



# User Manual FIP11/FIP11W/FIP11WP

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# **About This User Guide**

Thank you for choosing FlyingVoice FIP11/FIP11W/FIP11WP. The FIP11/FIP11W/FIP11WP which has 8 lines is a full-featured VoIP (Voice over Internet Protocol) phone that provides voice communication over an IP network.

This phone functions not only much like a traditional phone, allowing to place and receive calls and enjoy other features that traditional phone has, but also it own many data service which you could not expect from traditional telephone.

This guide will help you easily use the various features and services available on your phone.

The FIP11/FIP11W/FIP11WP, which has a 3dB internal antenna for 802.11n 1T1R, supports wireless.







#### This guide contains the following:

- Contact with FlyingVoice
- Purpose
- · Cross reference
- · Feedback
- · Product Declaration of Conformity
- · Warnings and cautions

# **Contact with FlyingVoice**

Main website: http://www.flyingvoice.com/

Sales enquiries: sales1@flyingvoice.com

Support enquiries: support@flyingvoice.com

Hotline: 010-67886296 0755-26099365

Address: Room508-509, Bldg#1, Dianshi Business Park, No.49 Badachu

Rd, Shijingshan District, Beijing, China

# **Purpose**

The documents are intended to instruct and assist personnel in the operation, installation and maintenance of the FlyingVoice equipment and ancillary devices. It is recommended that all personnel engaged in such activities be properly trained.

FlyingVoice disclaims all liability whatsoever, implied or express, for any risk of damage, loss or reduction in system performance arising directly or indirectly out of the failure of the customer, or anyone acting on the customer's behalf, to abide by the instructions, system parameters, or recommendations made in this document.

### **Cross references**

References to external publications are shown in italics. Other cross references, emphasized in blue text in electronic versions, are active links to the references.

This document is divided into numbered chapters that are divided into sections. Sections are not numbered, but are individually named at the top of each page, and are listed in the table of contents.

### **Feedback**

We appreciate feedback from the users of our documents. This includes feedback on the structure, content, accuracy, or completeness of our documents. Send feedback to support@flyingvoice.com.

# **Declaration of Conformity**

# **CE** certification

This device complies with the EU Directive 2014/35 / EU and the EMC Directive 2014/30 / EU.

### **FCC** certification

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

- This device may not cause harmful interference.
- This device must accept any interference received, including interference that may cause undesired operation.

### **Class B Digital Device or Peripheral**

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment can generate, use and radiate radio frequency energy. If not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. However, there is no guarantee that interference does not occur in a particular installation.

#### Note

Changes or modifications not expressly approved by the party responsible for compliance could void

the user's authority to operate the equipment.

If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interferences by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

# Warnings, cautions, and notes

The following describes how warnings and notes are used in this document and in all documents of the FlyingVoice document set.

### **Warnings**

Warnings precede instructions that contain potentially hazardous situations. Warnings are used to alert the reader to possible hazards that could cause loss of life or physical injury. A warning has the following format:

#### Warnings

Warning text and consequence for not following the instructions in the warning.

#### **Notes**

A note means that there is a possibility of an undesirable situation or provides additional information to help the reader understand a topic or concept. A note has the following format:

#### Note

Caution text and consequence for not following the instructions in the caution.

# **Chapter 1 Product description**

#### This chapter covers: :

- FIP11/FIP11W/FIP11WP
- Internet connection
- · Familiar with the phone

# FIP11/FIP11W/FIP11WP

Port/Model FIP11 FIP11W FIP11WP

#### **Product Picture**

	2.8 inch color LCD	2.8 inch color LCD	2.8 inch color LCD			
LED	screen	screen	screen			
	320 * 240 resolution	320 * 240 resolution	320 * 240 resolution			
Wi-Fi	×	ſ	I			
Line	8	8	8			
Ethernet Port	2*RJ45	2*RJ45	2*RJ45			
Ethernet Port	10/100Mbps	10/100Mbps	10/100Mbps			
Headset	RJ9	RJ9	RJ9			
USB	1	1	1			
Line Port	×	×	×			
Soft key	4	4	4			
Multi-function Key	8	8	8			
PoE	×	×	J			
Code	G.711(A-law,U-law),G.723,G.729A/B,G.722,iBLC					
TR069	J	ſ	ſ			
SNMP	J	ſ	J			
VPN		PPTP/L2TP/Open VPN				
Auto configuration	TPTP/HTT	P/HTTPS Tr069 Network ma	anagement			
Telephone book	J	ſ	ſ			
Call history	Ţ	ſ	ſ			
SMS	J	ſ	ſ			

# **Internet connection**

### **Phone Start**

Please refer to the quick installation manual for the telephone assembly Step 1. AC power adapter is plugged into an electrical outlet.

Note

Step 2. After blogging in the power, and the LED in the upper right corner of the phone will light up (red)

Step 3. When the phone has started normally, LED will go out and LCD will display the current status of the phone, including SIP registration information, network connection information and so on.

Note

If the phone does not display the above information, please re-confirm the installation and connection. If these operations are ineffective, you can try unplugging the power plug and plug, it in again. If you still can not see the show,

#### Connect to a wired network

- Step 1. Check the phone connection correctly and start successfully.
- Step 2. Connect one end of cable in the set to the internet port on the back of the phone, another end of the cable conectted to the available network adapter.
- Step 3. After the connection , the upper right corner of the LCD screen will appearwired network'logo,lke this: that means phone have been connected to a wired network for Internet access.

### Connect to a wireless network

You can configure a wireless connection in both the LCD and the web.

From the LCD menu:

Step 1. Press button and then use the ▲ and ▼ buttons or enter the number 12 to "12. Wireless Network".

Step 2. Select "2. Wireless network connection" and the phone will scan and display the surrounding wireless network.

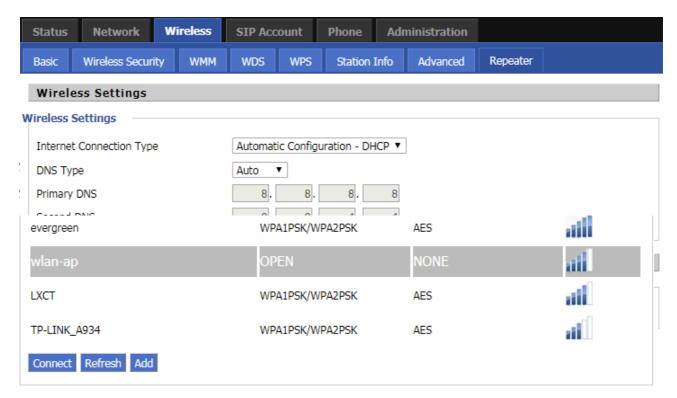
Step 3. Using the ▲ and ▼ buttons to navigate, use the softkey "Connect" connection under the LCD to select the network, the logo appears on the LCD, The line network is connected.

Notes

If you need Wi - Fi certification, please fill in the authentication and password.

#### From the page:

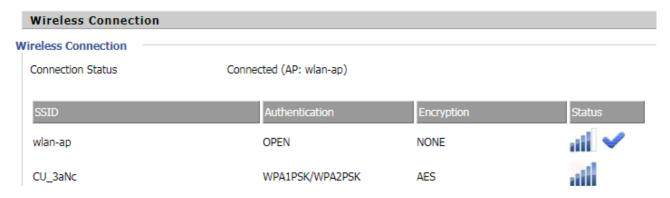
Step 1. Log in to the website and switch to the Network / Wireless page.



Step 4. If the connected wireless does not have a password, you can connect directly; if the wireless connection has a password, enter the password and click "OK" to confirm the connection.

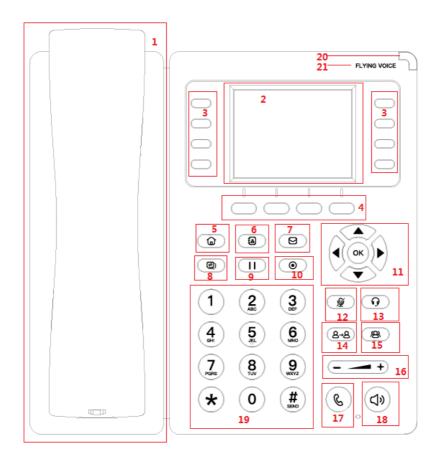


Step 5. If AP is connected, the wireless icon on the main screen of the LCD will appear as connected



# Familiar with the phone

# Front panel



No.	Name	Description
		answer the phone
		Pick up or hang up
	Handle	Handle bracket, to prevent the vertical installation of the phone when the handle drop
1		RJ-11 telephone line one end interface on the left side of the IP phone
		RJ-11 telephone line one end interface at the bottom of the IP phone handle
		Handle, handle handle included
2	LED	The LCD screen is used to display the status of the phone, such as the IP address of the Internet port. Phone number and line status
3	Multi-function key/Line key	1. These keys can be used for multi-function keys. For details, refer to Chapter 4, Multifunctional Key 2. Can also be used as a line key, do line button to use, select the appropriate line call to answer the phone.  The corresponding LED area is used to display the status of the current key. For details, refer to the manual section 3 LCD display
4	Soft key	You can select or control the operation items displayed below the LCD screen. The function of the soft keys depends on the contents of the corresponding LCD display at that time.
5	Main menu	login the main menu
6	Phonebook	phone book shortcuts
7	Voicemail	Voicemail shortcuts
8	Horizontal page	Horizontal switch screen shortcuts. The phone supports the key expansion board, multi-function keys can be increased to up to 32
9	Hold	To hold the current call, press the Hold key again to restore the current call
10	Recording	One-click recording (currently does not support recording function)

11	Navigation key	<ol> <li>Navigation keys, including up, down, turn left, right turn and confirm keys</li> <li>Turn left and right turn is also the volume of the shortcut keys</li> <li>The confirmation key is also a shortcut to view the current network status</li> </ol>		
12	Mute / delete	<ol> <li>When the phone is on, press to switch to silent mode</li> <li>When entering text or numbers, this key is used to delete a character or number from right to left</li> </ol>		
13	Headphones	Switch to headset mode		
14	Call Transfer	Call transfer: blind transfer and attended transfer		
15	3 way Conference	3 way Conference, press this key during the call to start the 3 way conference		
16	Volume button	Increase or decrease the volume		
17	Redial	Quickly dial the number of the last call		
18	Handsfree	Switch to handsfree mode		
19	Dial key	Used to make a phone call or the corresponding configuration		
20		he color		
Later				
	LINE O-C-O			

### As shown in the picture, from left to right:

Name	Description
LINE	Connect the telephone line
DC 5V	Power port
USB	Connect the USB device
PC	Connect PC interface
Internet	Connect switches, routers, the Internet

# **LED** indicator

Color	Status	Description	
0	Not shiny,off	<ul><li>1.The phone is in power off / idle / busy state</li><li>2.The call is in the hold / mute state</li></ul>	
•	Red is always bright	The phone is in an initialized state	
•	Red flash (300 milliseconds / time)	The phone is in a ringing state (any line)	
•	Red slow flash (1 second / time)	The phone receives a short message or voice message or line registration is successful	

# **LED** status



lcon	Description
09:45	Time: Displays the current time of the phone, the format is: minutes
	Wired network connection status
কি	Wireless network connection status
✓*	Missed call
	New message
A	Auto answer
$\Theta$	DND
<b>Q</b>	Call mute

8	Handle pick up / dial / call
€	Call forward: all froward;busy forward;no answer forward
1234	The screen number, the current screen is 1
<b>3</b>	The front line is already registered
<b>3</b>	The line registration failed
<u>&amp;</u>	The line goes off-hook / call
<b>©</b>	Ringing
history pbook status DND	4 soft keys The current functional description

# **Chapter 2 Basic functions**

#### This chapter contains the following:

- · SIP registration
- · Dial number
- · Answer the phone
- · Mute the call
- · Hold
- · 3-way conference
- · Call transfer
- · Call forward
- Call waitting
- · Auto Answer
- · DND
- · End the call
- · Blacklist
- · SMS
- · Voicemail

# **SIP** registration

There are two ways to register a SIP account: register from the LCD and register from the web page From the LCD:

Step 1. Press



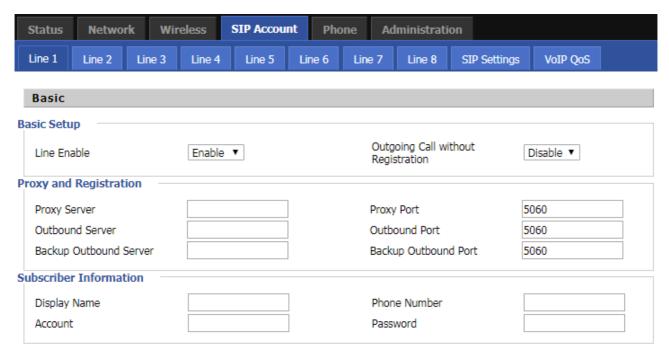
button.

