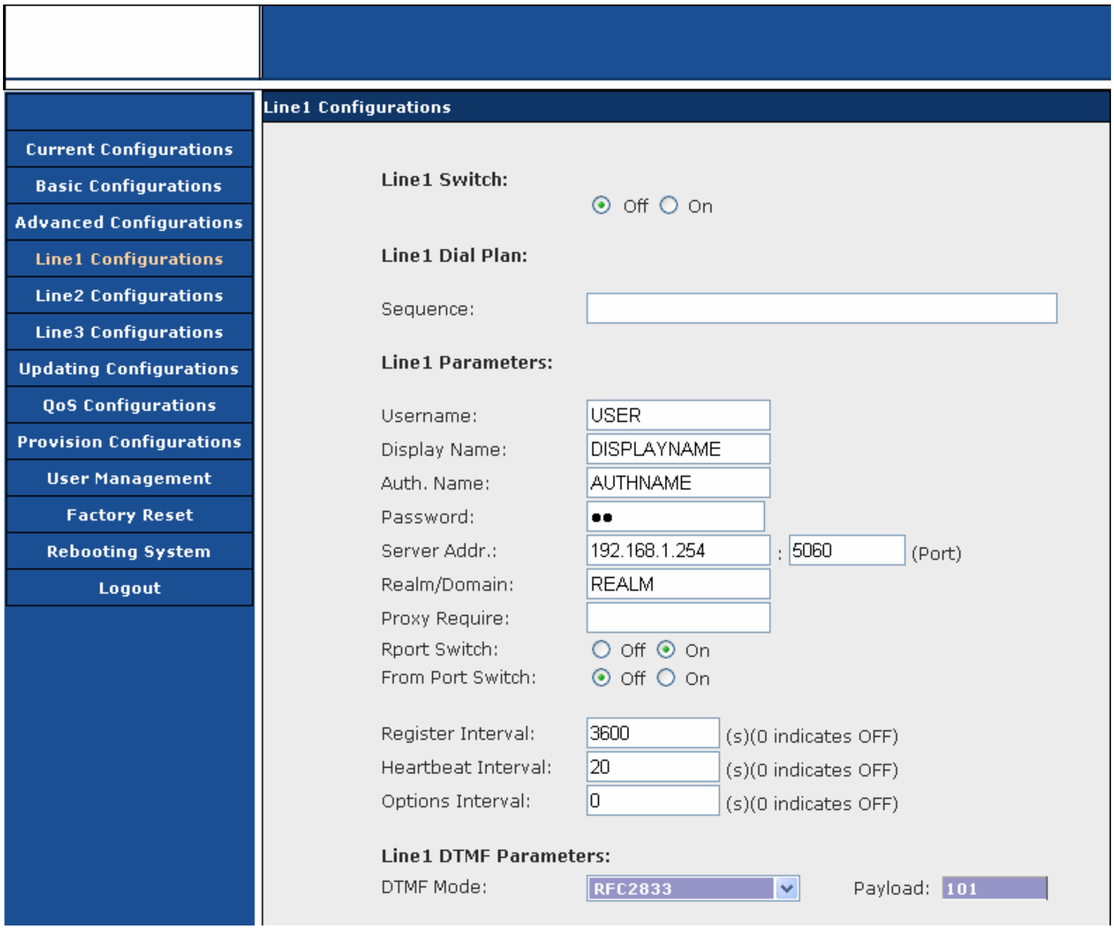
MIC Equalizer: Adjust the quality of sound.

