



User Manual

FIP16

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About This User Guide

Thank you for choosing FlyingVoiceFIP16. The FIP16 which have one line 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 services which you could not expect from traditional telephone. This guide will help you easily use the various features and services available on your phone.

TheFIP16, which support 2.4GHZ&5GHZ, It can suooprt 802.11 b/g/n, 802.11ac, 2T2R, And the FIP16 have 1.8" 126*160 TFT Color Screen.



This guide contains the following:

- [Contact with FlyingVoice](#)
- [Purpose](#)
- [Cross reference](#)
- [Feedback](#)
- [Product Declaration of Conformity](#)
- [Warnings and cautions](#)

Contact with Flying Voice

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

Sales enquiries: sales1@flyingvoice.com

Support enquiries: support@flyingvoice.com

Hotline: [010-67886296](tel:010-67886296) [0755-26099365](tel:0755-26099365)

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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

Part 15 FCC Rules

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

“This equipment complies with FCC RF radiation exposure limits set forth for an uncontrolled environment.

This equipment should be installed and operated with a minimum distance of 20 centimeters between the radiator and your body.”



Warning

Changes or modifications to this unit not expressly approved by the party responsible for compliance

could void the user's authority to operate the equipment.

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 generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation.

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 interference 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 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:



Warning

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



Notes

Notes text and consequence for not following the instructions in the Notes.

Chapter 1 Product description

This chapter covers:

- [FIP16](#)
- [Internet connection](#)
- [Familiar with the phone](#)

FIP16/FIP16/FIP16P

Port/Model	FIP16
------------	-------

Product Picture



LED	1.8 inch color LCD screen 126 * 160 resolution
Wi-Fi	2.4G&5G
Line	1
Ethernet Port	x
Headset	3.5mm
USB	1*USB for power adapter
Soft key	2
Code	G.711(A-law,U-law),G.723,G.729A/B,G.722,iBLC
TR069	√
SNMP	√
VPN	PPTP/L2TP/Open VPN
Auto configuration	TPTP/HTTP/HTTPS Tr069 Network management
Telephone book	√
Call history	√
SMS	√

Internet connection

Phone Start

Please refer to the quick installation manual for the telephone assembly

Step 1. Press the power button more than 3s to power on the FIP16.



Note

Make sure you had recharge the FIP16 enough power for power on .

Step 2. After power on the FIP16, the LCD screen of the phone will display the signature of "FlyingVoice".

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, wifi 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.

Connect to a wireless network

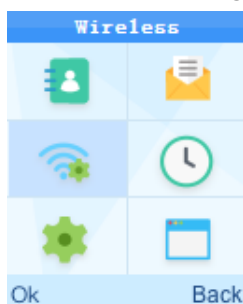
You can configure a wireless connection in both the LCD and the web.

From the LCD menu:

Step 1. Press menu and then use the ▲ and ▼ buttons or enter " **Wireless Network**" or just press the wireless in the right of LCD.

Step 2. Select " **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.

The screenshot shows the 'Wireless' tab selected in the top navigation bar. Below it, the 'Wireless Settings' section is active. It contains the following fields:

- Internet Connection Type: Automatic Configuration - DHCP (dropdown)
- DNS Type: Auto (dropdown)
- Primary DNS: 8. 8. 8. 8
- Second DNS: 8. 8. 4. 4

Below the settings, the 'Wireless Connection' section shows the 'Connection Status' as 'Disconnected'.

Step 2. choose one wireless network to connect.

Step 3. Use "connect" button at the bottom of the page.

The screenshot shows a list of available wireless networks with their names, security types, and encryption methods. Each entry has a signal strength icon on the right.

Network Name	Security Type	Encryption	Signal Strength
evergreen	WPA1PSK/WPA2PSK	AES	Full
wlan-ap	OPEN	NONE	Full
LXCT	WPA1PSK/WPA2PSK	AES	Full
TP-LINK_A934	WPA1PSK/WPA2PSK	AES	Full

At the bottom, there are three buttons: 'Connect', 'Refresh', and 'Add'.

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.

The screenshot shows the configuration screen for the selected network 'TP-LINK_9D97'. It includes the following fields:



- Authentication: WPA2PSK (dropdown)
- Encryption: ☐ TKIP ☒ AES
- Password: (empty text box)

At the bottom, there are two buttons: 'OK' and 'Cancel'.