Step 2. Use the ▲ and ▼ navigation keys or enter the number 7 to 7. Account, you need to enter the login password.

- Step 3. Select the line to register and fill in the relevant configuration as indicated by the LCD
- Step 4. Use the second soft key from the left to "save" the current configuration.
- Step 5. Check the registration of the corresponding line on the display.

#### From the web page:

- Step 1. Log in to the web page. Please refer to section 4.1 for details.
- Step 2. Switch to the SIP Account configuration page.



- Step 3. The account enable is set to "On" and the line can be used after opening.
- Step 4. The registration server fills in the IP address of the SIP server.
- Step 5. Display Name Fill in the content is the name of the number displayed on the LCD.
- Step 6. The registration account is filled with the account provided by the SIP server.
- Step 7. The name of the authentication is the SIP account provided by the SIP server.
- Step 8. The password is filled with the password provided by the SIP server registration account.
- Step 9. When you are finished, click the Save button at the bottom of the page to make the configuration take effect.
- Step 10. Check the registration of the corresponding line on the display / web status page.

**Notes** 

Step 3-9 is to fill in the required content, other parameters fill in the required

# Dial number

You can make a call by either of the following methods:

#### Use the handle

- Step 1. The handle goes off-hook, the LCD display shows "Enter number"
- Step 2. Press the dial key to enter the destination number.
- Step 3. Use the "Dial" / # / wait for 5 seconds to dial the phone number.

### Use handsfree

- Step 1. Press the handsfree off-hook, the LCD display shows "Enter the number"
- Step 2. Press the dial key to enter the destination number.
- Step 3. Use the "Dial" / # / wait for 5 seconds to dial the phone number.

### Use headphones

- Step 1. Press the headset key to go off-hook, the LCD display shows "Enter number"
- Step 2. Press the dial key to enter the destination number.
- Step 3. Use the "Dial" / # / wait for 5 seconds to dial the phone number.

# Use the redial key

In the standby mode, redial the number of the last call:

- Step 1. Press the "History" softkey to quickly select the call log.
- Step 2. Use the softkey to select "Redial List".
- Step 3. Select a phone number and press the "Dial" softkey to redial the called number.

### Call from the phone book

#### Add phone book:

#### From the LCD menu:

Step 1. Press button to enter the main menu and use the softkey to select 1 phone book. (or you can just press second softkey "pbook" to enter phoneebook.)

Step 2. Follow the prompts to add a phone book and press the "abc" softkey to toggle the input method (numbers, uppercase and lowercase).

#### Use the phone book:

Step 1. Press lacktriangle button to enter the menu item and use the lacktriangle and lacktriangle navigation keys or enter

the number 1 to **1. Phonebook**; Or use the phonebook shortcut keys in the standby mode; Or use

multi-function phone button to enter the phone book

Step 2. Use the  $\triangle$  and  $\nabla$  keys to select the number you want to dial, press the "Call" softkey to make a call immediately.

### Call from call log

Step 1. Press button to enter the main menu and use the ▲ and ▼ navigation keys or enter

the number 2 to **2. Call history**,Or in the standby or dial-up interface, use the softkey to enter the **"history"** 

Step 2. Select the dialed number of the dialed call / missed call / missed call. Press the "dial" softkey to dial the call immediately.

# **Answer the phone**

When there is a call, the LED in the upper right corner of the phone will flash and the phone rings. You can answer the call in the following ways:

- 1. Press the corresponding line button.
- 2. Press the Handsfree key to answer.
- 3. Pick up the handle and answer.
- 4. Press the earphone key to answer.
- 5. Press "answer" softkey to answer.
- 6. Auto Answer: If the phone is enabled for automatic answering, the device will automatically answer the call when there is an incoming call.

### Call mute

Mute mute: During a call, press button, the MIC will be deactivated, the handle is available,

and the icon is displayed on the screen. This can prevent the caller from hearing your or your background sound.

Unmute: Press button again, the icon disappears and the sound can be sent normally.

### Hold

When the phone is talking:

Step 1. When A and B talk, A presses the HOLD button to keep the current call, and then the A handset can hear the dial tone, and B will play keep music.

Step 2. At this point the A phone can enter another phone number to make a call.

Step 3. A Press the HOLD button again to release the current hold status and resume the previous call.

# 3-way Conference

Step 1. Start a conference call, A and B phone have a call, during the call, A phone press button, the current call is held, A phone to hear the dial Tone, B phone play to keep music.



Step 2. Dial the phone number of the C telephone.

Step 3. When the C phone answers the call, the A telephone presses the "conference key" as the conference presenter to open the conference call. Same conference call.

Step 4. If the A phone (conference host) hangs up, the other two calls will be disconnected; if the

Step 4. If the A phone (conference host) hangs up, the other two calls will be disconnected; if the non-hosting party B / C side hangs first, A phone can still talk to the other party.

# Call transfer

### Attended call transfer

#### **Functional Description:**

Attended call transfer: the phone as a middle side of attended call transfer, after asking the destination phone, then it will make the call which is connectting transferred to the destination phone.

#### How to Use:

Step 1. A and B phone call, A phone button. The current call is held, A phone to hear the dial tone, B phone to keep playing hold music.

Step 2. A call C.

Step 3. C phone answers, speaks to A (and A asks if C want to connect with B).

Step 4. If C agrees to answer the transfer call, the A phone presses the "Transfer" button of use softkey to complete the transfer. Then A will be disconnect from all call.

Step 5. If C don't agree to answer the transfer call, after A / C hangs up after the call, A presses



#### Blind call transfer

#### **Functional Description:**

Blind call transfer: the phone as a middle side of blind call transfer, will be connected directly to the destination phone without asking.

#### How to Use:

Step 1. A and B phone is making a call, A presses the "transfer" button or soft key during the call, A phone will hear the dial tone, B phone will hear hold music.

Step 2. A call C.

Step 3. C answer the call, then talking to B, A will automatically hang up.

# **Call Forwarding**

### **All Forwarding**

#### **Functional Description:**

Transfer all calls to another number. Can be configured from an LCD or Web page **From the LCD**:

Step 1. Press button.

Step 2. Use the  $\triangle$  and  $\nabla$  navigation keys or enter the number 8 to 8. Call Forward and select one of them

#### 1. CFWD All.

Step 3.Fill the relevant configuration according to LCD instructions: whether open all forward; target number; on code; off code.

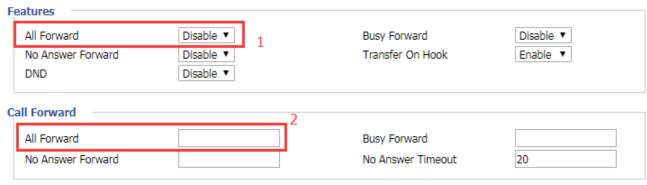
Step 4. Use the second soft key from the left to "save" the current configuration.

#### From Web page:

Step 1. Log in to the phone web page and switch to the "Phone - Preferences" page.



Step 2. There is a call item in the middle of the page.



- Step 3. Enable all forward at position 1 and fill the destination number at position 2.
- Step 4. At the bottom of the page, click 'Save' / 'Save& Apply', and the configuration is complete.

#### **How to Use**

- Step 1. The A handset assumes that the unconditional transfer has been configured and the target number is the C telephone.
- Step 2. B Telephone Calls a telephone
- Step 3. The C ringing, off-hook and B calls. A phone in the process without any reaction.

### **Busy Forward**

#### **Functional Description:**

When the line is busy, transfer all calls to another number. Can be configured from LCD or Web page From LCD:

Step 1. Press



Step 2. Use the ▲ and ▼ navigation keys or enter the number 8 to 8. Call Forward and select

#### 2.CFWD When Busy.

Step 3. Fill the relevant configuration according to LCD instructions: whether to open the busy forward; target number; on code; off code.

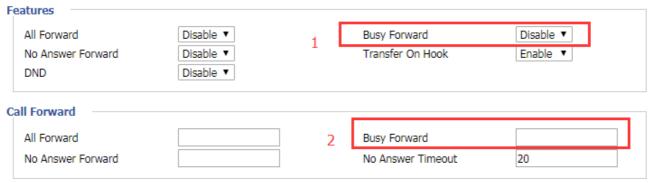
Step 4. Use the second soft key from the left to "save" the current configuration.

#### From Web page:

Step 1. Log in to the phone web page and switch to the "Phone - Preferences" page.



Step 2. There is a call item in the middle of the page.



- Step 3. Enable the busy forward at position 1 and fill destination number at position 2.
- Step 4. At the bottom of the page, click 'Save' / 'Save&Apply', and the configuration is complete.

#### How to use:

- Step 1. The A phone's busy forward has been configured successfully, and the target phone is C.
- Step 2. The A phone is calling with other phone (not B / C).
- Step 3. B Calls A.
- Step 3. Then C ringing, off-hook to make a connect with B, all this process A phone without any response.

#### No answer Forward

#### **Functional Description:**

When no one answers, the call is routed to another number. Can be configured from an LCD or Web page

#### From LCD:

Step 1. Press button.

Step 2. Use the ▲ and ▼ navigation keys or enter the number 8 to 8. Call Forward and select

#### 3.CFWD When No Answer.

Step 3. Fill the relevant configuration according to LCD instructions: whether to open the no answer forward; target number; on code; off code.

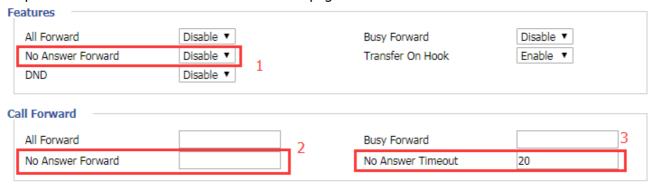
Step 4. Use the second soft key from the left to "save" the current configuration.

#### From Web page:

Step 1. Log in to the phone web page and switch to the "Phone - Preferences" page.



Step 2. There is a call item in the middle of the page.



- Step 3. Enable the no answer forward at position 1 and fill destination number at position 2. Potion 3 is no answer to the forward time, default 20s.
- Step 4. At the bottom of the page, click 'Save' / 'Save&Apply', and the configuration is complete. How to use:
- Step 1. The A handset assumes that no forwarding is configured and the destination number is the C telephone.
- Step 2. B Telephone Calls a telephone.
- Step 3. A phone rings, but nobody answers.
- Step 4. After 20 seconds, the handset stops ringing and the C phone rings and goes off-hook with B.

# **Call Waiting**

#### **Functional Description:**

Call waiting function: During the call, if a third party calls, the phone has a prompt, and the third party phone has been ring back tone.

