New Rock Technologies, Inc.

NRP1012 Series IP Phone

User Manual

NRP1012

NRP1012/P

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Safety Notices

Please read the following safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

- Please use an external power supply that is included in the package. Other power supplies may cause damage to the phone, affect the behavior or induce noise.
- Before using the external power supply in the package, please check with home power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it, it may cause fire or electric shock.
- The plug-socket combination must be accessible at all times because it serves as the main disconnecting device.
- Do not drop, knock or shake it. Rough handling can break internal circuit boards.
- Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposure the phone to high temperature, below 0℃ or high humidity. Avoid wetting the unit with any liquid.
- Do not attempt to open the device. Non-expert handling could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock and breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug or phone line, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place.
- You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

Contents

Contents of Figure

Contents of Table

1 Introduction

1.1 Appearance of an IP Phone

Figure 1-1 Appearance of an IP Phone

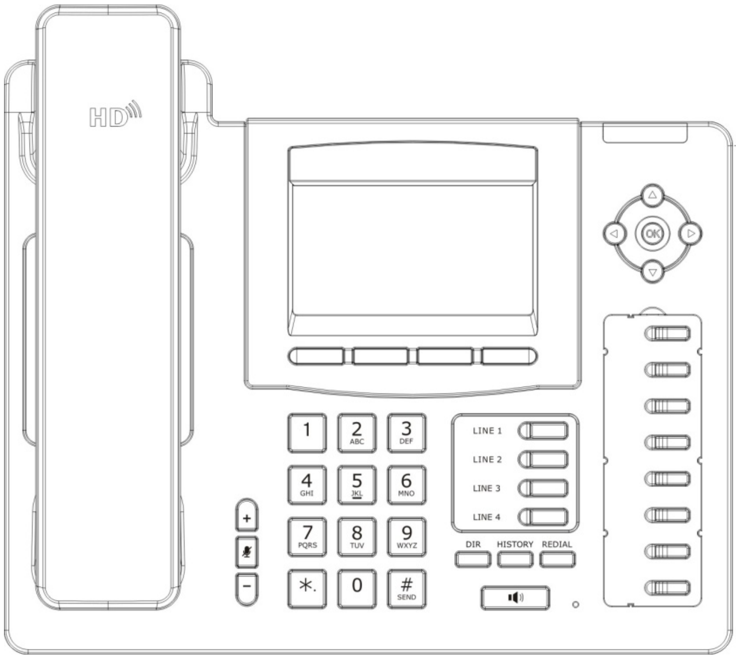
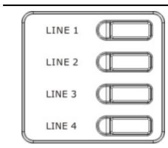
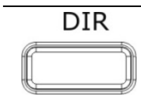


Table 1-1 Keypad description

Key	Name	Description
	Line key	The default is defined as the SIP line key, the user can select keys SIP line call, the call can also be defined as a line key, the call for a different way selector.
	Soft key 1/2/3/4	Executes different functions with different softkeys, such as PBook/DND/Menu/Del/Redial/Send/ Quit/Answer/Reject/Hold/Transfer/Conf and so on.
	Navigation	Helps to carry out the selection operation. When the IP phone is in the standby state, each navigation key has a special function. You can configure navigation keys with different functions according to habit.
	Phone book	Accesses to phone book, checks the contact list, adds new contacts, edits a contact, and returns to the standby page.

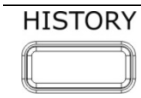
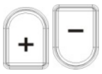
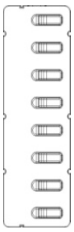
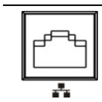
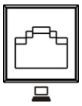
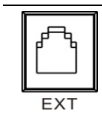
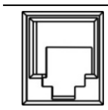
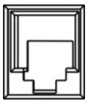
Key	Name	Description
	History	Checks information about the missed call, incoming call and outgoing call.
	Redial/Send	• Dials the last call number when the IP phone is in the hook off/hands-free mode. • Checks the outgoing call when the IP phone is in the stand-by mode. • Finds a specific contact in the phone book/call records and makes a speed dial.
	Hands-free	Make the phone into hands-free mode.
	Mute	Prevents the caller from hearing any sound of the callee when the IP phone is in the calling state, Note that the sound from the caller will not be affected and still be heard. Quits the mute mode, allowing the caller and callee to hear each other.
	Volume +/-	Turns the sound volume down or up.
	Indicator light	Blinks to prompt a missed call
	Digital keyboard	Inputs the phone number or numbers
	DSS keys	Configures each key with different functions in the Web page.

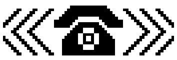
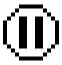
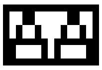
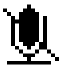
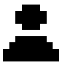
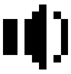
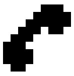
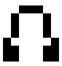
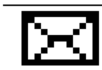
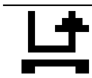
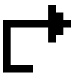
Table 1-2 Connection port description

Port	Name	Description
	Power interface	Input: 5V AC, 1A
	WAN	10/100M self-adaptive Ethernet interface for network connection
	LAN	10/100M self-adaptive Ethernet interface for PC connection
	Expansion board interface	Port type: RJ-45direct connector
	Headset	Port type: RJ-9 connector

Port	Name	Description
	Earpiece	Port type: RJ-9 connector

1.2 Icon Description

Table 1-3 Icon description

Icon	Description
	Call out
	Call in
	Call hold
	Auto answer
	Mute
	Contact
	Do not Disturb (DND)
	Hand-free
	Hook
	Headset
	SMS
	Missed call
	Call forward

1.3 LED Indicator

Table 1-4 Programmable key LED indicator status for BLF

LED Status	Description
Steady green	The object is in idle status.
Slow blinking red	The object is ringing
Steady red	The object is active
Off	The object is failed subscribe. Or No subscribe.

Table 1-5 Programmable key LEDs for Presence

LED Status	Description
Steady green	The object is online.
Slow blinking red	The object is ringing.
Steady red	The object is active.
Off	The object is failed./No subscribed.

Table 1-6 Programmable key LEDs for line

LED Status	Description
Steady green	The account is active.
Fast blinking green	There is an incoming or outgoing call to the account.
Slow blinking green	The call is on hold/the registration fails.
Slow Blinking red	Registration fails.
Off	The line is not registered or idle.

Table 1-7 Programmable key LEDs for MWI

LED Status	Description
Blinking green	There are new voice mails.
Off	There are no new voice mails.

Table 1-8 Power Indication LED

LED Status	Description
Steady red	The power connection is established/there is a missed call (the power indicator is on).
Slow blinking red	There is an incoming call.
Off	The power connection is not established/the power indicator is off.

1.4 Hardware Specifications

Table 1-9 Hardware specification

Item		Description
Adaptor		Input: 100-240V Output: 5V 1A
Port	WAN	10/100Base- T RJ-45 1 PORT
	LAN	10/100Base- T RJ-45 1 PORT
	Headset	RJ-9
	EXT	RJ-11

Item	Description
Power consumption	Idle: 2.5W/Active: 2.8W
LCD size	80 x 42mm
Operation temperature	0 to 40 ℃
Relative Humidity	10 to 65 %
CPU	Broadcom VoIP chipset
SDRAM	16 MB
Flash	4 MB
Weight	1.2 Kg

1.5 Character Mapping Table

Table 1-10 Mapping relation between the keypad and character

Keypad	Character	Keypad	Character
1	1 @	7 PQRS	7 P Q R S p q r s
2 ABC	2 A B C a b c	8 TUV	8 T U V t u v
3 DEF	3 D E F d e f	9 WXYZ	9 W X Y Z w x y z
4 GHI	4 G H I g h i	*.	*/.
5 JKL	5 J K L j k l	0	0
6 MNO	6 M N O m n o	# SEND	#/SEND

2 Initial Connection and Settings

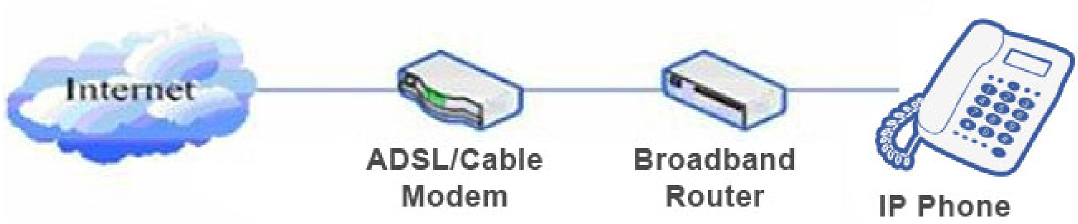
2.1 Power and Network Connections

2.1.1 Connecting to a Network

Prior to the configuration, make sure that broadband Internet access services are available in the current environment.

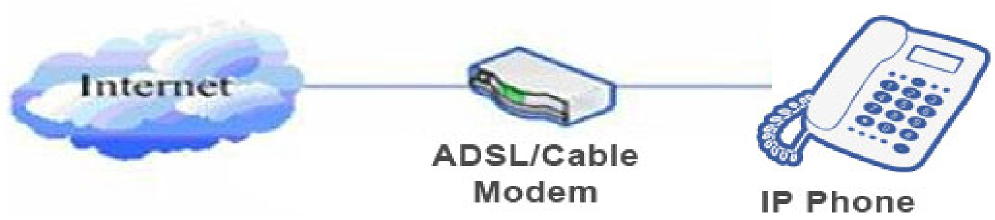
With a broadband router

Establish connections between the network hardware by connecting one end of the network cable to the WAN port of the IP phone and the other end to the LAN port of your broadband router. In most cases, configure your network settings to DHCP mode.



Without a broadband router

Establish connections between the network hardware by connecting one end of the network cable to the WAN port of the IP phone and the other end to the LAN port of your broadband modem. If a TV cable broadband is used, in most cases, you need to set the network mode of your IP phone to DHCP. If ADSL is used, you need to set the network mod to PPPOE.



With a substitute broadband router

NRP1012/P itself has the broadband routing capability. As long as the WAN port of the NRP1012/P is properly connected to the broadband modem and the LAN port of the NRP1012/P is connected to your computer or other devices with Internet capability, you can access the Internet.

For detailed settings, see 2.2.1 Network Settings.

2.1.2 Connecting to the Power Outlet

Prior to the configuration, make sure that the power connector matches the power outlet, and both the voltage and electric current are comply with the requirement.

- Plug the direct current (DC) connector of the power adapter into the DC 5V port in the rear of the NRP1012/P.
- Plug the alternating current (AC) connector of the power adaptor into the power outlet and start the NRP1012/P up.
- You can view the system progress and the **INITIALIZING** prompt on the screen. After the startup, you can view the current date, system time, and device model on the screen of the NRP1012/P.
- You can configure your account and make phone calls after the NRP1012/P is registered to a server

2.2 Basic Initialization

NRP1012/P provides a plenty of functions and parameters for configuration. To make a proper configuration, you need to know basic information about the network and Session Initiation Protocol (SIP) to understand the meanings of each parameter. To help you enjoy the high-quality VoIP services with low costs, this chapter describes items must be configured, making the configuration easy and intuitive…

2.2.1 Network Settings

Prior to the configuration, make sure that the Internet broadband connection is normal and the hardware is properly connected. The factory network mode of the NRP1012/P is DHCP, allowing the NRP1012/P to access the network as long as there is a DHCP server. If there is no DHCP server available, you need to change the network mode of the NRP1012/P to Static IP or PPPoE according to the network type.

PPPOE mode (for dial-up ADSL)

Step1 Obtain the PPPoE account and password.

Step2 Press **Menu -> Settings -> Advanced Settings** and enter the password. Choose **Network -> WAN settings -> Connection Mode**. Then, choose PPPoE through navigation keys and press **Save**.

Step3 Press **Back** to return to the parent menu, choose **PPPoE Setting**, and press **Enter**.

Step4 Press **Del** to delete the current information displayed on the screen, enter the username and password, and press **Save**.

Step5 Press **Back** six times to return to the stand-by screen.

Step6 Check the status. If the screen shows "Negotiating…", the NRP1012/P is trying to access the PPPoE Server; if the screen shows an IP address, the PPPoE mode takes effect.

Static IP mode (for static ADSL/Cable or no PPPOE/DHCP network)

Step1 Obtain information about the network parameters , such as the IP address, subnet mask, default gateway/router, and Domain Name System (DNS) from the network provider or network technical personnel.

Step2 Press **Menu -> Settings -> Advanced Settings**, and enter the passwords. Choose **Network -> WAN settings -> Connection Mode**. Then, choose **Static** through navigation keys and press **Save**.

Step3 Press **Back**, to return to the parent menu, choose **Static Setting**, and press **Enter**.

Step4 Press **Del** to delete the current information displayed on the screen, enter the IP address, subnet mask, gateway, and DNS, and press **Save**.

Step5 Press **Back** six times to return to the standby screen.

Step6 Check the status. If the screen shows **Static** and the IP address and gateway just set, the Static mode takes effect.

DHCP mode

Step1 Press **Menu -> Settings -> Advanced Settings** and enter the passwords. Choose **Network -> WAN settings -> Connection Mode**. Then, choose DHCP through navigation keys and press **Save**.

Step2 Press **Back** six times to return to the standby screen.

Step3 Check the status. If the screen displays **DHCP** in the first line, and the IP address and gateway just set in the second line, the DHCP mode takes effect.

3 Phone Function

3.1 Basic Function

3.1.1 Making a Call

Call Device

You can make a phone call via the following modes:

- If you pick up the handset, the icon will display in the upper left corner of the screen.
- If you press the **Speaker** key, the icon will display in the upper left corner of the screen.
- If you press the **Headset** key in the scenario where a headset is connected, the icon will display in the upper left corner of the screen.

You can dial the number first, and choose a call mode as needed.

Call Mode

If there is more than one account, you can press an available **line** key to make a phone call. Then dial the number you want to call.

- Dial the number and press the **Send** key.
- Press the **History** key, and use navigation keys to choose the number to dial (press the navigation key or to choose the missed calls, incoming calls, and outgoing calls.
- Press the REDIAL/SEND key to enter the outgoing call list.
- Press the programmable keys which are set as different speed dial keys for dial-up.

3.1.2 Answering a Call

Methods for answering an incoming call are as follows：

- If there are no other calls on the line, you can pick up the handset or press the **Speaker** or **Answer** key to answer the call If a headset is connected, press the **Headset** key.
- If there are other calls on the line, press the navigation key or to answer the call. During the conversation, you can also use the navigation key or to open the incoming call page, and press the corresponding key or pick up the handset to change the Headset, Handset or Speaker mode as needed.

3.1.3 DND

Press the **DND** key to activate the DND mode. In DND mode, the icon will display in the middle of the screen and all the incoming calls will be rejected. The rejected incoming calls will be prompted in the incoming call page, and you can check such information in the **Call History**. If the DND mode is already enabled, you can press the **DND** key again to quit the DND mode.

3.1.4 Call Forward

This function forwards an incoming call to another phone. Once this function is enabled, the icon displays in the upper right corner of the screen.

The status of the call forwarding includes:

- **Off**: Call forward is disabled by default.
- **Always**: Incoming calls are immediately forwarded.
- **Busy**: Incoming calls are immediately forwarded when the phone is busy.
- **No Answer**: Incoming calls are forwarded when the phone is not answered after a specific period.

Do as follows to configure the call forward function through a phone:

- Press **Menu** -> **Features** -> **Enter** -> **Call Forwarding** -> **Enter**.
- Choose any of the four options: **Disabled**, **Always**, **Busy**, and **No Answer.**
- If you choose any of the option (except for **Disabled**), enter the phone number incoming calls will be forwarded to and press **Save** to save the setting.

3.1.5 Call Hold

Press the **HOLD** key to put an active call on hold.

- If there is only one call on hold, press **HOLD** to retrieve the call.
- If there are more than one call on hold, press the corresponding **line** key, and the navigation key or to select a call, and then press **HOLD** to retrieve the call.

3.1.6 Call Waiting

- Press **Menu** -> **Features** -> **Enter** -> **Call Waiting** -> **Enter**.
- Use the navigation key or to enable or disable the call waiting function.
- Press **Save** to save the setting.

3.1.7 Mute

If you press **Mute** during the conversation, the icon will display in the LCD screen. This function prevents the caller from hearing you, but you can still hear the caller. You can press **Mute** again to resume the normal conversation.

3.1.8 Call Transfer

Blind Transfer

During conversion, you can press the **Transf** key, enter the number to transfer to, and press "**#**" to transfer the current call to a third party. After the transfer, you are disconnected with the caller. Note that you cannot select the SIP line when the call transfer is enabled.

Attended Transfer

To make an attended transfer, press the **Transf** key, enter the number to transfer to, and press **SEND** during the conversion. After the third party answers, press **Transf** to complete the operation. Note that to enable this function, you need to enable call waiting and call transfer first.

Alert Transfer

During the conversation, press the key **Transf**, enter the number to transfer to, and press **SEND** (the connection is not established yet). As soon as the call is answered, press **Transf** to complete the operation. Note that to enable this function, you need to enable call waiting and call transfer first.

3.1.9 Three-Way Conference Call

- Press the **Conf** key during the conversation.
- The first call is placed on hold. Enter the conference ID to join after hearing a dial tone, and press **SEND**.
- When the call is answered, press **Conf** and add the first call to the conference.
- If you want to leave the conference, press the **Split** key.

Note: This function operates normally only when your VoIP service provider supports RFC3515. To enable this function, you need to enable the call waiting function.

3.1.10 Multiple-Way Call

If theNRP1012/P is in the hook/handsfree/headset mode, press **HOLD** set the delay period (which is displayed as “--” on the screen). Each–“-“represents two seconds. For example, if you enter "123-45”, the phone will send out “45” two seconds after “123”; if you enter “123---45”, the phone sends out “45” six seconds after “123”.

Note: The function key must be configured as a HOLD.

3.1.11 Multi-line

By default, you can register and concurrently use a maximum of 6 SIP accounts on the NRP1012/P. Lines 1-4 corresponding to the NRP1012/P are SIP line, enabling you to select an SIP account to make a call. If the SIP account corresponding to a certain line key fails to be registered, the NRP1012/P prompts you the registration failure.

The NRP1012/P allows enterprise users to answer multi-way calls and make multi-line call. You can answer a maximum of 10 incoming calls, with 9 calls being in held and only one call being answered for the moment. In addition, you use the navigation key or to select a call to retrieve. After pressing **HOLD**, the selected call is retrieved and the current call is held automatically. You can define the four line keys as multi-line keys on the Web page, with each line key corresponding to a call. The line key lights on during the conversation and blinks after the call is held.

3.2 Advanced Function

3.2.1 Call Pickup

Call pickup is implemented by simulating the pickup function of private branch exchange (PBX). That is, when A calls B, but B does not answer for the moment, C can thus pick up the phone and answer the call after inputting the specified prefix code and B's number.

The following chart shows how to configure an appointed prefix in dial peer to have call pick up function.

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
*1*T	0.0.0.0	5060	SIP	rep:pickup	no suffix	3

1 represents a specified prefix code. C can answer A's call by inputting *1* and B's phone number. Users can set prefix in random, in the case of no affecting current dialing rules.

3.2.2 Join Call

If there is a conference call and A wants to join it, A needs to input a specific prefix code and the conference number.

The following chart shows how to configure an appointed prefix in dial peer to have join call function.

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
*2*T	0.0.0.0	5060	SIP	rep:joincall	no suffix	3

2 means an appointed prefix code. After completing the above configuration, A can dial *2* plus B or C number to join B and C's call. Users can set prefix in random, in the case of no affecting current dialing rules.

3.2.3 Redial/Unredial

In the scenario where A calls B, but B is busy, A is prompted to hang up the phone. To ensure that B can answer A's call as soon as B is idle, A can enable the redial function.

The redial function requires A to dial a specific prefix code before B's number. If B is idle, B can answer A's call by picking up the phone; if B is busy and A hangs up the phone as prompted, A will subscribe B's call status every 60 seconds. As soon as B becomes available, A is prompted to hook off and calls B automatically. B can answer A's call after picking up the phone. If B becomes available, but A is occupied and unwilling to contact B for the moment, A can disable the redial function by dialing the prefix code before B's number.

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
*3*T	0.0.0.0	5060	SIP	rep:redial	no suffix	3
*4*T	0.0.0.0	5060	SIP	rep:unredial	no suffix	3

3 represents a specified prefix code. To enable the redial function, A needs to dial *3* and B's phone number in sequence.

4 represents a specified prefix code. To disable the redial function, A needs to dial *4* and B's phone number in sequence.

3 or *4* can be selected randomly, as long as the current dialing rule is not affected.

3.2.4 Click to Dial

When browsing in an appointed Web page, user A can click to call user B via a link (this link to user B), then user A's phone will ring, after A hooks off, the phone will dial to B.

Note: External software supporting the click to dial function is required.

3.2.5 Call Back

The function allows you to call back the last number that called you.

3.2.6 Auto Answer

If an incoming call is not answered until the answer time expires, the phone enabled with this function will answer the call automatically.

3.2.7 Hotline

You can set a hotline number for each SIP line in the dialer interface. After Warm Line Time, the phone will call out the hotline number automatically.

3.2.8 Application

SMS

- Press **Menu** -> **Applications** -> **Enter** -> **SMS** -> **Enter**.
- Use the navigation keys to select and read messages in the Inbox/Outbox.
- After checking a message, you can press **Reply** to reply it. Use the 2aB softkey to select an input method. Then, compose a reply message, press **OK**, select a line for message sending, and press **Send**.
- To compose a new message, press **New** and select an input method using the 2aB softkey. Compose the message, press **OK**, select a line for message sending, and press **Send**.
- To delete a message, press **Delete**. There are three options for you to choose: Yes, All, and No. Choose any of them as needed.

Memo

Memos remind you of important things.

Press **Menu** -> **Application** -> **Memo** -> **Enter** -> **Add**.

There are some options to configure: **Mode**, **Date**, **Time**, **Text**, and **Ring Tone**. After the configuration, press **Save**.

Voice Mail

- Press **Menu** **->** **Application** **->** **Voice Mail** **->** **Enter**.
- Select the SIP line to be configured with a voice mail through the navigation key or , and press **Edit**. Then, use the navigation key or to enable voice mail, and input the number of the voice mail…
- Press **Save** to save the configuration.
- Press **Voicemail** to check the voice mail. Then, press **Dial** to call the voice mail number. After the connection is established, you need to enter the password to check voice messages.

3.2.9 Ping

- Press **Menu** **->** **Ping** **->** **Enter**.
- Enter the IP address to be pinged and then press **Start**. If the IP address is incorrect, press **Delete** to delete the IP address and change the input method.
- If the screen displays **OK** several seconds after the IP address is input,, the ping is successful; if the screen displays **Failed**, the ping fails.

3.2.10 Programmable Key Configuration

The phone has four line keys and eight programmable keys. You can configure different functions for each key. Functions that can be configured are listed as follows. The default configuration for each key is N/A, indicating that the key is not configured with any function.

Setting the key as Memory Key

Press **Menu** **->** **Settings** **->** **Basic Settings** **->** **Enter** **->** **Keyboard** **->** **DSS Key Settings**. You can choose to configure the line key or memory key using the navigation key or . If you choose to configure the memory key, set the phone number, line name, call mode, speed dial, interphone, BLF, Presence, MWI, and call park.

- **Speed dial**: You can configure the key as a speed dial key. Input the number wanted and save the configuration. After that, press this key to call the preset number in the speed dial mode.
- Interphone: You can configure the key with the interphone function useful. This function allows an operator or a secretary to quickly answer the call and is thus widely use in the office environment.
- BLF: BLF is short for “Busy lamp field”, and is used to inform the user of the status of the object subscribed.. This function can cooperate with the server to pick up the phone call. BLF can help

you monitor the status (idle, ringing, or busy) of other SIP account. For information about the LED status, see 1.3 LED Indicator.