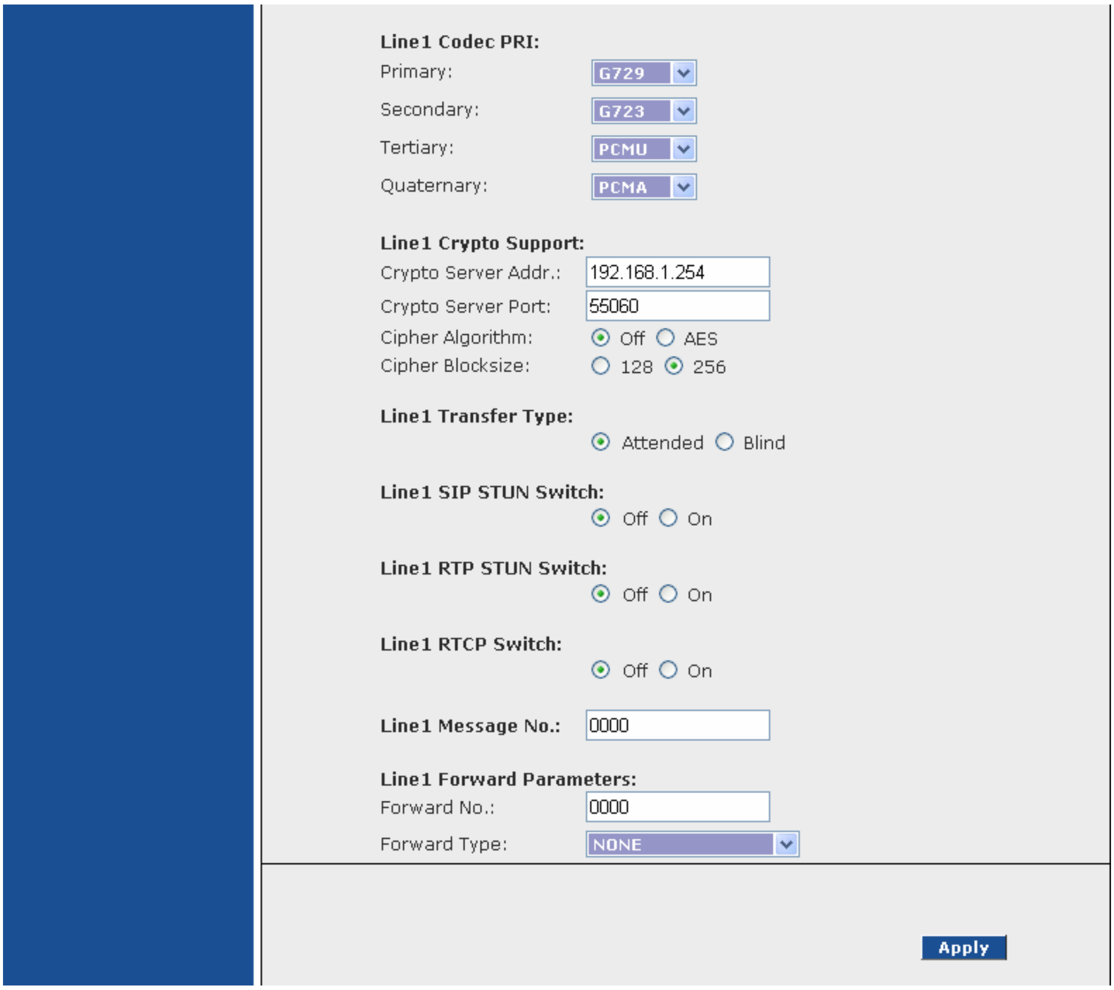
SNMP Server: Enable or disable the SNMP service and set the service address.

3.1.6 Line Configurations

Line1 Configurations, Line2 Configurations and Line3 Configurations are used to set parameters of accounts from Line1 to Line3. If you have more than 1 account, configure them in separate lines and make sure the line switch is “on”. If you just have one account, please set the line switch of others line to “off”. If you don’t need STUN when registering or talking, please set the “SIP STUN Switch” and “RTP STUN Switch” to off.

1 Check with your service provider, whether or not it need Stun, whether it need to support “rport” or port in from section of signal.





Line1 Switch: Enable or disable this line's register.

Line1 Dial plan: Set the sequence of this line's dial plan.

Username: Username of your SIP account (Commonly the same as the phone number).

Display Name: The display name sent to callee, if sip provider doesn't send a caller ID, this will be shown on the LCD of callee.

Auth name: Commonly the same as the phone username.

Password: Password of your SIP account.

Server Addr.: Register address of SIP server and port.

Realm/Domain: SIP domain. Enter the sip domain if you know, otherwise the GF502 VoIP phone will get the domain from signal maybe it is not correct.

Proxy Require: Some SIP service need this parameter.

Register Interval: Register expires time. The GF502 VoIP phone will auto configure this expire time to the server recommended setting if it is longer from the SIP server.

Heartbeat Interval: Send heartbeat pack periodically to keep the channel.

Options Interval: Send option signal to SIP server periodically to detect if the SIP server is available.