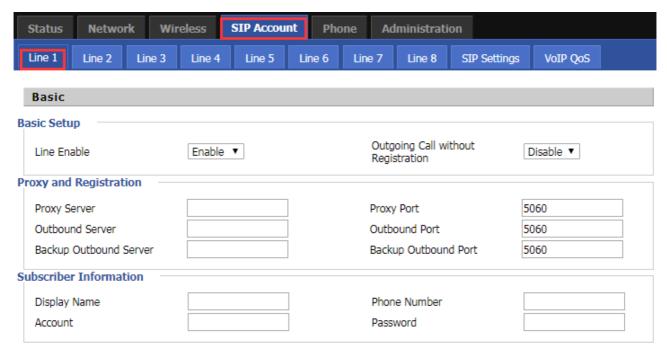
#### From LCD:

Step 1. Press button

Step 2. Use the  $\triangle$  and  $\nabla$  navigation keys / enter the number 5 to the **5. Perference** settings and select the **2.call waiting.** 

Step 3. Fill the relevant configuration according to LCD instructions: whether to enable call waiting. From Web page:

Step 1. Log in to the web page and switch to the 'VolP-Line 1' Configuration page.



Step 2. There is a call waiting conifiguration item in the 'Supplementary Service Subscription' on this page.



Step 3. At position 1, you can configure whether to enable the call waiting function.

#### How to Use

- Step 1. The A phone call waiting function is configured.
- Step 2. C calls A When A is connectting with B.
- Step 3. A phone LED flashes, there will be a tone in the handle (prompted a new call), the screen will display a new call.
- Step 4. When prompted with a new call, use the appropriate line key or hold button to answer the new call C and hold the call with B.
- Step 5. When there is a new call, A ends the call with B, and C is still calling A, then ringing, picking up and talking to C.
- Step 6. If the call waiting function is off, then after C dials A, it will prompt the call to fail and have a busy tone.

### **Auto Answer**

#### **Functional Description:**

Auto Answer function: automatic answer when there is a call.

#### From LCD:

Step 1. Press button.

Step 2. Use the  $\triangle$  and  $\nabla$  navigation keys or enter the number 5 to the **5. Perferencess** settings and select the **3 Auto Answer**.

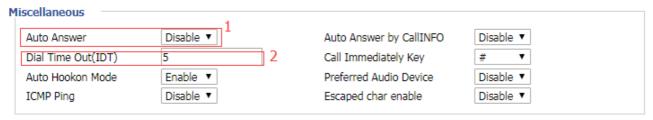
Step 3. Fill the relevant configuration according to LCD instructions: whether to enable auto answer

#### From Web page

Step 1. Log in to the web page and switch to the "Phone - Preferences" configuration page.



Step 2. At the bottom of the page, there is auto answer configuration item.



Step 3. At position 1, you can configure whether to enable the auto answer function.; position 2 is configured to wait for a few seconds after the automatic answer, for example setting 10, phone will auto answer after ringing 10s .

#### How to Use:

- Step 1. The A phone auto answer function is configured and the auto answer delay time is 10s.
- Step 2. B Calls A.
- Step 3. After 10s, the A will automatically answer the call.

### **DND**

#### **Functional Description:**

DND:Do not disturb, reject all calls.

#### From LCD:

Step 1. In the main screen of the phone, from left to right the fourth software is DND softkey, use it, the screen

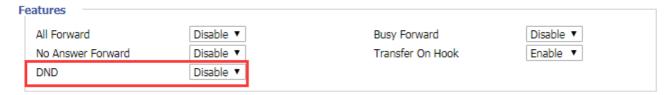
will appear a free play DND icon that means DND is enabled.

#### From Web page:

Step 1. Log in to the web page and switch to the "Phone - Preferences" configuration page.



Step 2. In the middle of the page there is a DND configuration item.



Step 3. You can configure whether to enable DND at position 1.

#### How to use:

- Step 1. The A phone DND is enabled, the icon displayed on the screen
- Step 2. B calls A.
- Step 3. The A phone doesn't have any respond, and the B phone prompts the call to fail and has a busy tone.

# End the call

Method 1: To end the handle call, hang up.

Method 2: To end the handsfree call, press the Handsfree key.

### **Blacklist**

#### **Functional Decription:**

Blacklist: If a number is blacklisted, the phone will block incoming calls to the number.

#### From LCD:

Step 1. Press buttons.

Step 2. Use the  $\triangle$  and  $\nabla$  navigation keys or enter the number 4 to select 4 Blacklist.

Step 3. Select "New Entry" and follow the prompt to add a blacklist.

#### From Web page:

Step 1. Log in to the web page and switch to the "Phone - Phonebook" configuration page.



Step 2. At the bottom of the page there are blacklist uploads and add the configuration.



第3步. 上传黑名单时, 注意文件的格式为.csv文件

- 4	А	B	C	D	E	F	G	H	I
1	NAME	NUMBER							
2	A	123							
3									
4									
5									
6									
7									
8									

Step 4. When adding a blacklist, click the "Add" button and follow the prompts to add a blacklist.

Blacklist				
Index	Name	Number	-	
Name				
Number	OK Cancel			

Step 5. Click "Save&Apply" / "Save" to make the configuration take effect

# **SMS**

# Send text messages

Step 1. Press buttons,

Step 2. Use the  $\triangle$  and  $\nabla$  navigation keys or enter the number 3 to 3 SMS.

Step 3. Select 1. SEND and follow the prompts to edit the text message.

Notes

When the input text, according to 'abc' softkey to switch input method (number or letters),'123' means number;'abc' means lowercase letters;'ABC' means capital letters

Step 4. Press "OK" softkey to enter the recipient's phone number.

Note

SMS has group messagefunction, user can send a message to up to 10 numbers; press the "OK" softkey to enter the next received phone number input, if the recipient has entered the contact, contact press twice to "complete"; you can also press the "directory" in the phone book to find the phone number.

Step 5. Set the sender's phone number, the default setting is set to Line1 phone number.

Step 6. Select the "select" softkey to send text messages.

### Read text messages

Step 1. When there is a new message, the phone screen will have a new message prompt, you can use the softkey to read the message.

Step 2. Also you can press



outton

Step 3. Use the  $\triangle$  and  $\nabla$  navigation keys or enter the number 3 to 3 SMS.

Step 4. select 2 Recived Box, or 3. Send Box.

#### Note:

The message in the Recived Box received from someone else, including new or old messages. Send Box is a sent message

Step 5. Use the  $\triangle$  and  $\nabla$  navigation keys to select the message and press the "read" softkey to read the message.

# Voicemail

Note:

The registration server needs to configure the relevant functions first.

#### From LCD:

Step 1. Press



button to enter the menu item and then select 10. Voice Mail.

Step 2. Enter the login password and select the line you want to configure.

Step 3. Then follow the prompts to enable voicemail, and enter the number in the "Voice Mail Number" (this number is the number configured in the registration server code, not free to fill).

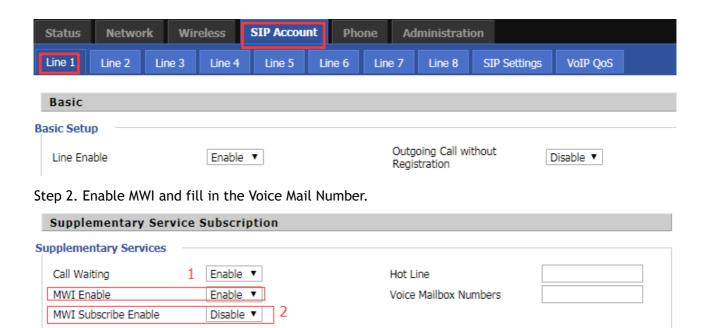
Step 4. Press the Save soft key to save your changes.

#### Note:

Voice mail number is when there is a voice message, off-hook dial to enter the number of voice mail, and this number is offered from registration server.

#### From Web page:

Step 1. Log in to the phone page and switch to the 'Phone / Line1' Configuration page.



Step 3. Use the 'Save&Apply' / 'Save' button to save your changes for the configuration to take effect.

#### How to Use:

- Step 1. When there is a voice message, the LED in the upper right corner of the phone will flash.
- Step 2. Press voice mail button to enter the mailbox
- Step 3. Also you can use 'Voice Mail Number': phone goes off-hook and dials the Voice Mail Number. Step 4. After entering the voicemail box, there will be a voice prompting how to operate. The user can listen to new voicemail or old voicemail or reply to voice mail based on voice prompts.

# **Chapter 3 LCD Configuration**

#### This chapter contains:

- · Change language
- · Change ringtone
- · Change the date and time
- · Reboot
- · Restore Factory
- · Other parameters of the phone

# Change language

Step 1. Use

button, then select **5 Preferences**.

- Step 2. Use the ▲ and ▼ navigation keys or enter the number 5 to select 5 Preferences.
- Step 3. Select 1. Language, use the "select" softkey to enter the language selection interface
- Step 4. Use the  $\triangle$  and  $\blacktriangledown$  keys to select the language and use the "select" softkey to save the selected language.

# **Change ringtone**

- Step 1. Use button and then select 6 Ring Tones.
- Step 2. Use the ▲ and ▼ navigation keys or enter the number 6 to select 6 Ring Tones.
- Step 3. Use the  $\triangle$  and  $\nabla$  keys to select a ringtone.
- Step 4. Use the softkey "select" to save the selected ringtone.

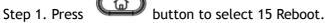
# Change the date and time

Usually the phone's date and time is server automatically set the current date and time. If you need to change the date and time, you can follow these steps:

#### Set the date and time:

- Step 1. Press button to select 9 Time / Date.
- Step 2. Select the date adjustment date and enter the date format: MM / DD / YY.
- Step 3. Select the time to adjust the time; enter the time format:  $HH \ / \ MM \ / \ SS.$
- Step 4. Press the "OK" softkey to save the changes.

### Reboot



Step 2. Press the "select" button to restart.

Step 3. Then there will be "Confirm System Reboot?" Prompt, press the "select" softkey to confirm the restart.

# **Factory Reset**

You can restore the factory settings on the LCD menu or on the phone's page. This setting is used when configuring and troubleshooting the network. Please do not easily recover, if necessary or need to contact the administrator.

#### From LCD:

- Step 1. Press button to select 16 Factory Reaet.
- Step 2. Press the "Select" softkey to select the factory settings.
- Step 3. Then confirm the prompt, press the "OK" softkey to continue, the device automatically restore the factory settings and restart.

# Other parameters of the phone

Main menu (LCD display)	Menu item (LCD display)	Submenu item (LCD display)	Functional description
	New Entry		using four soft keys, you can add numbers to the phone book.and up to 100 records can be saved
1.Phonebook	Contact		<ol> <li>Select a contact to view the contact number directly.</li> <li>Use the four soft keys can dial someone, sending text messages, editing, viewing, deleting, moving to blacklist</li> </ol>
	Redial List		1. List the last 60 calls
2.Call History	Answered Calls		2. Use the four soft keys to complete the quick call, send text messages, save to the
z.can mistory	Missed Calls		phone book and other functions
	SEND		editor and send text messages
3 . T e x t	Recived Box		can list the latest send and receive 100
Message	Send Box		messages
	New Entry		With four soft keys, you can add numbers to the blacklist, Up to 100 records can be saved
4.Black List	Contact		<ol> <li>Select a blacklist contact to view the contact number directly</li> <li>Use the four soft keys to complete the call, send text messages, edit, view, delete, move to the phone book and other functions</li> </ol>
	Language		Select a different language
	Call Waiting		enable or disable call waiting
5.Preference	Auto Answer		enable or disable auto answer
	Preferred Audioa Device		Choose whether to use the hands-free or headset when answering automatically
	BellType 1		
	BellType 2		
	BellType 3		
	BellType 4		
	BellType 5		
	BellType 6		Select a ringtone type, you can listen to the
6.Ring Tone	BellType 7		ringtones, easy to distinguish and choose the
-	BellType 8		ringtones
	BellType 9		
	BellType 10		
		Line Enable	Enable or disable lines
		Password	The password for the SIP account