Note: In the Web interface, you can also set the pickup number to activate the pickup function. For example, if you set the BLF number to 212, and the pickup number to 189, when there is an incoming call to 212, you can press the **BLF** key to pick up the incoming call of 212. Note that to enable the BLF function, you need to input *8 in the pickup number of the Web page.

- Presence: In addition to subscribe the object status, this function can check whether object online.

  Note: You can subscribe either the BLF or presence information of the same number.
- MWI: When the key is configured with the MWI function, you can press this key to quickly check unread messages.
- Call park: When the call park is enabled, you need to set a server number. If you feel inconvenient to answer a call in a place, you can press the key to transfer the call to the server number. Then, you can change a place and input the server number to recover the call.

Setting the key as a Line key

You can set these keys to line keys. Pressing these key will enter the dialer interface.

Setting the key as a Key Event

You can set the function type of these keys to Key Event, with multiple sub-type options available.

Choose any of the following functions for the key.

- None
- MWI
- DND
- Hold
- Transfer
- Phone Book
- Redial
- Pick up
- Join
- Auto-redial-on
- Auto-redial-off
- Call Forward
- History
- Flash
- Memo
- Headset
- Release

- Lock
- SMS
- Call Back
- Power Light
- Hide DTMF
- Prefix
- Hot Desking-

Setting the key as a DTMF key

You can configure the key as a DTMF key. This key function allows you to dial a preset number or enter the number during the conversation.

Setting the key as a URL key

This function requires you to configure a XML Phonebook address. After that, you can press the key to access the corresponding remote phonebook.

Setting the key as a BLF List Key

BLF List Key cooperates with the Broadsoft server. The traditional BLF is that every number will need to be subscribed, so if the numbers that subscribed is so many that it will cause to obstruction. However, BLF List Key will put the numbers that needed to be subscribed in a group, and the phone use the URL of the group to subscribe and analyze the specific information of each number such as number, name, state and so on according to the notifications from the server. Then set the idle Memory key as BLF List Key, later if the state of an object changes, the corresponding LED will change.

3.3 Other Functions

3.3.1 Auto Handdown

- Press **Menu** -> **Features** -> **Enter** -> **Auto Handdown** -> **Enter**.
- Enable the auto handdown function using the navigation keys and . Set the time to play the alert tone (in seconds) and press **Save**.
- When the call ends, the phone will back to the standby interface after the handdown time expires.

3.3.2 Ban Anonymous Call

- Press **Menu** -> **Features** -> **Enter** -> **Ban Anonymous Call** -> **Enter**.
- Select the SIP line to be enabled to the Ban Anonymous Call function and press **Enter**. Choose **Enabled** or **Disabled** with navigation keys, and press **Save**.
- If you choose **Enabled**, the phone will reject anonymous calls. If you choose **Disabled**, the phone

can receive anonymous calls.

3.3.3 Ban Outgoing

- Press **Menu** -> **Features** -> **Ban Outgoing** -> **Enter**.

  Enable or disable this function using the navigation keys and . If you choose **Enabled**, you cannot call any number.

3.3.4 Dial Plan

- Press **Menu** -> **Features** -> **Enter** -> **Dial Plan** -> **Enter**.
- Available plan include: Pressing # to send, Timeout to send, Timeout, Fixed Length Number, Press # to Do BXFER, BXFER On Onhook, AXFER On Onhook. You can enable or disable each dial plan using the navigation keys and .

3.3.5 Dial Peer

- Press **Menu** -> **Features** -> **Enter** -> **Dial Peer** -> **Enter**.
- Press **Add** and input the number and destination in the page displayed. For example: the number is 1 and the destination is 1234. Then, press **Save**.
- Input **1#** in the dial interface, you call the number 1234 in speed dial mode.

3.3.6 Auto Redial

- Press **Menu** -> **Features** -> **Enter** -> **Auto Redial** -> **Enter**.
- Enable the auto redial function using the navigation keys and . If you choose **Enabled**, set values for **Interval** and **Times**, and then press **Save**.
- After auto redial is enabled, if the callee does not answer your call for the moment, you are prompted whether use the auto redial function. If you press **OK**, the phone will call out the callee according the Interval and Times you set.

3.3.7 Call Complete

- Press **Menu** -> **Features** -> **Enter** -> **Call Complete** -> **Enter**.
- Enable this function using the navigation keys and .
- If the callee does not answer your call for the moment, the phone prompts you "**Call Complete Waiting number**?" Press **OK**. When in the callee becomes idle, the phone prompts you **"Call Complete Waiting number**?" (in 30s). If you press **OK**, the phone will call out the number automatically.

3.3.8 Headset Ringing

- Press **Menu** -> **Features** -> **Enter** ->**Headset Ringing** -> **Enter**.

- Enable this function using the navigation keys and and insert a headset. After that, the headset rings when the phone has an incoming call.

3.3.9 Power Light

- Press **Menu** -> **Features** -> **Enter** -> **Power Light** -> **Enter**.
- Switch on or off the power light using the navigation keys and .

3.3.10 Hide DTMF

- Press **Menu** -> **Features** -> **Enter** -> **Hide DTMF** -> **Enter**.
- You can choose any of the following options using the navigation keys: **Disabled**, **All**, **Delay**, **Last Show**. If you input **DTMF** while in conversing, the DTMF will be displayed as you have set.
-

3.3.11 Password Dial

- Press **Menu** -> **Features** -> **Enter** -> **Password Dial** -> **Enter**.
- Enable this function using the navigation key. After that, you can also set **Prefix** and **Length** of the password.
- When you dial a number beginning with the preset prefix, several figures of the number will be hidden (the number figures to be hidden is determined by the **Length** value you set).

3.3.12 Pre Dial

- Press **Menu** -> **Features** -> **Pre Dial** -> **Enter**.
- Enable this function using the navigation keys.

4 Phone Settings

4.1 Basic Settings

4.1.1 Keyboard

- Press **Menu** -> **Settings** -> **Enter** -> **Basic Settings** -> **Enter** -> **Keyboard** -> **Enter**.
- You can set the **DSS key settings**, **Programmable keys**, **Desktop Long Pressed**, **Soft Key** as needed. Press each key to enter the corresponding setting interface, then use the navigation keys to select the function for each key.
- Press **OK** to save the setting.

4.1.2 Screen

- Press **Menu** -> **Settings** -> **Enter** -> **Basic Settings** -> **Enter** -> **Screen** -> **Enter**.
- You can set **Contrast**, **Contrast Calibration** and **Backlight** as needed. Press the key to open the setting interface and use the navigation keys to set functions. After that, press **OK** to save the setting.

4.1.3 Ring Tone

- Press **Menu** -> **Settings** -> **Enter** -> **Basic Settings** -> **Enter** -> **Ring Tone** -> **Enter**.
- You can set the volume and tone of the ring. Press the key to open the setting interface and set the volume and tone using the navigation keys. Then, press **OK** to save the setting. By default, the system has nine ring tones available. There are three customized ring tones which can be set in the Web page.

4.1.4 Voice Volume

- Press **Menu** -> **Settings** -> **Enter** -> **Basic Setting** -> **Enter** -> **Voice Volume** -> **Enter**.
- Turn the voice volume down or up using the navigation keys and press **Save**.

4.1.5 Time & Date

- Press **Menu** -> **Settings** -> **Enter** -> **Basic Settings** -> **Enter** -> **Time & Date** -> **Enter**.
- You can set the time and date in **Auto** or **Manual** mode. Choose either of the modes using the navigation keys and press **Save**.

4.1.6 Greetings

- Press **Menu** -> **Settings** -> **Enter** -> **Basic Settings** -> **Enter** -> **Greetings** -> **Enter**.
- Compose the greeting words and press **Save**. After that, the greetings will be displayed on the screen after the phone startup.

4.1.7 Language

- Press **Menu -> Settings -> Enter -> Basic Settings -> Enter -> Language -> Enter**.
- Choose the language using the navigation keys. The default two languages are English and Chinese.

4.2 Advanced Settings

4.2.1 Accounts

Press **Menu -> Enter -> Advanced settings**, and input the password to enter the interface. The default password is 123 and can be changed the password through the Web page. Then, choose **Account** and press **Enter** to carry out the account setting, including the SIP setting.

4.2.2 Network

Press **Menu -> Enter -> Advanced settings**, and input the password to enter the interface. Then choose **Network** and press **Enter** to carry out network settings. For details, see 2.2.1 Network Settings.

4.2.3 Security

Press **Menu -> Enter -> Advanced settings**, and input the password to enter the interface. Then, choose **Security** and press **Enter** to configure **Menu Password**, **Key lock Password**, **Key lock Status** and **ban Outgoing**.

4.2.4 Maintenance

Press **Menu -> Enter -> Advanced settings**, and input the password to enter the interface. Then, choose **Maintenance** and press **Enter** to configure **Auto Provision**, **Backup**, and **Upgrade**.

4.2.5 Factory Reset

Press **Menu -> Enter -> Advanced settings**, and input the password to enter the interface. Then, choose **Factory Reset** and press **Enter**. System prompts a dialog box for you to determine whether to restore the factory settings, and you can choose **Yes** or **No** as needed.

5 Web Page Setting

5.1 Configuration Introduction

5.1.1 Configuration Methods

Users with different using habits can configure the NRP1012/P by means of:

- Phone keypad
- Web browser (recommended)
- Telnet with CLI command

5.1.2 Password Configuration

The browser and command lines of the phone can be set in either of the two modes: guest mode and administrator mode. In the administrator mode, users can both view and modify all options as needed; whereas in the guest mode, users can only view the SIP (1-2) and IAX 2 options, and the IP address and port of the server. The username and password for entering different modes are different.

Guest mode:

- Username: **guest**
- Password: **guest**

Administrator mode:

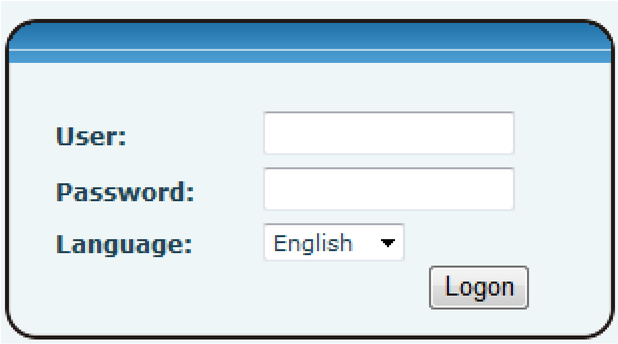
- Username: **admin**
- Password: **admin**

The default password of phone screen menu is 123.

5.2 Setting the NRP1012/P through a Web Browser

When both the NRP1012/P and your PC are connected to the network, enter the IP address of the WAN port of the NRP1012/P as the address bar. The IP address of the NRP1012/P can be obtained by pressing the navigation key (such as http://xxx.xxx.xxx.xxx/ or http://xxx.xxx.xxx.xxx:xxxx/).

Figure 5-1 Login interface



After configuring the IP phone, you need click **Save** in the **config** tag of **Maintenance** to save your modification. Otherwise, the phone restores the configuration before modification after reboot.

5.3 Basic Configuration

5.3.1 Status

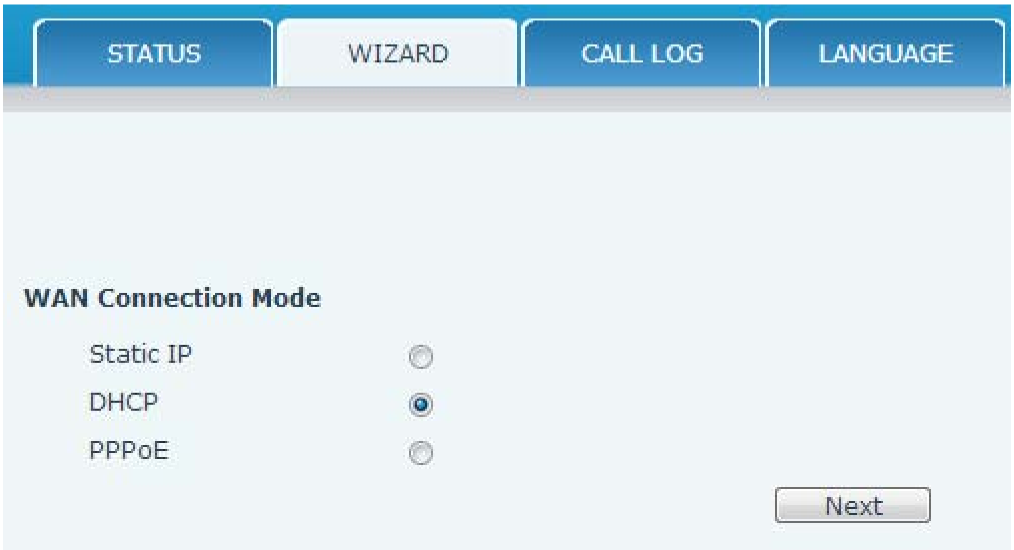
Figure 5-2 Status interface

Table 5-1 Status parameters

Name	Description
Network	Contains configuration information about the WAN and LAN ports, including the port connection mode (static, DHCP, or PPPoE), MAC address, IP address, IP gateway, and status (Enabled or Disabled) of the DHCP service and bridge mode.
Accounts	Contains information about the phone numbers corresponding to SIP lines 1-6 and IAX2. Information at the bottom of the page is the version number.

5.3.2 Wizard

Figure 5-3 Wizard interface



The connection mode of the phone varies with the network condition. You need to select a proper mode according to the real world situation. The NRP1012/P provides three network connection modes:

- **Static IP**: If your ISP server provides you a static IP address, you can choose this mode, and fill in the corresponding information including the static IP address, subnet mask, network gateway, and DNS. The preceding information can be obtained from the ISP service provide or network administrator.
- **DHCP**: In this mode, the system will obtain the network information from the DHCP server automatically, and no action is required.
- **PPPoE**: In this mode, you need to input your ADSL account and password.

To set your network quickly, see 2.2.1 Network Settings

Static IP

Choose **Static IP**, and click **Next** to configure and browse the network address and SIP parameter(by default, SIP line 1 is used). Click **Back** to return to the previous page.

Figure 5-4 Static IP settings interface

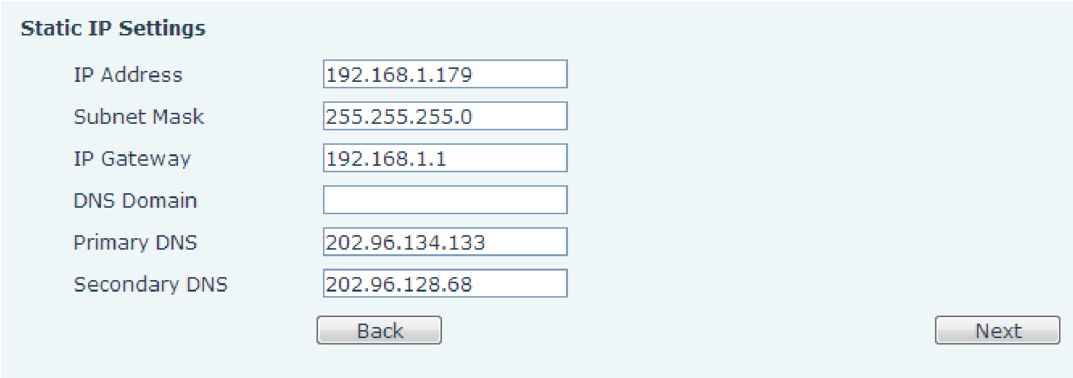


Table 5-2 Static IP settings parameters

Name	Description
IP Address	Inputs the IP address distributed to you.
Subnet Mask	Inputs the subnet mask distributed to you.
IP Gateway	Inputs the gateway address distributed to you.
DNS Domain	Sets the suffix of a DNS domain name. When a domain address that a user input cannot be parsed by DNS, the phone will automatically add this parameter to the end of the domain address and parse it again.
Primary DNS	Inputs the IP address of the primary DNS server.
Secondary DNS	Inputs the IP address of the secondary DNS server address.

Figure 5-5 Quick SIP settings interface

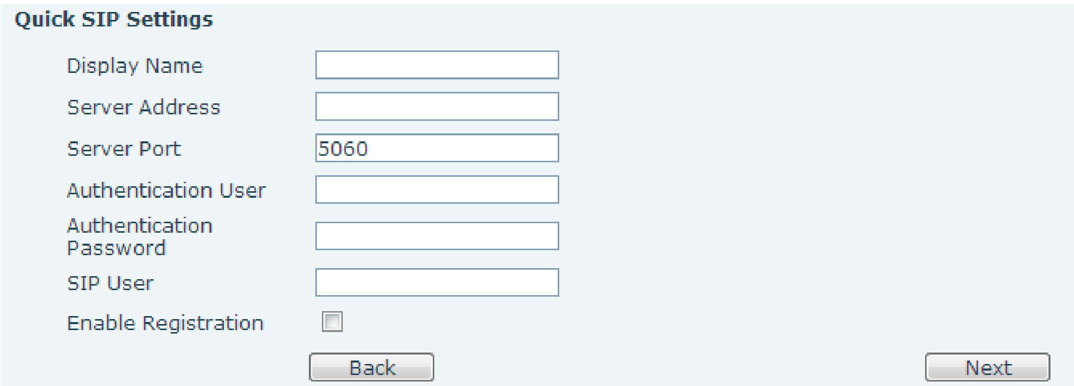
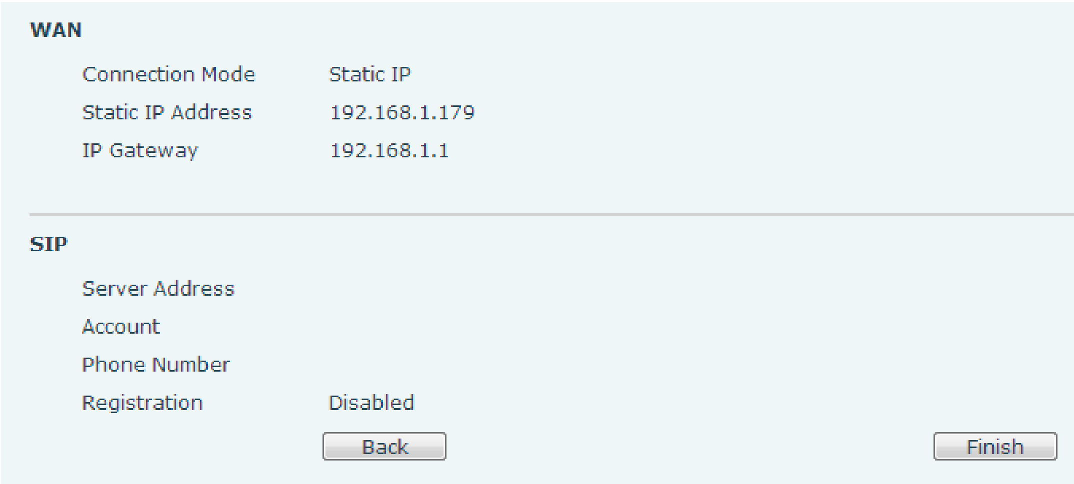


Table 5-3 Quick SIP settings parameters

Name	Explanation
Display Name	Sets the display name.
Server Address	Inputs the IP address of your SIP server (the address can be in the form of domain name).
Server Port	Inputs the signaling port of your SIP server.
Authentication User	Inputs the account name for SIP registration.

Name	Explanation
Authentication Password	Inputs the password for SIP registration.
SIP User	Inputs the phone number registered to the SIP server.
Enable Registration	Enable/disable registration.

Click **Next** to display detailed configuration information just set.



DHCP

Choose **DHCP** and click **Next** to configure and browse the account, password, SIP parameter (by default, SIP line 1 is used). Click **Back** to return to the previous page. The configuration details are identical with that of the Static IP mode.

PPPoE

Choose **PPPoE** and click **Next** to configure and browse the account, password, SIP parameter (by default, SIP line 1 is used). Click **Back** to return to the previous page. The configuration details are identical with that of the Static IP mode.

Figure 5-6 PPPoE settings interface

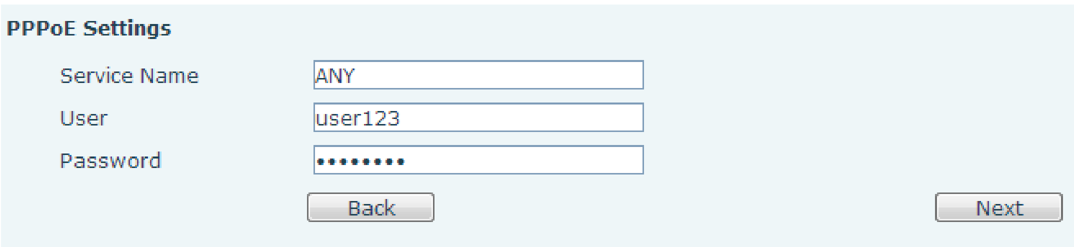


Table 5-4 PPPoE settings parameters

Name	Description
Server Name	Inputs the name of a PPPoE server. If the PPPOE service provider has no special requirements , you can use the default value.
User	Inputs your ADSL account.
Password	Inputs your ADSL password.

Note

After the configuration, click **Finish** and the IP phone will save the current setting and reboot automatically. After reboot, you can dial by the SIP account.

5.3.3 Call Log

You can check all the call information in this interface.

Figure 5-7 Call information interface

Table 5-5 Parameters

Name	Description
Start Time	Displays the start time of a call record.
Duration	Displays the conversation duration of a call record.
Dialed Calls	Displays the account/protocol/line of outgoing call record.

5.3.4 Language

Figure 5-8 Language setting interface

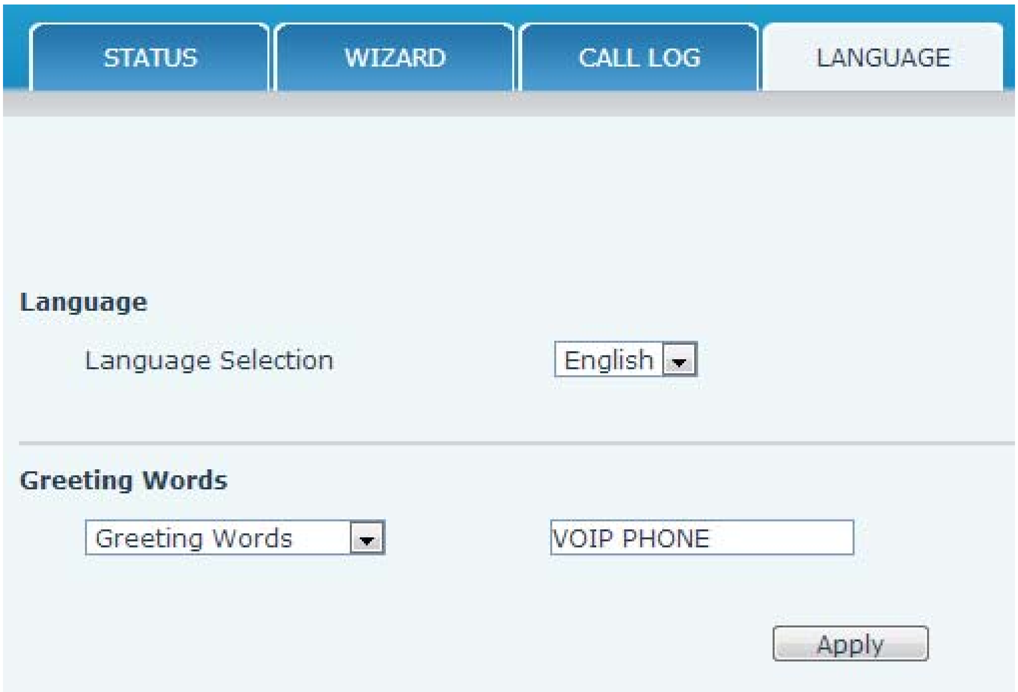


Table 5-6 Language parameters setting

Name	Description
Language	Sets the language of the phone. The default language is English.
Greeting Words	Sets the greeting words to be displayed on LCD screen.