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

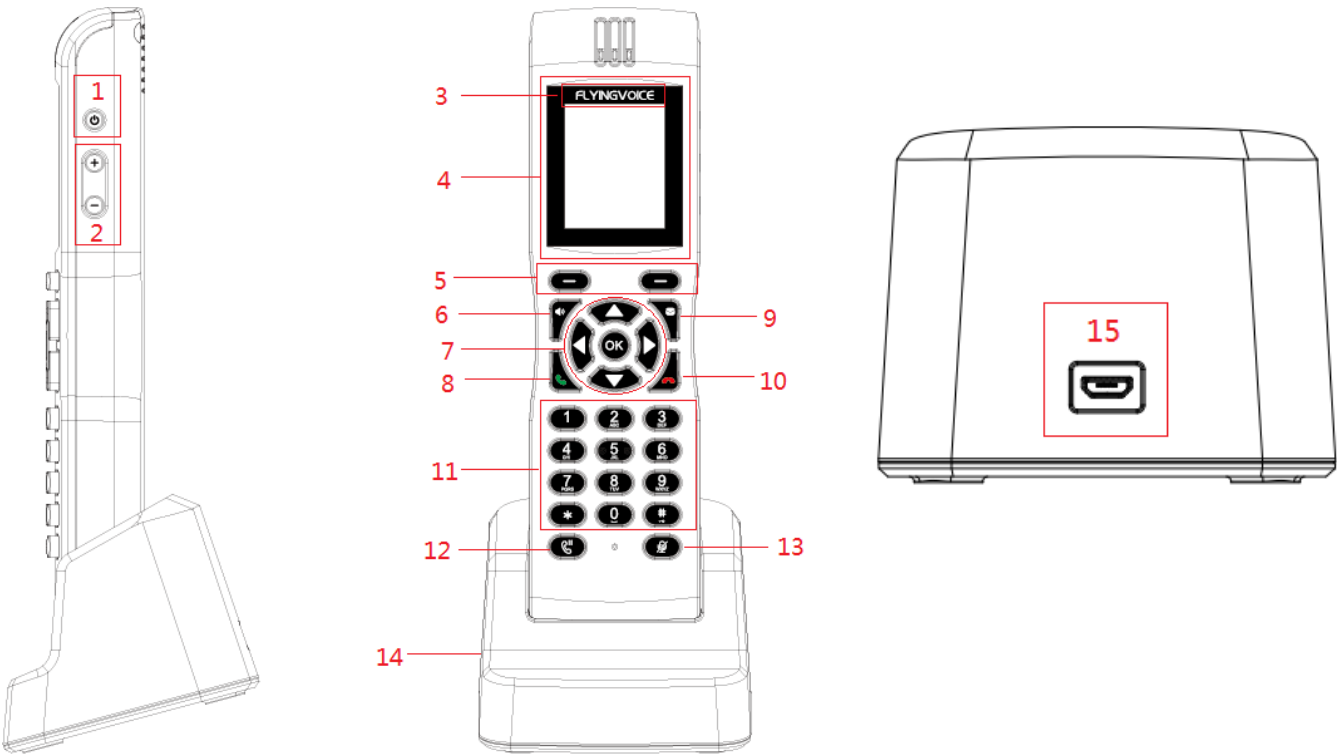
Wireless Connection

Wireless Connection

Connection Status		Connected (AP: wlan-ap)		
SSID	Authentication	Encryption	Status	
wlan-ap	OPEN	NONE		
CU_3aNc	WPA1PSK/WPA2PSK	AES		





Familiar with the phone

Front panel



No.	Name	Description
1	Power	Press the button for more than 3S to power on or power off
2	Volume button	Increase or decrease the volume
3	Logo	Flyingvoice Logo
4	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
5	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.
6	Handsfree	Switch to handsfree mode
7	Navigation key	1. Navigation keys, including up, down, turn left, right turn and confirm keys 2. Turn left and right turn is also the volume of the shortcut keys 3. The confirmation key is also a shortcut to view the current network status
8	Pick up	Pick up the call
9	Voicemail	Voicemail shortcuts
10	Hang up	Press the button to hang up the call
11	Numeric Keypad	Enters numeric digits for initiating a call or for entering configuration information.
12	Hold	To hold the current call, press the Hold key again to restore the current call
13	Mute	When the phone is on, press to switch to silent mode
14	Base	Base is used to charge to the phone
15	USB	Connect to the power cord to charge the phone

The description of icons in the home screen

Icons	Description
	The wifi is connected (There is little signal)
	The wifi is connected (A lattice signal)
	The wifi is connected (Two lattice signal)
	The wifi is connected (Three lattice signal)
	The wifi is connected (Full lattice signal)
	The wifi is disconnected
	Missed Call
	Short message
	Headset
	Battery charging state (charge 20%)
	Battery charging state (charge 40%)
	Battery charging state (charge 60%)
	Battery charging state (charge 80%)
	Battery charging state (charge 100%)
	Power surplus 20%
	Power surplus 40%
	Power surplus 60%
	Power surplus 80%
	Power surplus 100%
	Enable auto answer
	Do not disturb
	Mute
	Call forward is enable

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 Menu.

Step 2. Use the ▲ and ▼ navigation keys to **Setting--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 soft key from 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.

Status	Network	Wireless	SIP Account	Phone	Administration
Line	SIP Settings	VoIP QoS			

Basic			
Basic Setup			
Line Enable	Enable ▼	Outgoing Call without Registration	Disable ▼
Proxy and Registration			
Proxy Server	<input type="text"/>	Proxy Port	5060
Outbound Server	<input type="text"/>	Outbound Port	5060
Backup Outbound Server	<input type="text"/>	Backup Outbound Port	5060
Subscriber Information			
Display Name	<input type="text"/>	Phone Number	<input type="text"/>
Account	<input type="text"/>	Password	<input type="text"/>

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. Press the dial key to enter the destination number.

Step 2. 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 menu button to enter the main menu and use the softkey to select phone book.(or you can just press second softkey "pbook" to enter phonebook.)

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 menu button to enter the menu item and use the ▲ and ▼ navigation keys or enter the number 1 to **1. Phonebook**; Or use the phonebook shortcut kkeys in the LCD.

Step 2. Use the ▲ and ▼ 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 menu 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**" 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 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 connecting 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" 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  button to resume the call with B.

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" 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 ▲ and ▼ 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.

The screenshot shows the LCD configuration interface. At the top, there are tabs: Status, Network, Wireless, SIP Account, **Phone**, and Administration. Below these are sub-tabs: Preferences, Multi-Functional Key, Dial Rule, Phonebook, Call Log, Action URL, and Web Dial. The 'Preferences' sub-tab is selected. Under 'Preferences', there is a section titled 'Volume Settings'. It contains six items: Handset Input Gain (5), Speakerphone Input Gain (5), Ringer Volume (5), Handset Volume (5), Speaker Volume (5), and Speakerphone Mic Boost (Disable).

