X306 User Manual

Software Version: 1.0.0.0

Release Date：2025/9/26

Directory

1 Picture

2 Table

3 Safety Instruction

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

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

4 Overview

4.1 Overview

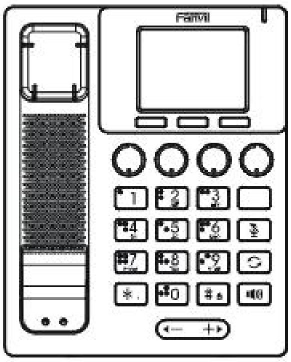
X306 is a network phone designed specifically for the internet age, suitable for small and medium-sized enterprises and households. The device provides excellent user experience for home and office users with its simple design. The device is not just a desktop phone, but also a masterpiece placed in the living room or office.

X306 is the latest generation of enterprise level network telephones developed on the basis of the X series research and development, inheriting many excellent features of the previous X series telephones, such as high-definition voice, headphones, high-performance echo cancellation full duplex speakers, 100 Mbps Ethernet, QoS, encrypted transmission, automatic configuration, etc; A brand new system, smooth operation, flat interface settings, and many other advantages; In addition,the X306 is also equipped with a HELP emergency call button, which can be used to seek help in emergency situations with the HELP button. Its buttons break the size design of conventional phones and add Braille to each number button, making it a versatile phone that can serve special groups.

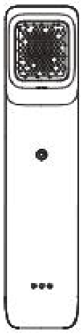
For enterprise users, the device is a cost-effective office device that provides convenient operation while achieving environmental protection; For home users, the device is a highly efficient communication device, allowing users to flexibly configure and customize the functions of the DSS key, saving space and costs. For enterprise and home users who pursue high quality and efficiency, it will be a very ideal choice.

In order to help some interested users better understand the details of the product, this user manual can serve as a reference guide for the use of the device. This document may not be applicable to the latest version of the software. If you have any questions, you can use the built-in help prompt interface of the device phone, or download and update your user manual from the official Fanvil website.

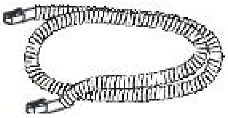
4.2 Packing Contents



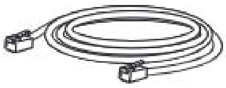
IP Phone



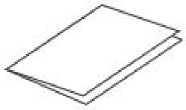
Handset



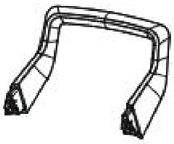
Handset Cord



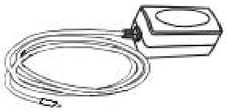
Ethernet Cord



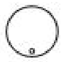
Quick Installation Guide



Stand



Power Adapter (Optional)



Contact Card*4pcs

5 Desktop Installation

5.1 PoE And the use of external power adapters

X306, called as ‘the device’ hereafter,supports two power supply modes, power supply from external power adapter or over Ethernet (PoE) complied switch.

PoE power supply saves the space and cost of providing the device additional power outlet. With a PoE switch, the device can be powered through a single Ethernet cable which is also used for data transmission. By attaching UPS system to PoE switch, the device can keep working at power outage just like traditional PSTN telephone which is powered by the telephone line.

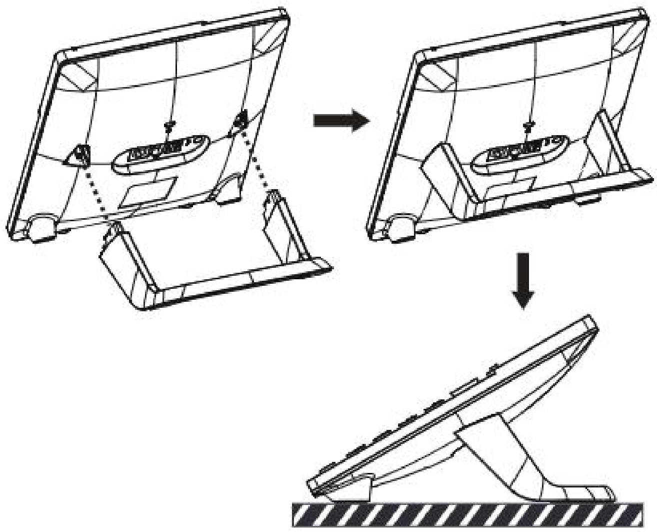
For users who do not have PoE equipment, the traditional power adaptor should be used. If the device is connected to a PoE switch and power adapter at the same time, the PoE power will be used in priority and will switch to the power adapter supply once it fails.

Please use the power adapter supplied by Fanvil and the PoE switch met the specifications to ensure the device work properly.

5.2 Desktop mounted method

Please install the phone according to the instructions in the picture below.