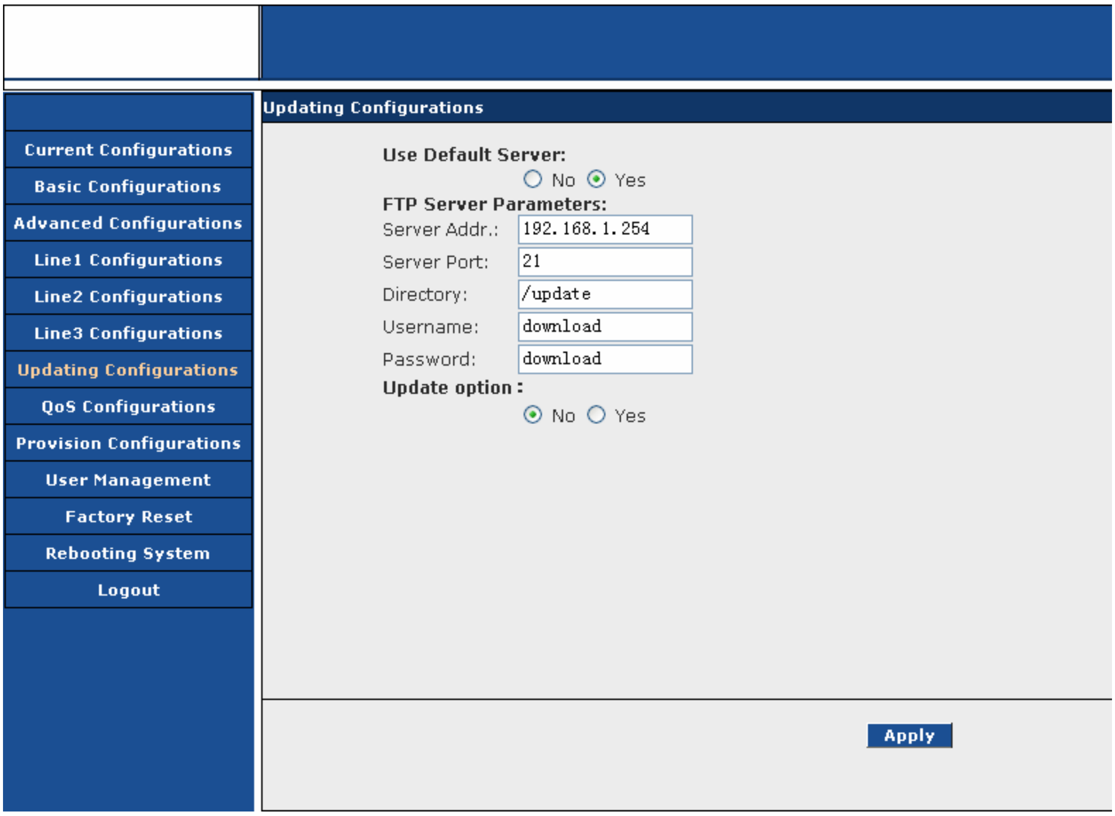
Line1 DTMF Parameters: DTMF signal sending mode and pay load. Support Inband, RFC2833 and Info.

Line1 Codec PRI: Setting the prefer codec in order. Support G729, G723, PCMU(G711u) and PCMA(G711a).

Line1 Crypto Support: Set the parameters to encrypt the signal. Need the SIP server support.
Line1 Transfer Type: Set the transfer type of this line: Attend or blind.
Line1 SIP STUN Switch: Stun switch of this line for signal.
Line1 RTP STUN Switch: Stun switch of this line for codec of voice.
Line1 RTCP Switch: Enable or disable the RTCP.
Line1 Message NO.: Set the line1 voice mail number.
Line1 Forward Parameters: Set the Forward type and number.

3.1.7 Updating Configurations page

The default path is our FTP server. If you want to use the default path, turn "Use Default Server" to “Yes” and press “Apply”. If you want to use another server, select “No” and input the parameters needed, and then press “Apply”.



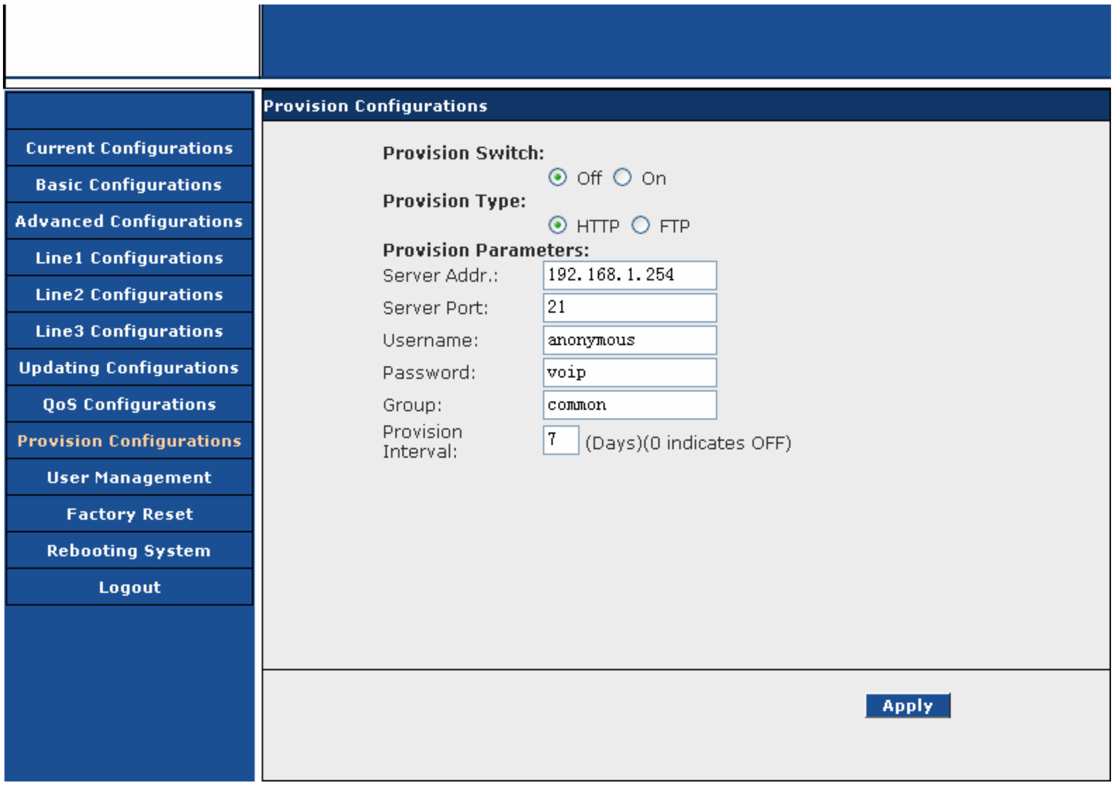
Use Default Server: Set Using the default update server or not.
Server Addr.: The FTP server address of update server.
Server Port: The FTP server port of update server.
Directory: The file path of the version.ini in the update server. If the file is in the root path of FTP server, fill in “/” or keep space.
Username: The login username of the FTP server.
Password: The login password of the FTP server.
Update option: Set “Yes” and press “Apply”, the phone will update immediately.