7.Accounts	Line1-8	Account	The account number of the SIP account
		Display Name	The display name on the LCD after registration is successful
		Phone number	SIP phone number
		SIP Proxy Server	SIP server IP address / domain name
		SIP Proxy Port	SIP server port
		O u t b o u n d Server	Proxy server IP address / domain name
		O u t b o u n d Server port	Proxy server port
		Backup Outbound Server	Backup Outbound Server IP Address / Domain Name
		B a c k u p Outbound Port	Backup Outbound server port
8.Call Forward	CWFD All	CFWD All	Enable or disable all forward
		Target Number	Set the target phone number for all forward
		On Code	Used to enable the all forward
	CFWD When Busy CFWD When No Answer	Off Code	Used to disable the all forward
		CFWD When Busy	Enable or disable busy forward
		Target Number	Set the target phone number for busy forward
		On Code	Used to enable the busy forward
		Off Code	Used to disable the busy forward
		CFWD When Busy	Enable or disable no answer forward
		Target Number	Set the target phone number for no answer forward
		CFWD No Ans Delay	Set the waiting time for no answer forward
		On Code	Used to enable the no answer forward
		Off Code	Used to disable the busy forward
9.Time/Date	Time(H:M:S)		Set the current time of the phone
	Date(M/D/Y)		Set the current date of the phone
10.Voice Mail	Line 1-8	MWI Enable	Enable or disable voice mail notification
		Voice Mail Number	Set the key to enter the voicemail number

	W A N Connection Type		<ol> <li>1.View or change the current Internet port connection type,</li> <li>2. Internet port connection type can be set to static, DHCP and PPPoE</li> </ol>
	Current IP		Change the IP address of the current Inernet port
	C urrent Netmask		Change the subnet mask of the current network,
	C u r r e n t Gateway		Change the gateway of the current network
	Primary DNS		Change the Primary DNS of the current network
11.Network	Secondary DNS		Change the Seconary DNS of the current network
	Enable WAN Login		Enables or disables users to log on to the device web page from the Internet port
	Web Port		View or change the Web port
	SIP QoS		View or change SIP QoS
	RTP QoS		View or change RTP QoS
	Data QoS		View or change data QoS
	VLAN Tag		Enable or disable VLAN
	VLAN ID		View or change the VLAN ID
	802.1p Priority		View or change the priority of 802.1p
	Wireless Setting	Wifi Country Region	View or change the current Wifi area
		W i f i Connection Type	<ol> <li>View or change the Wifi connection type</li> <li>Currently supports DHCP and static types</li> </ol>
		Current IP	View or change the IP address of the current wireless network
12.Wireless		Current Netmask	View or change the subnet mask for the current wireless network
		Current Gateway	View or change the current wireless network gateway
		Primary DNS	View or change Primary DNS for the current wireless network
		Primary DNS Secondary DNS	wireless network
	Wireless Connection	-	View or change the Secondary DNS of the

13.Product INFO	S o f t w a r e Version H a r d w a r e Version MAC Address Serial Number		To view the current information of Product Name, Software Version, Hardware Version and MAC Address.
	Internet Port Status		View the current network information, including: WAN connection type, IP address, subnet mask, default gateway, Primary DNS,Second DNS, WAN port status, WIFI connection status
	VPN Status		View VPN Status: VPN Status, VPN IP Address
14.Status	Registration Status	T h e registration status of lines 1-8	View the current registration status of lines 1-8
15.Reboot			Reboot Phone
16.Factory Reset			To set phone factory default.
1 7 . S e t Password			To reset password. The password of LCD is same as the one of Webpage. Default is null.
18.Login/ Logout			Press it to logout LCD.  Phoen will memory the password if user has input the password when access to 9Accounts and phone will not request the password if user access to it again.  Press 20Login/Logout to erase memory, and then user should input password when user access to 9Accounts again.

# **Chapter 4 Web Configuration**

## This chapter contains:

- · Login
- · SIP account
- · Line 1
- · Network
- · Phone
- · Dial Plan
- · Management

# Login

There are two modes to login Web:

- (1) admin mode: the default user name is admin, the password also is admin, you can view and configure the Web interface, all settings.
- (2) user mode: the default user name is user, password also is user, can only view and configure part of the device parameters

### Login steps:

Step 1. Connect the phone correctly and make sure that the device and the computer in the same

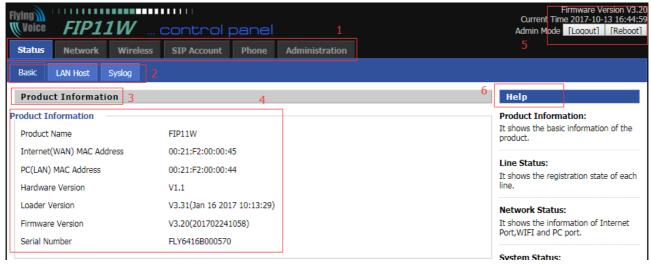
network.

Step 2. View the device IP: Enter the main menu, use the ▲ and ▼ navigation keys or enter the number 14 to select the 14 status, the select network view the device IP; you can also press the device IP.

Step 3. Enter "http: // IP address of the phone" in the address field of the computer browser and press Enter to display the device login page.



Step 4. Enter 'admin' or 'user' as the user name and enter the appropriate password. Click the "Login" button to enter the configuration page.



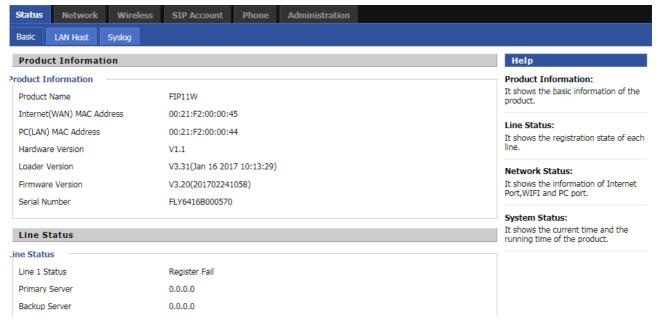
Serial number	Name	Description
Postition 1	Main navigation bar	Click this navigation bar to bring up the corresponding child navigation bar
Postition 2	navigation bar	Click the sub navigation bar to enter the configuration page
Postition 3	Product Information	Device Information Configuration Title
Postition 4	Product Information	Show product information
Postition 5	Login/Logout	main information shows the firmware version, DSP version, current time and management mode.
Postition 6	Help	help to display help information, users can get some help here
	Save & Apply	Use this button, conifg will be saved and apply

Save	After changing the parameters, you need to click this button to save. After you click Save, there is a need to restart the device.
Cancel	Click to cancel the change
Reboot	Click to restart
Refresh	Refresh current page

# Status

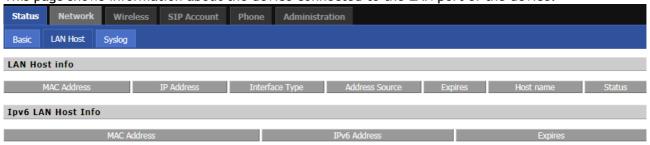
# **Basic**

This page shows the basic status of the phone: including product model, SIP account registration status, network status, VPN status, PC port status and system status. Click the Refresh button to refresh.



# **LAN Host**

This page shows information about the device connected to the LAN port of the device.



# **Syslog**

This page displays the system log: the user can use the 'Clear' button to delete all logs, clear all information; use the 'Refresh' button to refresh the system log; Use the 'Save' button Save Log to

the local computer to export the



# **SIP Account**

This page includes: account settings, SIP Settings and VoIP QoS settings, the following picture is the navigation bar:

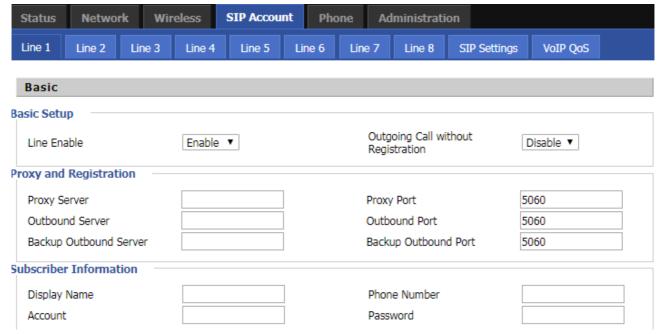


# **Account setting**

The user can configure the parameters of line 1-8 in this page, including the following four parts: Basic, Audio Configuration, Supplementary Service Subscription and Advanced.

#### **Basic**

Set up your VoIP service provider to provide basic information such as phone number, account number, password, SIP proxy, and so on.



Parameter name	Description
Line Enable	Whether to enable the corresponding Line
Proxy Server	SIP server domain name or IP
Poxy Port	SIP server supports VoIP service port, the default is 5060
Outbound Server	Proxy server IP or domain name
Outbound Port	The service port of the proxy server
Backup Outbound Server	Back up the proxy server
Backup Outbound Port	Back up the proxy server port
Display Name	This value will be displayed on the LCD screen
Phone Number	The phone number provided by the SIP server
Account	SIP server provided by the SIP server

**Audio Configuration**Select audio encoding format:

ec Setup			
udio Codec Type 1	G.711U ▼	Audio Codec Type 2	G.711A ▼
Audio Codec Type 3	G.729 ▼	Audio Codec Type 4	G.722 ▼
Audio Codec Type 5	G.723 ▼	Audio Codec Type 6	G726-32 ▼
Audio Codec Type 7	iLBC ▼		
G.723 Coding Speed	5.3k bps ▼	Packet Cycle(ms)	20 ▼
Silence Supp	Disable ▼	Echo Cancel	Enable ▼
Auto Gain Control	Enable ▼	Use First Matching Vocoder in 2000K SDP	Disable ▼
Codec Priority	Remote ▼	Packet Cycle Follows Remote SDP	Disable ▼

Parameter name	Description
Audio Codec Type 12,3,4,5,6,7	Select the corresponding encoding mode from G. 711A, G.711U, G.722, G.729 and G.723 five encoding modes
G.723 Coding Speed	Select the encoding rate of G.723, there are two kinds of 5.3kbps and 6.3kbps
Packet Cycle(ms)	Set the RTP packaging cycle, the default configuration is 20ms
Silence Supp	Whether it is enabled.
Echo Cancel	Whether to enable echo cancellation, the default is enabled
Auto Gain Control	Whether the automatic gain control is activated, the automatic gain control is the automatic control method that automatically adjusts the gain of the amplifier circuit with the signal strength
Use First Matching Vocoder in 2000K SDP	Whether to use the first matching speech code
Codec Priority	Coding priority strategies include local and remote
Packet Cycle Follows Remote SDP	Whether the packaging cycle to the main end

# **Supplementary Service Subscription**

# Supplementary Service Subscription Supplementary Services Call Waiting Enable ▼ Hot Line MWI Enable Enable ▼ Voice Mailbox Numbers MWI Subscribe Enable Disable ▼

Parameter name	Description
Call Waiting	Whether to enable call waiting
Hot Line	Fill in the hotline number. After the user is set up, go off-hook and immediately dial the hotline number automatically

MWI Enable	Whether to enable MWI (message waiting indication), if the user needs to use voice mail, please enable the function
Voice Mailbox Numbers	Fill in the SIP server to provide voice mail signature code to Elatix platform, for example, the voice mailbox signature * 97
MWI Subscribe Enable	Whether to enable MWI subscription

## **Advanced**

#### Advanced SIP Advanced Setup Enable ▼ Carry Port Information Disable ▼ Domain Name Type 56049 Signal Port DTMF Type Inband RFC2833 Payload(>=96) Register Refresh Interval(sec) 3600 101 • Caller ID Header FROM Remove Last Reg Enable ▼ UAC ▼ Session Refresh Time(sec) Refresher SIP 100REL Enable Disable ▼ SIP OPTIONS Enable Disable ▼ Disable ▼ Initial Reg With Authorization Reply 182 On Call Waiting Disable ▼ NAT Keep-alive Interval(10-15 Disable ▼ Anonymous Call 60s) Anonymous Call Block Disable ▼ Proxy DNS Type A Type Disable ▼ Use OB Proxy In Dialog Disable ▼ Complete Register Rea Subscribe Enable Disable ▼ Reg Subscribe Interval(sec) Dial Prefix User Type Phone ▼ Hold Method ReINVITE ▼ Request-URI User Check Enable ▼ Only Recv Request From Disable ▼ Server Address Server Disable ▼ VPN Disable ▼ SIP Received Detection Disable ▼ RTP Encrypt Type Disable ▼ SIP Encrypt Type Country Code Remove Country Code Disable ▼ Tel URL Use Random SIP Port Enable ▼ Disable ▼ 50000 Max Random SIP Port Min Random SIP Port 60000 Prefer Primary SIP Server Disable ▼ Hold SDP Attribute Inactive Disable ▼ BLF List URI BLF PickUp Code RTP Advanced Setup (0 50000 RTP Port Min RTP Port Max means auto select)