Note

The maximal length of a greeting message is twelve English characters or five Chinese characters.

5.4 Network

5.4.1 WAN

Figure 5-9 WAN setting interface

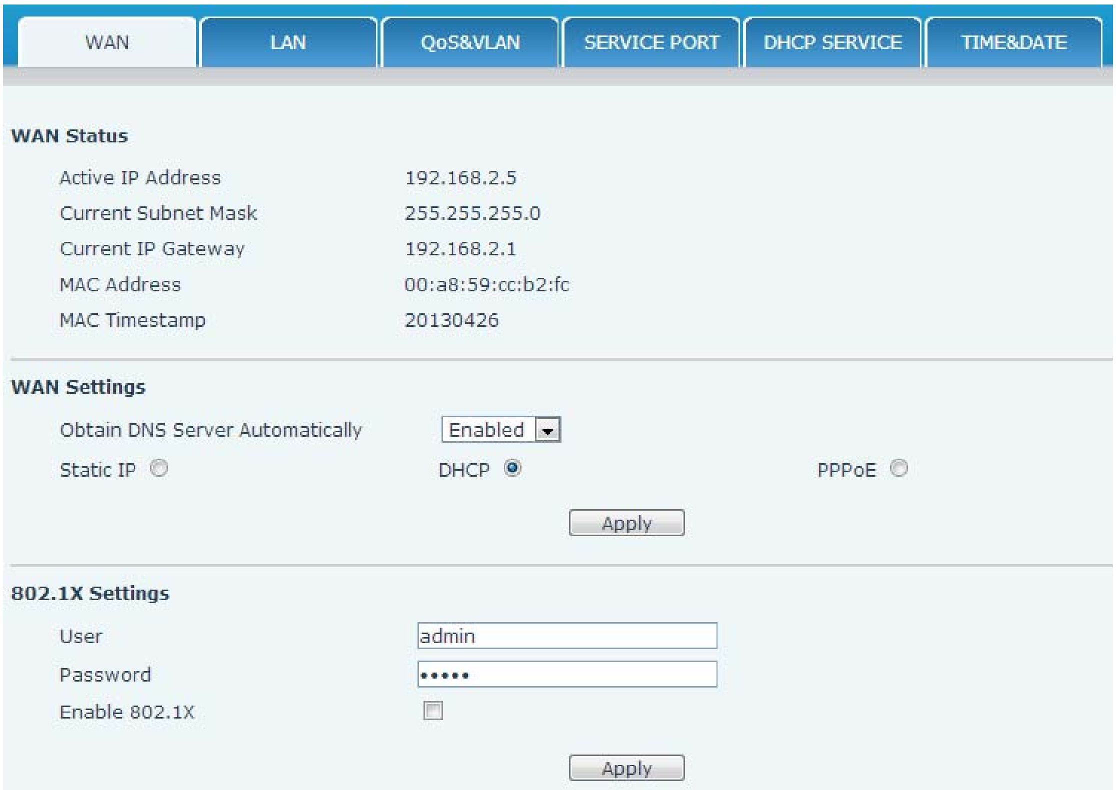


Table 5-7 WAN parameters setting

Name	Description
WAN Status	
Active IP Address	Specifies the current IP address of the phone.
Current Subnet Mask	Specifies the subnet mask of the phone.
Current IP Gateway	Specifies the current gateway IP address of the phone.
MAC Address	Specifies the MAC address of the phone.
MAC Timestamp	Specifies the time when the MAC address is obtained.
WAN Settings	
Obtain DNS Server Automatically	Indicates whether to obtain the DNS address in DHCP mode. If the value is set to Disabled, you will obtain the DNS address in static IP mode. The default value is **Enabled**.
Static IP	Indicates whether to enable the static IP mode. If you can obtain a fixed IP address from the ISP service provider, you can enable this mode and fill up information such as the static IP address, subnet mask, gateway, and DNS domain name. The preceding information can be obtained from the ISP service provide or network administrator.

Name	Description
DHCP	Indicates whether to enable the DHCP mode. If you choose this mode, the phone can obtain the network-related information automatically from the DHCP server automatically, and no action is required.
PPPoE	Indicates whether to enable PPPoE mode. If you choose this mode, you need to input the ADSL account and password.
802.1x Settings	
User	
Password	
Enable 802.1x	

Note

- After parameter setting, click **Apply** to validate these parameters.
- If you change the IP address, you need to input a new IP address in the address bar so that you can log in to the phone.
- If you use an IP address obtained during the system startup, and the IP address of the DHCP server is identical with the LAN IP address, the system increases the last octet of the LAN IP address by 1 and modifies the LAN IP address segment assigned by the DHCP server. If the system attempts to accesses DHCP services after startup, and the IP address assigned by the DHCP server is identical with the LAN IP address, the WAN port cannot obtain an IP address and therefore the phone cannot access the network through the WAN port.

5.4.2 LAN

Figure 5-10 LAN setting interface

Table 5-8 LAN parameters setting

Name	Description
IP Address	Specifies a static LAN IP address.
Subnet Mask	Specifies a LAN subnet mask.

Name	Description
DHCP Service	Enables the DHCP server of the LAN port. If you modify the LAN IP address, the phone will automatically modify contents in the DHCP Lease Table according to the IP address and subnet mask, and then save the modification. To validate the DHCP service settings, you need to reboot the phone.
NAT	Enables NAT.
Port Mirror	Enables the Port Mirror function. This function takes effect only in bridge mode. Port mirror is used to copy the data stream from the WAN port to the LAN port of the phone.
Enable Bridge Mode	Enables the bridge (transparent) mode. With this function, the phone will no longer set an IP address for the LAN physical port, and LAN and WAN port are connected to the same network. After the setting, click **Apply**, and the phone will reboot.

Note

- The LAN configuration no long takes effect in bridge mode.
- The system reboots after the LAN IP address or bridge mode changes.

5.4.3 QoS & VLAN

The NRP1012/P supports 802.1Q/P and DiffServ configurations. The VLAN function can be divided into Voice VLAN and Data VLAN by VLAN IDs. If Data VLAN is configured, signaling, voice streams, and other data streams are tagged with different VLAN IDs to make the VLAN application more flexible.

Figure 5-11 QoS & VLAN setting interface

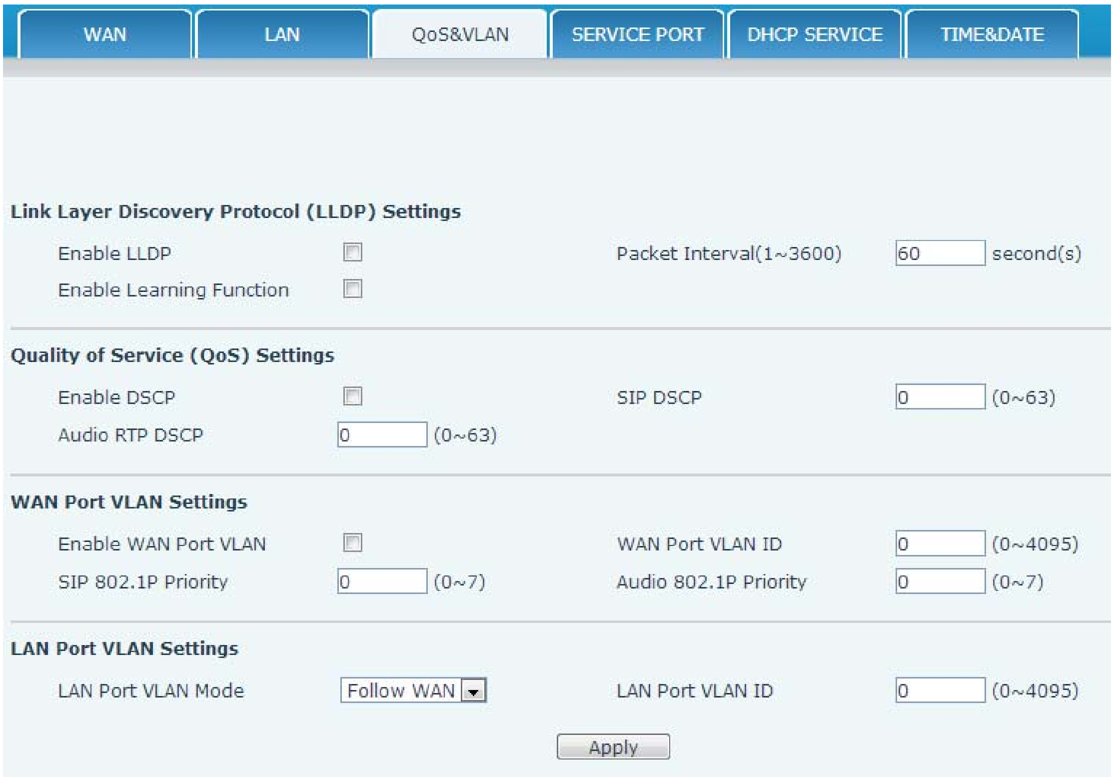


Table 5-9 QoS & VLAN parameters setting

Name	Description
Link Layer Discovery Protocol (LLDP) Settings	
Enable LLDP	Enables the LLDP function.
Package Interval(1-3600)	Specifies the interval at which LLDP packets are sent, in seconds. The default value is 60.
Enable Learning Function	Enables the LLDP Learn function. If enabled, the phone can automatically learn values of the QoS, VLAN ID, and 802.1p from the switch. If the data is different from that of the LLDP server, phone synchronizes its values with that of the switch.
Quality of Service(QOS) Settings	
Enable DSCP	Enables DSCP.
SIP DSCP	Sets a value for SIP DSCP.
Audio RTP DSCP	Sets a value for Audio RTP DSCP.
WAN Port VLAN Settings	
Enable WAN Port VLAN	Enables VLAN of the WAN port.
WAN Port VLAN ID	Sets a VLAN ID. The value ranges from 0- to 4095.
SIP 802.1p Priority	Sets a value for SIP 802.1P priority. The value ranges from 0 to 7.
Audio 802.1p Priority	Sets a value for the audio 802.1P priority. The value ranges from 0 to 7.
LAN Port VLAN Settings	
LAN Port VLAN Mode	Enables/Disables the Port VLAN function. If enabled, you need to input a VLAN ID different from that of the WAN port.
LAN Port VLAN ID	Sets a VLAN ID. The value ranges from 0 to 4095.

5.4.4 Service Port

You can set the Telnet/HTTP/RTP port in this page.

Figure 5-12 Service port setting interface

Table 5-10 Service port parameters setting

Name	Description
Service Port Settings	
Web Server Type	Sets a web service protocol. The value can be HTTP or HTTPS.
HTTP Port	Sets a Web browser port. The default port number is 80. To increase the system safety, you are recommended to change the port to a non-80 standard port, and save the configuration.
HTTPS Port	Before using HTTPS, you need to download and then install the HTTPS authentication certification on the phone. After that, choose HTTPS and configure the Web browser port. The default port number is 443. To increase the system safety, you are recommended to change the port to a non-443 standard port, and reboot the phone after saving the configuration..
Telnet Port	Sets a Telnet port. The default port number is 23.
RTP Port Range Start	Set the RTP Start port. The port number is allocated automatically.
RTP Port Quantity	Sets the maximum number of RTP ports that can be allocated to the phone The default number is 200.

Note

- You are recommended to change the Telnet/HTTP port number to a value greater than 1024. Because ports with their number smaller than 1024 are reserved for the system.
- If you set the HTTP port number to 0, HTTP services are disabled.

5.4.5 DHCP Server

Figure 5-13 DHCP server setting interface

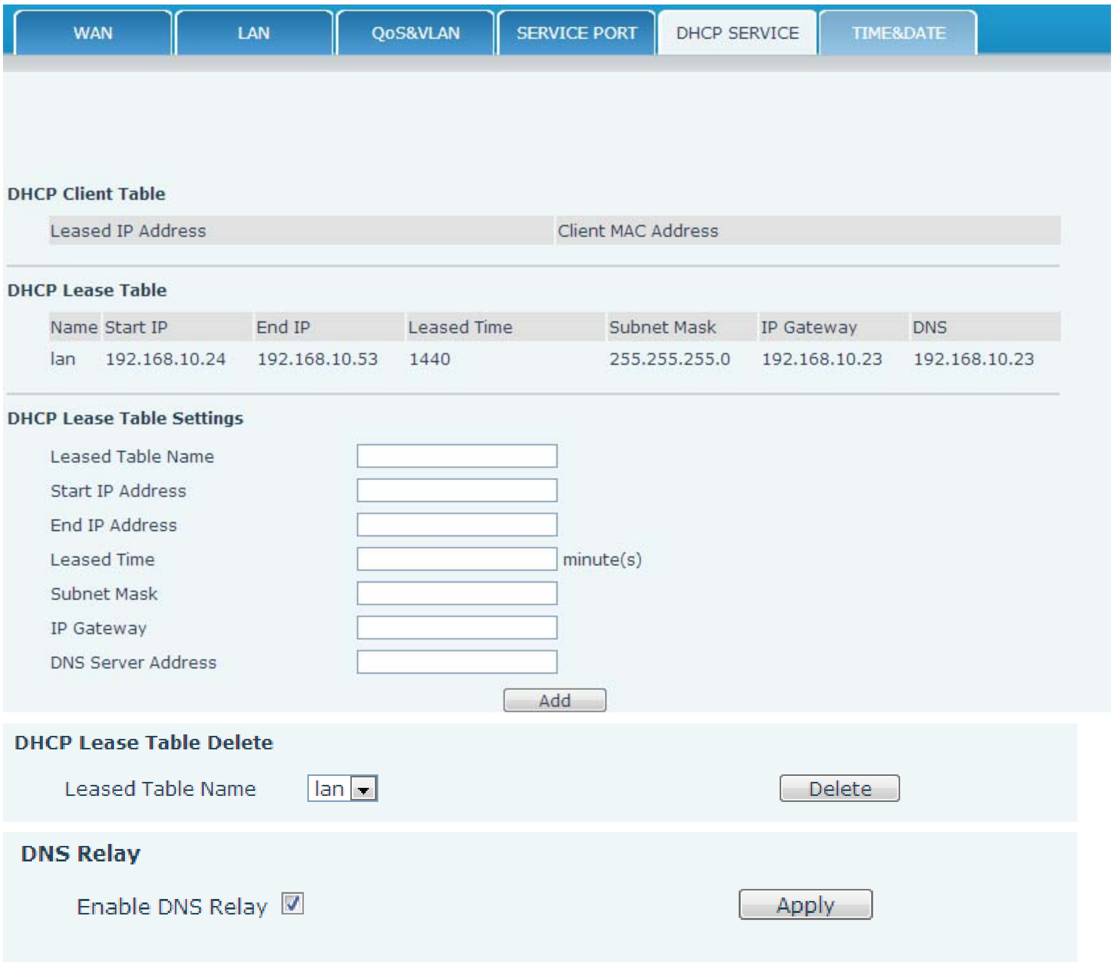


Table 5-11 DHCP server parameters setting

Name	Description
DHCP Client Table	Lists the IP address and MAC address allocated through DHCP. If the LAN port of the phone is connected with a device, this table shows the IP and MAC address of this device.
DHCP Lease Table	Shows the DHCP Lease Table, the unit of Lease time is Minute.
DHCP Lease Table Settings	
Lease Table Name	Specifies the name of a lease table.
Start IP Address	Specifies the start IP address of the lease table. From this address on, the LAN port searches for IP addresses not occupied and allocates them to devices applying for DHCP services.
End IP Address	Specifies the end IP address of the lease table. The number of IP addresses from the start IP address to the end IP address specifies the number of devices that can access DHCP services through the LAN port.
Leased Time	Sets the lease duration of an IP address of the lease table.

Name	Description
Subnet Mask	Sets the subnet mask of the lease table.
IP Gateway	Sets the default IP gateway of the lease table.
DNS Server Address	Sets the default IP address of the DNS server of the lease table. Click **Add** to add entries to the lease table.
DHCP Lease Table Delete	
Leased Table Name	Select name of lease table, click the Delete button will delete the selected lease table from DHCP lease table.
DNS Relay	
Enable DNS Relay	Select DNS Relay, the default is enabled. Click the Apply button to become effective.

Note

- The number of entries in the release table cannot exceed the number of Class C IP addresses. You are recommended to use the default release table without any modifications.
- If you need to modify the release table, reboot the HTTP server to validate the modification.

5.4.6 TIME&DATE

You can configure the zone according to your location and enable Simple Network Time Protocol (SNTP) to obtain the date and day-light saving time automatically. Alternatively, you can set the time manually.

Figure 5-14 TIME&DATE setting interface

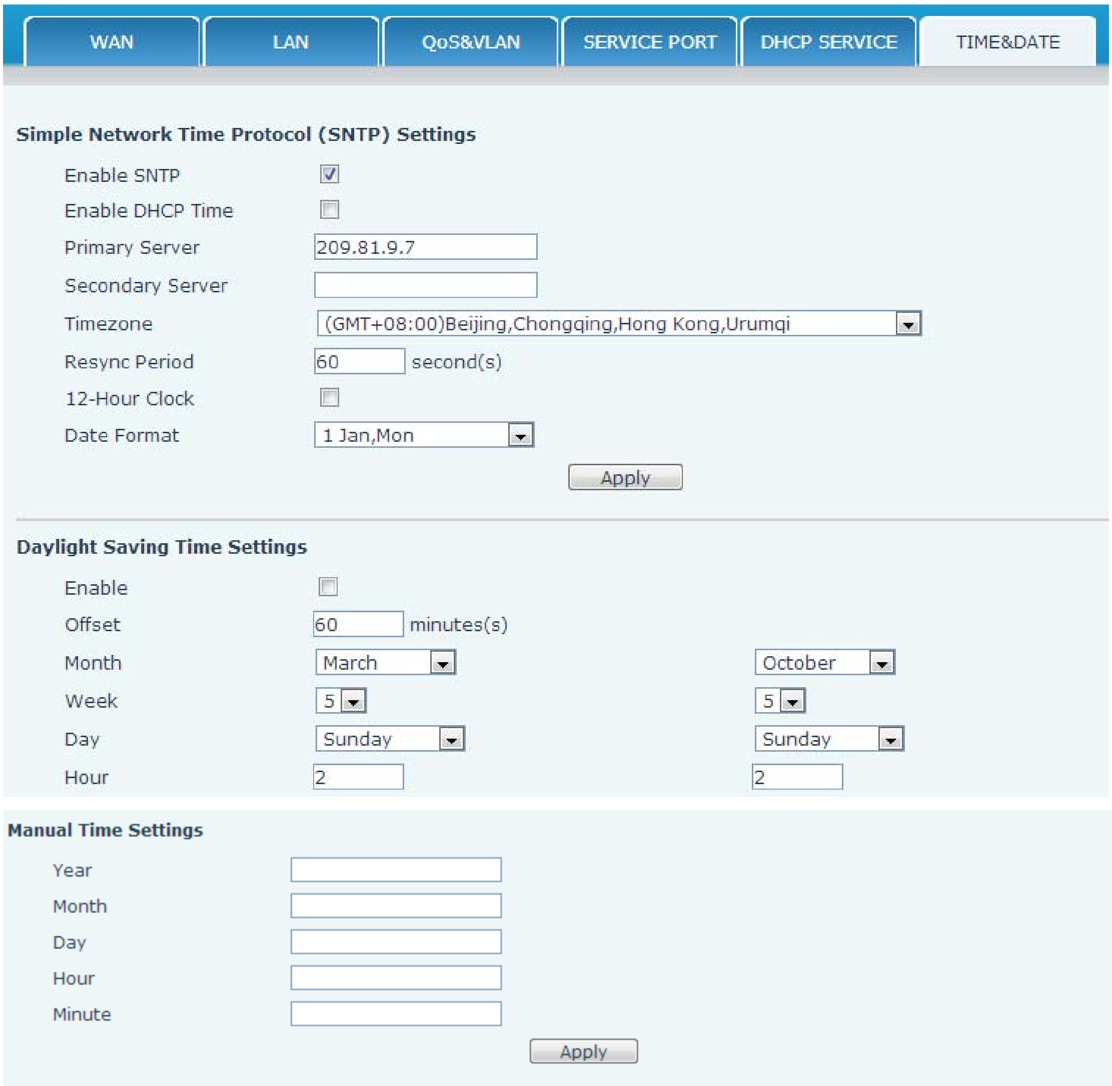


Table 5-12 TIME&DATE parameters setting

Name	Description
Simple Network Time Protocol (SNTP) Settings	
Enable SNTP	Enables SNTP.
Enable DHCP Time	Enables DHCP Time so that the phone will automatically synchronize the standard time.
Primary Server	Sets an SNTP primary server IP address.
Secondary Server	Sets an SNTP secondary server IP address.
Time Zone	Selects a time zone according to your location.
Resync Period	Sets the timeout, period. The default value is 60, in seconds.
12 -Hour Clock	Switches the time mode between 12 hours and 24 hours. The default mode is 24 hours.
Date format	Specifies the date format.
Daylight Saving Time Settings	
Enable	Enables the daylight saving time.

Name	Description
Offset	Sets the variety length.
Month	Sets the start and end month.
Week	Sets the start and end week.
Day	Sets the start and end day.
Hour	Sets the start and end hours.
Manual Time Settings	
Manual Time Settings	

Note

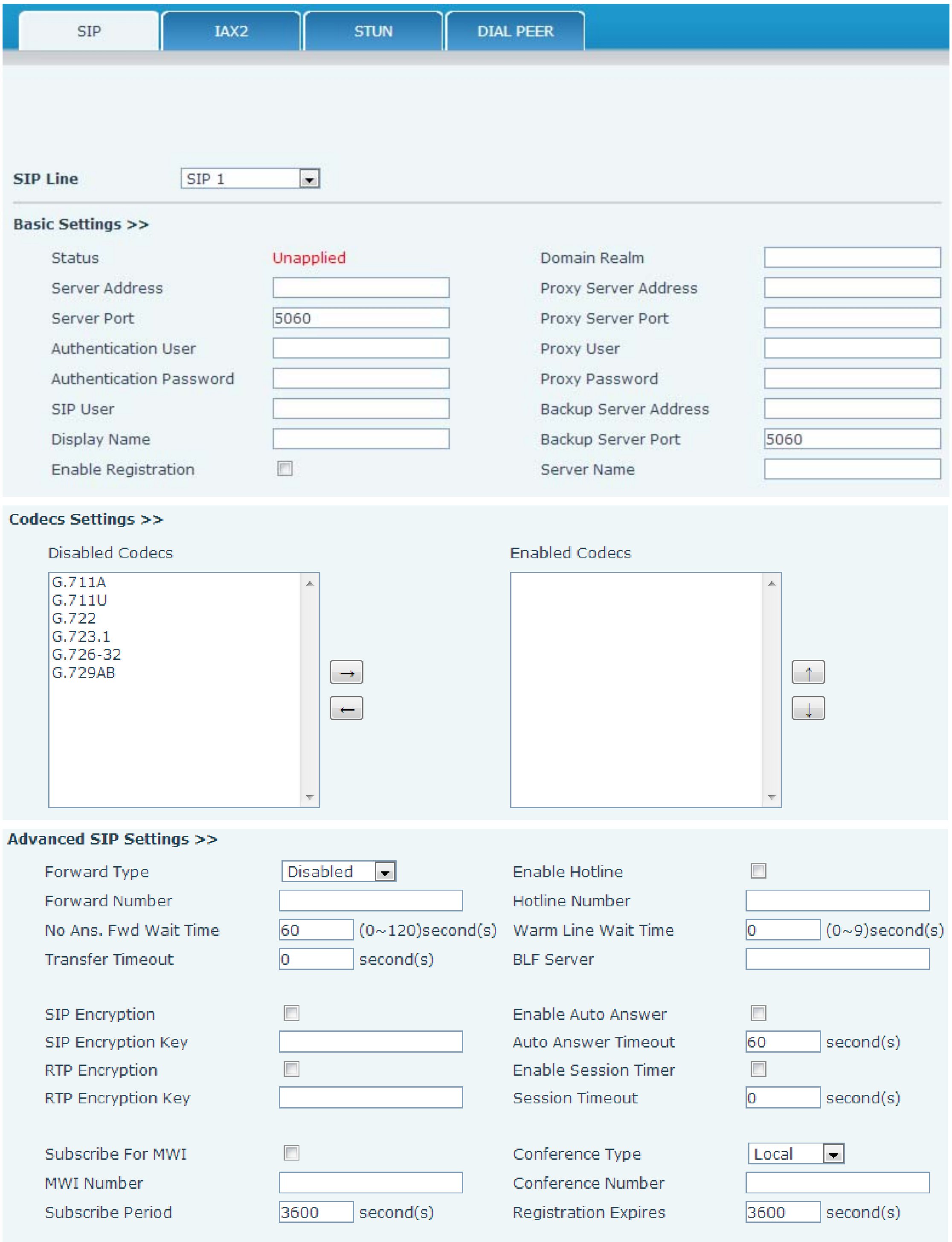
First of all, you need to disable the SNTP service, and above the date hours minutes each of which is required to complete and submit to make manual.