3.1.8 QoS Configurations page

The GF502 VoIP phone implement QoS based on 802.1p, The QoS is used to mark the network communication priority in the data link/MAC sub-layer. In this page parameters about QoS can be configured.

3.1.9 Provision Configurations page

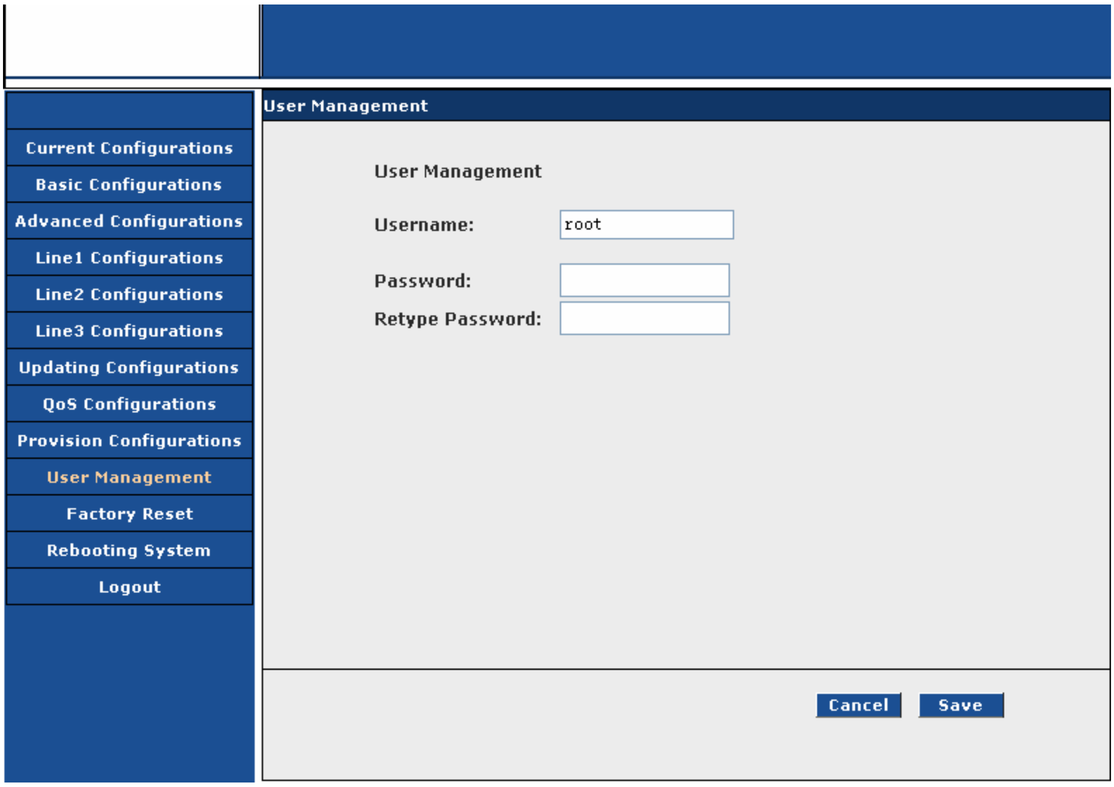
If you do not use provision function, set the switch to “off”, else, set it to “on”. There are two type provisions: HTTP and FTP mode. Select a mode and input correct value of provision parameters. Do not forget to press “Apply” after changing.



Provision Switch: Enable or disable the provision function.
Provision Type: Select the provision type.
Server Addr.: Provision server address.
Server Port: Provision server access port.
Username: Provision server login username.
Password: Provision server login password.
Group: The folder’s name which you put the setting.ini in.
Provision Internal: Set the internal of provision function.