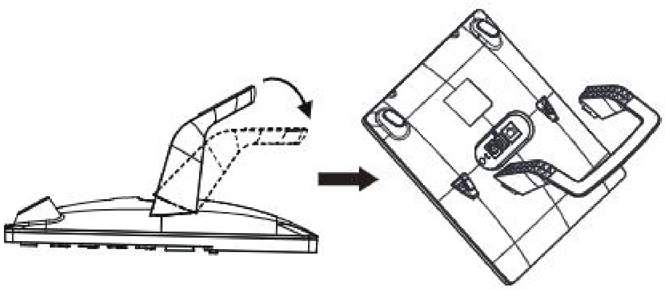
1）Bracket desktop installation



Picture 1 - Desktop installation

2）Disassemble the desktop stand

Pull the bracket in the direction of the arrow shown in the diagram (towards the screen) to remove it.



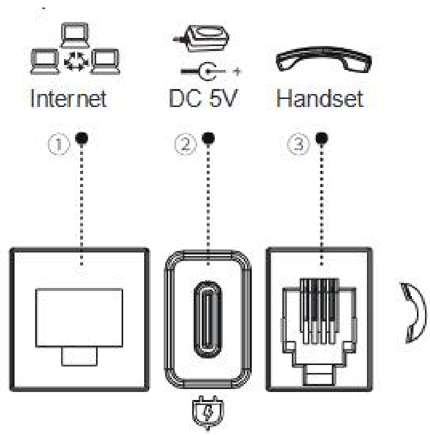
Picture 2 - Desktop disassembly

Connect the power adapter, network, PC, phone and Headset to the appropriate port as shown in the picture below.

① Network interface: connect to LAN or Internet

② Power interface: Connect power adapter

③ Controller interface: Connect the phone controller



Picture 3 - Connecting to the Device

6 Appendix Table

6.1 Appendix I - Icon

Table 1 - Keypad Icons

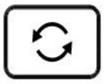
	increase or decrease ringer volume Left (-) or Right (+) navigation keys
	Redial key
	Hand-free key
	HELP speed dial key (red)

Table 2 - Screen status icon display

	In hands-free mode
	In headset mode
	In handset mode
	Mute activated
	Silent mode
	Call is on hold
	Auto-answering activated
	Call forward activated
	VLAN activated
	VPN activated
	Bluetooth device paired connection
	New SMS
	New VM messages
	Forward call(s)
	Missed call
	Received call(s)
	Dialed call(s)

	Internet connected
	Internet is disconnected
	No IP address
	Wireless network connected
	Wireless network disconnected
	Wireless network failure

6.2 Appendix II - Keyboard character query table

Table 3 - Look-up Table of Characters

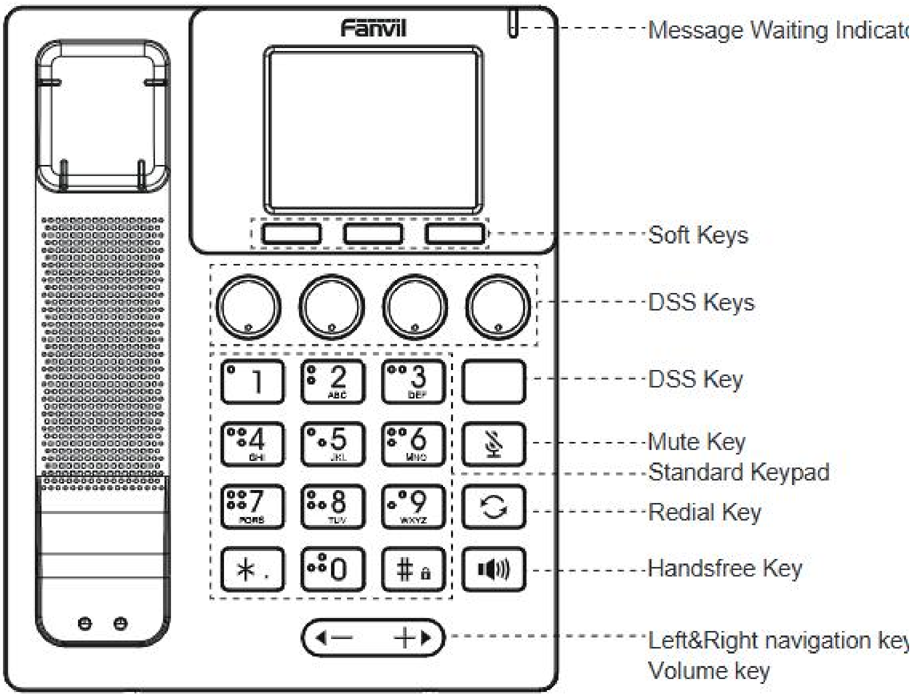
Mode Icon	Text Mode	Key Button	Characters Of Each Press
123	Numeric	1	1
		2	2
		3	3
		4	4
		5	5
		6	6
		7	7
		8	8
		9	9
		0	0
		*	*.:/@[],-_='?\"()<>{}
		#	#
abc	Lower Case Alphabets	1	@:;()<>[]{}
		2	a b c
		3	d e f
		4	g h i
		5	j k l
		6	m n o
		7	p q r s
		8	t u v
		9	w x y z
		0	(space)
		*	. , */+-:_=' ?\" |
		#	# ^!&$% £ ¥ ¤ ~ ¡ ¿ §