5.5 VoIP

5.5.1 SIP

Configure the SIP server in the following interface:

Figure 5-15 SIP setting interface



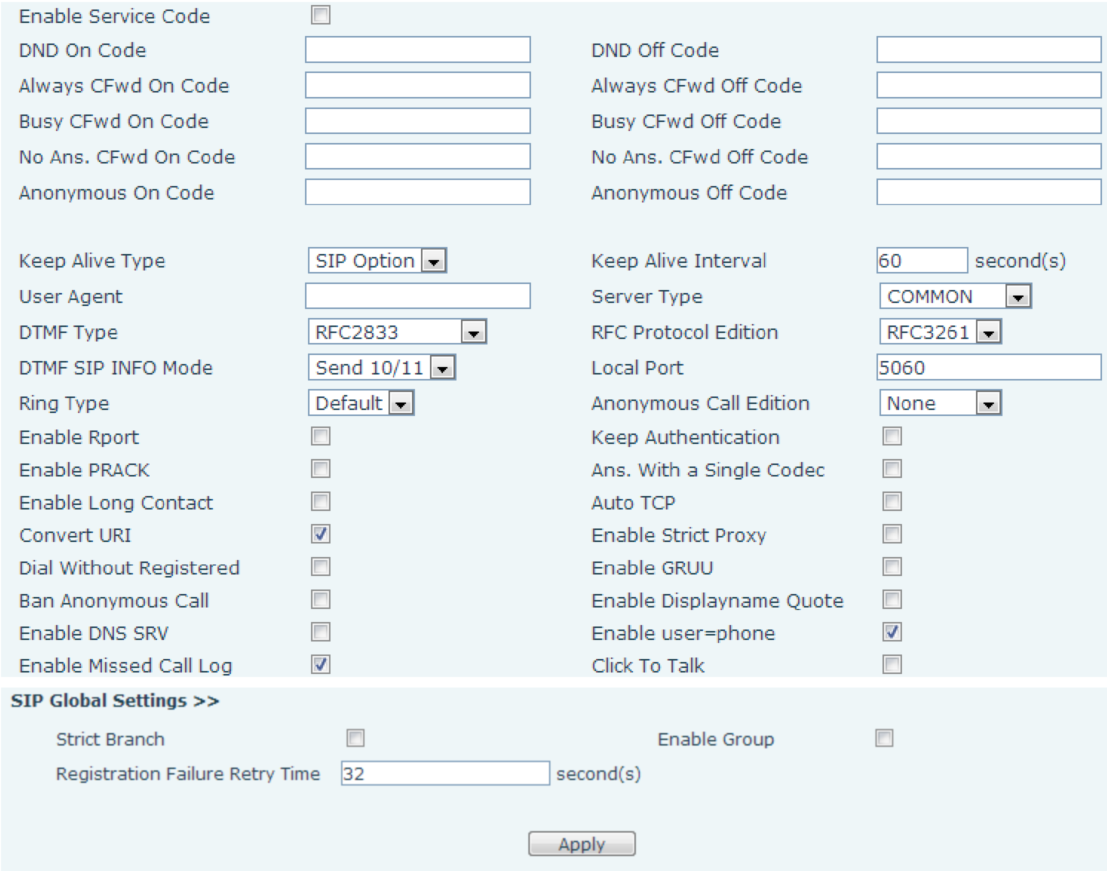


Table 5-13 SIP parameters setting

Name	Description
SIP Line	In this part, you can choose one of the four SIP lines and configure a SIP account for it.
Basic Settings	
Status	Indicates whether the phone is registered to the SIP server. Available values include Applied and Unapplied.
Server Address	Specifies the IP address of the SIP server.
Server Port	Specifies the signaling port of the SIP server.
Authentication User	Specifies the account name for logging in to the SIP server.
Authentication Password	Specifies the password for logging in to the SIP server.
SIP User	Specifies the phone number registered to the SIP server. If the value is None, the registration cannot be initiated.
Display Name	Specifies the name to be displayed.
Enable Registration	Enables or disables registration.
Domain Realm	Specifies a domain name of the SIP server. If the domain name of the SIP terminal is not specified as the domain realm, you can use the IP address or domain name of the SIP server.

Name	**Description**
Proxy Server Address	Specifies the IP address of a proxy server. (Usually, the SIP service provider provides a server with configurations of the proxy server and registration server being identical. If configurations (such as the IP address) of the proxy server and registration server are different, you need to make modifications as needed.
Proxy Server Port	Specifies the signaling port of the SIP proxy server.
Proxy User	Specifies the account of the SIP proxy server.
Proxy Password	Specifies the password of the SIP proxy server.
Backup Server Address	Specifies the IP address of the backup server. I If the IP address of the primary server is unavailable, the phone will use the backup server address.
Backup Server Port	Specifies the port of the backup server.
Server Name	
Codecs Settings	
Disable Codecs/Enable Codecs	Adds or deletes codecs, and changes the codecs priority through the navigation keys.
Advanced SIP Settings	
Forward Type	Specifies the call forward mode. By default, call forward is disabled. **Disabled**: disables call forward. **Busy**: forwards the call to a specified number if the phone is busy. **No answer**: forwards the call not answered within the specified time to a specified number. **Always**: forwards all incoming calls to a specified number under any conditions. The phone will prompt you of the incoming before carrying out the forwarding operation.
Forward Number	Specifies the number the incoming call will be forwarded to.
No Ans. Fwd Wait Time	Forwards the incoming call to a specified number at a certain delay in the scenario where the forward type is set to No answer and the, incoming call is not answered in time.
Transfer Timeout	For the phone supports the transfer of certain special features server, set interval time between sending “bye” and hanging up after the phone transfers a call.
Enable Hotline	Enables the hot line function.
Hotline Number	Sets a hot line number for the phone to dial automatically after the handset is picked up for a certain period of time.
Warm Line Wait Time	Sets the hot line delay time.
BLF Server	Sets this parameter if your server does not support subscription package to separates the registration server from the subscription server. Because ordinary BLF application will send subscription package to the registration server
SIP Encryption	Enables or disables SIP encryption.
SIP Encryption Key	Sets the key for SIP encryption.
RTP Encryption	Enables or disables RTP encryption.
RTP Encryption Key	Sets the key for RTP encryption.
Enable Auto Answer	Enables Auto Answer.

Name	Description
Auto Answer Timeout	Sets the timeout period for the phone to automatically answer the incoming call. For example, if the value is set to 60, the phone will automatically answer the call that is not answered within 60 seconds.
Enable Session Timer	Set Enable/Disable Session Timer, whether support RFC4028.It will refresh the SIP sessions.
Session Timeout	Sets the session timeout period.
Subscribe for MWI	Enables the phone to subscribe the peer status or voice messages..
MWI Number	Sets an MWI number to receive voice message and notifications from the SIP server.
Subscribe Period	Sets the interval for sending subscription packets.
Conference Type	Specifies the type of a conference. If the value is set to Local, the conference ID is not required.
Conference Number	Specifies the number of a conference, which can be obtained from the system administrator.
Registration Expire	Sets the registration expiration time of the SIP server. The default value is 60, in seconds. If the registration expiration time required by the SIP server is different (longer or shorter) than the value configure on the phone, change the value on the phone to the time recommended by the server and register again.
Enable Service Code	Enables the service code function.
DND On Code	Enables the DND On Code function. When you press the **DND** key, the phone will send a message to the server, and the server will turn on the DND function. After that, the server will reject any incoming call and information about the incoming call will not be displayed in the Call History or on the LCD screen of the phone.
DND Off Code	Disables the DND Off Code function. When you press the **DND**, the phone will send a message to the server, and the server will turn off the DND function accordingly.
Always CFwd On Code	Enables the phone to send a message for the server to turn on the function immediately. After that, the server will always forward the incoming call to a preset number automatically and information about the incoming call will not be displayed in the Call History or on the LCD screen of the phone.
Always CFwd Off Code	Disables the Always CFwd Off Code function. The phone will send a message to the server, and the server will turn off the function immediately.
Busy CFwd On Code	Enables the phone to send a message to the server and instruct the server to turn on the function immediately. In the scenario where the phone is in calling and the call waiting function is disabled, the server will automatically forward the new incoming call to a preset number . Information about the incoming call will not be displayed in the Call History or on the LCD screen of the phone.
Busy CFwd Off Code	Disables the Busy CFwd Off Code function. The phone will send a message to the server, and the server will turn off the function immediately.
No Ans. CFwd On Code	Enables the on answer forward function. The phone will send a message to the server, and the server will turn on the function immediately. When an incoming call is not answered after the timeout period, the server will forward the call to a preset number automatically. Information about the incoming call will not be displayed in the Call History or on the LCD screen of the phone.
No Ans. CFwd Off Code	Disables the No Answer CFwd Off Code function. The phone will send a message to the server, and the server will turn off the function immediately.
Anonymous On Code	Enables the anonymous call function. The phone will send a message to the server, and the server will turn on the function immediately.
Anonymous Off Code	Disables the anonymous call function. The phone will send a message to the server, and the server will turn off the function immediately.

Name	Description
Keep Alive Type	Sets the keep alive type. If the value is set to Option, the phone will send an option SIP message at a set interval, and the server will send "200OK" in response. If the value is set to UDP, the phone will send a UDP message at a set interval to the server.
Keep Alive Interval	Sets the interval at which the phone sends a message to check whether the server operates normally.
User Agent	Enables a user agent.
DTMF Type	Sets the DTMF sending mode. The value can be: DTMF_RELAY DTMF_RFC2833 DTMF_SIP_INFO Different VoIP Service providers may use different modes.
DTMF SIP INFO Mode	Sets the type to 10/11 or * / #.
Ring Type	Sets the ring tone for of each SIP line.
Enable Rport	Enables/Disables the system to support RFC3581. Via rport is a special way to realize SIP NAT.
Enable PRACK	Enables or disables the SIP PRACK function, which is specialized for the Ringback tone (RTB). The default configuration is recommended.
Enable Long Contact	Enables the Contact field to contain more parameters. This function is used together with SEM services.
Convert URI	Converts **#** to **%23** before sending URI messages.
Dial Without Registered	Allows proxy-based call without registration.
Ban Anonymous Call	Disables the anonymous call function.
Enable DNS SRV	Supports DNS looking up with sip.udp mode.
Enable Missed Call Log	Enable the missed call log by it, the phone will save the missed call log into the call history record and display the missed calls on the idle screen, or won't save the missed call log into the call history record and display the missed calls on the idle screen.
Server Type	Sets the signaling encryption mode or the type of a special server.
RFC Protocol Edition	Sets a SIP version. If the phone needs to communicate with the gateway enabling with SIP 1.0 (such as CISCO5300), you need to set the value to RFC2543. The default value is RFC3261.
Local Port	Sets a port for each SIP line.
Anonymous call Edition	Enables the phone to make anonymous calls safely. Both RFC3323and RFC3325 are supported.
Keep Authentication	Enables the phone to send a registration message containing an authentication field to the server. In this case, the phone does not need to send a separate authentication message to the server, and the server is required to respond only to the registration message, which simply the operation process
Ans. With A Single Codec	Enables the phone to respond to the SIP message with just one codec supported.
Auto TCP	Enables the phone to automatically use TCP for message transport in the scenario where the message size exceeds 1300 bytes.
Enable Strict Proxy	Enables the phone to be compatible with special SIP servers. In composing the message in response to the server, the phone uses the source IP address of the server, not the IP address in the **via** field of the message received.
Enable GRUU	Enables the phone to support GRUU.

Name	Description
Enable Display name Quote	Quotes the display name in output signaling message for server compatibility.
Enable user=phone	Enables the **invite sip** message to contain the **user=phone** field for server compatibility.
Click to talk	Enables the click to Talk function. This function needs application support of the software.
Transport Protocol	Sets a transport protocols, TCP or UDP.
Enable BLF List	Monitors the status of multiple accounts
SIP Global Settings	
Strict Branch	Enables the phone to strictly check the Branch field in the received SIP message. If the Branch value of **via** field in a SIP message is not beginning with z9hG4k, the phone will not respond to the SIP message. Notice: This configuration takes effect for all SIP accounts.
Enable Group	Enables the group function, which is used for SIP group backup. by selecting it, then the phone enable the sip group backup function. Notice: This configuration takes effect for all SIP accounts.
Registration Failure Retry Time	Sets an interval at which the phone registers to the SIP server in case of registration failure. Notice: This configuration takes effect for all SIP accounts.

5.5.2 IAX2

Figure 5-16 IAX2 setting interface

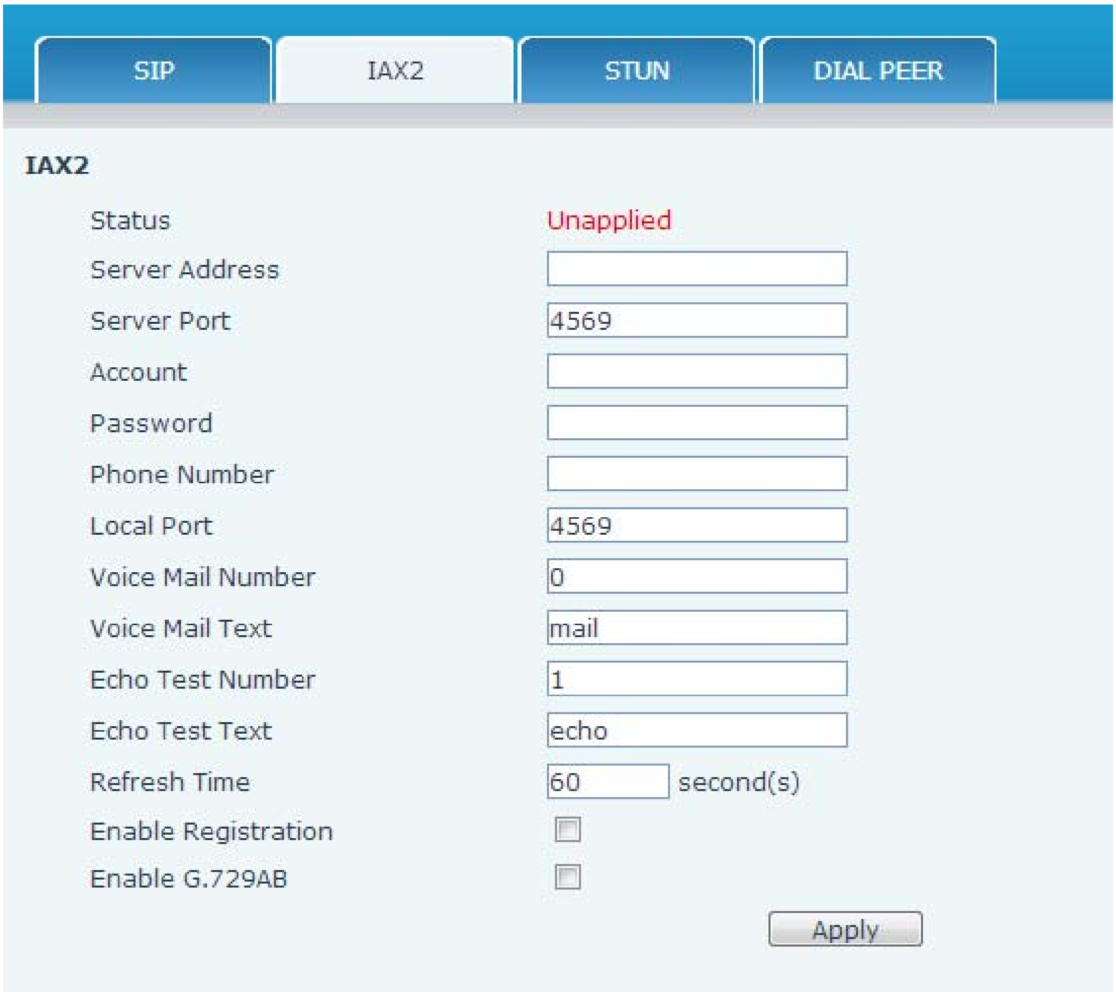


Table 5-14 IAX2 parameters setting

Name	Description
Status	Indicates whether the phone has been registered to the IAX2 server. The value can be **Applied** or **Unapplied**.
Server Address	Specifies the IP address of the IAX2 server. The value can be in the form of domain name.
Server Port	Specifies the port number of the IAX2 server.
Account	Specifies the account name for logging in to the IAX2 server.
Password	Specifies the password for logging in to the IAX2 server.
Phone Number	Specifies the phone number assigned.
Local Port	Set your local sport, the default is 4569.
Voice Mail Number	Specify the voice mail's number.
Voice Mail Text	Specifies the name of a voice mail.

Name	Description
Echo Test Number	Enables the IAX2 server to support echo test, a function to check whether the communication initiated by a terminal to a selected platform is normal. If the value is set to Enabled, the echo test number is in text format, the phone uses the echo test number to replace the echo test text..
Echo Test Text	Specifies the name of the echo test text.
Refresh Time	Sets the interval for refreshing the registration information of the IAX2 server. The value can be an integer ranging from 60 to 3600.
Enable Registration	Enables the phone to register to the IAX2 server or disables the phone from registering to the IAX2 server.
Enable G.729AB	Enables the phone to support G.729.

5.5.3 Stun

In this Web page, you can configure SIP STUN.

With a STUN server, the phone in private network can obtain the type of NAT and the NAT mapping IP and port of SIP. The phone might register itself to the SIP server with global IP address and port to enable the device to make a call and be called in private network.

Figure 5-17 Stun setting interface

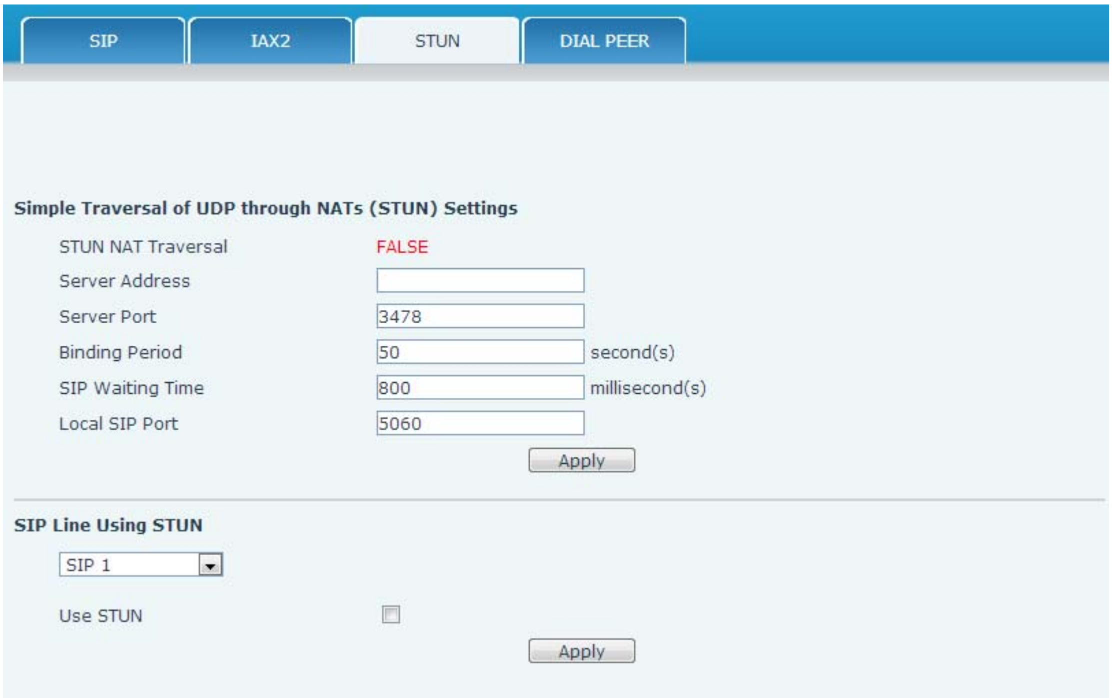


Table 5-15 Stun parameters setting

Name	Description
Simple Traversal of UDP through NATs (STUN) Settings	
STUN NAT Traversal	Shows STUN NAT Transverse estimation, true means STUN can penetrate NAT, while False means not.
Server Address	Sets the IP address of a SIP STUN server.

Name	**Description**
Server Port	Sets the port of a SIP STUN server.
Blinding Period	Sets the interval at which the STUN server identifies the NAT type. If the NAT server finds that a NAT mapping is idle after a certain period of time, it will release the mapping. Therefore, the system needs to send a STUN packet to keep the mapping effective and alive.
SIP Waiting Time	Specifies the time the SIP server waits for a response from the STUN server. You can set the value according to your network condition.
Local SIP Port	Specifies the local SIP Port. The default value is 5060. Note that this port takes effect immediately after being configured. If modified, the SIP server will use the new port for communication.
Sip Line Using STUN	
SIP n	Choose line to set info about SIP, There are 6 lines available.
Use STUN	Enables/Disables the SIP STUN function.

Note

- SIP STUN is used to realize the SIP penetration to NAT. If you configure the IP address and port(which defaults to 3478) of the STUN server , and enable SIP STUN on the phone, you can use an ordinary SIP server to realize penetration to NAT.

5.5.4 Dial Peer

This function offers you a flexible dial mode, allowing you to make phone calls through Internet or based on a dial peer table. For example, if you know the mapping relationship between the number (156) and IP address (192.168.1.119) of a peer, and want to call him in P2P mode, you can dial 156 to make conversation with him.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
156	192.168.1.119	5060	SIP	no alias	no suffix	0

Another example is that if you want to dial a long distance call to Beijing, you need to dial the area code 010 before the local phone number, such as 01062213123. To make the dialup process easier, you can use the dial peer table to set a dialing rule: using 1 to replace the area code 010, which simplifies 01062213123 to 162213123.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
1T	0.0.0.0	5060	SIP	rep:010	no suffix	1

To save the memory space and avoid abundant input, the following functions are added:

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
13xxxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0
13[5-9]xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0

- The sign “**x**” is added to represent a digit number. If user makes the above configuration, after user dials 11 digit numbers starting with 13, number 0 will be automatically added to the dialed numbers.

- **Square brackets** "**[]**" are added to specify the value range. It may be a range, a list of ranges separated by commas, or a list of digits.

If a user dials 11 digit numbers started from 135 to 139, the phone will automatically add 0 before the dialed numbers.

Use this phone you can dial out via different lines without switch in web interface.