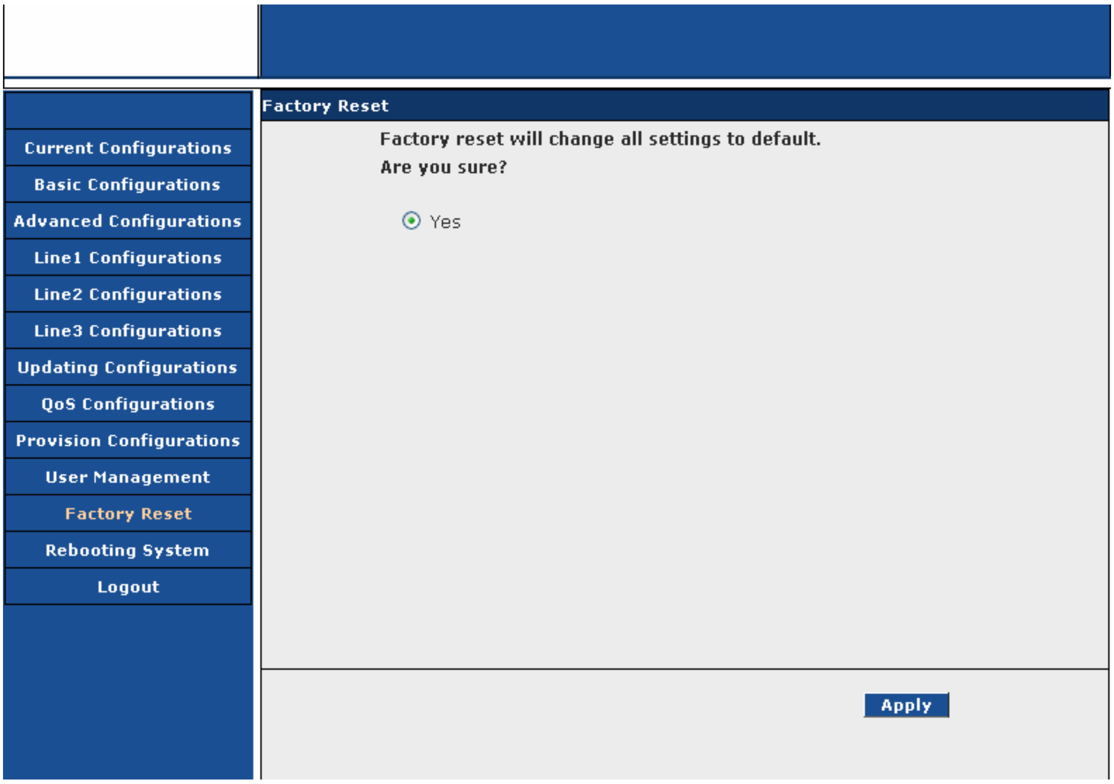
3.1.10 User Management page

Set WEB and keypad access account password of the GF502 VoIP phone.



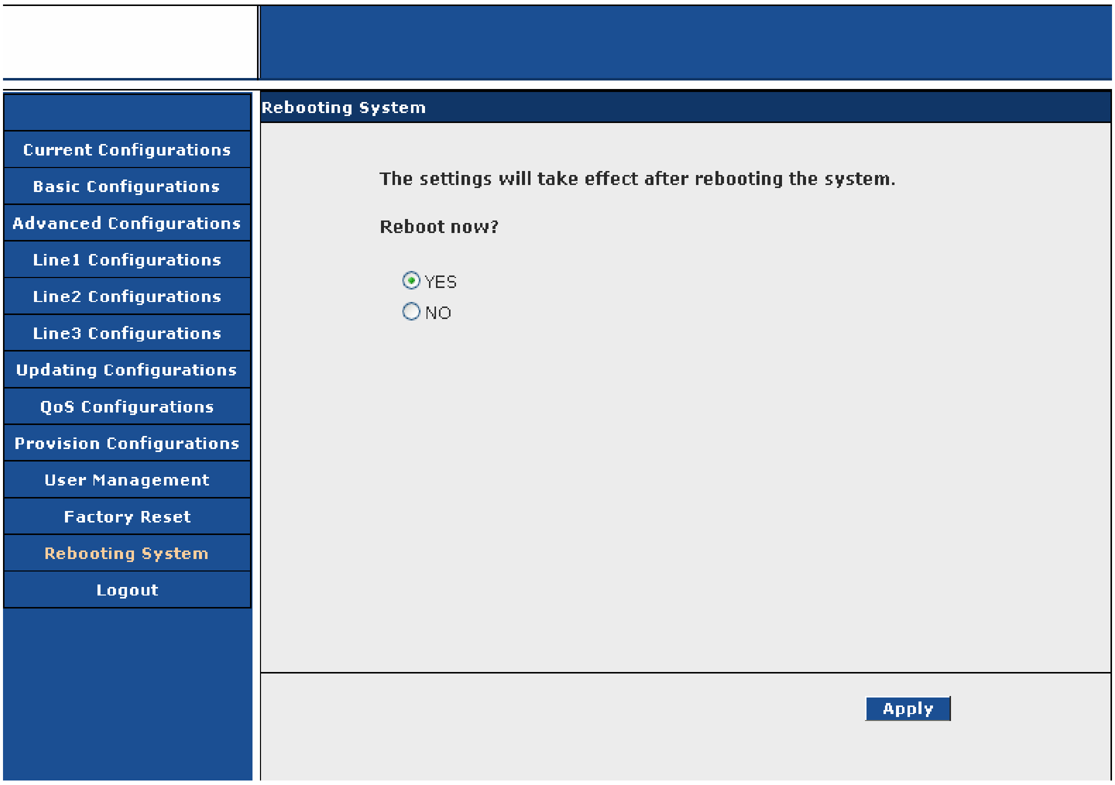
3.1.11 Factory Reset page

In this page press “Apply” will restore all parameters to the factory default.