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

The screenshot shows the 'Features' and 'Call Forward' sections. In the 'Features' section, 'All Forward' is set to 'Disable' (marked with a red box and '1'), 'No Answer Forward' is 'Disable', 'DND' is 'Disable', 'Busy Forward' is 'Disable', and 'Transfer On Hook' is 'Enable'. In the 'Call Forward' section, 'All Forward' is empty (marked with a red box and '2'), 'No Answer Forward' is empty, 'Busy Forward' is empty, and 'No Answer Timeout' is '20'.

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 mu button.

Step 2. Use the ▲ and ▼ navigation keys to **Setting---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.

The screenshot shows the LCD configuration interface. At the top, there are tabs for Status, Network, Wireless, SIP Account, **Phone**, and Administration. Below these, there are sub-tabs: Preferences, Multi-Functional Key, Dial Rule, Phonebook, Call Log, Action URL, and Web Dial. The 'Preferences' sub-tab is selected. Below the sub-tabs, there is a 'Preferences' section. Under this section, there is a 'Volume Settings' section. It contains two columns of settings: Handset Input Gain (5), Speakerphone Input Gain (5), Ringer Volume (5), Handset Volume (5), Speaker Volume (5), and Speakerphone Mic Boost (Disable).

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

The screenshot shows the LCD configuration interface. Under the 'Features' section, there are three rows: All Forward (Disable), No Answer Forward (Disable), and DND (Disable). To the right of these, there is a red box around the 'Busy Forward' option (Disable) with a red number 1 next to it. Below this, there is a 'Call Forward' section. It contains two rows: All Forward (empty field) and No Answer Forward (empty field). To the right of these, there is a red box around the 'Busy Forward' option (empty field) with a red number 2 next to it. Below this, there is a 'No Answer Timeout' option (20).

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 Menu button.

Step 2. Use the ▲ and ▼ navigation keys to **Setting---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.

The screenshot shows the LCD Configuration interface. At the top, there are tabs for Status, Network, Wireless, SIP Account, **Phone**, and Administration. Below these are sub-tabs: Preferences, Multi-Functional Key, Dial Rule, Phonebook, Call Log, Action URL, and Web Dial. The 'Preferences' sub-tab is selected. Under 'Preferences', there is a 'Volume Settings' section with the following fields:

Handset Input Gain	5 ▼	Handset Volume	5 ▼
Speakerphone Input Gain	5 ▼	Speaker Volume	5 ▼
Ringer Volume	5 ▼	Speakerphone Mic Boost	Disable ▼

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

The screenshot shows the 'Features' and 'Call Forward' sections. In the 'Features' section, the 'No Answer Forward' option is highlighted with a red box and labeled '1'. In the 'Call Forward' section, the 'No Answer Forward' field is highlighted with a red box and labeled '2', and the 'No Answer Timeout' field is highlighted with a red box and labeled '3'.

Step 3. Enable the no answer forward at position 1 and fill destination number at position 2. Position 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 Menu button.

Step 2. Use the ▲ and ▼ navigation keys to the **Setting- Preference** 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 **'VoIP-Line '** Configuration page.

Status **Network** **Wireless** **SIP Account** **Phone** **Administration**

Line SIP Settings VoIP QoS

Basic

Basic Setup

Line Enable Outgoing Call without Registration

Proxy and Registration

Proxy Server Proxy Port
 Outbound Server Outbound Port
 Backup Outbound Server Backup Outbound Port

Subscriber Information

Display Name Phone Number
 Account Password

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

Supplementary Service Subscription

Supplementary Services

Call Waiting Hot Line
 MWI Enable Voice Mailbox Numbers
 MWI Subscribe Enable


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 connecting 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 Menu button.

Step 2. Use the ▲ and ▼ navigation keys **Setting---- Preferences** 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.

Preferences			
Volume Settings			
Handset Input Gain	5 ▼	Handset Volume	5 ▼
Speakerphone Input Gain	5 ▼	Speaker Volume	5 ▼
Ringer Volume	5 ▼	Speakerphone Mic Boost	Disable ▼

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

Miscellaneous	
Auto Answer	Disable ▼ ¹
Dial Time Out(IDT)	5 ²
Auto Hookon Mode	Enable ▼
ICMP Ping	Disable ▼
Auto Answer by CallINFO	Disable ▼
Call Immediately Key	# ▼
Preferred Audio Device	Disable ▼
Escaped char enable	Disable ▼

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:

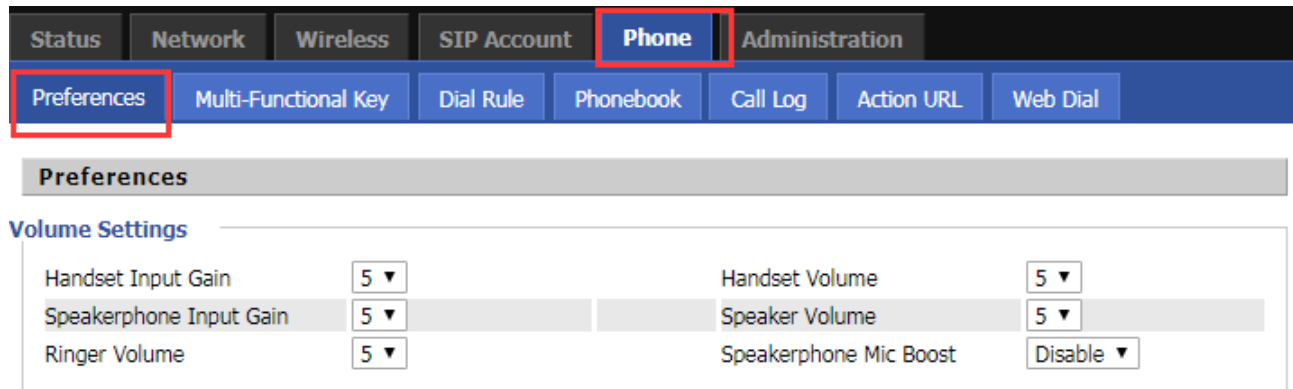
DND:Do not disturb, reject all calls.