Domain Name Type Whether to enable domain name recognition in SIP URIS Carry Port Information Whether to carry the SIP URI port information Signal Port The local port number of the SIP protocol  DTMF Type Select the second way of dialing, optional items are In-band, RFC2833 and SIP Info.  RFC2833 Payload(>=96) The user can use the default settings  Register Refresh Interval(sec) The time interval between two normal registration messages. The user can use the default settings.  When enabled, an unregistered message will be sent before the registration is disabled, and no unregistered messages will be sent before registration; should be set according to the different server requirements  Remove Last Reg Whether to remove the last registration message  Session Refresh Time(sec) The interval between two sessions, the user can use the default settings  Refresher Select Refresh from UAC and UAS  If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold Interval" parameter.  SIP OPTIONS Enable Whether to open the SIP OPTION function  Initial Reg With Authorization Whether to carry the certification information when registering  Reply 182 On Call Waiting Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s) The time interval for sending empty packets  Anonymous Call Block Whether to enable anonymous call blocking  Proxy DNS Type Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog Whether the OB agent is used in the conversation  Complete Register Whether to enable full registration  Reg Subscribe Enable When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix  User Type Whether the end user is IP or Phone  Hold Method Call hold is REINVITE or INFO	Parameter name	Description
Signal Port The local port number of the SIP protocol  DTMF Type Select the second way of dialing, optional items are In-band, RFC2833 and SIP Info.  RFC2833 Payload(>=96) The user can use the default settings  Register Refresh Interval(sec) The time interval between two normal registration messages. The user can use the default settings  When enabled, an unregistered message will be sent before the registration is disabled, and no unregistered messages will be sent before registration; should be set according to the different server requirements  Remove Last Reg Whether to remove the last registration message Session Refresh Time(sec) The interval between two sessions, the user can use the default settings  Refresher Select Refresh from UAC and UAS  If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold interval" parameter.  SIP OPTIONS Enable Whether to open the SIP OPTION function  Initial Reg With Authorization Whether to carry the certification information when registering  Reply 182 On Call Waiting Whether to carry the certification information when registering  Reply 182 On Call Waiting Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s) The time interval for sending empty packets  Anonymous Call Whether anonymous calls are enabled Anonymous Call Block Whether to enable anonymous call blocking  Proxy DNS Type Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog Whether to enable full registration  Reg Subscribe Enable When enabled, the subscription message is sent after the registration hessage; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix User Type Whether the end user is IP or Phone	Domain Name Type	Whether to enable domain name recognition in SIP URIs
DTMF Type  Select the second way of dialing, optional items are In-band, RFC2833 Payload(>=96)  The user can use the default settings  Register Refresh Interval(sec)  The time interval between two normal registration messages. The user can use the default settings  When enabled, an unregistered message will be sent before the registration is disabled, and no unregistered messages will be sent before registration; should be set according to the different server requirements  Remove Last Reg  Whether to remove the last registration message  Session Refresh Time(sec)  The interval between two sessions, the user can use the default settings  Refresher  Select Refresh from UAC and UAS  If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold Interval" parameter.  SIP OPTIONS Enable  Whether to open the SIP OPTION function  Initial Reg With Authorization  Whether to carry the certification information when registering  Reply 182 On Call Waiting  Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s)  The time interval for sending empty packets  Anonymous Call  Whether anonymous calls are enabled  Anonymous Call Block  Whether to enable anonymous call blocking  Proxy DNS Type  Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog  Whether to enable full registration  When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	Carry Port Information	Whether to carry the SIP URI port information
RFC2833 and SIP Info.  RFC2833 Payload(>=96) The user can use the default settings  Register Refresh Interval(sec) The time interval between two normal registration messages. The user can use the default settings.  When enabled, an unregistered message will be sent before the registration is disabled, and no unregistered messages will be sent before registration; should be set according to the different server requirements  Remove Last Reg Whether to remove the last registration message  Session Refresh Time(sec) The interval between two sessions, the user can use the default settings  Refresher Select Refresh from UAC and UAS  If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold Interval" parameter.  SIP OPTIONS Enable Whether to open the SIP OPTION function  Initial Reg With Authorization Whether to carry the certification information when registering  Reply 182 On Call Waiting Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s) The time interval for sending empty packets  Anonymous Call Whether to enable anonymous calls blocking  Proxy DNS Type Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog Whether to enable full registration  Reg Subscribe Enable Whether to enable full registration  Reg Subscribe Enable Whether to enable full registration  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix  User Type Whether the end user is IP or Phone  Hold Method Call hold is REINVITE or INFO	Signal Port	The local port number of the SIP protocol
Register Refresh Interval(sec) The time interval between two normal registration messages. The user can use the default settings.  When enabled, an unregistered message will be sent before the registration is disabled, and no unregistered messages will be sent before registration; should be set according to the different server requirements  Remove Last Reg Whether to remove the last registration message  Session Refresh Time(sec) The interval between two sessions, the user can use the default settings  Refresher Select Refresh from UAC and UAS  If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold Interval" parameter.  SIP OPTIONS Enable Whether to open the SIP OPTION function  Whether to open the SIP OPTION function  Whether to carry the certification information when registering  Reply 182 On Call Waiting Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s) The time interval for sending empty packets  Anonymous Call Anonymous Call Block Whether to enable anonymous calls are enabled  Anonymous Call Block Whether to enable anonymous call blocking  Proxy DNS Type Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog Whether the OB agent is used in the conversation  Complete Register Whether to enable full registration  Reg Subscribe Enable Whether to enable full registration  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix User Type Whether the end user is IP or Phone  Hold Method Call hold is REINVITE or INFO	DTMF Type	
The user can use the default settings.  When enabled, an unregistered message will be sent before the registration is disabled, and no unregistered messages will be sent before registration; should be set according to the different server requirements  Remove Last Reg Whether to remove the last registration message  Session Refresh Time(sec) The interval between two sessions, the user can use the default settings  Refresher Select Refresh from UAC and UAS  If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold Interval" parameter.  SIP OPTIONS Enable Whether to open the SIP OPTION function  Initial Reg With Authorization Whether to carry the certification information when registering  Reply 182 On Call Waiting Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s) The time interval for sending empty packets  Anonymous Call Block Whether anonymous calls are enabled  Anonymous Call Block Whether to enable anonymous call blocking  Proxy DNS Type Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog Whether the OB agent is used in the conversation  Complete Register Whether to enable full registration  Reg Subscribe Enable When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix  User Type Whether the end user is IP or Phone  Hold Method Call hold is REINVITE or INFO	RFC2833 Payload(>=96)	The user can use the default settings
the registration is disabled, and no unregistered messages will be sent before registration; should be set according to the different server requirements  Remove Last Reg Whether to remove the last registration message  Session Refresh Time(sec) The interval between two sessions, the user can use the default settings  Refresher Select Refresh from UAC and UAS  If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold Interval" parameter.  SIP OPTIONS Enable Whether to open the SIP OPTION function  Initial Reg With Authorization Whether to carry the certification information when registering  Reply 182 On Call Waiting Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s) The time interval for sending empty packets  Anonymous Call Whether anonymous calls are enabled  Anonymous Call Block Whether to enable anonymous call blocking  Proxy DNS Type Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog Whether the OB agent is used in the conversation  Complete Register Whether to enable full registration  Reg Subscribe Enable When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix  User Type Whether the end user is IP or Phone  Hold Method Call hold is REINVITE or INFO	Register Refresh Interval(sec)	
Session Refresh Time(sec)  The interval between two sessions, the user can use the default settings  Refresher  Select Refresh from UAC and UAS  If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold Interval" parameter.  SIP OPTIONS Enable  Whether to open the SIP OPTION function  Initial Reg With Authorization  Whether to carry the certification information when registering  Reply 182 On Call Waiting  Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s)  Anonymous Call  Whether anonymous calls are enabled  Anonymous Call Block  Whether to enable anonymous call blocking  Proxy DNS Type  Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog  Whether the OB agent is used in the conversation  Complete Register  Whether to enable full registration  Reg Subscribe Enable  Whether to enable full registration  When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	Caller ID Header	the registration is disabled, and no unregistered messages will be sent before registration; should be set according to the
Refresher  Select Refresh from UAC and UAS  If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold Interval" parameter.  SIP OPTIONS Enable  Whether to open the SIP OPTION function  Initial Reg With Authorization  Whether to carry the certification information when registering  Reply 182 On Call Waiting  Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s)  The time interval for sending empty packets  Anonymous Call  Whether anonymous calls are enabled  Anonymous Call Block  Whether to enable anonymous call blocking  Proxy DNS Type  Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog  Whether the OB agent is used in the conversation  Complete Register  Whether to enable full registration  Reg Subscribe Enable  When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	Remove Last Reg	Whether to remove the last registration message
If this option is enabled, the IP phone will send SIP-OPTION to the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the "NAT Hold Interval" parameter.  SIP OPTIONS Enable Whether to open the SIP OPTION function  Initial Reg With Authorization Whether to carry the certification information when registering  Reply 182 On Call Waiting Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s) The time interval for sending empty packets  Anonymous Call Whether anonymous calls are enabled  Anonymous Call Block Whether to enable anonymous call blocking  Proxy DNS Type Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog Whether to enable full registration  Complete Register Whether to enable full registration  Reg Subscribe Enable When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix  User Type Whether the end user is IP or Phone  Hold Method Call hold is REINVITE or INFO	Session Refresh Time(sec)	
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Initial Reg With Authorization  Whether to carry the certification information when registering  Reply 182 On Call Waiting  Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s)  The time interval for sending empty packets  Anonymous Call  Whether anonymous calls are enabled  Anonymous Call Block  Whether to enable anonymous call blocking  Proxy DNS Type  Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog  Whether the OB agent is used in the conversation  Complete Register  Whether to enable full registration  Reg Subscribe Enable  When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	SIP 100REL Enable	the server instead of sending Hello messages on a regular basis. The interval for sending is the parameter set for the
Reply 182 On Call Waiting Whether or not to send 182 when the call is waiting  NAT Keep-alive Interval(10-60s) The time interval for sending empty packets  Anonymous Call Whether anonymous calls are enabled  Anonymous Call Block Whether to enable anonymous call blocking  Proxy DNS Type Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog Whether the OB agent is used in the conversation  Complete Register Whether to enable full registration  Reg Subscribe Enable When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix  User Type Whether the end user is IP or Phone  Hold Method Call hold is REINVITE or INFO	SIP OPTIONS Enable	Whether to open the SIP OPTION function
NAT Keep-alive Interval (10-60s)  Anonymous Call  Anonymous Call Block  Whether anonymous calls are enabled  Anonymous Call Block  Whether to enable anonymous call blocking  Proxy DNS Type  Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog  Whether the OB agent is used in the conversation  Complete Register  Whether to enable full registration  Reg Subscribe Enable  When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	Initial Reg With Authorization	
Anonymous Call Whether anonymous calls are enabled  Anonymous Call Block Whether to enable anonymous call blocking  Proxy DNS Type Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog Whether the OB agent is used in the conversation  Complete Register Whether to enable full registration  Reg Subscribe Enable When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix  User Type Whether the end user is IP or Phone  Hold Method Call hold is REINVITE or INFO	Reply 182 On Call Waiting	Whether or not to send 182 when the call is waiting
Anonymous Call Block  Whether to enable anonymous call blocking  Proxy DNS Type  Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog  Whether the OB agent is used in the conversation  Complete Register  Whether to enable full registration  Reg Subscribe Enable  When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	NAT Keep-alive Interval(10-60s)	The time interval for sending empty packets
Proxy DNS Type  Set the DNS server type, the optional items are Type A, DNS SRV, and Auto  Use OB Proxy In Dialog  Whether the OB agent is used in the conversation  Complete Register  Whether to enable full registration  Reg Subscribe Enable  When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	Anonymous Call	Whether anonymous calls are enabled
Use OB Proxy In Dialog  Whether the OB agent is used in the conversation  Complete Register  Whether to enable full registration  Reg Subscribe Enable  When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	Anonymous Call Block	Whether to enable anonymous call blocking
Complete Register  Whether to enable full registration  When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	Proxy DNS Type	
Reg Subscribe Enable When enabled, the subscription message is sent after the registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix Dial before prefix  User Type Whether the end user is IP or Phone  Hold Method Call hold is REINVITE or INFO	Use OB Proxy In Dialog	Whether the OB agent is used in the conversation
registration message; the subscription message is not sent when disabled  Reg Subscribe Interval(sec)  Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	Complete Register	Whether to enable full registration
Dial Prefix  Dial before prefix  User Type  Whether the end user is IP or Phone  Hold Method  Call hold is REINVITE or INFO	Reg Subscribe Enable	registration message; the subscription message is not sent
User Type Whether the end user is IP or Phone Hold Method Call hold is REINVITE or INFO	Reg Subscribe Interval(sec)	
Hold Method Call hold is REINVITE or INFO	Dial Prefix	Dial before prefix
	User Type	Whether the end user is IP or Phone
Request-URI User Check Whether to allow the user to check	Hold Method	Call hold is REINVITE or INFO
	Request-URI User Check	Whether to allow the user to check