3.1.12 Rebooting System page

After all changes have been done, click "Rebooting System", select "Yes", then click "Apply" to restart the device.



◎*Configure via keypad*

There are several keypad menus on GF502 VoIP phone: main menu, update menu, forward menu and other functions menus. To enter some menus you should input the password of "root" or "user". User password is for the "User" of main menu and forward menu; Root password is for the "Root" of main menu and update menu. The passwords are the same to access to WEB. The default usernames and passwords are root/111111 for administrator and user/000000 for user.

3.2.1 Function keys introduction

When using keypad and LCD to configure the settings of GF502 VoIP phone, following keys will be used:

Keys	***Function***	***Keys***	***Function***
OK	Enter main menu and submenu; Confirm change	CANCEL	Exit current menu; Exit setting mode; Del the char at left of cursor
LEFT	Turn over menu up;	RIGHT	Turn over menu

	Move cursor left		down; Move cursor right
HOLD	Enter the update menu	FORWARD	Enter the forward menu
REDIAL	Switch input type when entering letters		

3.2.2 Text input

When configuring the GF502 VoIP phone by keypad, sometime it is necessary to input something. To select input method in the input interface by press **REDIAL** key: char (with indicator “abc”) and numeric (with indicator “123”).

3.2.3 Main menu

At the idle mode, press **OK** to enter the main menu.

GF502 VoIP Phone Keypad Main Menu			
Level 1	*Level 2*	*Level 3*	*Level 4*
User	Network	External Mode	STATIC
			DHCP
			PPPoE
		External NIC	Ext ip address
			Ext gw address
			Ext subnet mask
			Ext dns1
			Ext dns2
			Ext dns3
		PPPoE	PPPoE user ID
			PPPoE password
			PPPoE service name
		NAT Settings	NAT
			DHCP Server
			Int IP From
			Int IP To
			Int ip address
			Int subnet mask
	Tones	Ringing Tone	
		Ringing Volume	
		Side tone Volume	
	Clock	Date	
		Time	

		GMT Offset	
		SNTP Server	
		Synchronization	
	User Password		
Root	Logo Settings	Logo	
		Logo2	
	SIP1 Settings	Sip Switch	
		Sip1 Params	Proxy Server
			Proxy Port
			UserName
			Auth Name
			Password
			Realm
			Support rport
			Register Interval
			Heartbeat Interval
			Options Interval
			DTMF Mode
			RFC2833 Payload
		Crypto1 Params	Encrypt server ip
			Encrypt server port
			Sip Encrypt depth
			SIP crypt
		Codec	First Codec
			Second Codec
			Third Codec
			Fourth Codec
		Forward1 Params	Forward Type
			Forward No
	SIP2 Settings (similar to SIP1 Settings)		
	SIP3 Settings (similar to SIP1 Settings)		
	Rtp	RTP Port From	
		RTP Port To	
	Conference	Conference Type	
		Conference Room	
	Httpd	Httpd Switch	
	Provision	Provision Switch	
		Provision Type	
		Provision Server	
		Provision Port	
		Provision User	
		Provision Password	
		Provision Group	

		Provision Interval
	SNMP	SNMP switch
		SNMP Server ip
	Root Password	
	Reset	
	HD_TEST	

3.2.4 Update menu

At the idle mode, press and hold the key **HOLD** for three seconds then input the root password to enter the update menu.

UPDATE MENU	Update config	Use default
		FTP Serveraddr
		FTP Port
		FTP Path
		FTP User
		FTP Password
	Update	
	Version	

3.2.5 Forward menu

At the idle mode, press **FORWARD** then input the user password to enter the forward menu.

FORWARD MENU	Forward1 Parameters	Forward Type
		Forward No
	Forward2 Parameters	Forward Type2
		Forward No2
	Forward3 Parameters	Forward Type3
		Forward No3

3.2.6 Other functions menu

At the idle mode, press associated keys to enter the other functions menu

OTHER FUNCTIONS	Phone Book	List
		Search
		Add
		Erase
	Call Log	Missed

		Received
		Dialed
		Erase
	Speed Dial	
	Ping	

Chapter 4: Update the Firmware

5 Do not cut off power during the upgrade. Maybe it will lead to fatal error. The LCD of phone will show the upgrade process, when done the phone will reboot automatically. If the phone doesn't reboot, please reset power after the information "Please switch off" to the LCD.

The GF502 VoIP phone uses a FTP server to download the firmware. The FTP server should be ready and be able to accept connections.

First you need to configure the update parameters via WEB or keypad (refer to Chapter 3). After the configuration has been done, there are two way to update.

◎Update via keypad

Enter the update menu, move the cursor point to "Update" option, Press **OK**. The phone will check new version and automatically download the firmware.