ABC	Upper Case Alphabets	1	@:;()<>[]{}
		2	A B C
		3	D E F
		4	G H I
		5	J K L
		6	M N O
		7	P Q R S
		8	T U V
		9	WZYX
		0	(space)
		*	.,*/+-:_= ￡ ￥¤~¡¿§
		#	# ^!&$%
2aB	Mixed type input	1	1
		2	2 a b c A B C
		3	3 d e f D E F
		4	4 g h I G H I
		5	5 j k l J K L
		6	6 m n o M N O
		7	7 p q r s P Q R S
		8	8 t u v T U V
		9	9 w z y x W Z Y X
		0	0
		*	.,*/+-:_='?\"|；()<>[]{}
		#	# ^!&$% ￡ ￥¤~¡¿§
Abc	Alphabets	1	@:;()<>[]{}
		2	a b c
		3	d e f
		4	g h i
		5	j k l
		6	m n o
		7	p q r s
		8	t u v
		9	w x y z

		$\boxed{0}$	(space)
		$\boxed{*}$	.,*/+-:_=’ ?\” |
		$\boxed{\#}$	# ^!&$% £ ￥ ¤~¡ ¿ §

7 Introduction to the User

7.1 Instruction of Keypad



Picture 4 - Instruction of Keypad

The picture above shows the keypad layout of the phone.Each button provides its own specific function.Users can refer to the instructions for the keys in the illustration in this section to operate the phone.

Table 4 - Instruction of Keypad

Number	**The keypad names**	**Instruction**
①	Soft Keys	These three buttons provide the functions displayed on the screen corresponding to them
②	DSS Keys	Can be set as line key/function key/speed dial key, etc. (default is speed dial key)
③	HELP	Quick dial key, after configuring the number, you can call out for help with just one click through the HELP button in emergency situations
④	Mute Key	During a call, the user can press this button to mute the microphone

⑤	Standard Keypad	These 12 standard phone buttons provide standard phone button functions. At the same time, long pressing certain buttons can trigger the provision of special functions
⑥	Redial Key	Press the 'redial' button to redial the last dialed number
⑦	Handsfree Key	Users can press this button to activate the hands-free speaker audio channel
⑧	Left&Right navigation key/Volume key	In standby mode, on the ringtone and ringtone configuration interface, press this button to decrease/increase the volume of the ringtone; In the call or tone adjustment interface, press this button to decrease/increase the volume. Left (-) or Right (+) navigation keys: Users can move the cursor in the screen list left/right by pressing the volume. On some settings and text editing pages, users can press the left/right navigation keys to change options or move the cursor left/right in the screen list.

7.2 Using Handset / Hands-free Speaker / headset

- **Using Handset**

To talk over handset, user should lift the handset off the device and dial the number, or dial the number first, then lift the handset and the number will be dialed. User can switch audio channel to handset by lifting the handset when audio channel is opened in speaker or headset.

- **Using Headset**

For headset usage, users can first press the headset button to dial the number, or dial the number first and then press the headset button. When the handset or speaker channel is active, users can switch the phone's audio channel by pressing the headset button.

- **Using Hands-free Speaker**

To talk over hands-free speaker, user should press the hands-free button then dial the number, or dial the number first then press the hands-free button. User can switch audio channel to the speaker from handset by pressing the hands-free button when audio channel is opened in handset.

7.3 Idle Screen

Picture 5 - Screen layout/default home screen

The image above shows the default standby screen, which is the user interface most of the time.
The upper half of the home screen shows the status of the device, information and data that can be edited (such as voice messages, missed calls, auto answer, do not disturb, network connection status, etc).The lower part of the area is the function menu button, which is also the first layer of the function menu button. Users can operate the phone through them.
Users can restore the phone to the default standby screen interface by picking up and dropping the handle.
The icon description is described in 6.1 appendix I-icon.

7.4 Phone Status

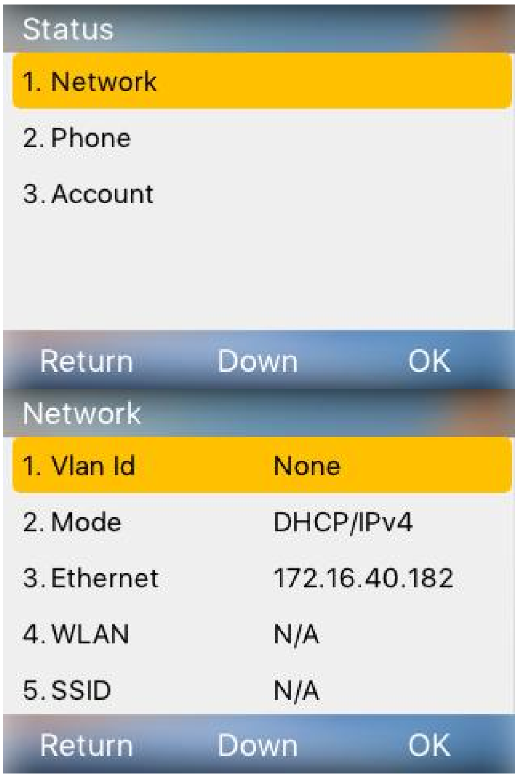
The phone status includes the following information about the phone:

- Network :
  VLAN ID
  Mode
  Ethernet
  WLAN
  SSID
- Phone:
  MAC
  Wi-Fi MAC
  Model
  Hardware
  Software
  RAM
  ROM
  UPtime
  Uboot
- Account

SIP Account

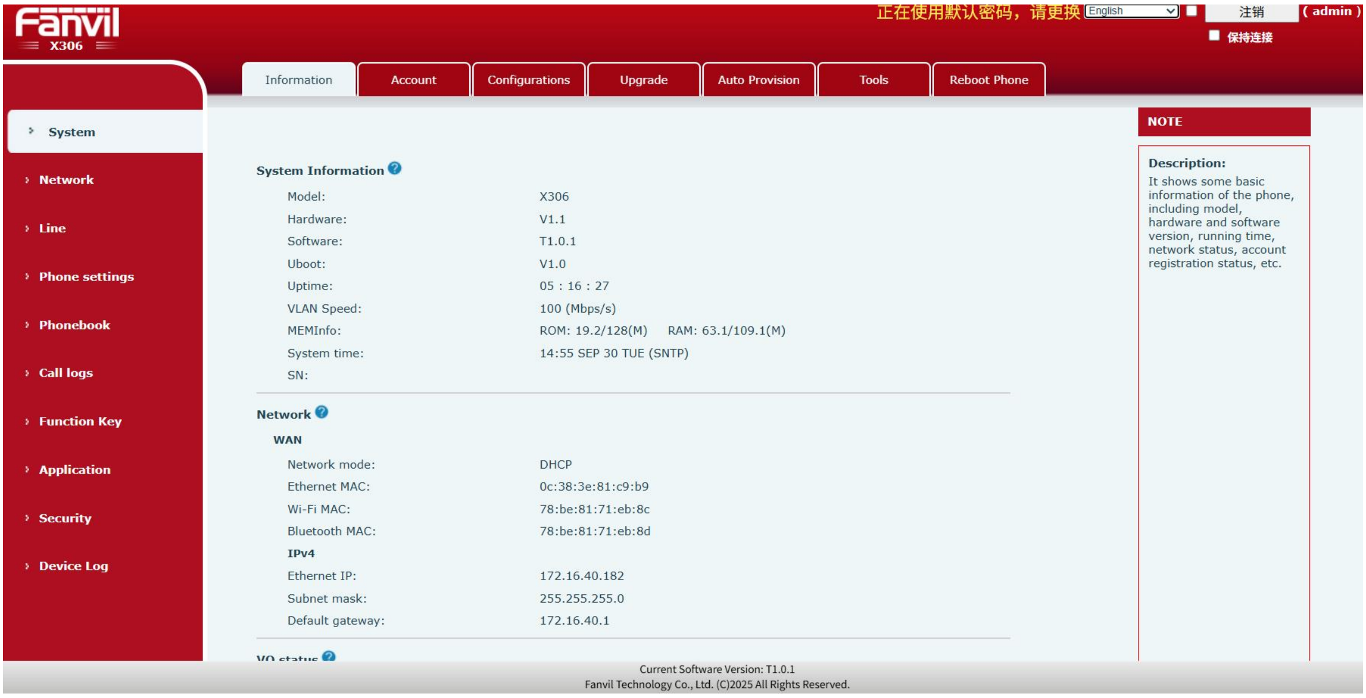
SIP Account Status (registered / Inactive / uncommitted / trying / time out)

Phone interface: When the phone is in standby mode,enter the “ # * 107” to enter the phone menu and select [Status]-[Network], or press and hold the # key to quickly enter the phone [Status], as shown in the figure:



Picture 6 - The Phone status

- WEB interface: Refer to 7.5 Web management to log in the phone page, enter the 【**System**】 >> 【**Information**】 page, and check the phone status, as shown in the figure:

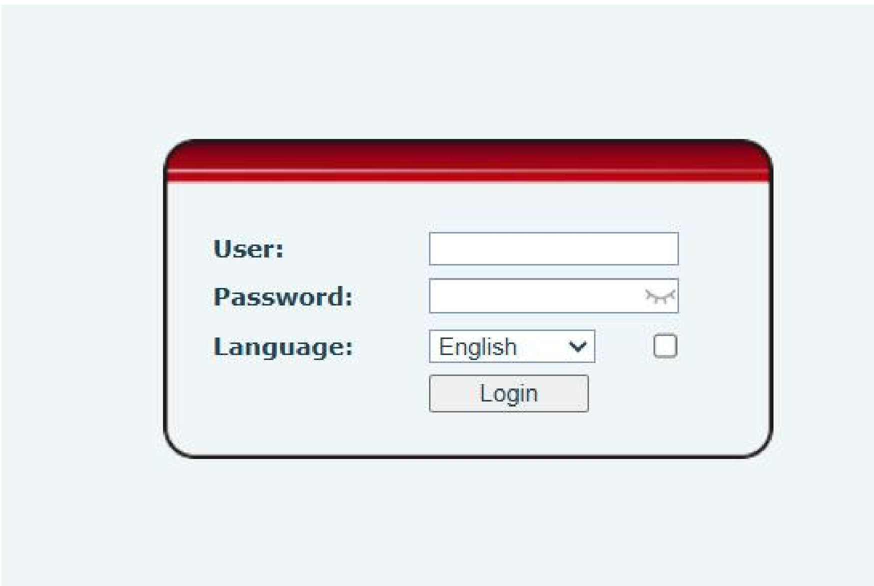


Picture 7 - WEB phone status

7.5 Web Management

Phone can be configured and managed on the web page of the phone. The user first needs to enter the IP address of the phone in the browser and open the web page of the phone. The user can check the IP address of the phone by pressing the ” # * 107” to enter the phone menu and select [Status]>>[Network], or press and hold the # key to quickly enter the phone [Network].