Only Recv Request From Server	If enabled, will only accept requests from the server, do not accept other requests
Server Address	SIP server address
SIP Received Detection	Whether to allow SIP receive detection
VPN	Whether to enable VPN
SIP Encrypt Type	Whether to allow SIP message encryption
RTP Encrypt Type	Whether to allow RTP message encryption
Country Code	Country code
Remove Country Code	Whether to allow the removal of national codes
Tel URL	Whether to open the Tel URL
Use Random SIP Port	Whether to use random port
Min Random SIP Port	SIP minimum random port
Max Random SIP Port	SIP maximum random port
Prefer Primary SIP Server	Whether to enable the preferred primary server
Hold SDP Attribute Inactive	Whether to enable the call to keep the inactive attribute
BLF List URL	
BLF PickUp Code	
RTP Port Min	RTP minimum port
RTP Port Max	RTP's maximum port

# **SIP settings**

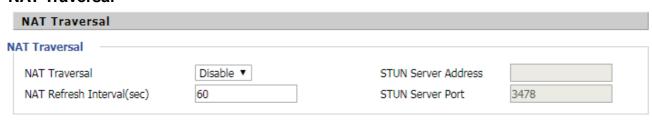




Parameters name Description

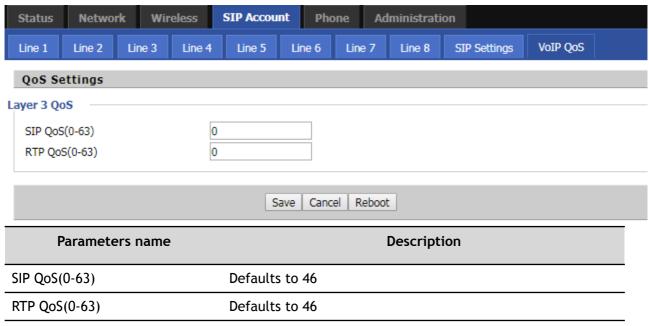
SIP T1	The default value is 500
SIP User Agent Name	Enter the SIP User Agent header field
Max Forward	Modify the maximum hop value, the default is 70
Max Auth	Change the number of authentication failures, the default value is 2
Reg Retry Intvl	Registration failed again registration interval, default is 30
Reg Retry Long Intvl	Registration failed Register again for the long interval Default 1200
Mark All AVT Packets	The default enable is on
RFC 2543 Call Hold	The default enable is on
SRTP	The default is disabled
SRTP Prefer Encryption	Support for AES_CM and ARIA_CM
Service Type	Default general
DNS Refresh Timer	Modify the DNS refresh time, the default value of 0
Transport	The transmission type defaults to UDP

# **NAT Traversal**



Parameters name	Description
NAT Traversal	Whether to enable NAT mode, or select STUN to penetrate
STUN Server Address	STUN server IP address
NAT Refresh Interval(sec)	Refresh interval
STUN Server Port	STUN port, the default is 3478

# VoIP QoS



Configuration can be based on the scene environment to modify the parameters

# Network

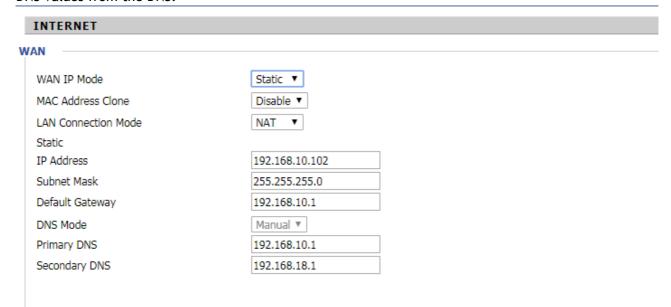
This page includes: WAN port settings, LAN port settings, VPN, DMZ, wireless, QoS and DDNS settings, the following picture for the navigation bar:



# WAN

WAN connection There are three ways to connect to the Internet: DHCP, static and PPPoE **Static:** 

In static mode, the user should fill in the IP address, subnet mask, default gateway, primary DNS and DNS values from the DNS.



Parameters name	Description
WAN IP Mode	Select static IP.
MAC Address Clone	Clone MAC Address
LAN Connection Mode	LAN connection mode Selects bridging or routing
IP Address	The IP address of the Internet port.
Subnet Mask	The subnet mask of the Internet port.
Default Gateway	The default gateway for Internet ports.
DNS Mode	The DNS type is specified: the user should manually set the preferred DNS and alternate DNS.  The DNS type is Automatic: The IP phone will automatically obtain the preferred DNS and alternate DNS from the DHCP server.
Primary DNS	The preferred DNS address for the Internet port
Secondary DNS	Alternative DNS for Internet ports.

## DHCP:

In DHCP mode, the IP phone is a DHCP client that obtains the IP address, subnet mask, and default gateway from the DHCP server.



Parameters name	Description
WAN IP Mode	Select DHCP
DHCP Server	DHCP Server
MAC Address Clone	Clone MAC Address
LAN Connection Mode	LAN connection mode Selects bridging or routing
DNS Mode	The DNS type is specified: the user should manually set the preferred DNS and alternate DNS.  The DNS type is Automatic: The IP phone will automatically obtain the preferred DNS and alternate DNS from the DHCP server.
Primary DNS	The preferred DNS address for the Internet port
Secondary DNS	Alternative DNS for Internet ports.

# PPPoE:

## INTERNET WAN PPPoE ▼ WAN IP Mode MAC Address Clone Disable ▼ LAN Connection Mode NAT • DNS Mode Auto Primary DNS Secondary DNS PPPoE PPPoE Account PPPoE Password ••••• Confirm Password ••••• Service Name Leave empty to autodetect Operation Mode Keep Alive ▼ Keep Alive Redial Period(0-3600s)

Parameters name	Description
WAN IP Mode	Select PPPoE.
MAC Address Clone	Whether to allow MAC cloning
LAN Connection Mode	LAN connection mode Select bridging or routing.
DNS mode	The DNS type is specified: the user should manually set the preferred DNS and alternate DNS. The DNS type is Automatic: The IP phone will automatically obtain the preferred DNS and alternate DNS from the DHCP server.
Primary DNS	The preferred DNS address for the Internet port.
Secondary DNS	Alternative DNS for Internet ports.
PPPoE Account	Fill out the PPPoE account obtained from the internet service provider.
PPPoE Password	Fill in the password obtained from the Internet service provider for the PPPoE account.
Confirm Password	Re-enter the PPPOE account password.
Service Name	Fill in the name of the service.
Operation Mode	Run mode, active, on-demand and manual settings. The default run mode mode remains active.
Keep Alive Redial Period(0-3600s)	The dialing interval for the redial dial.

# LAN

ı	PC Port(LAN)				
P	C Port(LAN)				
	Local IP Address		192.168.1.1		
	Local Subnet Mask		255.255.255.0		
	Local DHCP Server		Enable ▼		
	DHCP Start Address		192.168.1.2		
	DHCP End Address		192.168.1. 254		
	DNS Mode		Auto ▼		
	Primary DNS		192.168.1.1		
	Secondary DNS		192.168.10.1		
	Client Lease Time(0-86400s)		86400		
			DHCP Client List		
	DHCP Static Allotment				
	NO.	MAC		IP Address	
	1				
	2				
	3				
	DNS Proxy		Enable ▼		

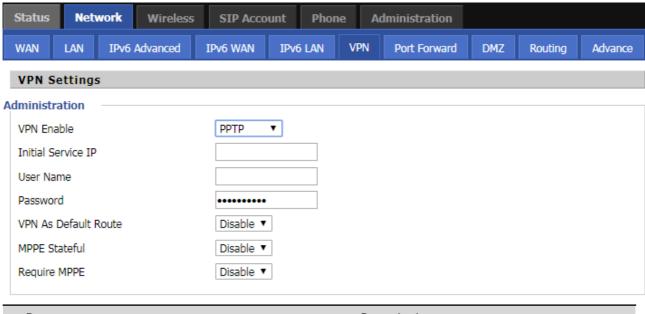
Parameters name	Description
Local IP Address	Set the IP address of the PC port.
Local Subnet Mask	Set the subnet mask.
Local DHCP Server	Whether to enable the DHCP server. If the LAN port is not in NAT mode, the user can not enable the DHCP server.
DHCP Start Address	Start IP address, assign the address of the address to the DHCP client.
DHCP End Address	End the IP address and assign the address of the address to the DHCP client.
DNS Mode	The DNS type is specified: the user should manually set the preferred DNS and alternate DNS.  The DNS type is Automatic: The IP phone will automatically obtain the preferred DNS and alternate DNS from the DHCP server.
Primary DNS	The preferred DNS address for the Internet port.
Secondary DNS	Alternative DNS for Internet ports.
Client Lease Time(0-86400s)	DHCP Client to obtain IP lease, in seconds, the default is 86400s.

# **VPN**

VPN is a technology that builds a private network on a public network. The connection between any two nodes of the VPN network does not have the end-to-end physical link required by the traditional

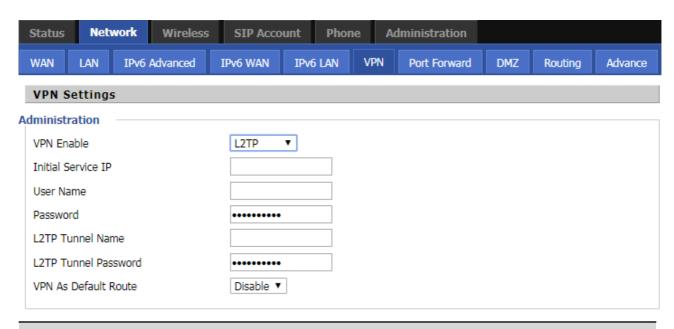
private network, but rather the network platform provided by the public network service provider, and the user data is transmitted in the logical link. With VPN technology, you can establish private connections and transfer data between any two devices on the public network.