Figure 5-18 Dial peer interface

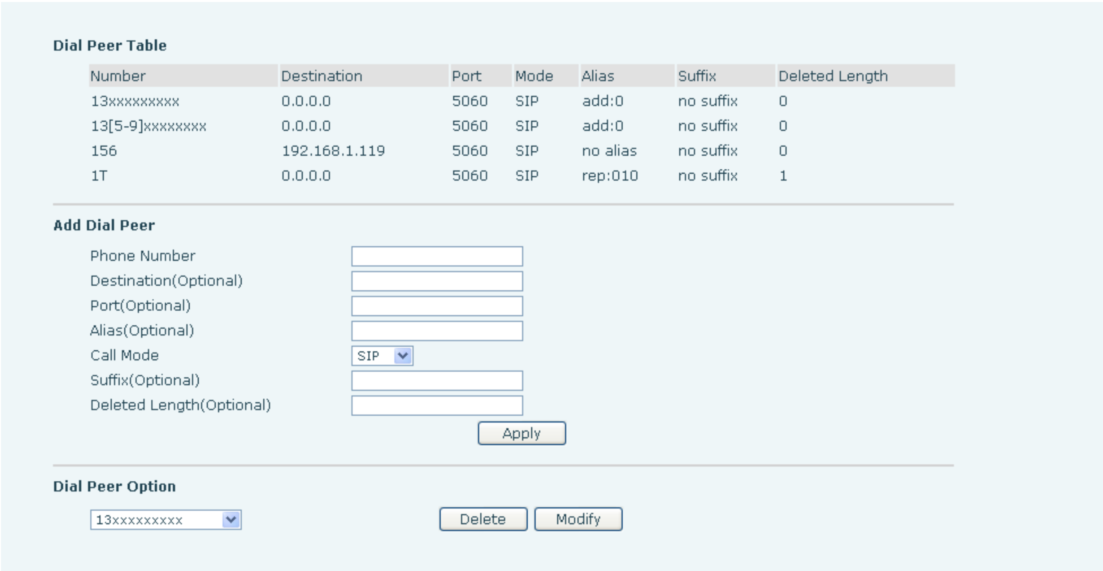


Table 5-16 Dialpeer parameters setting

Name	Description
Phone Number	Adds an outgoing number. The outgoing number can be set in two modes: full matching or prefix matching. In full matching mode the phone will make a call unless the outgoing number is identical with
Destination	Sets a destination address. In case of P2P mode, you can input the IP address or domain name of the peer. The DNS server of the phone will parse the value to a specific IP address. This parameter is optional.
Port	Sets the signaling port, with the default value of 5060. This parameter is optional.
Alias	Sets an alias. This parameter is optional. There are four types of aliases. 1) **Add: xxx**: indicates that you need to dial xxx in front of the phone number, which will reduce the length of the dialed number. 2) **All: xxx**: indicates that xxx will replace some phone number. 3) **Del**: indicates that the number with length appointed will be deleted. 4) **Rep**: indicates that that phone will replace the number with the length and number appointed.
Mode	Selects a signaling protocol, SIP or IAX2
Suffix	Sets the suffix. This parameter is optional.
Delete Length	Sets the digits in the phone number to be deleted or replaced. This parameter is optional.

The following describes how to configure the dial peer table to achieve the configuration of multiple accounts simultaneously:

8T	0.0.0.2	5060	SIP	del	no suffix	1
9T	0.0.0.0	5060	SIP	del	no suffix	1

9T indicates that if the phone is configured with a SIP1 server and has been registered to the server, all users attempting to make phone calls through the SIP1 server need to add 9 before the phone number.

8T indicates that if the phone is configured with a SIP2 server and has been registered to the server, all users attempting to make phone calls through the SIP2 server need to add 8 before the phone number

2T	0.0.0.0	4569	IAX2	del	no suffix	1

2T indicates that if the phone is configured with a an IAX2 server and has been registered to the server, all users attempting to make phone calls through the IAX2 server need to add 2 before the phone number

Note: To compatible with functions in version 1.6, the "**Dialpeer With Line:**" field is added to the configuration file of version 1.7 version, indicating whether to enable the on-line inquiry function. 0 for **Disabled** and 1 for **Enabled**. The default value is 0.

Compared with version 1.6, functions added in version 1.7 include: **Not enabled on-line inquiry**

The function is the same as that in version 1.6.

Type: specifies the protocol on which the rule takes effect.

Destination: specifies the destination IP address.

0.0.0.1: indicates that SIP line 1 is used.

0.0.0.2: indicates that SIP line 2 is used.

0.0.0.x: indicates that SIP line x is used.

(To compatible with codes in earlier versions, **0.0.0.0** indicates that SIP line 1 is used; **255.255.255.255** indicates that SIP line 2 is used.

Configuration examples are as follows:

2T	255.255.255.255	5060	SIP	del	no suffix	1
3T	0.0.0.0	4569	IAX2	del	no suffix	1

If the phone dials the number 21111, the call is sent out through SIP line2, with the called number of 1111.

If the phone dials the number 32222, the call is sent out through IAX2, with the called number of 2222.

Enable on-line query capabilities

The precondition for this function is that the phone must have multiple lines, allowing the line and protocol to be selected before dial-up.

Dialpeer table in the query, the first comparison dialing protocol is selected in the table and dialpeer agreement, if the same, continue down the match, otherwise, check the next one.

Step match line information, comparing the selected dial-up line is a line in the table and dialpeer is the same, if the same, continue down the match, otherwise the next query.

The third step is for a prefix or exact match.

- **Mode**: If the mode is set to SIP, this rule takes effect only for SIP-based calls. If the mode is set to IAX2, this rule takes effect only forIAX2 calls.
- **Destination**: indicates the destination IP address.
- **0.0.0.1**: indicates that this rule takes effect only for calls on SIP line 1.
- **0.0.0.2**: indicates that this rule takes effect only for calls on SIP line 2.
- **0.0.0.x**: indicates that this rule takes effect only for calls on SIP line x.
- **0.0.0.0**: indicates that this rule takes effect for calls on all SIP lines.

Configuration examples are as follows:

3T	0.0.0.0	4569	IAX2	del	no suffix	1
2T	0.0.0.0	5060	SIP	del	no suffix	1

- If you dial 21111 and SIP1 is registered successfully (which will thus be the default line), the call will be sent out through SIP line 1, with the called number of 21111.
- If you dial 32222 and SIP1 is registered successfully (which will thus be the default line), the call will be sent out through SIP line 1, with the called number of 32222.

To activate the configured dial peer function,

- Choose SIP2 and dial 21111. The call will be sent out through SIP line 2, with the called number of 1111.
- Choose IAX2 and dial 32222. The call will be sent out through IAX2, with the called number of 2222.

Table 5-17 Examples of different alias application

Set by web	Description	Example
Phone Number: 9T Destination (optional): 255.255.255.255 Port(optional): Alias(optional): del Call Mode: SIP Suffix(optional): Delete Length (optional): 1	You need set phone number, Destination, Alias and Delete Length. Phone number is XXXT; Destination is 255.255.255.255 (0.0.0.2) and Alias is del. This means any phone No. that starts with your set phone number will be sent via SIP2 line after the first several digits of your dialed phone number are deleted according to delete length.	If you dial “93333”, the number that the SIP2 server receives is “3333”.
Phone Number: 2 Destination (optional): Port(optional): Alias(optional): all:33334444 Call Mode: SIP Suffix(optional): Delete Length (optional):	This setting will realize speed dial function, after you dialing the numeric key “2”, the number after all will be sent out.	When you dial “2”, the number that the SIP1 server receives is 33334444.

Set by web	Description	Example
Phone Number: 8T Destination (optional): Port(optional): Alias(optional): add:0755 Call Mode: SIP Suffix(optional): Delete Length (optional):	The phone will automatically send out alias number adding your dialed number, if your dialed number starts with your set phone number.	When you dial “8309“, the number that the SIP1 server receives is “07558309”.
Phone Number: 010T Destination (optional): Port(optional): Alias(optional): rep:0086 Call Mode: SIP Suffix(optional): Delete Length (optional): 3	You need set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is rep: xxx If your dialed phone number starts with your set phone number, the first digits same as your set phone number will be replaced by the alias number specified and New phone number will be send out.	When you dial “0106228”, the number that the SIP1 server receives is “86106228”.
Phone Number: 147 Destination (optional): Port(optional): Alias(optional): Call Mode: SIP Suffix(optional): 0011 Delete Length (optional):	If your dialed phone number starts with your set phone number. The phone will send out your dialed phone number adding suffix number.	When you dial “147”, the number that the SIP1 server receives is “1470011”.

5.6 Phone

5.6.1 Audio

In this page, you can configure voice-relevant parameters.

Figure 5-19 Audio setting interface

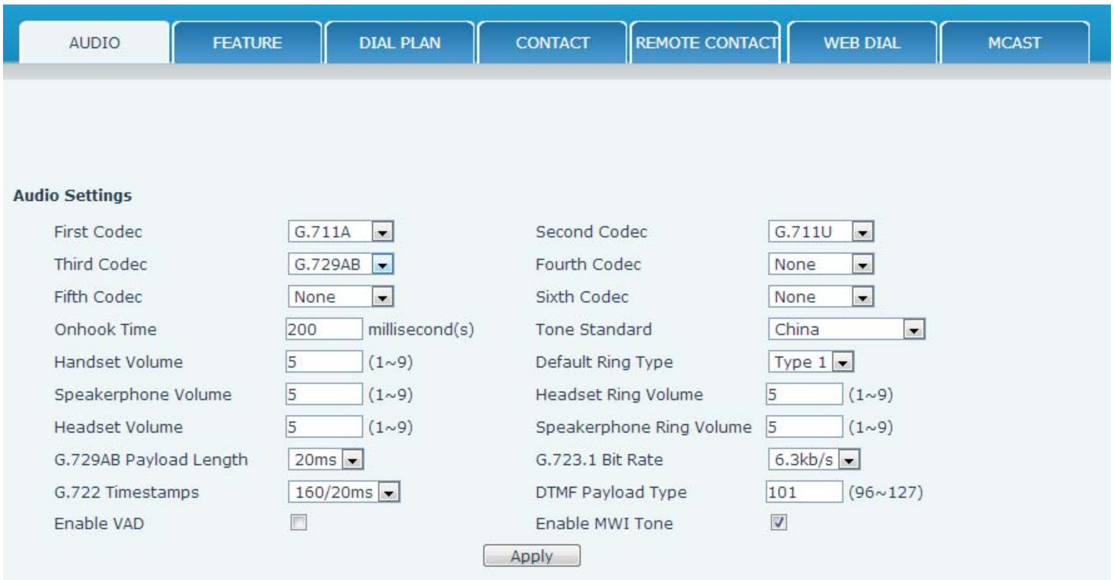


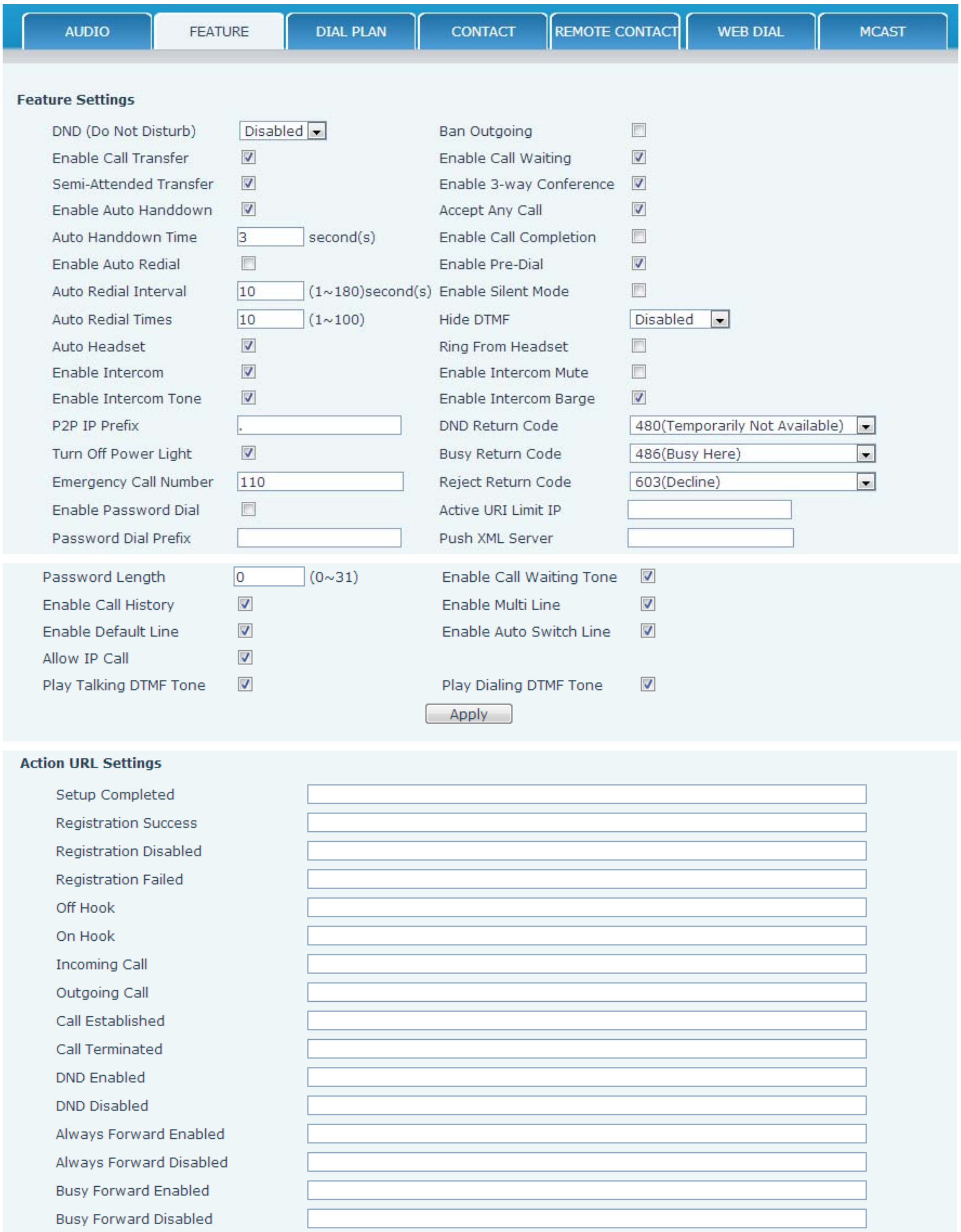
Table 5-18 Audio setting parameters

Name	Description
First Codec	Specifies the first preferential DSP codec. The value can be G.711A/u, G.722, G.723.1, 726-32 G.729AB, or None.
Second Codec	Specifies the second preferential DSP codec. The value can be G.711A/u, G.722, G.723.1,726-32 G.729AB, or None.
Third Codec	Specifies the third preferential DSP codec. The value can be G.711A/u, G.722, G.723.1, 726-32 G.729AB, or None.
Fourth Codec	Specifies the forth preferential DSP codec. The value can be G.711A/u, G.722, G.723.1, 726-32 G.729AB, or None.
Fifth Codec	Specifies the fifth preferential DSP codec. The value can be G.711A/u, G.722, G.723.1, 726-32 G.729AB, or None.
Sixth codec	Specifies the sixth preferential DSP codec. The value can be G.711A/u, G.722, G.723.1, 726-32 G.729AB, or None.
Onhook Time	Specifies the minimum response time of Hand down. The default value is 200, in ms.
Tone Standard	Sets the tone standard.
Handset Volume	Sets the sound volume of the handset.
Default Ring Type	Sets a default ring tone.
Speakerphone volume	Sets the sound volume of the speaker.
Headset Ring Volume	
G729AB Payload Length	Sets the payload length of the G.729 voice codec.
G723.1 Bit Rate	Sets the G.723 bit rate. The value can be5.3 Kbit/s or 6.3 Kbit/s.
G722 Timestamps	Sets the G.722 timestamp. The value can be 160/20 ms or 320/20 ms.
DTMF Payload Type	Set DTMF Payload Type.
Enable VAD	Enables the voice activation detection (VAD) function. If enabled, theG.729 payload length cannot exceed 20 ms.
Enable MWI Tone	

5.6.2 Feature

In this Web page, you can configure parameters listed in Figure 5-20:

Figure 5-20 Feature setting interface



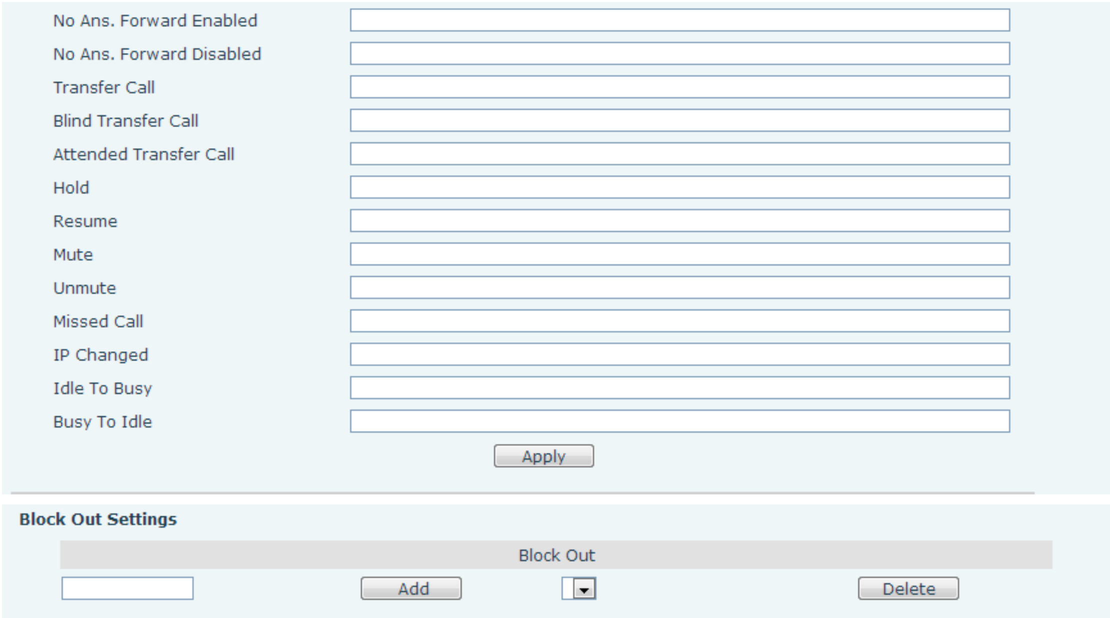


Table 5-19 Feature parameters setting

Name	Description
DND (Do Not Disturb)	Enables the phone to reject any incoming call and prompt the caller that the phone is busy. The outgoing call of the phone is not affected.
Ban Outgoing	If you select Ban Outgoing to enable it, and you cannot dial out any number.
Enable Call Transfer	Enables the phone to transfer incoming calls to a specified number. Call Transfer by selecting it.
Enable Call Waiting	Enables Call Waiting by selecting it. Then the phone prompts the caller to redial or that the phone is busy or the call is rejected. When it's ok and the phone finds out that the caller is idle by sip message, it will prompt the user to redial.
Semi-Attended Transfer	Enables the phone to transfer incoming calls to a specified number in semi-attended mode.
Enable 3-way Conference	Enables the user to start a 3-way conference by selecting it.
Enable Auto Hand down	The phone will hang up and return to the idle status automatically in the hands-free mode.
Accept Any Call	If you select it, all calls will be accepted even if the calling number is untraceable.
Auto Hand down Time	Specifies Auto Hand down Time, the phone will hang up and return to the idle status automatically after Auto Hand down Time in the hands-free mode, and play dial tone Auto Hand down Time in the handset mode.
Enable Call Completion	Enables Call Completion by selecting it.
Enable Auto Redial	Enables the phone to prompt you whether to dial the number again in the case that the peer is busy or the call is rejected (the code 486 is received).
Enable Pre-Dial	After this feature is disabled, the number dialed in standby interface, will be sent out over the time; After the feature is enabled, the number will not be send out over the time.
Auto Redial interval	Sets the auto redial interval of the phone.
Enable Silent Mode	Enables Silent Mode by selecting it, the phone light will blink to prompt the user that there is an incoming call instead of playing ring tone.

Name	Description
Auto Redial Times	Sets the auto redial times of the phone.
Hide DTMF	Specifies the hide DTMF mode.
Auto Headset	
Ring From Headset	Enables Ring From Handset by selecting it, the phone plays ring tone from handset.
Enable Intercom	Enables Intercom Mode by selecting it.
Enable Intercom Mute	Enables mute mode during the intercom call.
Enable Intercom Tone	If the incoming call is an intercom call, the phone plays the intercom tone.
Enable Intercom Barge	Enables Intercom Barge by selecting it, the phone automatically answers the intercom call during a call. If the current call is an intercom call, the phone will reject the second intercom call.
P2P IP Prefix	Sets Prefix in peer to peer IP call. For example: what you want to dial is 192.168.1.119, If you define P2P IP Prefix as 192.168.1., you dial only #119 to reach 192.168.1.119. Default is “.”. If there is no “.” Set, it means to disable dialing IP.
DND Return Code	Specifies DND Return code.
Turn Off Power Light	Enables Turn Off Power Light by selecting it.
Busy Return Code	Specifies Busy Return Code.
Emergency Call Number	Specifies the Emergency Call Number. Despite the keyboard is locked, you can still make an emergency call.
Reject Return Code	Specifies Reject Return Code.
Enable Password Dial	Enables Password Dial by selecting it, when the number entered starts with the password prefix, the following N numbers After the password prefix is hidden as *, N stands for the value which you enter in the Password Length field. For example: after you set the password prefix as 3 and the Password Length as 2, when you enter the number 34567, it will display 3**67 on the phone.
Active URI Limit IP	Specifies the server IP address that remotely controls thexcorresponding operations of the phone.
Password Dial Prefix	
Push XML Server	Specifies the Push XML Server, when the phone receives a request, it will determine whether to display the corresponding content on the phone that are sent by the specified server or not.
Password Length	Specifies the Password length.
Enable Call Waiting Tone	Enables this function you will not hear the tone “beep” when there are multiple incoming calls
Enable Call History	
Enable Multi Line	
Enable Default Line	

Name	Description
Enable Auto Switch Line	
Allow IP Call	
Play Talking DTMF Tone	
Play Dialing DTMF Tone	
Action URL Settings	
Action URL Settings	Specifies the Action URL that reports the operation of the phone to the server, for example, url: http://InternalServer /FileName.xml? (Internal Server is the server IP address. Filename is the name of xml that contains the action message).
Block Out Settings	
Block out	Sets the Add/Delete Limit List. Please input the prefix of the phone numbers that you forbid the phone to dial. For example, if you want to forbid to dial the phones with prefix 001, you need to input 001 in the blank of limit list, and then you cannot dial any of these phone numbers with prefix 001. X and are wildcard x means matching any single digit. For example, 4xxx that represents any number with prefix 4 and the length of which is 4 will be forbidden to dial out, means matching any arbitrary number digit. For example, 6 that represents any number with prefix 6 will be forbidden to dial out.

5.6.3 Dial Plan

This setting page consists of two parts **Basic Settings** and **Dial Plan Table**. Configurations of each part are as follows:

Basic Settings:

- End with "#": dial your desired number, and then press #.
- Fixed Length: the phone will intersect the number according to your specified length.
- Time Out: After you stop dialing and waiting time out, the system will send the number collected.
- Press # to Do Blind Transfer: input the number you want to transfer to, and then press"#", you can transfer the current call to the number.
- Blind Transfer on OnHook: input the number you want to transfer to, and then hang up the handle or press the speaker, you can transfer the current call to the number.
- Attend Transfer on OnHook: hang up the handle or press the speaker you can realize the blind transfer function
- Press the DSS key Blind: Press the DSS key, the current call will turn out blind.

Dial Plan Table:

In this part, you can add dial plans and customize the number prefix.

In order to maintain the end-user pbx secondary dial for dialing call mode. When requested to enter a phone number prefix, the system according to the rules in the closing number configuration rules, re-issues the dial tone, the user continues to enter the number, after the end of the closing number, the phone number will be prefixed and analog secondary dial tone is sent to the back of the numbers together with the server.