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.

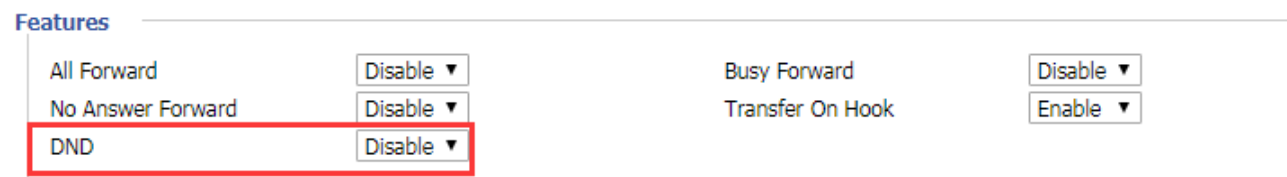
From Web page:

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



Preferences			
Volume Settings			
Handset Input Gain	5 ▼	Handset Volume	5 ▼
Speakerphone Input Gain	5 ▼	Speaker Volume	5 ▼
Ringer Volume	5 ▼	Speakerphone Mic Boost	Disable ▼


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



Features			
All Forward	Disable ▼	Busy Forward	Disable ▼
No Answer Forward	Disable ▼	Transfer On Hook	Enable ▼
DND	Disable ▼		

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 Menu buttons,
- Step 2. Use the ▲ and ▼ navigation keys to **Setting- 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.

- Step 3. When uploading the blacklist, please note that the file format is .csv file

	A	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

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

SMS

Send text messages

Step 1. Press Menu buttons,

Step 2. Use the ▲ and ▼ navigation keys to **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 message function, 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

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 Menu button

Step 3. Use the ▲ and ▼ 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 ▲ and ▼ 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 menu button to enter the menu item and then select **Setting---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.

The screenshot shows the 'SIP Account' configuration page for 'Line 1'. The top navigation bar includes 'Status', 'Network', 'SIP Account' (highlighted), 'Phone', and 'Administration'. Below this, the 'Line 1' tab is selected, showing 'SIP Settings' and 'VoIP QoS' sub-tabs. The 'Basic Setup' section contains two settings: 'Line Enable' with a dropdown menu set to 'Enable', and 'Outgoing Call without Registration' with a dropdown menu set to 'Disable'.


Step 2. Enable MWI and fill in the Voice Mail Number.

The screenshot shows the 'Supplementary Service Subscription' page. The 'Supplementary Services' section contains three settings: 'Call Waiting' with a dropdown menu set to 'Enable' (marked with a red '1'), 'MWI Enable' with a dropdown menu set to 'Enable' (highlighted with a red box), and 'MWI Subscribe Enable' with a dropdown menu set to 'Disable' (highlighted with a red box and marked with a red '2'). To the right, there are two empty input fields for 'Hot Line' and 'Voice Mailbox Numbers'.

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 background](#)
- [Reboot](#)
- [Restore Factory](#)
- [Other parameters of the phone](#)

Change language

Step 1. Use menu button, then select **Setting---Preferences**.

Step 2. Use the ▲ and ▼ navigation keys to select **Setting---Preferences**.

Step 3. Select **1. Language**, use the "select" softkey to enter the language selection interface

Step 4. Use the ▲ and ▼ keys to select the language and use the "select" softkey to save the selected language.

Change ringtone

Step 1. Use Menu button and then select **Setting---Sound--- Ring Tones**.

Step 2. Use the ▲ and ▼ navigation keys or enter the number 6 to select **6 Ring Tones**.

Step 3. Use the ▲ and ▼ keys to select a ringtone.

Step 4. Use the softkey "select" to save the selected ringtone.

Change Background

Change the background:

Step 1. Press menu button to select **Setting--Display--Background**.

Step 2. Select the background picture that you like.

Step 4. Press the "OK" softkey to save the changes.

Reboot

Step 1. Press menu button to select **Setting---Reboot**, Or just press the power button for a long time to 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 menu button to select **Setting---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
1.Phonebook	New Entry		using four soft keys, you can add numbers to the phone book.and up to 100 records can be saved
	Contact		1. Select a contact to view the contact number directly. 2. Use the four soft keys can dial someone, sending text messages, editing, viewing, deleting, moving to blacklist
2.Call History	Redial List		1. List the last 60 calls
	Answered Calls		2. Use the four soft keys to complete the quick call, send text messages, save to the phone book and other functions
	Missed Calls		
3.Text Message	SEND		editor and send text messages
	Recived Box		can list the latest send and receive 100 messages
	Send Box		
4.Black List	New Entry		With four soft keys, you can add numbers to the blacklist,Up to 100 records can be saved
	Contact		1. Select a blacklist contact to view the contact number directly 2. Use the four soft keys to complete the call, send text messages, edit, view, delete, move to the phone book and other functions
5.Preference	Language		Select a different language
	Call Waiting		enable or disable call waiting
	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		

6.Ring Tone	BellType 3		Select a ringtone type, you can listen to the ringtones, easy to distinguish and choose the ringtones
	BellType 4		
	BellType 5		
	BellType 6		
	BellType 7		
	BellType 8		
	BellType 9		
	BellType 10		
7.Accounts	Line	Line Enable	Enable or disable lines
		Password	The password for the SIP account
		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
		Outbound Server	Proxy server IP address / domain name
		Outbound Server port	Proxy server port
		Backup Outbound Server	Backup Outbound Server IP Address / Domain Name
		Backup 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
		Off Code	Used to disable the all forward
	CFWD When Busy	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 No Answer	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
11.Network	WAN Connection Type		1.View or change the current Internet port connection type, 2. Internet port connection type can be set to static, DHCP and PPPoE
	Current IP		Change the IP address of the current Internet port
	Current Netmask		Change the subnet mask of the current network,
	Current Gateway		Change the gateway of the current network
	Primary DNS		Change the Primary DNS of the current network
	Secondary DNS		Change the Secondary 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
12.Wireless	Wireless Setting	Wifi Country Region	View or change the current Wifi area
		Wifi Connection Type	1. View or change the Wifi connection type 2. Currently supports DHCP and static types
		Current IP	View or change the IP address of the current wireless network
		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