Open the browser, enter the phone IP, log in to the phone webpage, and the first thing you see is the phone's login page.



Picture 8 - Landing page

Users must correctly enter the user name and password to log in to the web page. The default user name and password are "admin". For the specific details of the operation page, please refer to page 11 Web configuration.

7.6 Network Configurations

The device relies on IP network connection to provide service. Unlike traditional phone system based on a circuit switched wire technology, IP devices are connected to each other over the network and exchange data in packet basis based on the devices' IP address.

To enable this phone, you must first correctly configure the network configuration. To configure the network, users need to enter # * 107 to enter the [Menu] and find [Advanced Settings] .

The default password for advanced is "123".

NOTICE: If user saw a 'WAN Disconnected' icon flashing in the middle of screen, it means the network cable was not correctly connected to the device's network port. Please check the cable is connected correctly to the device and to the network switch, router, or modem.

The device supports three types of networks, IPv4/IPv6/IPv4&IPv6

There are three common IP configuration modes about IPv4

- Dynamic Host Configuration Protocol (DHCP) – This is the automatic configuration mode by getting network configurations from a DHCP server. Users need not to configure any parameters manually. All configuration parameters will be getting from DHCP server and applied to the device. This is recommended for most users.
- Static IP Configuration – This option allows user to configure each IP parameters manually, including IP Address, Subnet Mask, Default Gateway, and Primary/Secondary DNS servers. This is usually used in an office environment or by power users.
- PPPoE – This option is often used by users who connect the device to a broadband modem or router. To establish a PPPoE connection, user should configure username and password provided by the service provider.
- The device is default configured in DHCP mode.

There are two common IP configuration modes about IPv6

- DHCP - This is the automatic configuration mode by getting network configurations from a DHCP server. Users need not to configure any parameters manually. All configuration parameters will be getting from DHCP server and applied to the device. This is recommended for most users.
- Static IP configuration - this option allows users to manually configure each IP parameter, including IP Address,Subnet Mask,Prefix Length, Default Gateway,Primary/Secondary DNS Server. This usually applies to some Aprofessional network user environments.

Please see 10.6.2.1 network Settings for detailed configuration and use.

7.7 SIP Configurations

A line must be configured properly to be able to provide telephony service. The line configuration is like a virtualized SIM card. Just like a SIM card on a mobile phone, it stores the service provider and the account information used for registration and authentication. When the device is applied with the configuration, it will register the device to the service provider with the server's address and user's authentication as stored in the configurations.

The user can conduct line configuration on the interface of the phone or the webpage, and input the corresponding information at the registered address, registered user name, registered password and SIP user, display name and registered port respectively, which are provided by the SIP server administrator.

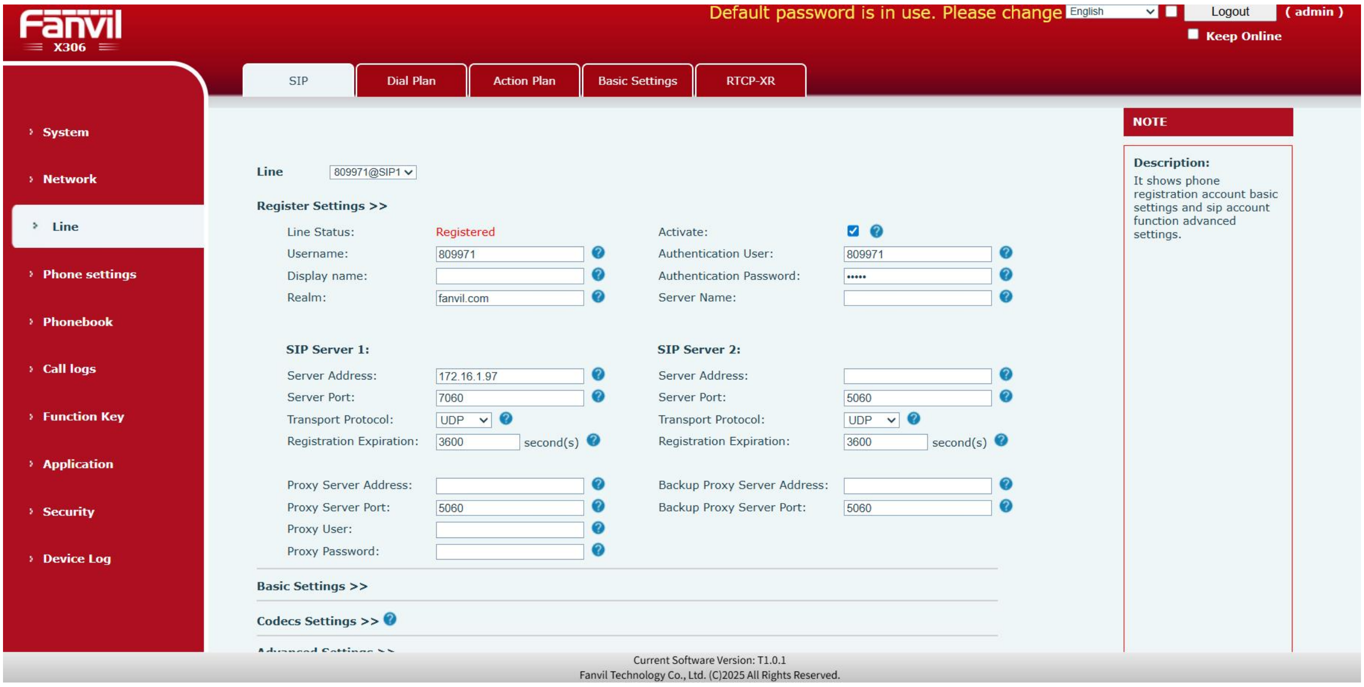
- Phone interface: To manually configure a line, the user can enter the phone menu by entering # * 107, select [Advanced]>>[Account]>>[SIP1]/[SIP2] Configuration, and click Confirm to save the configuration.