◎Update via WEB

Enter the Updating Configurations page with browser. Set "Update Option" to "Yes" and press "Apply". The phone will check new version and automatically download the firmware.

Chapter 5: Usage of the Phone

◎*Talk mode and operations*

5.1.1 Communication mode

During a call if you want switch the talk mode, you can do the operation like this: Press H.F key then replace handset from handset mode to hands free mode; just pick up handset from hands free mode to handset mode

- Handset mode
- Hands free mode
- Headset mode(optional)

5.1.2 Operation of off hook

- Picking up the handset
- Press H.F key
- Press Line1-Line3 key

5.1.3 Operation of on hook

- Replacing the handset when handset mode
- Press H.F key when hands free mode

◎*Main base call functions*

5.2.1 Making calls from the GF502 VoIP phone

On-hook dialing

In idle mode, there are several ways to show dial number.

- Use the keypad to input the number you wish to call
- Press **REDIAL** to display the last 99 phone numbers you called.
- Query phone book
- Query call log

And then you can trigger dialing by going off-hook

Off-hook dialing

Performing off-hook operation first, and then you can dial out by any one of following ways.

- Press digit keys
- Press **REDIAL** key

1 In the off-hook dialing by pressing digital keys method, after the number press "#", then the number will be dial out directly, or else the phone will wait five seconds to dial the number out. Maybe there will be impact when you dial number because of your local dial plan setting.

5.2.2 Speed dial

In the speed dial menu, select the memory key you wish to edit, press **OK**, key in the number and then press **OK** to store the entry.

Press and hold one of the 0-9 key which store the number you want to dial in idle mode, the GF502 VoIP phone will dial out the number.

1 The speed dial function support the first registered account only.

5.2.3 Answer a call

When a call is coming, the caller number or alias is displayed, if the caller number is stored in the phonebook, the associated name is displayed.

While the phone is ringing, you can perform off-hook action to answer the call. Incoming call during talking refer to multi-line section.

5.2.4 Reject a call

Press the **CANCEL** to reject a call when the phone is ringing.

◎Handling calls function

5.3.1 Call forward

The function of the Call Forward processed the line forwarding to another phone number. The user can set the system to another phone number. When a call is coming, the system will forward the line to another phone number that is set previously. The system supports four kinds of ways to implement this function including "All", "Busy", "No Answer" and "Busy & NO

Answer".

All: Provided that a call is coming, the system will forward the line immediately.

Busy: A call is coming and the user is on a busy line. Then the system will forward the calling line.

No Answer: On condition that the user does not respond to the calling, the system will forward the line to another phone number.

Busy & NO Answer: The system forwards the call if either the line is already in use or the call is not answered in time.

You can configure the forward function via web or keypad, Refer to **Chapter 3**: 3.1.6 or 3.2.5

5.3.2 Hold and retrieve a call

When you place a call on hold, only your phone can retrieve the call. To place a call on hold:

- Connect to the call(if not already connected)
- Press the **HOLD** key or press the relevant line key

When switch between calls, you do not have to press the **HOLD** key to go from one call to the next. The phone will automatically put your current call on hold as soon as you press a new line key.

To retrieve a held call, you can press **HOLD** or the relevant line key.

5.3.3 Transferring calls

Please set the transfer type via WEB or keypad: Attend Transfer or Blind Transfer.

Blind Transfer: A blind transfer is when you transfer a call directly to another extension without consulting with the person receiving the call.

- Connect to the call you wish to transfer (if not already connected).
- Press **TRANSFER**
- Dial the desired phone number to which you want to transfer the current call.
- Hang up

1 If you do not wish to transfer the call after pressing **TRANSFER**, press **HOLD** to retrieve the call.

Attend Transfer: You want to consult with the person you are transferring the call to, before you complete the transfer.

- Connect to the call you wish to transfer (if not already connected).
- Press **TRANSFER**
- Dial the desired phone number to which you want to transfer the current call.
- After connected, press **TRANSFER** again.
- If you do not wish to transfer the call, hang up the new connect. This disconnects the

new call, leaving the original call on hold, off-hook to retrieve the call.

1 If you wish to do attend transfer, an idle line with no account is needed to make the new call.

5.3.4 Mute

During conversation, press the **MUTE** key to mute. You will still be able to hear the caller, but they will not hear you. Press again will disable mute.

1 If you place a muted call on hold, the phone will not take the call off mute when you reconnect to the call.

5.3.5 DND

If DND "Do Not Disturb" function is enabled, when a call comes in, the caller hears a busy signal. In idle mode, press **DND** will enable this function, to disable it press DND again. If DND is enabled, DND will display on LCD.

5.3.6 Multi-line

The GF502 VoIP phone supports up to 3 lines. Each line can be separately operated various services, including answer, call, hold, transfer etc. To switch between line/call pressing line keys. Besides the current active line, all others are on hold by IP phone and can be resumed by pressing associated line key. Press active line key will hold the call.