		Secondary DNS	View or change the Secondary DNS of the current wireless network
	Wireless Connection	AP	To display the name of all APs.
13.Product INFO	Product Name		To view the current information of Product Name, Software Version, Hardware Version and MAC Address.
	Software Version		
	Hardware Version		
	MAC Address		
	Serial Number		
14.Status	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
	Registration Status	The 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.
17.Set Password			To reset password. The password of LCD is same as the one of Webpage. Default is null.

Chapter 4 Web Configuration

This chapter contains:

- [Login](#)
- [SIP account](#)
- [Line](#)
- [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:

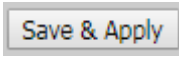

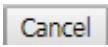
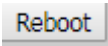
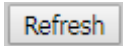
Step 1. Connect the phone correctly and make sure that the device and the computer in the same network.

Step 2. View the device IP: Enter the menu, use the ▲ and ▼ navigation keys to select the **Setting---status**, the select network view 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.

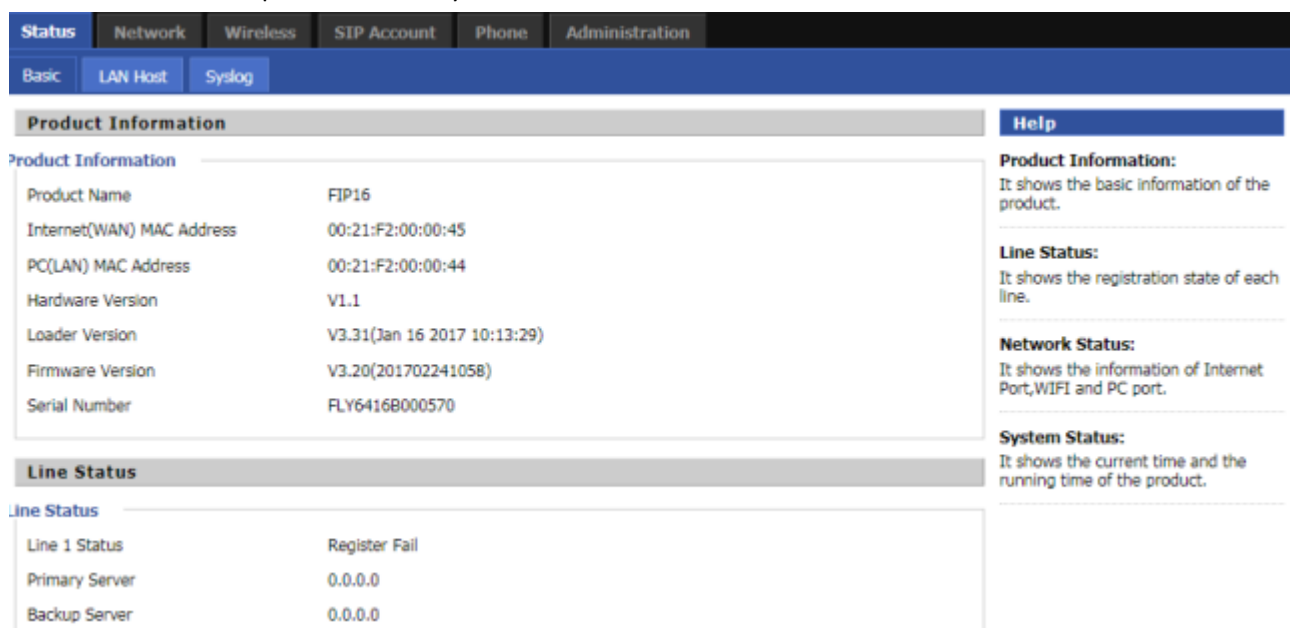
Serial number	Name	Description
Postition 1	Main navigation bar	Click this navigation bar to bring up the corresponding child navigation bar
Postition 2	navigation bar	Click the sub navigation bar to enter the configuration page

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
		Use this button,conifg will be saved and apply
		After changing the parameters, you need to click this button to save. After you click Save, there is a need to restart the device.
		Click to cancel the change
		Click to restart
		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.



Status | Network | Wireless | SIP Account | Phone | Administration

Basic | LAN Host | Syslog

Product Information

Product Name: FIP16

Internet(WAN) MAC Address: 00:21:F2:00:00:45

PC(LAN) MAC Address: 00:21:F2:00:00:44

Hardware Version: V1.1

Loader Version: V3.31(Jan 16 2017 10:13:29)

Firmware Version: V3.20(201702241058)

Serial Number: FLY6416B000570

Line Status

Line 1 Status: Register Fail

Primary Server: 0.0.0.0

Backup Server: 0.0.0.0

Help

Product Information:
It shows the basic information of the product.

Line Status:
It shows the registration state of each line.

Network Status:
It shows the information of Internet Port,WIFI and PC port.

System Status:
It shows the current time and the running time of the product.