NOTICE:User must enter correct PIN code to be able to advanced settings to edit line configuration. (The default PIN is 123)

The parameters and screens are listed in below pictures.

Picture 9 - Phone line SIP address and account information

- WEB interface: After logging into the phone page, enter [**Line**] >> [**SIP**] and select **SIP1/SIP2** for configuration, click apply to complete registration after configuration, as shown below:



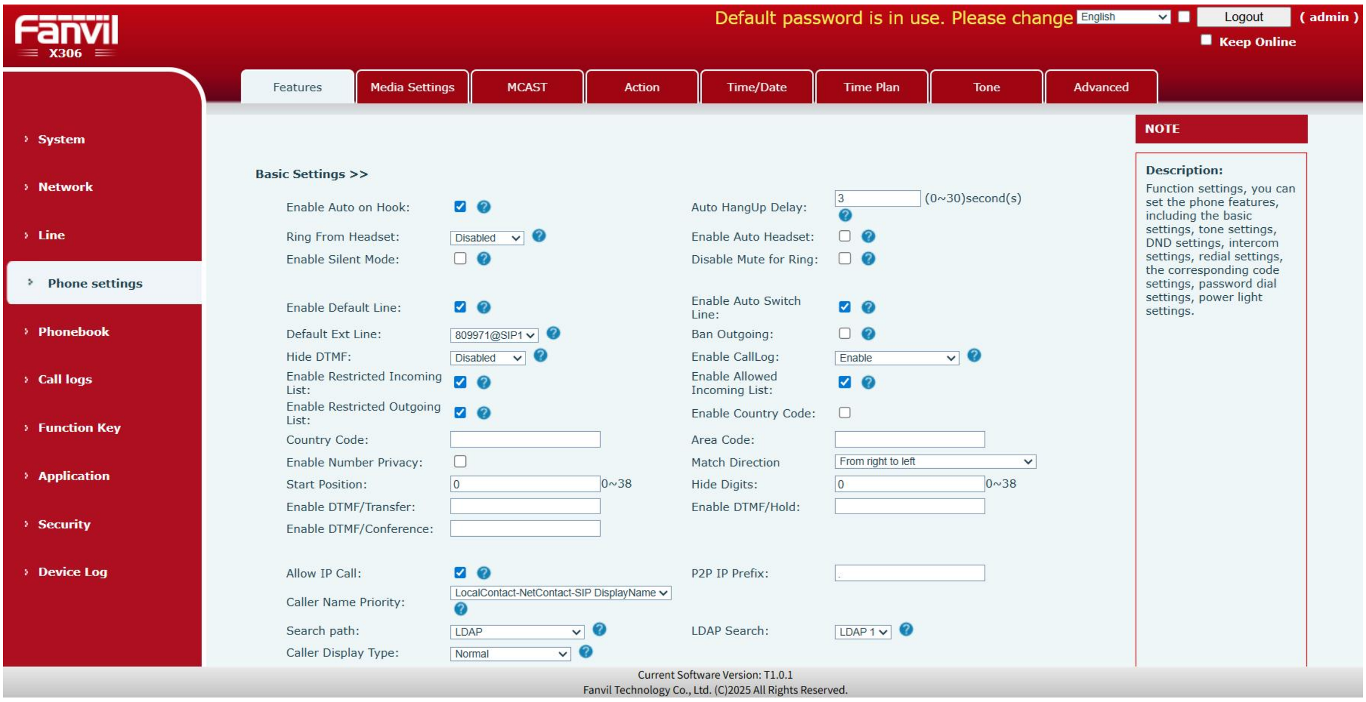
Picture 10 - Web SIP registration

8 Basic Function

8.1 Making Phone Calls

■ **Default Line**

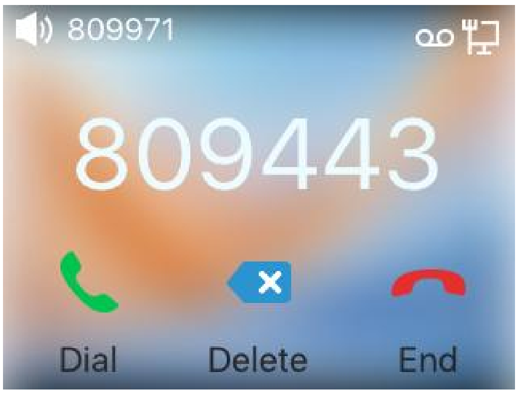
The device provides 2 line services(1 backup line). Enable or disable default line, user can press[**Web**] [**Phone settings**] >> [**Features**] >> [**Basic Settings**].



Picture 11 - Default line

■ **Dialing Number then Opening Audio**

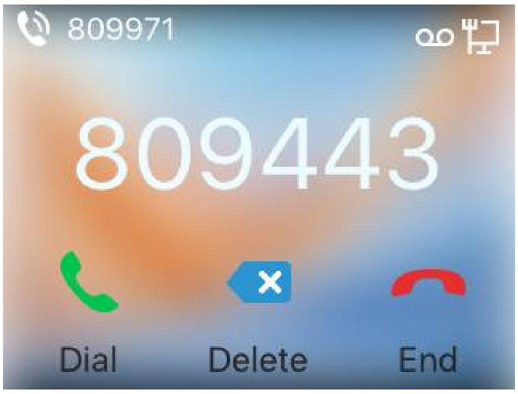
To make a phone call, users can dial numbers by directly pressing the number keys. When the dialed number is completed, user can press the hand-free button to turn on the speaker , or lift the handset to call out.



Picture 12- Enable voice channel dialing

■ **Opening Audio then Dialing the Number**

Another alternative is the traditional way firstly open the audio channel by lifting the handset,or turning on the hands-free speaker by pressing hands-free button, and then dial the number with one of the above methods. When number dialed completed, user can press [**Dial**] button to call out, or the number will be dialed out automatically after timeout.