For example:

<u>When you are communicating with one party and wish to make a brief to another party.</u>

- ➢ Press a free line key (then you will hear dial tone, and the original call is held)
- ➢ Key in number you want to call and dial out

<u>Answer an incoming call when another call is active</u>

- ➢ The phone will assign a free line dedicated to the incoming call, and a call waiting tone will ring.
- ➢ Press this line key, and the original line is held by phone automatically

5.3.7 Three-way

To make a three-way talk, make sure the conference type is set to three-way mode.

- ➢ Connect to the first party you wish to include in the conference (if not already connected).

- Change line to make a new call or pick up a new call.
- Press **CONF** to complete the three-way conference.

5.3.8 Conference

To make a conference by conference room number from service provider, make sure the conference type is switched to conference mode, and the room number is set. Please check the operations with your service provider

- Connect to the first party you wish to include in the conference (if not already connected).
- Press **CONF** to place the current party into the conference room.
- Dial the new number of the person who you wish to add to the conference.
- After connected, press **CONF** to place the new party into the conference room.
- Repeat the two steps until all persons who you wish to add are in the conference room.
- Press **CONF** again to add yourself into the conference.

5.3.10 Message

This function is used to access voicemail number. It is mostly equal a speed dial key. You can check the service and number with your service provider, and configure it via WEB.

Some call center show new messages and the number of messages in the LCD, for example: Msg: 1/5 means that you have one new message and five old messages.

5.3.11 Volume adjustment

You can adjust the ring tone and side tone in the tone menu via keypad. There are five types of ring tone and five levels of volume.

During conversation, you can adjust volume of microphone by "**MIC +**"or "**MIC -**" keys and adjust volume of speaker by "**VOL +**" or "**VOL –**" keys.

◎*Phone book and call log*

5.4.1 Phone book

In idle mode, press **PHONE BOOK** to access the phone book. You can view, add and delete entry in the phonebook. You can also dial from phonebook. You can enter up to 200 entries into the GF502 VoIP phone book by adding them manually, or by saving the number and name from other lists stored on your phone.

1 In the details interface of phone book, if the text can’t be display in one line, you can use the **MIC + -** or **VOL+ -** to see the rest information.

- Add a entry to phone book
  a. Add in idle mode: key in the number which you want to save when idle, then press **OK**. In the next interface select an option (Tel, Mobile) where you want save to and press **OK**. Input the name and press **OK** again to finish.
  b. Add in phone book submenu: Select Add in the phone book menu, key in the name and number step by step follow the direction on the LCD screen.
  c. Add from call log: Access an entry of call log which you want to save and press **OK** to operation options. Using the save option, select a location (Tel or Mobile) and then input the name. Press **OK** to finish.
- View the list of directory
  a. View in idle mode: Press **LEFT** or **RIGHT** key will enter the list of phone book directly.
  b. View in phone book submenu: Select List in the phone book menu.

1 If the phone book list is empty, “No record” is displayed.

- Search for an entry: To search for an entry, you can use the Search of phone book menu. Select to search by name or number.

1 To search, you should key in the whole name or number now.

- Edit contact details: Select the contact you want to edit by list or searching. Scroll to the desired name, TEL number or Mobile number. Press **OK** to edit.
- Delete contacts: In the Erase submenu of phone book, you can delete all contacts or one by one. Select “All” to erase all phone book memory. Select “One by one” and scroll to the desired entry you want to erase, press **OK** to delete it.
- Import and Export are not available now.
- Dial out from phone book: Dial from list of phone book, go off hook. The TEL number will be dialed out (if there is no TEL number, the Mobile number will be dialed out). Dial from details interface, scroll to the number you want to dial, and then go off hook.

5.4.2 Call log

In idle mode, press **HISTORY** to access the call log. You can view, edit, save and delete entries in the call log. You can also dial from call log. There are three kinds of call logs: Missed call, Received call and Dialed call. Each supports 99 entries. When reaching 99, the oldest will be erased to place a new one.

- View the call log: In the call log menu, scroll to the log you want to view and press **OK** to access the log. Use **LEFT** and **RIGHT** keys to scroll between log entries. Press **OK** will access the detail interface of the entry.

1 If there is new missed call, "[x] Missed Call" will display on LCD, and the call log submenu will directly access to missed call logs. In idle mode, press **REDIAL** key will directly access to dialed call logs.

- Edit and Save: In the detail interface, scroll to "edit", and press **OK** to edit the number. Saving the number to phone book refer to 5.4.1 Phone Book.
- Erase the call log: To delete one entry, enter the detail interface, scroll to "erase" and press **OK**. To delete a group or all entries, access to the main menu of call log. Select "Erase", scroll to the desired group which you want to delete, and then press **OK**.

◎Other functions

5.5.1 Ping

In idle mode, press **DIAG** to access this function. You can use it to test the network.

5.5.2 Lock

In idle mode, press and hold **MUTE** key for three seconds, the phone will be locked. Key in the password of root or user, the phone will be unlocked. If locked, you cannot dial out number but you can still receiving incoming calls.