LAN Host

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

Status	Network	Wireless	SIP Account	Phone	Administration	
Basic	LAN Host	Syslog				
LAN Host info						
MAC Address	IP Address	Interface Type	Address Source	Expires	Host name	Status
Ipv6 LAN Host Info						
MAC Address	IPv6 Address			Expires		

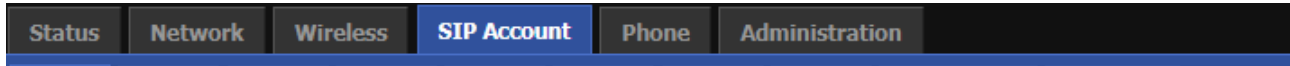
Syslog

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 log.

Status	Network	SIP Account	Phone	Administration
Basic	LAN Host	Syslog		
<div>Refresh Clear Save</div> <pre> Manufacturer:FLYINGVOICE ProductClass:FIP16W SerialNumber:FLY95186000030 BuildTime:201805171807 IP:192.168.1.1 HWVer:V4.1 SWVer:V3.20 <Thu Jul 19 10:46:59 2018> tr069[12835]: tr069.c <Thu Jul 19 10:47:24 2018> tr069[12835]: tr069.c <Thu Jul 19 10:47:26 2018> LinkStatus: WAN Link Down <Thu Jul 19 10:47:29 2018> wificlient: APscan FV_508_VOICE 00:21:f2:12:e2:bc channel[14] signal[83] <Thu Jul 19 10:47:30 2018> LinkStatus: WAN Link Down <Thu Jul 19 10:47:32 2018> tr069[12835]: tr069.c <Thu Jul 19 10:47:32 2018> tr069[12835]: tr069.c <Thu Jul 19 10:47:34 2018> udhcpc[6737]: Performing a DHCP renew <Thu Jul 19 10:47:34 2018> udhcpc[6737]: Sending renew... <Thu Jul 19 10:47:34 2018> udhcpc[6737]: bind(UDP): Cannot assign requested address <Thu Jul 19 10:47:34 2018> udhcpd[17880]: Received a SIGTERM <Thu Jul 19 10:47:35 2018> tr069[12823]: 9: Invalid argument <Thu Jul 19 10:47:35 2018> tr069[12823]: nat detect session fail! to start again after 10s. <Thu Jul 19 10:47:37 2018> udhcpc[6737]: Sending renew... <Thu Jul 19 10:47:37 2018> udhcpc[6737]: bind(UDP): Cannot assign requested address <Thu Jul 19 10:47:40 2018> LinkStatus: WAN Link Down <Thu Jul 19 10:47:40 2018> udhcpc[6737]: Sending renew... <Thu Jul 19 10:47:40 2018> udhcpc[6737]: bind(UDP): Cannot assign requested address <Thu Jul 19 10:47:42 2018> LinkStatus: WAN Link Down <Thu Jul 19 10:47:43 2018> udhcpd[28151]: udhcpd (v1.12.1) started <Thu Jul 19 10:47:45 2018> LinkStatus: WAN Link Down <Thu Jul 19 10:47:48 2018> LinkStatus: WAN Link Down <Thu Jul 19 10:47:50 2018> LinkStatus: WAN Link Down <Thu Jul 19 10:47:50 2018> udhcpc[6737]: Sending select for 192.168.10.143... </pre>				

SIP Account

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

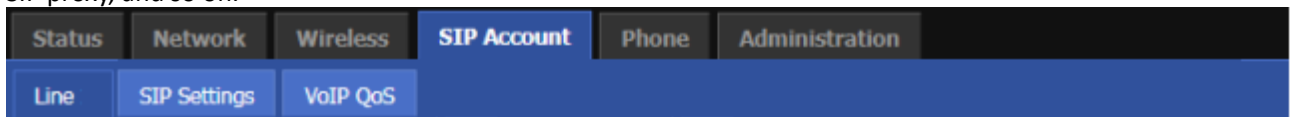


Account setting

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.



Basic

Basic Setup

Line Enable

Outgoing Call without Registration

Proxy and Registration

Proxy Server

Proxy Port

Outbound Server

Outbound Port

Backup Outbound Server

Backup Outbound Port

Subscriber Information

Display Name

Phone Number

Account

Password

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:

Audio Configuration

Codec Setup

Audio 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 200OK 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	<input type="text"/>
MWI Enable	Enable ▼	Voice Mailbox Numbers	<input type="text"/>
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

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

RTP Advanced Setup

RTP Port Min	<input type="text" value="0"/> (0 means auto select)	RTP Port Max	<input type="text" value="50000"/>
--------------	--	--------------	------------------------------------

Parameter name	Description
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.
Caller ID Header	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
SIP 100REL Enable	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)	Set Reg Subscribe Interval,default is 0
Dial Prefix	Dial before prefix
User Type	Whether the end user is IP or Phone
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	Set BLF List URL
BLF PickUp Code	Set BLF PickUp Code
RTP Port Min	RTP minimum port
RTP Port Max	RTP's maximum port

SIP settings

SIP Parameters

Status	Network	SIP Account	Phone	Administration
Line 1	SIP Settings	VoIP QoS		
Basic				
Basic Setup				
Line Enable	Enable ▼	Outgoing Call without Registration	Disable ▼	
Proxy and Registration				
Proxy Server	192.168.10.88	Proxy Port	5060	
Outbound Server		Outbound Port	5060	
Backup Outbound Server		Backup Outbound Port	5060	
Allow DHCP Option 120 to Override SIP Server	Disable ▼			
Subscriber Information				
Display Name	608	Phone Number	608	
Account	608	Password	••••••••	
Audio Configuration				
Codec Setup				
Audio 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 ▼	
Echo Cancel	Enable ▼			
Auto Gain Control	Enable ▼	Use First Matching Vocoder in 200OK SDP	Disable ▼	
Codec Priority	Remote ▼	Packet Cycle Follows Remote SDP	Disable ▼	

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

NAT Traversal

NAT Traversal

NAT Traversal	Disable ▼	STUN Server Address	<input type="text"/>
NAT Refresh Interval(sec)	60	STUN Server Port	3478