Picture 13 - Open the voice channel and dial the number

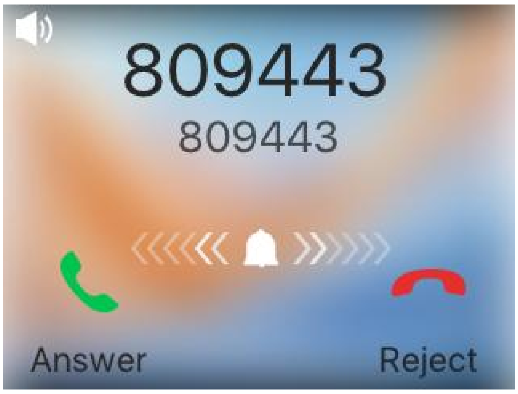
- **Cancel Call**

While calling the number, user can press end the audio channel by putting back the handset or pressing the hands-free button to drop the call.

Picture 14 - Call number

8.2 Answering Calls

When there is an incoming call while the device is idle, user will see the following incoming call alerting screen.



Picture 15 - Answering calls

User can answer the call by lifting the handset, open speaker phone by pressing the hands-free button, or the [Answer] button. To reject the incoming call, user should press [**Reject**] button.

8.2.1 Talking

When the call is connected, user will see a talking mode screen as the following figure.

Picture 16 - Talking interface

Table 5 - Talking mode

Number	Name	Description
①	Call the other end name	The name of the other party on the call.
②	Call the other end number	The number of the other party on the call.
③	Call duration	The duration of a call after it has been established.

8.3 End of the Call

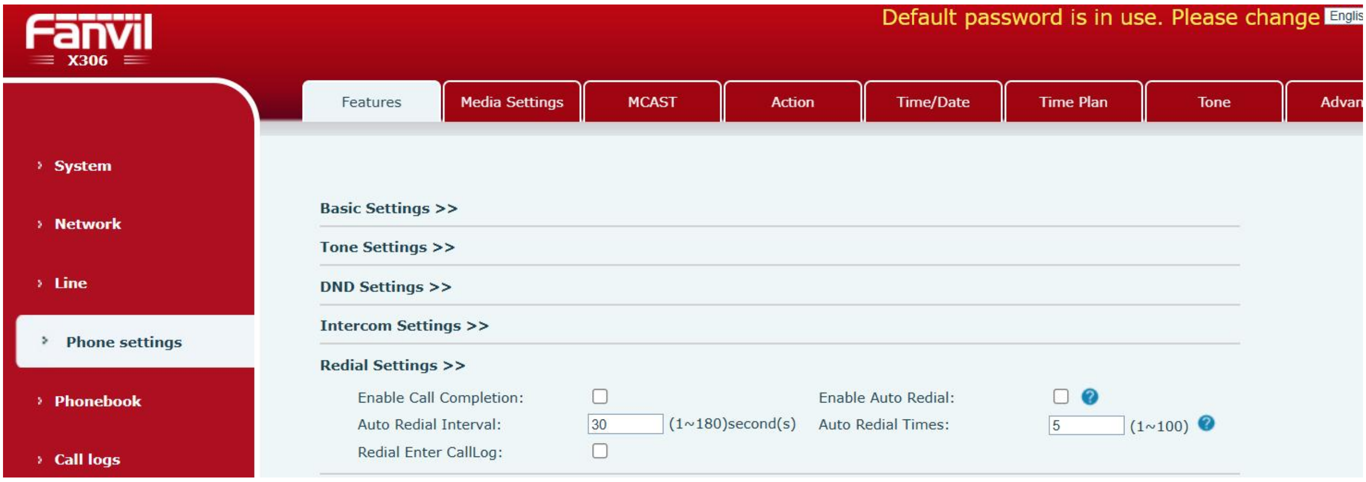
After the user finishes the call, the user can put the handle back on the phone, press the hands-free button or Softkey [**End**] key to close the voice channel and end the call.