For example:

If the plan **9, xxxxxxxx** is configured in the dial plan table, when you dial **9**, the system re-plays the dial tone, prompting you to continue to dial the number. After the dial-up, the phone sends out a 9-digit number containing the number **9**.

Figure 5-21 Dial plan setting interface

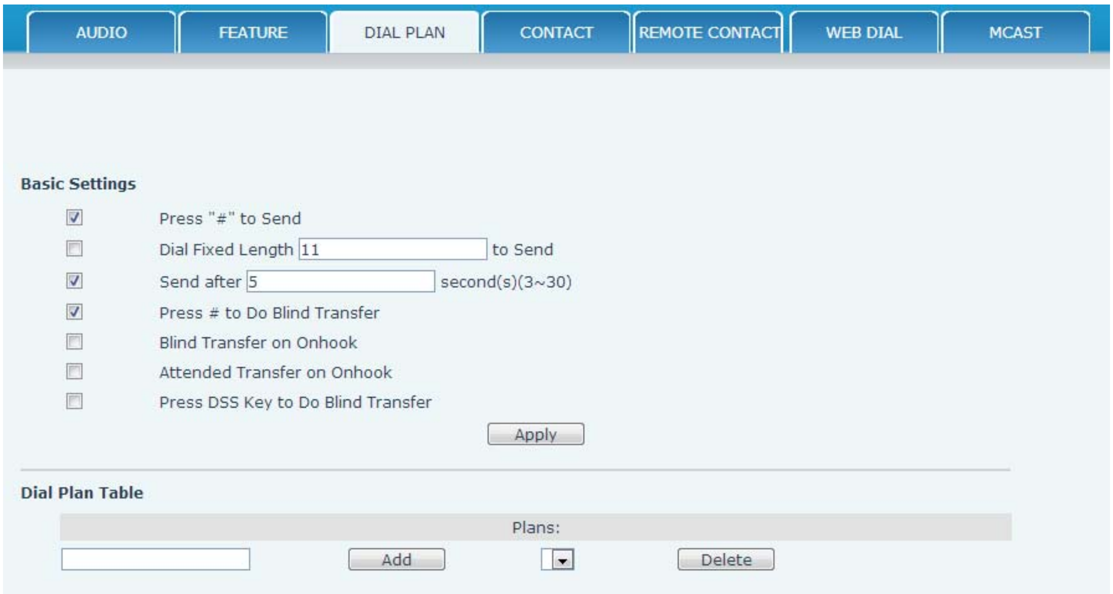


Table 5-20 Dial plan parameters setting

Name	Description
Basic Setting	
Press "#" to Send	Indicates that you need to press # after dialing a number.
Dial Fixed Length	Indicates that the phone will automatically call out after the number with a specified length is dialed. For example, if the value is set to 11,
Press # to Do Blind Transfer	Enables the phone to transfer the incoming call to a specified number (the third party) after # is pressed. This function requires you to press # after inputting the phone number of the third party.
Blind Transfer on OnHook	Enables the phone to transfer the incoming call to a specific number (the third party) after hang-up. This function requires you to input the number of the third-party before hang-up.
Attend Transfer on OnHook	Enables the phone to transfer the incoming call to a third party after the call is answered.

5.6.4 Contact

This function is equivalent to the phone book, recording the contact name, phone number, and ring tone.

Figure 5-22 Contact setting interface

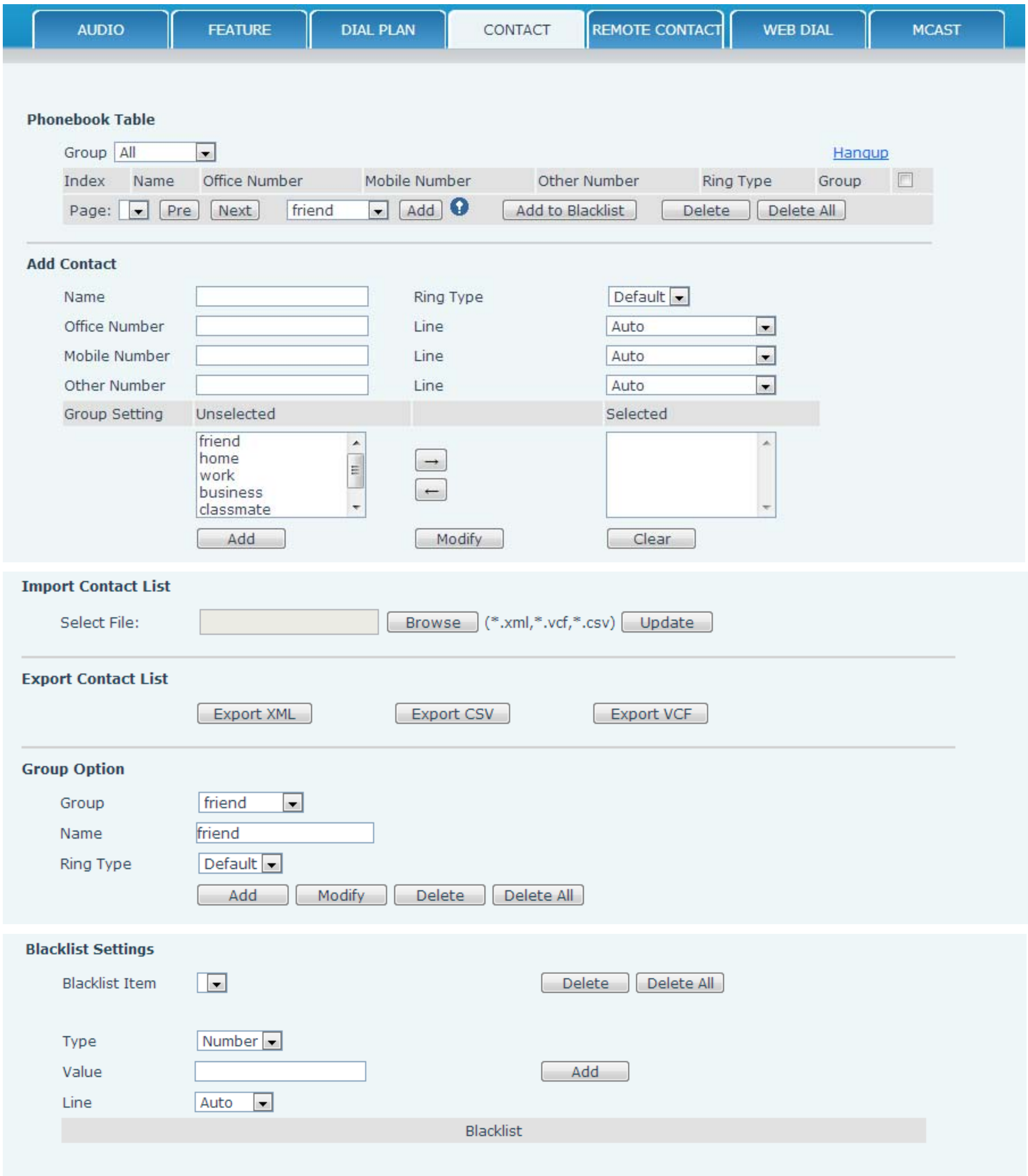


Table 5-21 Contact parameters setting

Name	Description
Phonebook Table	
Name	Displays the number of the incoming call in the form of contact name on the LCD screen.
Shows the detail of the current phonebook.	
Notice: the maximum capability of the phonebook is 500 items, you can select one or more contacts to group them, add them to the blacklist. And you candelete one or more contacts, or delete all.	
Add Contact List	
Name	Specifies the name corresponding to the phone number.

Name	**Description**
Office Number	Specifies the office number.
Mobile Number	Specifies the mobile number.
Other Number	Specifies the other number.
Ring Type	Specifies a ring type for the incoming calls.
Line	Specifies the sip line for each number.
Group setting	Selects a group for the contact from the unselected group; you can select multiple groups for one contact.
	Note: Button **Add** for adding a new contact; button **Modify** for modifying the added contact; tbutton **Clear** for emptying all input information of the contact.
Group Option	
Group	Modifies or deletes on the existing groups.
Name	Inputs the name of the group, then clicks button **Add**, you can add a new group.
Ring Type	Specifies a ring type for a group.
Blacklist Settings	
Type	Selects the blacklist type, you can choose the phone number or its prefix.
Value	Inputs the number or its prefix.
Line	Selects the sip line.

Note: Button Add for adding a new blacklist; button Delete for deleting one item; button Delete All for deleting all items.

If the user does not want to answer some phone calls, add these phone numbers to the Black List, and these calls will be rejected x and are wildcard x means matching any single digit. For example, 4xxx that reprents any number with prefix 4 and the length of which is 4 will be forbidden to be responded.

DOT (.) means matching any arbitrary number digit. For example, 6. that represents any number with prefix 6 will be forbidden to be responded.

If the user wants to allow a number or a series of number of incoming calls, he may add the number(s) to the list as the white list rule. The configuration rule is -number, for example, -123456, or -1234xx.

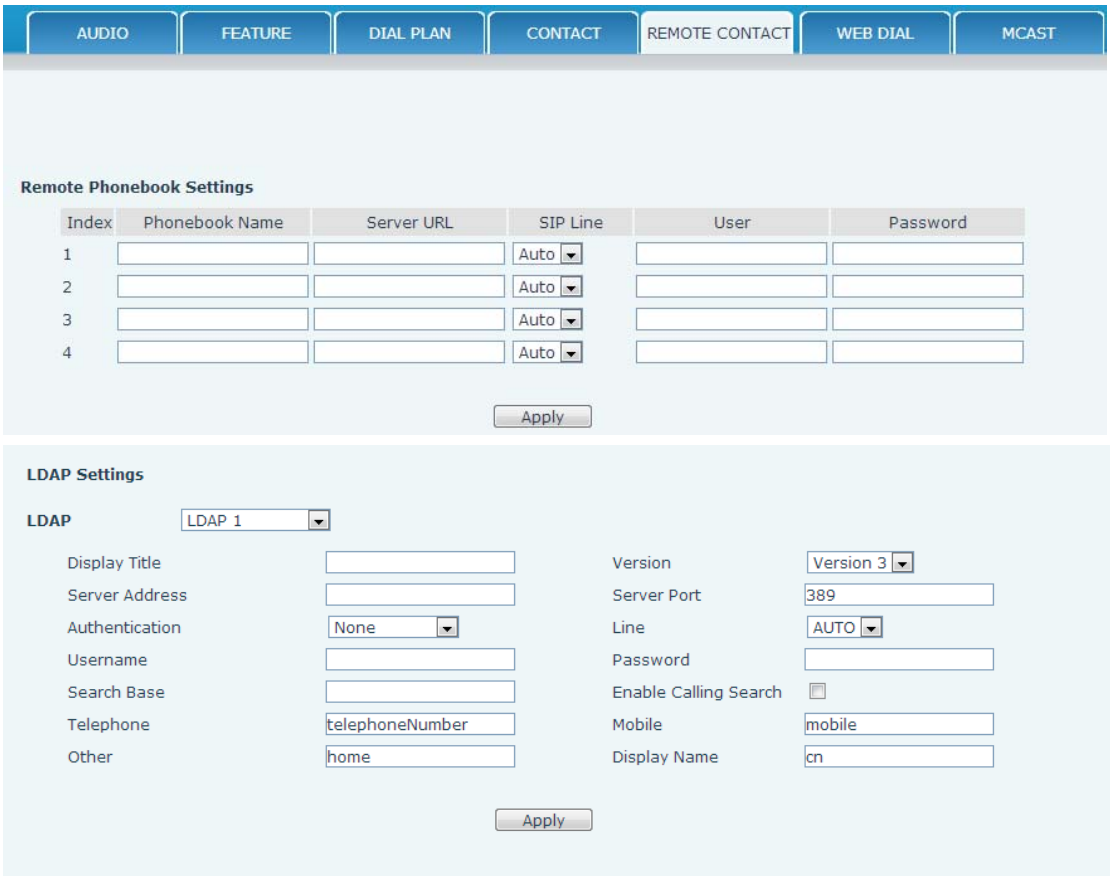
Black List
-4119
.

Means any incoming number is forbidden except for 4119

Note: End with DOT (.) when set up the white list.

5.6.5 Remote Contact

Figure 5-23 Rremote Contact setting interface



In this page, you can configure an address for the XML phonebook address. After that, you can directly access the corresponding remote phonebook after pressing the **remote contact** key on the phone.

For example:

You can set name of the phonebook to **fanvil** or **ldap**, with the respective server URL to tftp://192.168.1.3/admin/phonebook/index.xml and ldap://192.168.1.3/dc=winline,dc=com.

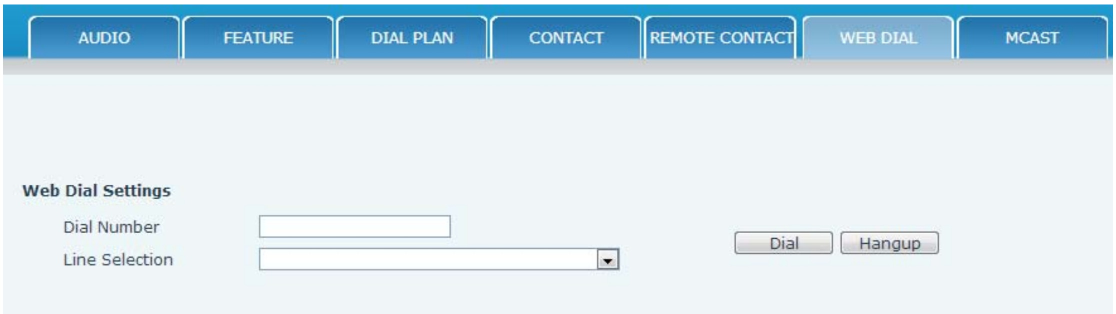
Table 5-22 Remote Contact parameters

Name	Description
Remote Phonebook Setting	
Phonebook Name	Specifies the name to be displayed on the phone.
Server URL	Specifies the server URL of the remote phonebook.
SIP Line	Specifies the SIP line for the remote phonebook.
Authentication	Specifies the authentication mode for the remote phonebook. This parameter is used for LDAP.
User/password	Specifies the authentication username and password.

5.6.6 Web Dial

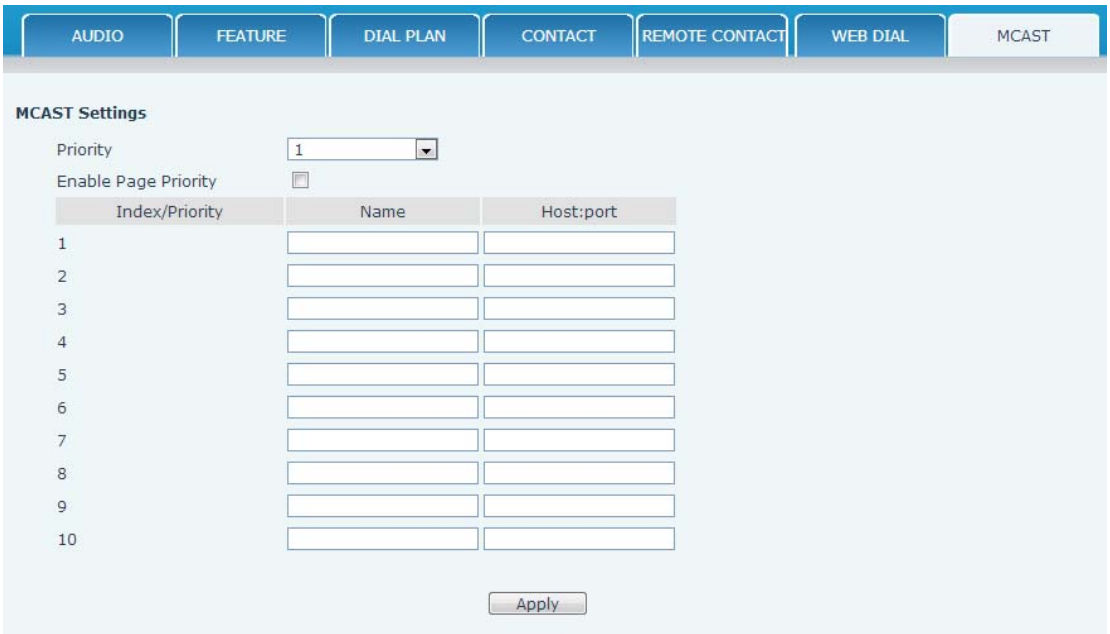
This function enables you to make a call through the Web page. After inputting the phone number and select a line, you can call out by clicking **Dial**. To terminate a call, click **Hangup**.

Figure 5-24 Wed dial setting interface



5.6.7 MCAST

Figure 5-25 MCAST interface



Use the multicast function to send notice to every member of the multicast is simple and easy. By setting the multicast key on your phone, you can send multicast RTP flow to the pre-configured multicast address. By listening multicast address is configured on the phone, listen and play the multicast address to send the RTP stream.

Send multicast setting

On the phone web page, function key-function key, set a function key， as shown

Value format IP:Port，the IP address of multicast is range from 224.0.0.0 to 239.255.255.255,port is greater than 1024

If multicast codec is G722, the LCD screen will displays "HD", which means the phone is sending high-definition voice stream

Operate steps:

When the phone is idle, press multicast key

Multicast RTP stream is sent to pre-configured multicast address (IP: Port)。The phone which listens to multicast address in the local network can receive the RTP stream. Multicast function key LED lights yellow.

LCD screen displays the following:

Press the hold softkey to hold the current multicast session

Press the end softkey again or multicast function key, multicast session can be stopped

Notice: RTP stream is one side, that is from a sender to a receiver. When the phone initiates a multicast RTP session in a call, the current call is on hold.

Receive multicast setting

You can set up the phone monitoring 10 different multicast addresses to receive these multicast RTP stream.

You have two method to receive RTP stream of multicast that can be set up through the web page: Enable priorities of normal calls and Enable page Priority:

Enable priorities of normal call by select it, if the incoming RTP stream priority of multicast lower than the priority of current for normal calls, the phone will ignore the RTP stream of multicast. If the incoming RTP stream priority of multicast higher than the priority of current for normal calls, the phone will receive the RTP stream of multicast, and hold the current call.

Disabled priorities of normal call by select disable, the phone will ignore all local network RTP stream of multicast.

Options as follows:

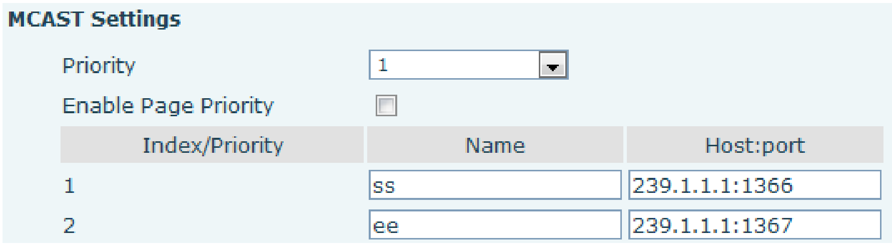
1-10:the priority defined for normal calls,1 the highest level,10 the lowest level

Disabled: Ignore all RTP stream of multicast

Enable Page Priority

Page priority determines the phone how to handle the newly received multicast RTP stream when in a multicast session. Enabled page priority, the phone will automatically ignore the low priority multicast RTP stream and receive the high priority multicast RTP stream and hold the current multicast session; If not enabled, the phone will automatically ignore all incoming multicast RTP stream.

Web page is set as follows:



Now multicast ss has higher priority than multicast ee, the highest priority is for normal calls

Notice: When a multicast session begins, multicast sender and receiver will beep

5.7 Function Key

5.7.1 Function Key

Figure 5-26 Function Key interface

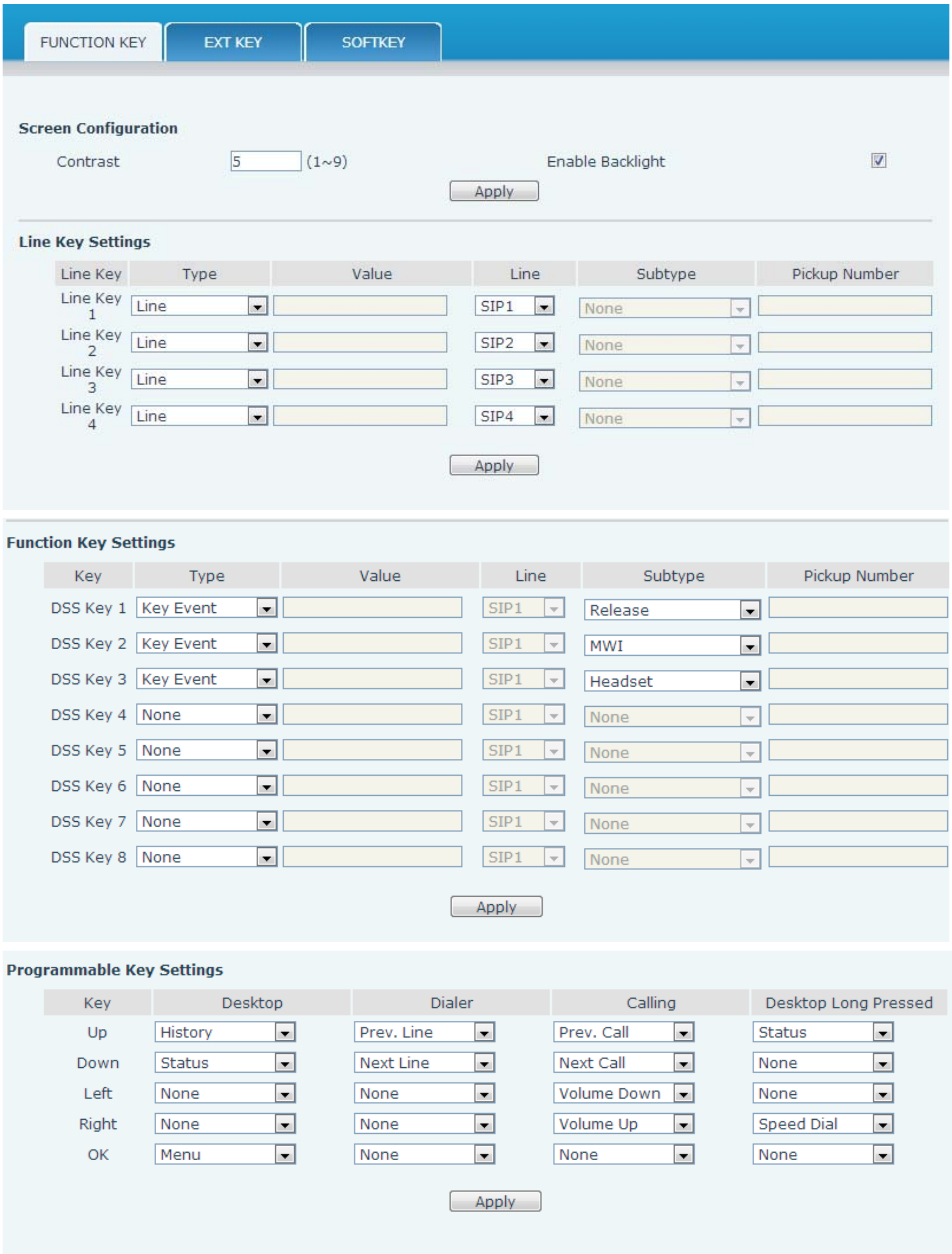


Table 5-23 Function Key parameters setting

Name	**Description**
Screen Configuration	
Contrast	Sets the contrast of the screen.
Enable Backlight	Turns the backlight on or off.
Line Key Settings	
In this part, you can set the Line: select Auto, SIP1, SIP2 or IAX2 in function key type. After you set it, you pick up handset or hands-free, press this function key, and then you can use the corresponding SIP line.	
Function Key Settings	
key	Shows the function key's serial number.
Type	Memory Key: settings can be stored in key storage for each number, the standby or off-hook, selects the function keys on the keyboard can call this number. Line, sets the dial mode (Auto, SIP1, SIP2, IAX2).Key Event functions, monitor state. DTMF: In the call, sends DTMF. URL: You can input remote book url.
Value	Sets the type parameter values.
Line	Chooses the lines for this feature.
Subtype	Selects the function parameters Key Event and Memory Event.
Pickup Number	Please input the pickup number When SubType is BLF or presence.