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

StatusNetworkWirelessSIP AccountPhoneAdministration

LineSIP SettingsVoIP QoS

QoS Settings

Layer 3 QoS

SIP QoS(0-63)

0

RTP QoS(0-63)

0

Save

Cancel

Reboot

Parameters name	Description
SIP QoS(0-63)	Defaults to 46
RTP QoS(0-63)	Defaults to 46

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

Network

LAN

PC Port(LAN)

PC 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:

Status	Network	Wireless	SIP Account	Phone	Administration				
WAN	LAN	IPv6 Advanced	IPv6 WAN	IPv6 LAN	VPN	Port Forward	DMZ	Routing	Advance

VPN Settings

Administration

VPN Enable PPTP ▼
Initial Service IP
User Name
Password
VPN As Default Route Disable ▼
MPPE Stateful Disable ▼
Require MPPE Disable ▼

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	Enable or disable the MPPE Stateful
Require MPPE	Enable or disable the Require MPPE

L2TP

StatusNetworkWirelessSIP AccountPhoneAdministration

WANLANIPv6 AdvancedIPv6 WANIPv6 LANVPNPort ForwardDMZRoutingAdvance

VPN Settings

Administration

VPN Enable

L2TP

Initial Service IP

User Name

Password

.....

L2TP Tunnel Name

L2TP Tunnel Password

.....

VPN As Default Route

Disable

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.
L2TP Tunnel Name	L2TP Tunnel Name
L2TP Tunnel Password	L2TP Tunnel Password
VPN As Default Route	Prohibited or open, the default is prohibited.

OpenVPN

Status	Network	Wireless	SIP Account	Phone	Administration				
WAN	LAN	IPv6 Advanced	IPv6 WAN	IPv6 LAN	VPN	Port Forward	DMZ	Routing	Advance

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 FIP16 / FIP16 / FIP16L / FIP16P 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.

Status	Network	Wireless	SIP Account	Phone	Administration				
WAN	LAN	IPv6 Advanced	IPv6 WAN	IPv6 LAN	VPN	Port Forward	DMZ	Routing	Advance

Demilitarized Zone (DMZ)

DMZ Setting

DMZ Enable	Enable ▼
DMZ Host IP Address	<input type="text"/>

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

Routing

StatusNetworkWirelessSIP AccountPhoneAdministration

WANLANIPv6 AdvancedIPv6 WANIPv6 LANVPNPort ForwardDMZRoutingAdvance

Static Routing Settings

Add a routing rule

Destination

Host/Net

Gateway

Interface

Comment

Host ▾

LAN ▾

Apply

Reset

Current Routing table in the system

No.	Destination	Mask	Gateway	Flags	Metric	Interface	Comment
-----	-------------	------	---------	-------	--------	-----------	---------

Delete Selected

Reset

Add a routing rule:

Parameters name	Description
Destination	The destination address of the routing rule to be added
Host/Net	Select the way you want to add
Gateway	The IP address of the gateway
Interface	Select whether the LAN port or other network interface
Comment	Fill in memo information
Current Routing table in the system	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.

StatusNetworkWirelessSIP AccountPhoneAdministration

PreferencesMulti-Functional KeyDial RulePhonebookCall LogAction URLWeb Dial

Preferences

VolumeSettings

Volume Settings

Handset Input Gain

Speakerphone Input Gain

Ringer Volume

Handset Volume

Speaker Volume

Speakerphone Mic Boost

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

Regional

Tone Type

Dial Tone

Busy Tone

Off Hook Warning Tone

Ring Back Tone

Call Waiting Tone

Min Jitter Delay(0-600ms)

Max Jitter Delay(20-1000ms)

Ringing Time(10-300sec)

Parameters name	Description
Tone Type	Select the tone type.
Dial Tone	Dial tone
Busy Tone	Busy tone.
Off Hook Warning Tone	Hang up warning tone
Ring Back Tone	Ring back tone
Call Waiting Tone	Call waiting tone
Min Jitter Delay(0-600ms)	jitter delay the minimum value, the phone jitter using the adaptive mechanism.
Max Jitter Delay(20-1000ms)	jitter delay the maximum value of the phone jitter using the adaptive mechanism.
Ringing Time(10-300sec)	Ring time.

Miscellaneous

Miscellaneous

Auto Answer	Disable ▾	Auto Answer by CallINFO	Disable ▾
Dial Time Out(IDT)	5	Call Immediately Key	# ▾
Auto Hookon Mode	Enable ▾	Preferred Audio Device	Disable ▾
ICMP Ping	Disable ▾	Escaped char enable	Disable ▾

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	Enable or disable the Auto Hookon Mode .
Preferred Audio Device	Enable or disable the 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.

bulk add

Status	Network	Wireless	SIP Account	Phone	Administration		
Preferences	Multi-Functional Key	Dial Rule	Phonebook	Call Log	Action URL	Web Dial	

Phonebook Upload && Download

Phonebook Upload && Download

Local File 未选择任何文件

Blacklist Upload && Download

Blacklist Upload && Download

Local File 未选择任何文件

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

Blacklist Upload &&

Upload or download the blacklist file in CSV format

The format of the batch is:

	A	B	C	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:

Phonebook				
Index	Name	Number	Ring	

Name
 Number
 Ring

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

Redial List				
Index	NUMBER	Start Time	Duration	<input type="checkbox"/>
1	601	10/17 19:54	00:00:01	<input type="checkbox"/>
2	1234	10/17 19:55	00:00:01	<input type="checkbox"/>
3	585852145865	10/17 19:55	00:00:01	<input type="checkbox"/>