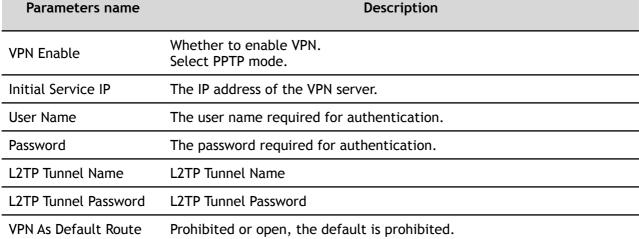
### PPTP:



Parameters name	Description
VPN Enable	Whether to enable VPN. Select PPTP mode.
Initial Service IP	The IP address of the VPN server.
User Name	The user name required for authentication.
Password	The password required for authentication.
VPN As Default Route	Prohibited or open, the default is prohibited.
MPPE Stateful	
Require MPPE	

## L2TP







## VPN Settings

#### Administration

VPN Enable

OpenVPN ▼

OpenVPN TLS Auth

Disable ▼

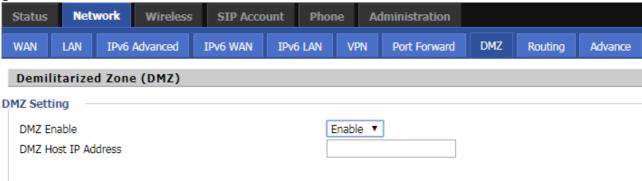
VPN As Default Route

Disable ▼

Parameters name	Description
VPN Enable	Whether to enable VPN. Select OpenVPN mode.
OpenVPN TLS Auth	Whether OpenVPN TLS authentication is enabled
VPN As Default Route	Prohibited or open, the default is prohibited.

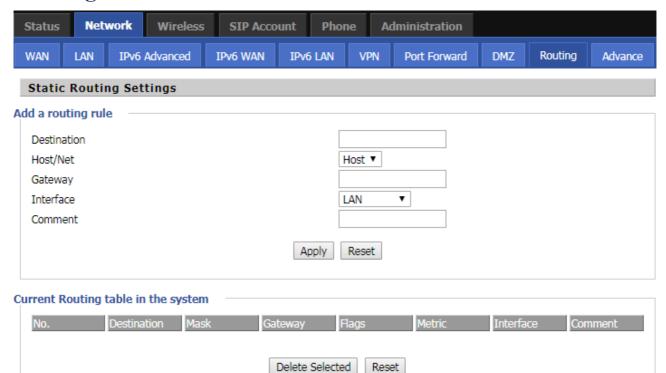
## **DMZ**

DMZ can be understood as a different from the external network or within the network of special network area, the role is to WEB, e-mail and other external access to the server alone in the area, so that the entire need to protect the internal network connected to the port, Do not allow any access, to achieve internal and external network separation, to meet user needs. The FIP11 / FIP11W / FIP11WL / FIP11WP provides such a DMS that can map external data based on any protocol to a single host (DMZ host) of the LAN. Ordinary Internet operations and other Internet activities of other clients will continue without undue disruption. The DMZ host allows the internal host to be completely exposed to the Internet, which usually helps some special applications such as NetMeeting or online games.



Parameters name		Description
DMZ Enable	Whether to enable DMZ.	
DMZ Host IP Address	Fill in the DMZ host IP	

# Routing



## Add a routing rule:

Description
The destination address of the routing rule to be added
Select the way you want to add
The IP address of the gateway
Select whether the LAN port or other network interface
Fill in memo information
You can view the routing rules for the current device

# **Phone**

This page user can configure parameters such as volume, call forwarding, dial plan, phone book, and so on.



# **Preferences**

## **VolumeSettings**

## **Volume Settings**

Handset Input Gain 5 ▼ Handset Volume 5 ▼

Speakerphone Input Gain 5 ▼ Speaker Volume 5 ▼

Ringer Volume 5 ▼ Speakerphone Mic Boost Disable ▼

Parameters name	Description
Handset Input Gain	Adjust the volume of the handset MIC.
Handset Volume	Adjust the volume of the handset.
Speakerphone Input Gain	Adjust the volume of the speaker MIC.
Speaker Volume	Adjust the volume of the speaker.
Ringer Volume	Adjust the volume of the ringer.
Speakerphone Mic Boost	Whether boost the volume of the speaker MIC by 20db.

## Regional

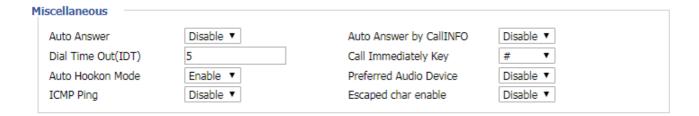
Ringing Time(10-300sec)

60



Select the tone type.  Dial tone
Busy tone.
Hang up warning tone
Ring back tone
Call waiting tone
jitter delay the minimum value, the phone jitter using the adaptive mechanism.
jitter delay the maximum value of the phone jitter using the adaptive mechanism.
Ring time.
Ri Ci ji m

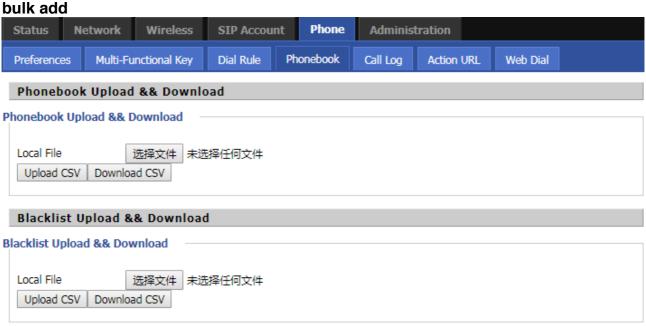
## **Miscellaneous**



Parameters name	Description
Auto Answer	Whether to enable automatic answering, if the phone automatically answer the phone.
Auto Answer by CallINFO	Whether to enable automatic answering under CallINFO.
Dial Time Out(IDT)	How long does the phone dial a dial tone?
Call Immediately Key	Use "#", "*" to do speed dialing, or disable.
Auto Hookon Mode	
Preferred Audio Device	
ICMP Ping	Whether ICMP Ping is enabled.  If enabled, the phone will ping the SIP server every interval; otherwise it will send "hello" empty packet to the SIP server.
Escaped char enable	Enable or disable, disabled by default.

# **Phonebook**

Currently supports two ways to upload phone books: bulk add, and a single way of adding.



Parameters name	Description	
Phonebook Upload && Download	Upload or download the phone file in CSV format	
Blacklist Upload && Download	Upload or download the blacklist file in CSV format	

The format of the batch is:

4	A	В	С	D	E
1	Name	Number	Bell Type		
2	A	123	Bell Type1		
3					
4					
5					
6					
7					

Parameters name	Description
Name	Enter the name
Number	Enter the number
Ring	Select the ringtone type

# Single add:

Phonel	oook			
Index	Name	Number	Ring	

Name	
Number	
Ring	Bell Type 1 ▼
	OK Cancel

## Steps:

## Add a phone book:

- Step 1. Click the Add button, and then the configuration table appears as shown.
- Step 2. Fill in the value of the parameter.
- Step 3. Press the Ok button to end the configuration.
- Step 4. Press the Save button to save your changes.

#### Edit Phonebook:

- Step 1. Check a number.
- Step 2. Click the Save button and the configuration table shown in Figure 2 appears.
- Step 3. Change the value of the parameter.
- Step 4. Press the OK button to end the configuration.
- Step 5. Press the Save button to save your changes.

#### Delete a phone call:

- Step 1. Check one or more numbers.
- Step 2. Click the "Delete" button to delete the phone.

## Move from phone book to blacklist:

- Step 1. Check one or more numbers.
- Step 2. Press the Move to Blacklist button to move the blacklist to the phonebook.

# Call log

You can view phone call log information, such as dialed lists, received calls, and missed calls.

# Redial List

Redia	Kediai List				
Index	NUMBER	Start Time	Duration		
1	601	10/17 19:54	00:00:01		
2	1234	10/17 19:55	00:00:01		
3	585852145865	10/17 19:55	00:00:01		

# Answered Calls Answered Calls

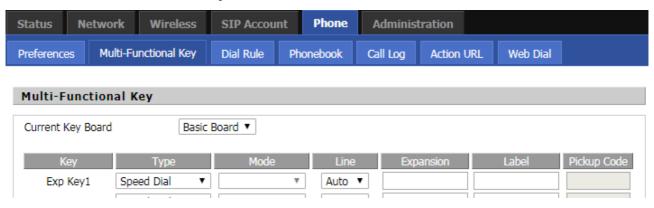
Index	NUMBER 601	Start Time 10/17 19:55	Duration 00:00:00	
Today	NUMBER	Charle Time	Duration	

### **Missed Calls**

#### Missed Calls

Index	NUMBER	Start Time	Duration	
1	601	10/17 19:56	00:00:00	

# **Multi-Functional Key**

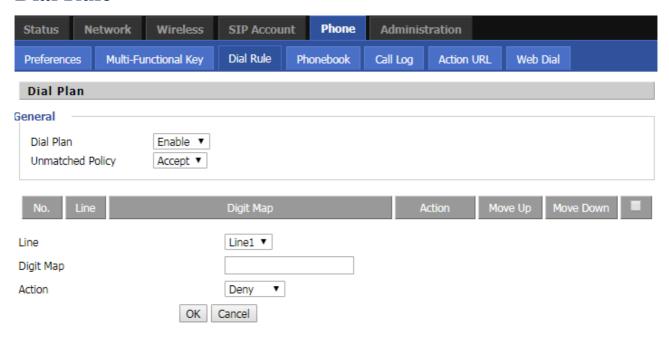


Parameters name	Description	
Key	Function key number.	
Туре	Function key type	
Mode	Select the corresponding shortcut key function on the Shortcut Keys.	
Line	Select Line / Auto Select.	
Expansion	Enter the number.	
Multi-function keys	Description	
disable	Hide the function key and disable it	
Speed dial	Do the emergency dial key to use, set the emergency call number, use the key to dial out directly	
BLF	Listen the status of the setup number	
shortcuts	Function shortcuts, including phone book, call log, SMS, login / logout	
Call Park	The call resides while the other party is on hold, transferring the call to another extension and continuing the conversation.	
Line keys	Can set this key as line key	
BLF Lists		
X-broadworks-callpark		

## BLF key status:

Icon	Description	Function	
<b>₽</b> B	Green	The monitored account is idle	
B	Red	The monitored account is in a call / busy state	
<b>₽</b> B	Yellow	Monitored account ringing	
B	Grey	Monitored account does not exist (or registration failed)	
<b>₽</b> B	Blue	When the BLA status of BroadSoft platform is Parking Lot	

# **Dial Rule**



Parameters name	Description	
Dial Plan	Whether to enable dial plan	
Unmatched Policy	nmatched Policy Choose to accept or reject	
Line	Select the account line	
Digit Map Fill in the expression for the dial plan		
Action	Grammar, please refer to the dial plan for the grammar	

#### step:

#### Add a dial plan:

- Step 1. Enable the dial plan
- Step 2. Click the "Add" button, will appear as shown in Figure 1 configuration table
- Step 3. Fill in the value of the parameter
- Step 4. Press the "Ok" button to end the configuration.
- Step 5. Press the Save button to save your changes

## Edit a dial plan:

- Step 1. Enable the dial plan
- Step 2. Select a dial plan
- Step 3. Click the "Edit" button, will appear as shown in Figure 2 configuration table
- Step 4. Change the value of the parameter
- Step 5. Press the "Ok" button to end the configuration
- Step 6. Press "Save" to save your changes

## Delete a dial plan:

- Step 1. Enable the dial plan
- Step 2. Select a dial plan
- Step 3. Click the "Delete" button to delete the dial plan

## Dial Rule grammar

Character	Description
-----------	-------------

0 1 2 3 4 5 6 7 8 9 * #	Legal characters
Х	The lowercase letter 'x' matches a legal character
[sequence]	Match a sequence For example: [0-9]: matches one of the numbers 0 to 9 [23-5*]: Match character 2 or 3 or 4 or 5 or *
х.	Matchx, xx, xxx, xxxx, xxxxx  For example: "01. "can match "0", "01", "011", "0111",, "01111"
<pre>&lt; d i a l e d : substituted&gt;</pre>	Replace For example: <8:1650>123456: input "85551212", output "16505551212"
х,у	Enter "x" will have a dial tone, enter "y" after the dial tone stops For example:  "9,1xxxxxxxxxxx": telephone input "9" after the dial tone, enter "1" after the dial tone stop  "9,8,010x": telephone input "9" after the dial tone, enter "0" after the dial tone stop
Т	Set the delay time For example: "<9: 111> T2": the phone will broadcast a valid number "2"



#### Example 1

If the user dials # 12 # 2, the call will call 010 # 12% 232.