NOTE：

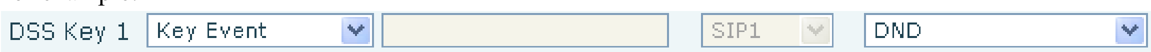
Memory keys can be configured as follows:
Speed Dial function, through the configuration of the key corresponding to the number of ways as shown below.

User can press the F1 key to allocate this number by line1 line.
Intercom function, you can press this key in standby to automatically answer the call and make each other.

User can be configured in accordance with push to talk function the way: 4116 was the other number; Then press the standby button to automatically answer the call 4116.

key can be configured through the following events:

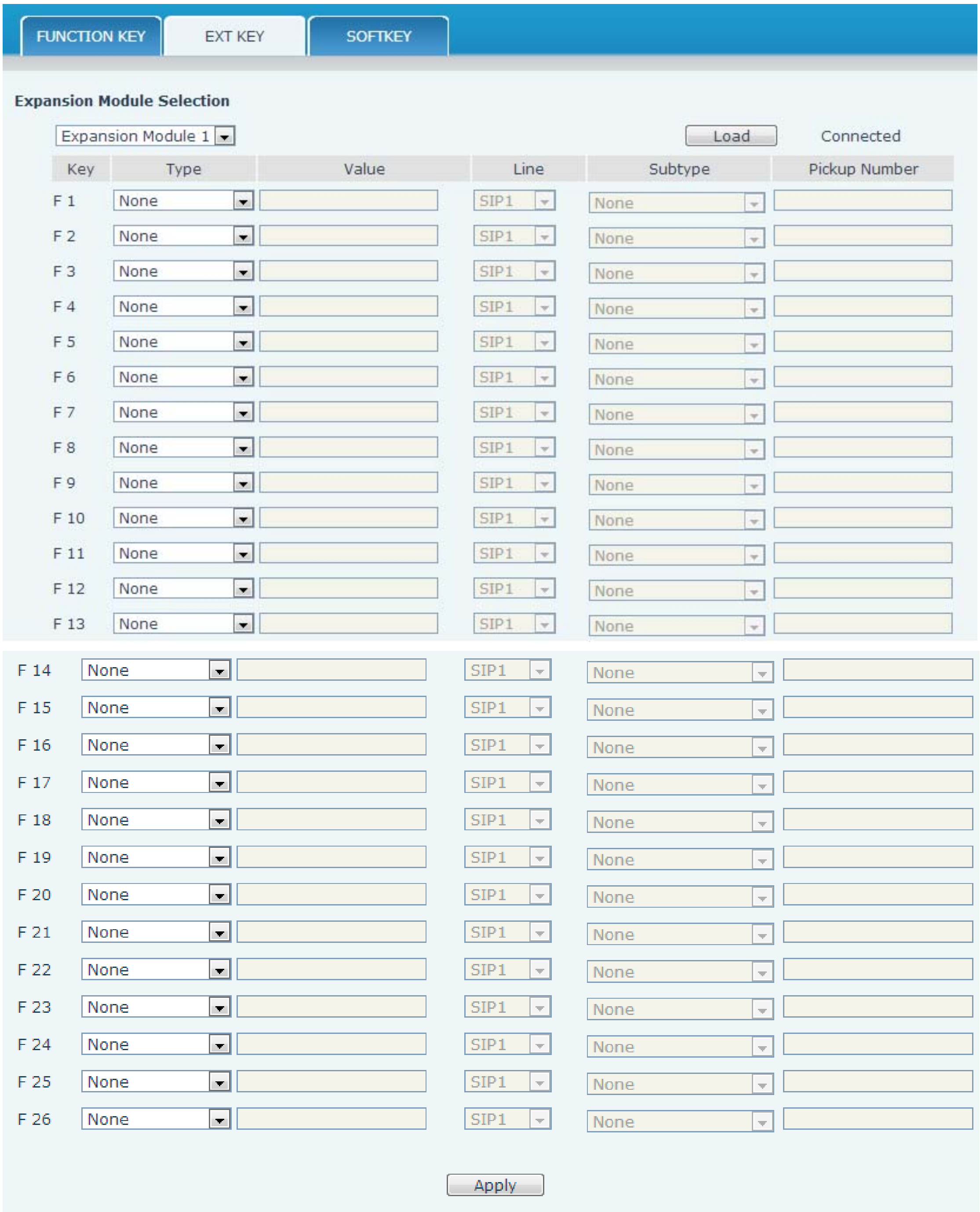
For example:



5.7.2 EXT KEY

EXT key has the same usage with the Function key. The IN port of the extension board is connected to the phone and the OUT port is connected to another extension board. If there is only one extension board, the power supply is not needed; if there are more than one extension board, the first board needs to connect to a 5V power supply through the RJ-45 direct connector.

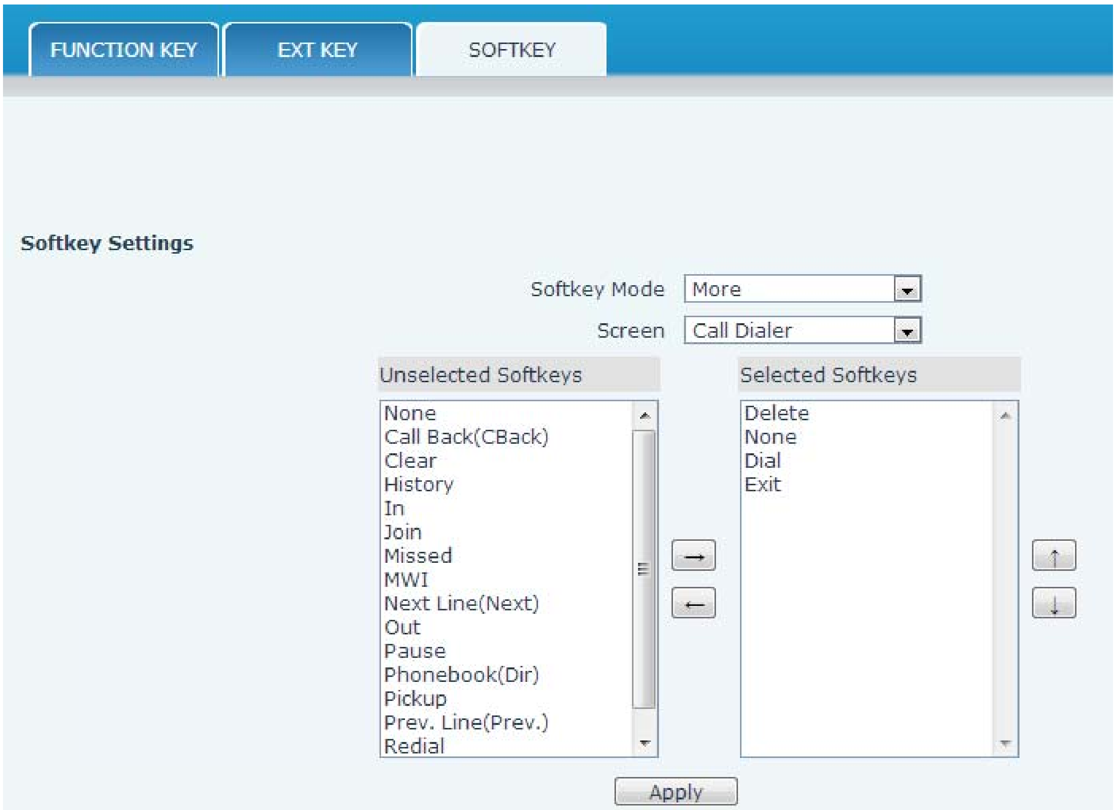
Figure 5-27 EXT KEY interface



5.7.3 Sofykey

You can configure different functions for each softkey in the screen interface.

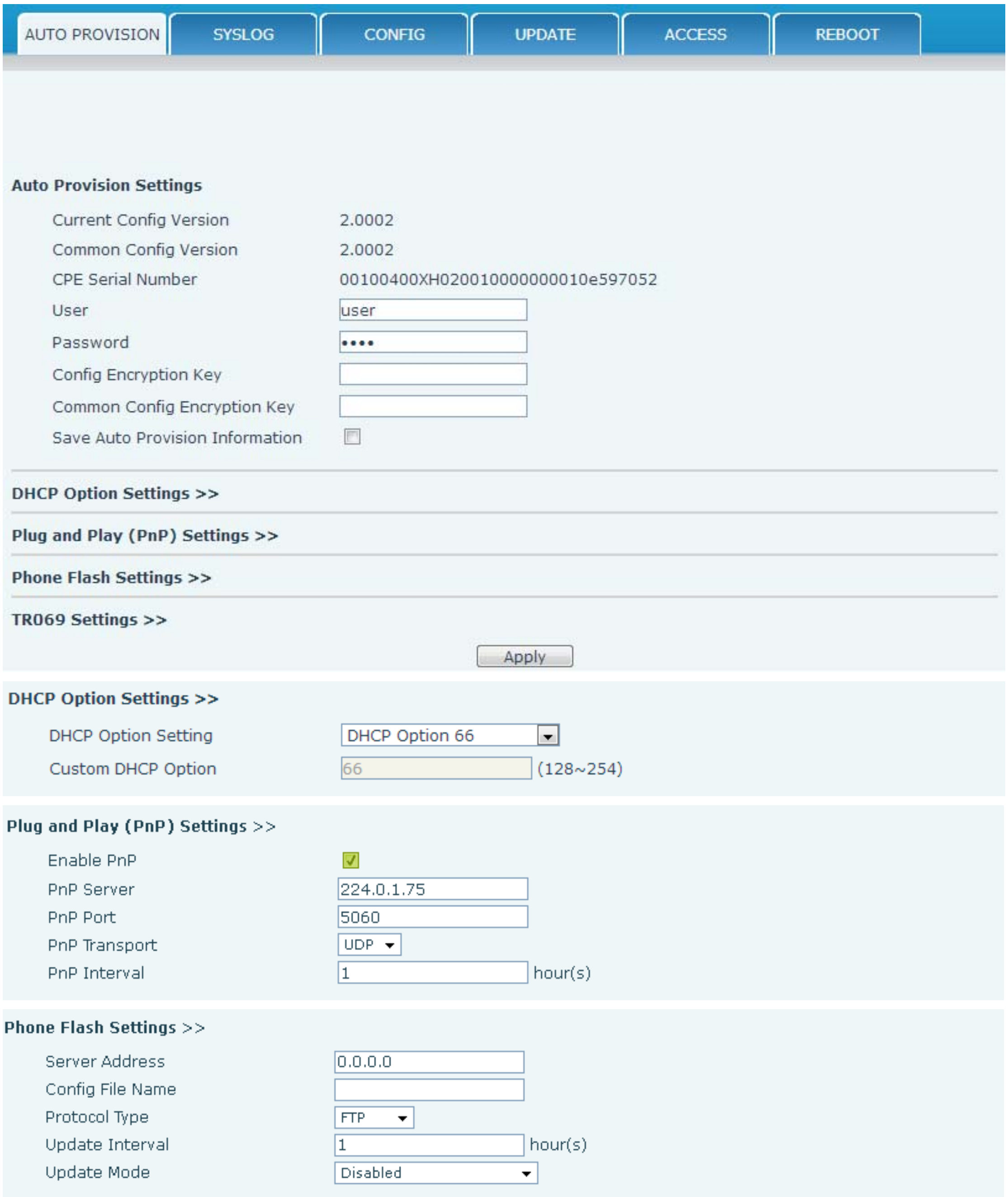
Figure 5-28 Softkey setting interface



5.8 Maintenance

5.8.1 Auto Provision

Figure 5-29 Auto provision setting interface



The NRP1012/P supports PnP, DHCP, and Phone Flash to obtain the parameters. After PnP, DHCP, and Phone Flash are all deployed, the terminal goes through the following process to obtain the server address and other parameters when it boots up:

DHCP option → PnP server → Phone Flash

Table 5-24 Auto provision parameters setting

Field name	Explanation
Auto Update Setting	
Current Config Version	Specifies the version of the current configuration file. If the version of the CFG configuration file downloaded is the same as same as that of the running one, the system does not enable **auto provision**. If the terminal confirms the configuration in Digest mode, the terminal wouldn't upgrade configuration unless the configuration on the server is different from the running configuration.
Common Config Version	Specifies the version of the common configuration file. If the configuration downloaded and the running configurations are the same, the auto provision would stop. If the terminal confirms the configuration in Digest mode, the terminal wouldn't upgrade configuration unless the configuration in the server is different from the running configuration.
CPE Serial Number	Specifies the CPE serial number.
User	Specifies the username of an FTP server If the download protocol is TFTP, this parameter does not need to be configured. If the download protocol is FTP, and no value is set to this parameter, the system uses the default value **anonymous**.
Password	Specifies the password for logging in to the FTP server.
Config Encrypt Key	Inputs a password if the configuration file to be upgraded is encrypted.
Common Config Encrypt Key	Inputs a password if the common configuration file to be upgraded is encrypted.
Save Autoprovision Information	Saves the username, password, authentication and ID information configured for HTTP/HTTPS/FTP. Information saved on the phone remains unchanged unless the URL information configured on the server changes.
DHCP Option Setting	
DHCP Option Setting	Specifies the DHCP option. The value can be DHCP custom option, DHCP option 66, or DHCP option 43. Select any of the preceding modes so that the application parameters can be obtained automatically. By default, the DHCP option is disabled.
Custom DHCP Option	The value ranges from 128 to 254. The custom DHCP option must be identical with the one defined in the DHCP server.
Plug and Play	
Enable PnP	Enables the PnP function. If PnP is enabled, the terminal will send a SIP SUBSCRIBE message to a multicast address when it starts up. Any SIP server supporting SIP SUBSCRIBE messages will reply with a SIP NOTIFY message containing the Auto Provisioning Server URL. The URL can be used to obtain the configuration files to be downloaded to the terminal.
PnP Server	Specifies a PnP server.
PnP Port	Specifies a PnP port.
PnP Transport	Specifies a PnP transfer protocol.
PnP Interval	Specifies the PnP timeout period, in hours.
Phone Flash Settings	
Server Address	Sets FTP/TFTP/HTTP server IP address for auto update. The address can be an IP address or Domain name with subdirectory.
Config File Name	Sets the name of the configuration file which needs to be updated. The system will use the MAC address as the config file name if the config file name is null. For example, 000102030405.
Protocol Type	Specifies the protocol type, which can be FTP, TFTP, HTTP or HTTPS.
Update Interval	Specifies the update interval, in hours.

Field name	Explanation
Update Mode	Specifies different update modes: 1. **Disable** 2. **Update after reboot** 3. **Update at time interval**
TR069 Settings	
Enable TR069	Enables TR069.
ACS Server Type	Specifies the type of an ACS server.
ACS Server URL	Specifies the URL address of an ACS server.
ACS User	Specifies the user for logging in to the ACS server.
ACS Password	Specifies the password for logging in to the ACS server.
TR069 Auto Login	Enable TR069 Auto Login by selecting it.
"Inform" Sending Period	Indicates that the **Inform** message will be sent every six minutes.

5.8.2 Syslog

Syslog is a protocol used to record the log messages in client/server mechanism. The syslog server receives the messages from clients, and classifies them based on the priority and type. Then these messages will be written into log under some rules configured by the administrator .

At present, the system debugging message can be divided into eight levels:

- Level 0: **Emergency**: indicates that the system cannot work. This is the debugging message of the highest level. The default level of the system debugging message is 0.
- Level 1: **Alert**: Your system has a deadly problem.
- Level 2: **Critical**: Your system has a serious problem.
- Level 3: **Error**: indicates that the system will be affected. Level 4: **Warning**: indicates that the system running is not affected but there are some potential dangers.
- **Level** 5: **Notice**: indicates that the system can work properly in certain conditions, but you need to check its working environment and parameter settings.
- Level 6: **Info**: indicates that the system will output daily debugging messages.
- Level 7: **Debug**: the lowest debug info Professional debugging info from R&D person.

At present, the **Info** messages is of the lowest level is . The **Debug** messages can be displayed through **Telnet**.

Figure 5-30 Syslog setting interface

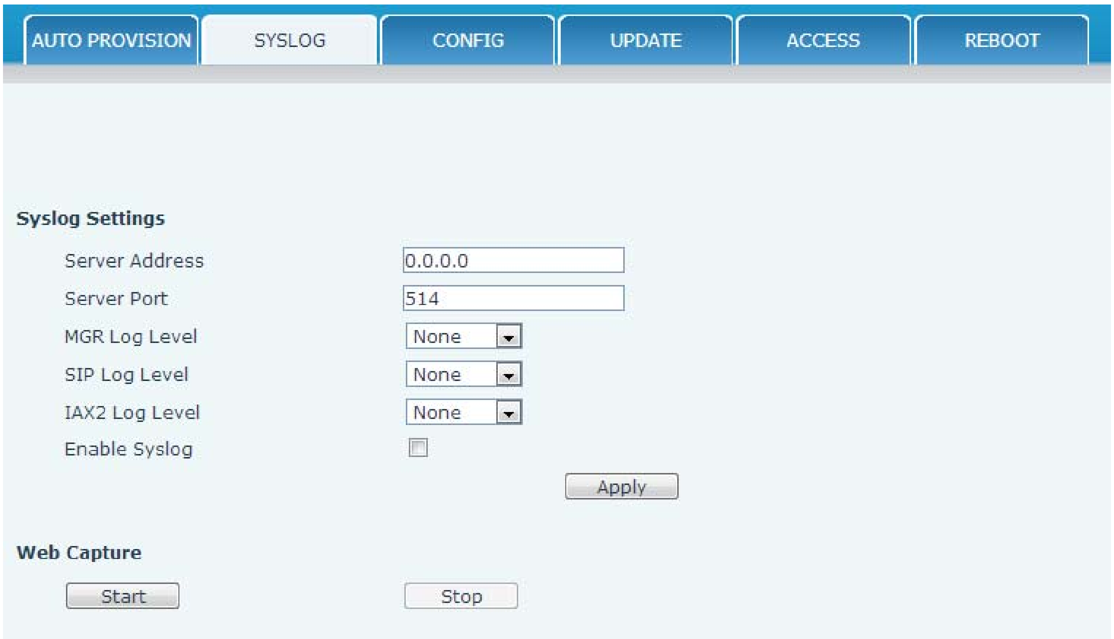


Table 5-25 Syslog parameters setting

Name	Description
Syslog Settings	
Server Address	Specifies the IP address or domain name of the Syslog server.
Server Port	Specifies the port of the Syslog server.
MGR Log Level	Specifies the level of the MGR log.
SIP Log Level	Specifies the of the SIP log.
IAX2 Log Level	Specifies the level of the IAX2 log.
Enable Syslog	Enables or disables the Syslog server.
Web Capture	
Start	Enables the phone to capture packets through the WAN port.
Stop	Disables the phone from capturing packets.

5.8.3 Config

Figure 5-31 Config setting interface

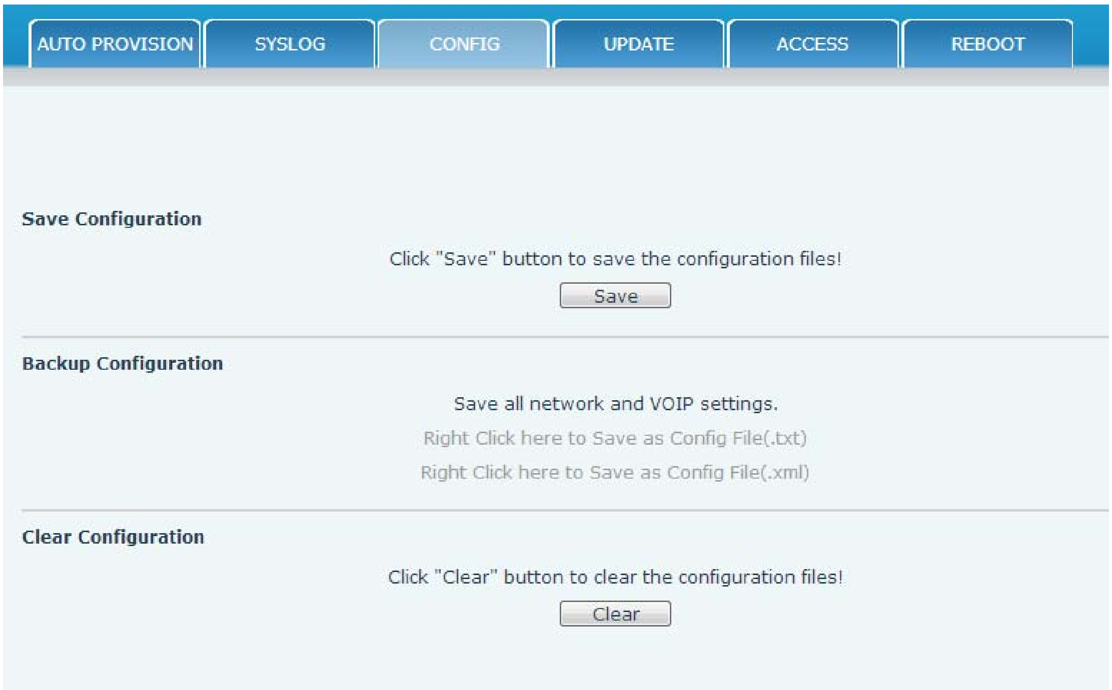


Table 5-26 Congif parameters setting

Name	Description
Save Configuration	Saves the current configuration. Note: You can save all changes of configurations. Click the Save button, all changes of configuration will be saved, and become effective immediately.
Backup Configuration	Backs up the configuration file. You can right click the mouse and click **Save as** to save the configuration file in **.txt** or **.xml** format. You can check the configuration file by clicking it.
Clear Configuration	Restores the factory setting and restarts the phone. **Note:** If you log in to the phone as Admin, the phone will delete all configurations and restore the default factory setting; if you log in to the phone as Guest, the phone will delete all the configurations except for VoIP accounts (SIP1-2 and IAX2) and version number.

5.8.4 Update

In this page, you can configure the phone according to an existing configuration file.