Answered Calls

Answered Calls				
Index	NUMBER	Start Time	Duration	<input type="checkbox"/>
1	601	10/17 19:55	00:00:00	<input type="checkbox"/>

Missed Calls

Missed Calls				
Index	NUMBER	Start Time	Duration	<input type="checkbox"/>
1	601	10/17 19:56	00:00:00	<input type="checkbox"/>

Dial Rule

Status	Network	Wireless	SIP Account	Phone	Administration
Preferences	Multi-Functional Key	Dial Rule	Phonebook	Call Log	Action URL
Web Dial					

Dial Plan

General

Dial Plan Enable ▼
 Unmatched Policy Accept ▼

No.	Line	Digit Map	Action	Move Up	Move Down	
Line	Line1 ▼		Action			
Digit Map						
Action		Deny ▼				
<div>OK Cancel</div>						

Parameters name	Description
Dial Plan	Whether to enable dial plan
Unmatched 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
x	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 *
x.	Matchx, xx, xxx, xxxx, xxxxx..... For example: "01."can match "0", "01", "011", "0111", , "01111..."
<diald: substituted>	Replace For example: <8:1650>123456: input "85551212", output"16505551212"
x,y	Enter "x" will have a dial tone, enter "y" after the dial tone stops For example: "9,1xxxxxxxxx": 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
T	Set the delay time For example: "<9: 111> T2": the phone will broadcast a valid number "2"

No.	Line	Digit Map	Action	Move Up	Move Down	
1	Line1	<:010>#12<#:%23>2	Dial Out	⬆	⬇	<input type="checkbox"/>
2	Line2	<5,:><:241333>8101	Dial Out	⬆	⬇	<input type="checkbox"/>
3	Line3	<[4-5]:>22xxxx<:333>	Dial Out	⬆	⬇	<input type="checkbox"/>
4	Line4	<2-3,:5:>622.	Dial Out	⬆	⬇	<input type="checkbox"/>
5	Line5	777x.8	Deny	⬆	⬇	<input type="checkbox"/>

Example 1 points to Line 1

Example 2 points to Line 2

Example 3 points to Line 3

Example 4 points to Line 4

Example 5 points to Line 5

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.

Status	Network	Wireless	SIP Account	Phone	Administration				
Management	Firmware Upgrade	Scheduled Tasks	Certificates	Provision	SNMP	TR069	Diagnosis	Operating Mode	

Management

Time/Date Setting

Time/Date Setting

NTP Settings

NTP Enable	Enable ▼
Option 42	Disable ▼
Current Time	2017 - 10 - 13 . 19 : 05 : 14
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	Set the 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	Admin User ▼
New User Name	admin
New Password	<input type="password"/> (The maximum length is 25)
Confirm Password	<input type="password"/>

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

Web Access

Remote Web Login	Enable ▼
Web Port	80
Web SSL Port	443
Web Idle Timeout(0 - 60min)	5
Allowed Remote IP(IP1;IP2;...)	0.0.0.0

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	You can set the web SSL port,default is 443
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

System Log Setting

Syslog Setting

Syslog Enable	Enable ▼
Syslog Level	INFO ▼
Remote Syslog Enable	Disable ▼
Remote Syslog Server	

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

Factory Defaults Setting

Factory Defaults Setting

Factory Defaults Lock	Disable ▼
Zero Config	Enable ▼

Factory Defaults

Reset to Factory Defaults Factory Default

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

Firmware Management

Status	Network	Wireless	SIP Account	Phone	Administration			
Management	Firmware Upgrade	Scheduled Tasks	Certificates	Provision	SNMP	TR069	Diagnosis	

Firmware Management

Firmware Upgrade

Local Upgrade 选择文件 未选择任何文件

Upgrade

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.

Status	Network	Wireless	SIP Account	Phone	Administration
Management	Firmware Upgrade	Scheduled Tasks	Certificates	Provision	SNMP
TR069	Diagnosis				

Provision

Configuration Profile

Provision Enable	Enable ▾
Resync On Reset	Enable ▾
Resync Random Delay(sec)	40
Resync Periodic(sec)	3600
Resync Error Retry Delay(sec)	3600
Forced Resync Delay(sec)	14400
Resync After Upgrade	Enable ▾
Resync From SIP	Disable ▾
Option 66	Enable ▾
Option 67	Enable ▾
Config File Name	\$(MA)
User Agent	
Profile Rule	http://prv1.flyingvoice.net:69/config/\$(MA)?mac=\$(MA)&

Firmware Upgrade

Upgrade Enable	Enable ▾
Upgrade Error Retry Delay(sec)	3600
Upgrade Rule	

Save & Apply Save Cancel Reboot

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	Set the 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

Status	Network	Wireless	SIP Account	Phone	Administration			
Management	Firmware Upgrade	Scheduled Tasks	Certificates	Provision	SNMP	TR069	Diagnosis	

TR069 Configuration

ACS

TR069 Enable	Enable ▾
CWMP	Enable ▾
ACS URL	<input type="text" value="http://acs1.flyingvoice.net:8080/tr069"/>
User Name	<input type="text" value="FLY6416B000570"/>
Password	<input type="password" value="....."/>
Periodic Inform Enable	Enable ▾
Periodic Inform Interval	<input type="text" value="1800"/>

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 the phone have enough .

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.

Audio Configuration			
Codec Setup			
Audio 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 ▼

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.

FCC Statement

This device complies with Part 15 of the FCC rules. Operation is subject to the following two conditions: 1) this device may not cause harmful interference, and 2) this device must accept any interference received, including interference that may cause undesired operation.

Note: 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 generates uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in particular installation. 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 interference 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.

Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

FCC Radiation Exposure Statement

This device complies with FCC RF radiation exposure limits set forth for an uncontrolled environment. This transmitter must not be co-located or operating in conjunction with any other antenna or transmitter.