#### Example 2

If the user dials 58101, the phone will call 2413338101.

The phone will press after 5 will play the sound, press 8 after dial tone stop.

#### Example 3

If the user dials 422xxxx or 522xxxx, the phone will dial 22xxxx333.

## Example 4

If the user dials 2622 or 26222 or 262222 or 362222.

The phone will dial 5622 or 56222 or 562222.

The phone will have a dial tone after pressing 2 or 3, and the dial tone will stop after pressing 6.

#### Example 5

If the user dials 777xxx ... x8, the phone rejects the number to dial out.

# Administration

In this page, the user can configure the time / date, password, system log and so on.



# Management

# Time/Date Setting

#### Time/Date Setting NTP Settings NTP Enable Enable ▼ Option 42 Disable ▼ 2017 - 10 - 13 . 19 : 05 Current Time Sync with host Sync with host NTP Settings (GMT+08:00) China Coast, Hong Kong • Primary NTP Server pool.ntp.org Secondary NTP Server cn.pool.ntp.org NTP synchronization(1 - 1440min) 60

Parameters name	Description
NTP Enable	Whether to enable NTP
Option 42	Whether to enable Option 42
Current Time	Show current time
Sync with host	Set the time zone
NTP Settings	
Primary NTP Server	Preferred IP address or domain name of the NTP server
Secondary NTP Server	The IP address or domain name of the alternate NTP server

NTP synchronization(1 - 1440min)

NTP synchronization cycle, the cycle time can be 1 to 1440 minutes of any one, the default setting is 60 minutes

### **Password Reset**

# Administrator Settings Password Reset User Type New User Name New Password Confirm Password (The maximum length is 25)

Parameters name	Description
User Type	Select an administrator or an ordinary user
New User Name	Enter a new username
New Password	Enter a new password
Confirm Password	Enter the new password again

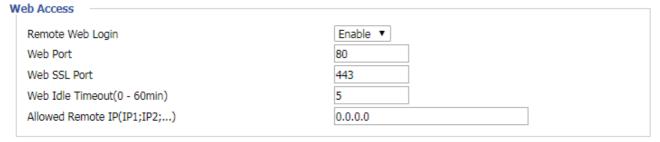
### **Admin Mode Change Password:**

- Step 1. Select the administrator from the drop-down list.
- Step 2. Enter the new password twice in the new password and confirm the password field.

#### User mode change password:

- Step 1. Select the user from the drop-down list.
- Step 2. Enter the new password twice in the new password and confirm the password field.

# Web Access



Parameters name	Description
Remote Web Login	Whether to enable remote Web logon
Web Port	Set the port to log in through the Internet port and PC port. The default value is 80
Web SSL Port	
Web Idle Timeout(0 - 60min)	Set the network idle timeout in minutes. If the network idle timeout without any operation, the page automatically log off
Allowed Remote IP(IP1;IP2;)	Allows remote connections to IP addresses

# **System Log Setting**



Parameters name	Description
Syslog Enable	Whether the system log is enabled
Syslog Level	Select the system day level, INFO and Debug, which Debug can get more information than INFO
Remote Syslog Enable	Whether to enable remote log service
Remote Syslog Server	Remote Syslog Server IP Address

The phone supports local and remote system logs.

#### local:

- Step 1. Disable the remote system log enable and select a log level, as shown in Figure 1.
- Step 2. Press the Save Settings button to save and press the Restart button to apply the settings.
- Step 3. The user can view the status / syslog page of the syslog.

#### Remotely:

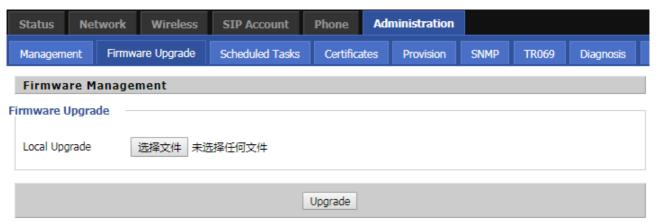
- Step 1. Enable remote system log enable and enter the IP address in Remote Syslog Server, as shown in Figure 2.
- Step 2. Select a log level.
- Step 3. Press the Save Settings button to save and press the Restart button to apply the settings.
- Step 4. The user can view the syslog server's system log and also view the records of the Status / Syslog web pages.

## **Factory Defaults Setting**



Click the Factory Default button to reset the phone to factory settings.

# Firmware Management



- Step 1. Select an upgrade file type for the upgrade software.
- Step 2. Press Browse to select the upgrade file.
- Step 3. Press Upgrade to start the upgrade and the LCD will show the prompts that are being upgraded.
- Step 4. Log in to the phone page by checking the status of the firmware in the firmware version of the firmware upgrade to determine whether the upgrade is successful.

# **Provision**

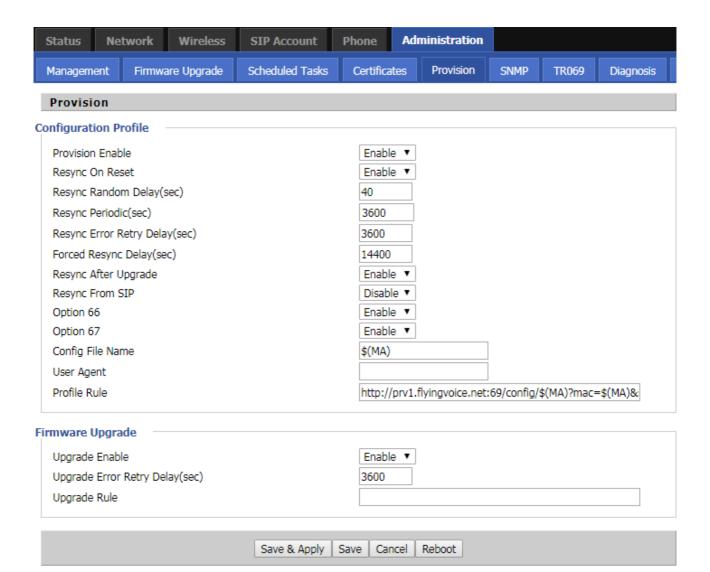
- 1) configuration allows the phone to automatically upgrade or automatically configure.
- 2) Phone support provides three ways: TFTP, HTTP and HTTPS.

Before testing or using TFTP, the user should have a TFTP server and upgrade files and configuration files.

Before testing or using HTTP, the user should have an HTTP server and upgrade files and configuration files.

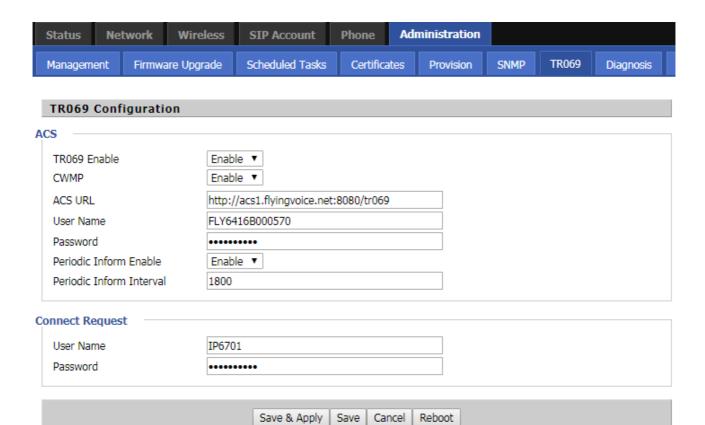
Before testing or using HTTPS, the user should have HTTPS servers and upgrade files and configuration files and CA certificate files (should be the same for the https server) and client certificate files and private key files.

- 3) The user can upload the CA certificate file and the client certificate file and the private key file management / certificate management page of the device.
- 4) For details, please refer to the file Provision user manual.



Parameters name	Description
Provision Enable	Whether provisioning is enabled
Resync On Reset	Whether DIV378 is restarted after restarting
Resync Random Delay(sec)	Set the maximum delay for requesting a synchronization file, which defaults to 40
Resync Periodic(sec)	Set the timing resynchronization, the default is 3600 seconds
Resync Error Retry Delay(sec)	If the last resynchronization is a failure, after the "Resync Error Retry Delay" time, ATA will retry the resynchronization, defaulting to 3600 seconds
Forced Resync Delay(sec)	If the time to re-sync, but ATA is busy, in this case, ATA will wait for some time, the longest is the "forced resynchronization delay", the default is 14400s, after time, ATA will be forced to re-sync
Resync After Upgrade	After re-synchronization, whether to enable the firmware update function, the default is enable
Resync From SIP	Whether to enable resynchronization from SIP
Option 66	It is only used in the company within the provisions of the model. When using TFTP with option 66 to implement the configuration, the user must enter the correct profile name on the ATA web page. When option 66 is disabled, this parameter does not work
Option 67	Whether to enable Option 67
Config File Name	Configuration file name
User Agent	
Profile Rule	The URL of the configuration file
Upgrade Enable	Note that the specified file path is relative to the root directory of the TFTP server
Upgrade Error Retry Delay(sec)	Turn on or off
Upgrade Rule	Set the upgrade error retry delay interval, the default 3600 seconds

# **TR069**



Parameters name	Description
TR069 Enable	Whether TR069 is enabled
CWMP	Whether CWMP is enabled
ACS URL	TR069 the server's URL, the default for the Fahrenheit ACS server address
User Name	The user name for the TR069 server connection
Password	The password for the TR069 server connection
Periodic Inform Enable	Whether to enable periodic information
Periodic Inform Interval	TR069 The interval at which the server sends information
User Name	TR069 The user name of the server connected to the phone
Password	TR069 The password for the server to connect to the phone

# **Chapter 5 Common Troubleshooting**

This section provides a solution to the problems that may arise during the installation and operation of IP telephones.

- · The power is not reflected
- · No dial tone
- · Can not make a call
- · Can not receive phone calls
- · no sound in the call
- · can not log in
- · forget password

# The power is not reflected

## solution:

Check that the power adapter is properly connected.

# No dial tone

#### solution:

Check that the telephone wiring and telephone line order are properly connected.

# Can not make a call

#### solution:

Check if the server supports the current audio codec type, or contact your administrator, vendor, or ITSP for more information or assistance.

# Can not receive phone calls

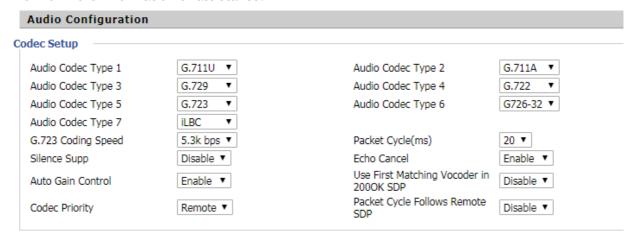
#### solution:

Check if the server supports the current audio codec type, or contact your administrator, vendor, or ITSP for more information or assistance.

# No sound in the call

### solution:

Check if the server supports the current audio codec type, or contact your administrator, vendor, or ITSP for more information or assistance.



# Can't login Web page

### solution:

Check that the Ethernet cable is properly connected.

Check whether the URL is correct, URL format: http://IP address.

Check if your firewall / NAT settings are correct.

If the IE version is checked by IE8, or use another browser such as Firefox or Mozilla, or contact your administrator, vendor or ITSP for more information or assistance.

# Forget password

The default password for the site and menu is admin.

If the user changes the password and then forget, you can not access the configuration site or menu item that requires the password.

## solution:

use "#\*06#" to factory reset. (Press 23646 to confirm)

If you choose the factory default, you will return to the original factory settings for the phone and will delete all current settings, including system logs and call records.