Figure 5-32 Update setting interface

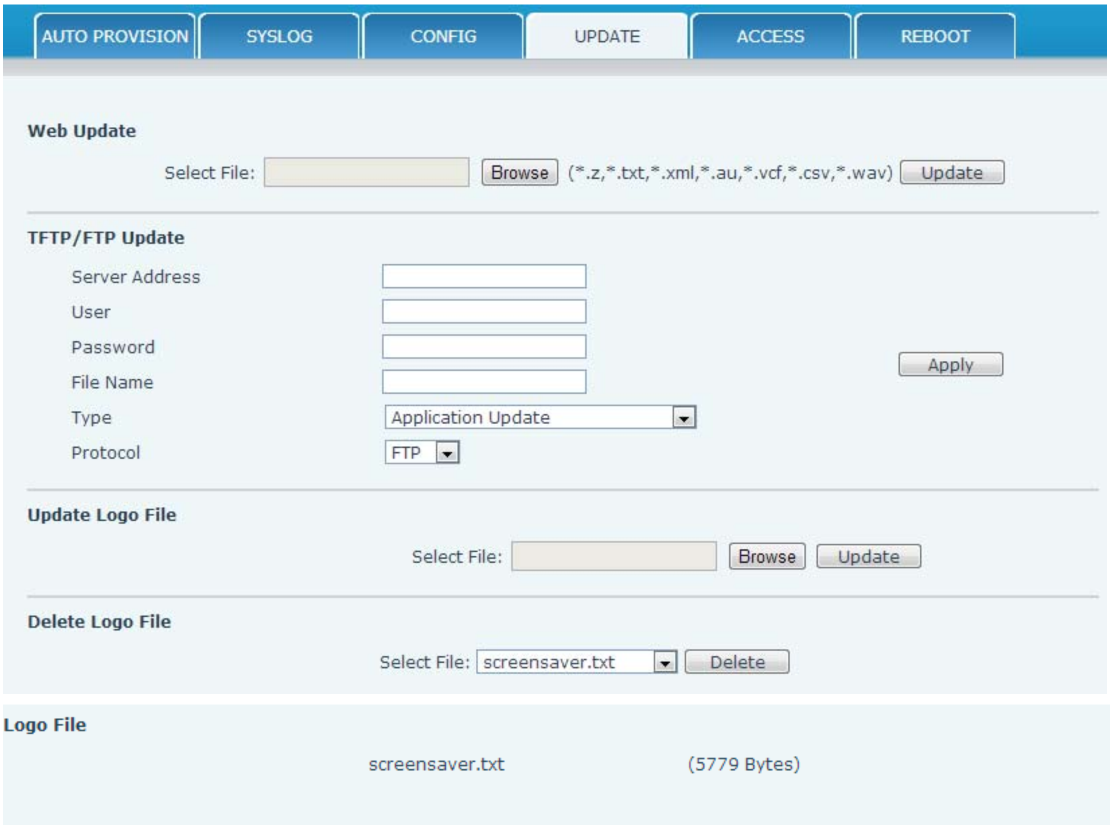


Table 5-27 Update parameters setting

Name	Description
Web Update	
Web Update	Updates the Web page. You can click the **Browse** button, find out the configuration file saved before or provided by manufacturer, and download the configuration file to the phone Alternatively, you can download the update system file of the phone . You can also update downloaded update file, logo picture, ring, mmiset file by web.
TFTP/FTP Update	
Server Address	Sets the FTP/TFTP server address for download/upload. The address can be an IP address or Domain name with subdirectory.
User	Specifies the username for logging in to the FTP server for download or upload. If the TFTP mode is used, the username and password are not required.
Password	Specifies the username for logging in to the FTP server for download or upload.
File name	Specifies the name of the update file or configuration file.
	Notice: You can modify the configuration file exported. And you can also download config file which includes several modules that need to be imported. For example, you can download a config file just keep with SIP module. After reboot, other modules of system still use previous setting and are not lost.

Name	Description
Type	Specifies the update type. The value can be: 1. **Application update**: downloads the system update file. 2. **Config file export**: uploads the configuration file to the FTP/TFTP server, name and save it. 3. **Config fie import**: downloads the configuration file to the phone from the FTP/TFTP server. The configuration will take effect after the phone is restarted. 4. **Phone book export** (*.vcf)
Protocol	Specifies the protocol running on the server. The value can be FTP or TFTP.
Update Logo File	
Select File	Specifies the URL address of the logo file to be updated.
Delete Logo File	
Select File	Specifies the logo file to be deleted.
Logo File	
Logo File	Specifies a logo file.

5.8.5 Access

In this page, you can add or delete a user as needed, change the user level, and configure the keyboard lock function.

Figure 5-33 Access setting interface

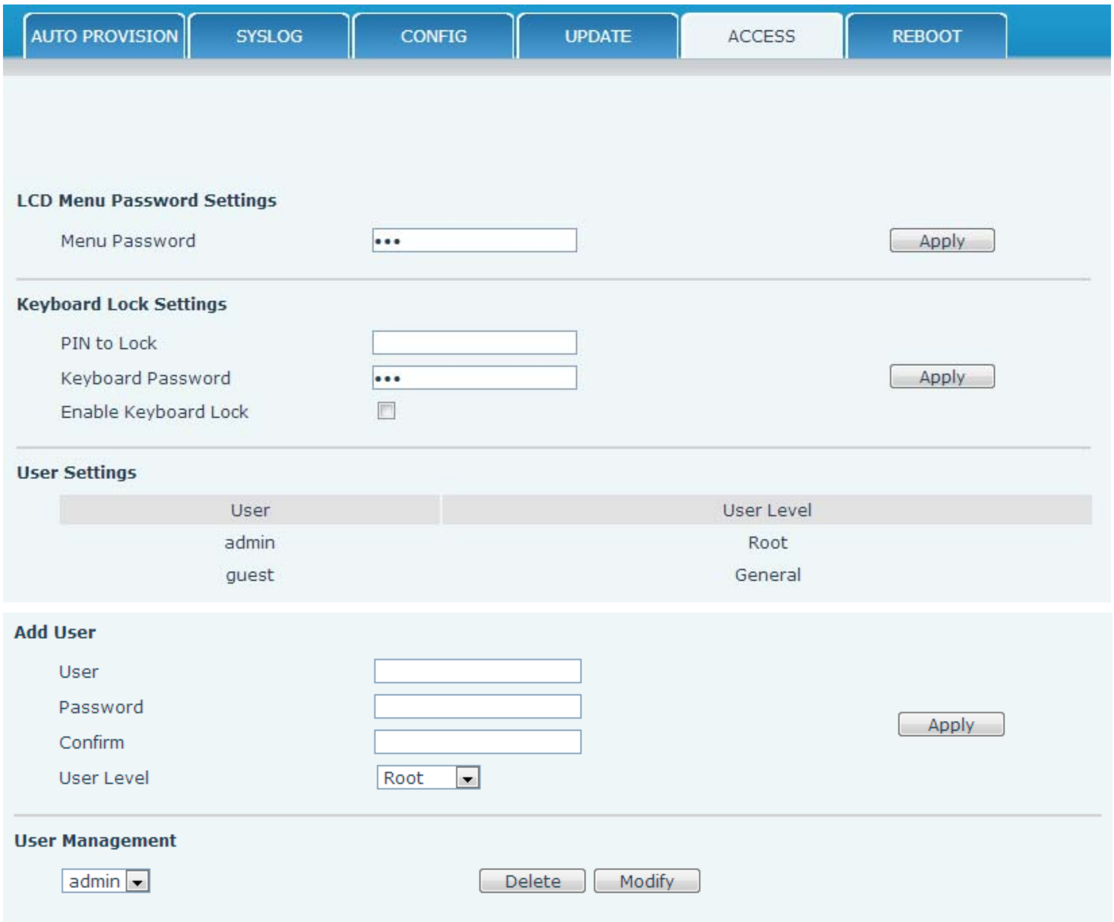
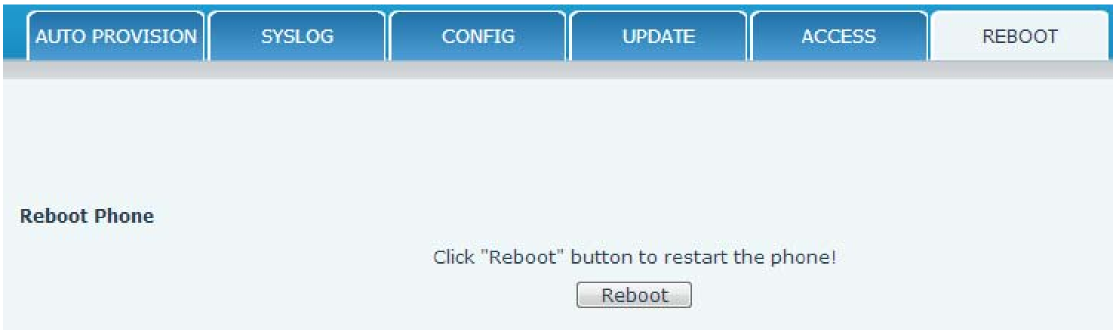


Table 5-28 Access parameters setting

Name	Description
Keyboard Password	Sets the password for entering the setting menu of the phone through the phone's key board. The password is digit.
Keyboard Lock Settings	
User	Sets the account user name.
User Level	Sets the user level, Root user has the right to modify the configuration, General can only read.
Password	Sets the password.
Confirm	Confirms the password.
Selects the account and clicks Modify to modify the selected account. Clicks Delete to delete the selected account. General user only can add the user whose level is General.	

5.8.6 Reboot

Figure 5-34 Reboot setting interface



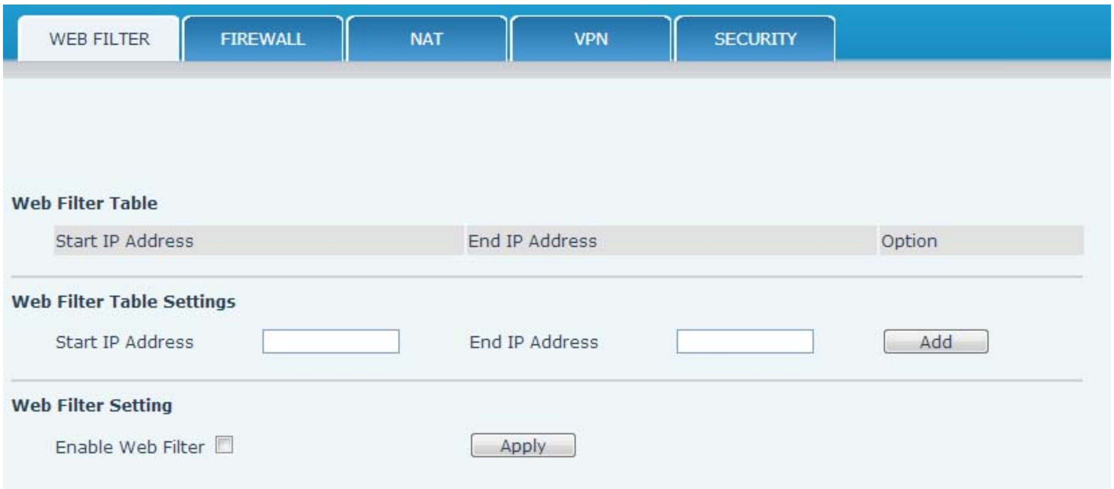
If you modify the phone configuration, you need to reboot the phone by clicking **Reboot** in this page to validate the modification.

Notice: Before reboot, you need confirm that you have saved all configurations.

5.9 Security

5.9.1 Web Filter

Figure 5-35 Web filter interface



In this page, you can configure devices in a certain IP address segment to access the MMI of the phone for configuration and management.

Table 5-29 Web filter parameters setting

Name	Description
Web Filter Table Settings	Adds or deletes the start and end IP addresses. IP addresses within this IP network segment can access the phone. To make the configuration effective, click **Apply**.
Web Filter Setting	Enables the Web filter function on the phone. You can tick off the item and click **Apply** to make the configuration effective.
Notice: Do not set your visiting IP outside the Web filter range, otherwise, you cannot logon through the web.	

5.9.2 Firewall

In this Web interface configure the firewall to prevent unauthorized Internet access (both input and output), improving the network security.

There are two types of rules, input and rules. Each rule is numbered and you can configure a maximum of 10 rules of each type.

Figure 5-36 Firewall setting interface

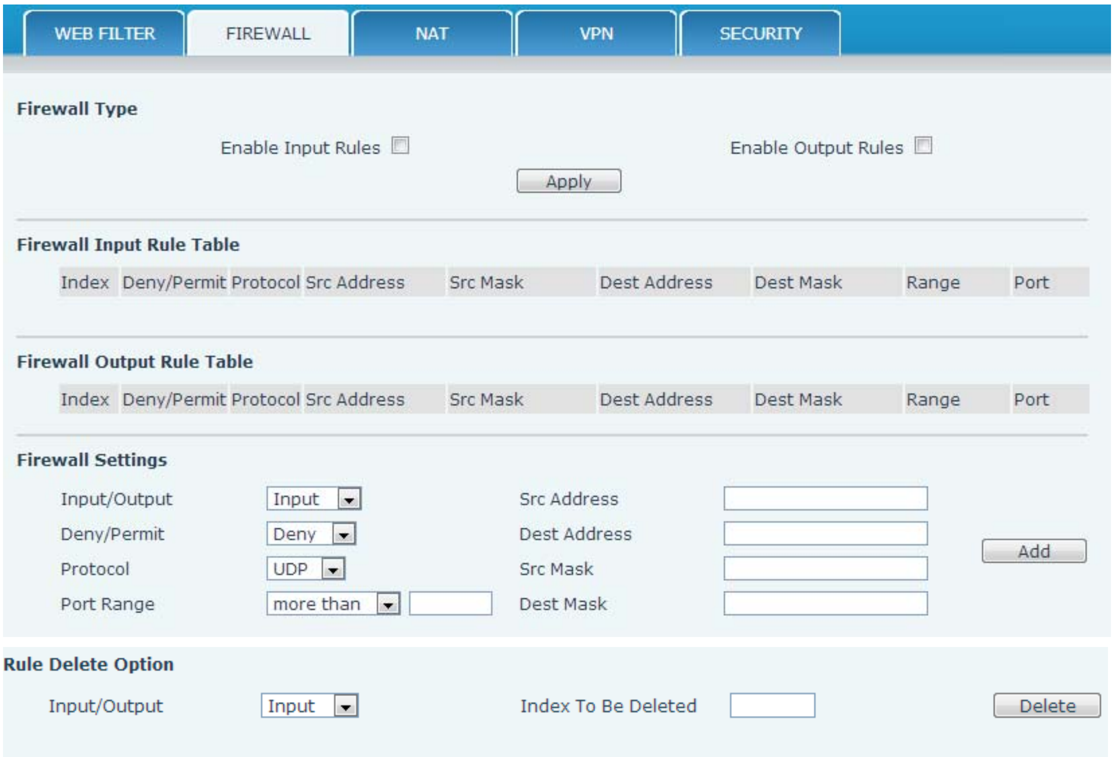


Table 5-30 Firewall parameters setting

Name	Description
Firewall Type	
Enable Input Rules	Enables the application of input rules on the firewall.
Enable Output Rules	Enables the application of output rules on the firewall.
Firewall Settings	
Input / Output	Adds an input or output rule as needed.
Deny/Permit	Permits or rejects the current rule configuration.
Protocol	Specifies the type of a filtering protocol, which can be TCP, UDP, ICMP, or IP.
Port Range	Specifies the range of port number to be filtered.
Src Address	Specifies a source address. The value can be a host address, network address, 0.0.0.0, or an address in the form similar to *.*.*.0 (such as 192.168.1.0).
Des Address	Specifies a destination address. The value can be an IP address, a network address, 0.0.0.0, or an address in the form similar to *.*.*.0 (such as 192.168.1.0).
Src Mask	Specifies the subnet mask of the source address. If the value is set to 255.255.255.255, the access requirement from a specific host will be filtered; if the value is set to 255.255.255.0, the access requirement from devices within a certain network segment will be filtered.
Dest Mask	Specifies the subnet mask of the destination address. If the value is set to 255.255.255.255, the access requirement from a specific host will be filtered; if the value is set to 255.255.255.0, the access requirement from devices within a certain network segment will be filtered.
Add	Adds an input or output rule as needed.

Name	Description
Delete	Deletes an input or output rule as needed.

5.9.3 NAT

NAT Settings

Net Address Translation (NAT) is an IP address translation (IP address mapping) protocol. It is used to transform the private IP address and port to public IP address and port.

DMZ Settings

Some intranet devices need to provide extranet services. To ensure that these devices can provide better services and the internal network is safe, you need to isolate devices with extranet services from others as needed. The DMZ setting help you build a DMZ to protect devices at the network level, reducing the security risk caused by providing service to unauthorized user.

Figure 5-37 NAT setting interface

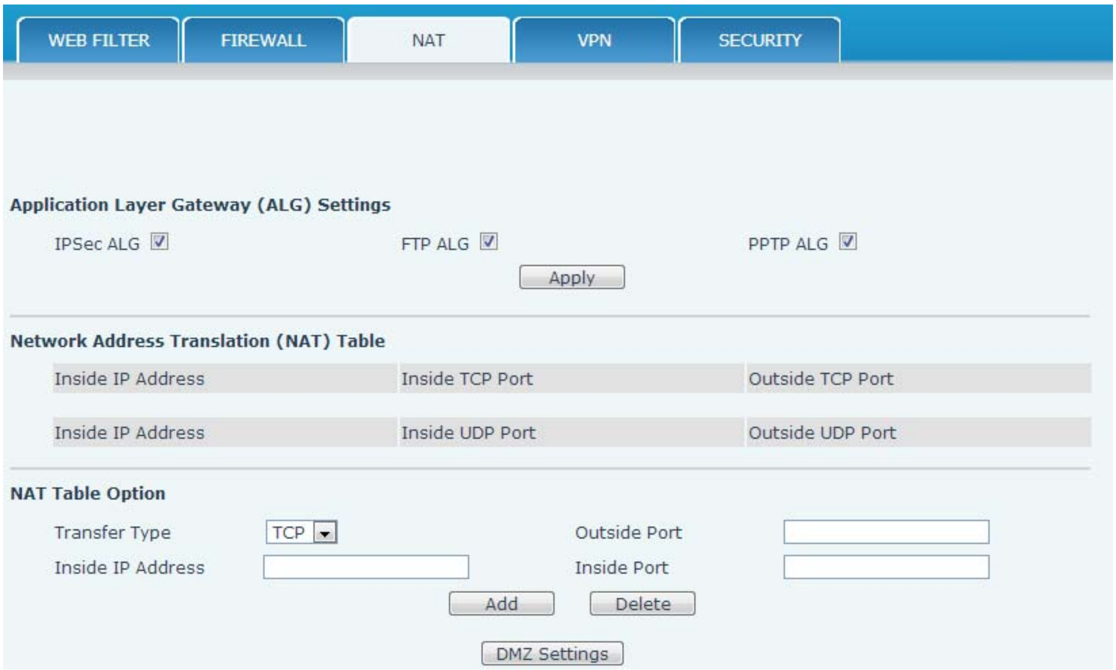


Table 5-31 NAT parameters setting

Name	Description
Application Layer Gateway (ALG) Settings	
IPSec ALG	Enables/disables IPSec ALG, an encryption/decryption technology. By default, IPSec ALG is enabled.
FTP ALG	Enables/disables FTP ALG, a connection-layer service used to transform the intranet IP of the packet into an extranet IP. By default, FTP ALG is enabled.

Name	Description
Application Layer Gateway (ALG) Settings	
PPTP ALG	Enables/disables point-to-point tunneling protocol (PPTP) ALG. By default, PPTP ALG is enabled.
NAT Table Option	
Transfer Type	Specifies the type of the protocol for NAT mapping, TCP or UDP
Outside Port	Specifies the WAN port for NAT mapping
Inside IP Address	Specifies the LAN IP address for NAT mapping.
Inside Port	Specifies the LAN port for NAT mapping
Note: After the setting, you can click Add to add a new mapping entry and click Delete to delete a mapping entry.	
DMZ Settings	Sets the LAN IP address (such as 192.168.10.23) corresponding to the WAN IP address (such as 192.168.1.119).
Note: 10M/100M adaptive means the physical consultation speed of the network card, and other equipment. The testing speed in bridge mode is near to 100M. The voice quality and real-time performance of the communications is ensured at the cost of the NAT transmission performance. The system performs the best-effort transmission only when being idle. Therefore, a100M transmission speed cannot be ensured.	

5.9.4 VPN

In this Web page, you can configure a safe remote access to the enterprise intranet network from the public network. That is to say, you can use a special tunnel to connect public networks in different areas to an inner network.

Figure 5-38 VPN setting interface

Table 5-32 VPN parameters setting

Name	Description
VPN IP	Specifies the IP address of the current VPN.

Name	Description
Selects L2TP. You can choose only one for current state. After you select it, you'd better save the configuration and reboot your phone.	
Enable VPN	Enable or disable VPN.
Layer 2 Tunneling Protocol (L2TP)	
VPN Server Address	Specifies the IP address of a VPN L2TP server.
VPN User	Specifies the username for logging in to the VPN L2TP server.
VPN Password	Specifies the password for logging in to the VPN L2TP server.

5.9.5 Security

Figure 5-39 Security setting interface

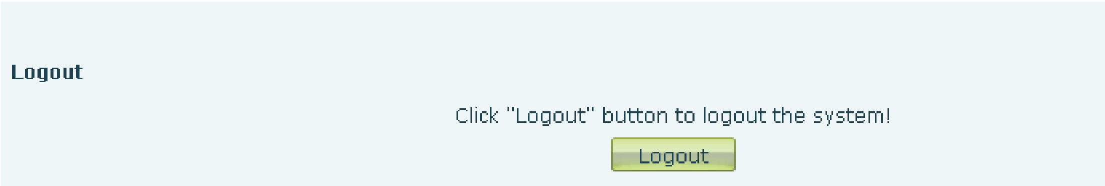
Table 5-33 Security parameters setting

Name	Description
Update Security File	
Select Security File	Specifies the security file to be updated. You can select a file and click the **Update** button.
Delete Security File	
Select Security File	Specifies the security file to be deleted You can select a file and click the **Delete** button.
SIP TLS File	Specifies the SIP TLS authentication certification file.
HTTPS File	Specifies the HTTPS authentication certification file.
Open VPN Files	Specifies the Open VPN File authentication certification file.

5.10 Logout

You can quit the Web page by clicking **Logout**. For the next access, you need to input the username and password again.

Figure 5-40 Logout setting interface



6 Appendix

6.1 Voice Features

The NRP1012/P can provide the following voice features:

- Six SIP servers
- SIP 2.0 (RFC 3261) and the corresponding RFCs
- IAX2
- Three SIP lines (users can simultaneously register to three SIP server and choose any one of them to carry out call-in and call-out services)
- Multiple call queuing
- IAX2 line key-based call
- Different codec modes: G.711A/u, G.723.1, G.729a/b, G.722, and G.726-32
- HD voice
- Echo cancellation (in support of G.168, with acoustic echo cancellation (AEC) reaching 96ms in hands-free mode)
- Voice Gain Setting, VAD, CNG
- Full duplex hands-free
- SIP domain, SIP authentication (in none, basic, or MD5 mode), DNS name of server, P2P IP call
- Different DTMF types: SIP info, DTMF Relay, and RFC 2833
- Eight DSS keys
- Different program modes :soft keys and function keys
- Customization of different languages (the default language is English)
- Different SIP applications: Call forward / transfer (blind transfer / attended transfer / Ringing Transfer) / Call hold / call waiting / conference call / paging and intercom / call park / then grab / interpolation / Automatic Callback / Click call / auto secondary dial /
- Flexible call control functions: flexible dialing, support hotline number, calling reject, reject blacklist, certification calls, white list barring, do not disturb, speakerphone automatic answer, caller ID, anonymous calls, outgoing calls etc.
- A maximum of 100 incoming calls / outgoing calls / missed calls records, respectively.
- A maximum of 500 contacts.
- SMS
- XML phonebook/browser
- Speed dial
- SRTP

- BLF
- Code synchronization via IP PBX/IMS
- Dialup via web phone book
- Voice codec setting for each SIP line
- Keypad lock, and emergency call during the keypad lock
- Customized lcd logo
- Selection of headset, speakerphone ringing mode
- Customized ringing tone setting
- Group listening

6.2 Network Features

The NRP1012/P can provide the following network features:

- Bridging and routing mode for WAN/LAN port
- Basic NAT and NAPT
- PPPoE for xDSL
- VLAN (Voice VLAN/ Data VLAN)
- NAT penetration (STUN penetration)
- DMZ
- VPN (L2TP/OPEN VPN) function
- Primary and secondary DNS server for the WAN port, dynamic obtaining of the DNS address in DHCP mode, or static configuration of the DNS address
- Configuration of the DHCP client on the WAN port
- Configuration of the DHCP server on the LAN port
- QoS with DiffServ
- Network tools including ping, trace route, and telnet client

6.3 Maintenance and Management

The NRP1012/P can provide the following maintenance and management functions:

- Application update in POST mode
- Web, Telnet and keypad management
- Management with different account right
- Language selection through LCD and WEB configuration, and dynamic multi-language shifts
- Software update and configuration file setting through HTTP, FTP or TFTP
- Syslog

- Auto provisioning (automatic configuration update and system maintenance)