Note: When the phone is in the reserved state, the user must press the [Resume] key to return to the call state end the call

8.4 Redial

- Redial the last outgoing number:
  When the phone is in standby mode, press the redial button and the phone will call out the last number dialed.
- Call out any number with the redial key:
  Enter the number, press the redial key, and the phone will call out the number on the dial.

- Press the redial key to enter the call record:
  Log in the phone page, enter [**Phone Settings**] >> [**Features**] >> [**Redial Settings**], check redial to enter the call record, press the redial button when standby to enter the call record page, and press again to call out the currently located number.



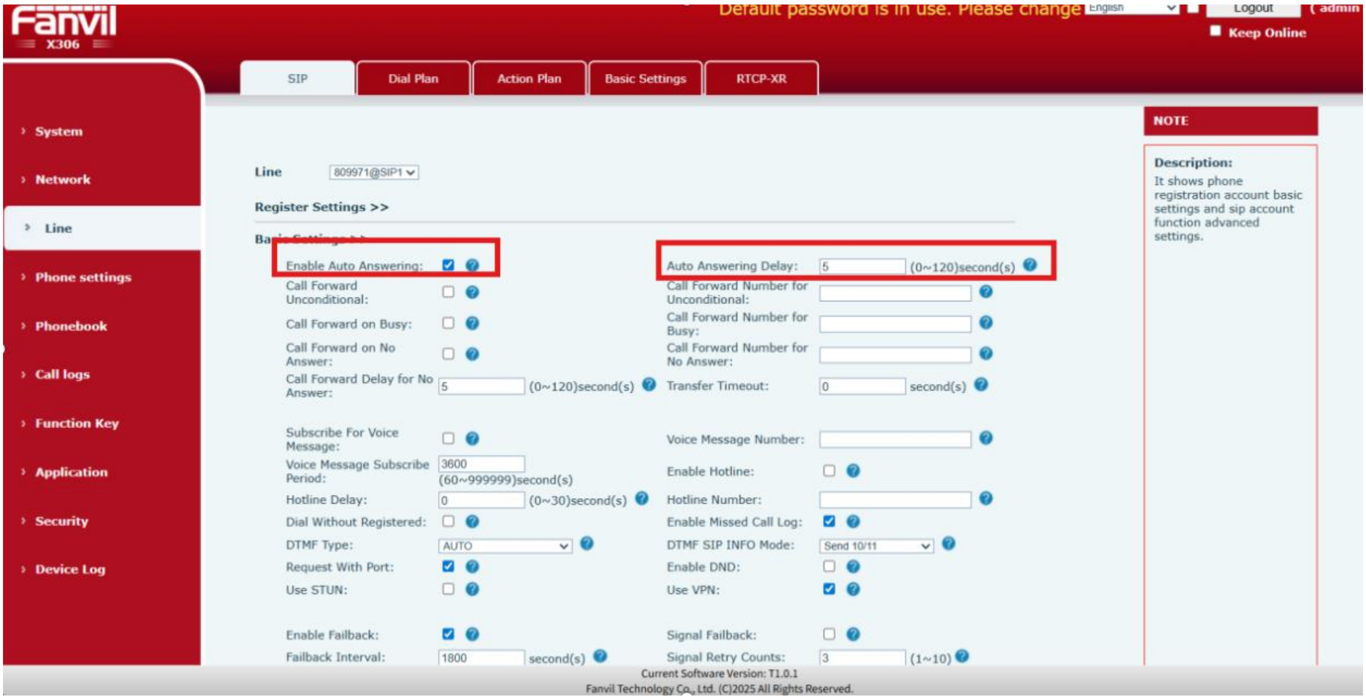
Picture 17 - Redial set

8.5 Auto-Answering

User may enable auto-answering feature on the device and any incoming call will be automatically answered (not including call waiting). The auto-answering can be enabled on line basis.

The user can start the automatic answer function in the webpage interface.

- **WEB interface:**
  Log in the phone page, enter [**Line**] >> [**SIP**], select [**SIP**] >> [**Basic settings**], start auto-answering, and click apply after setting the automatic answering time.



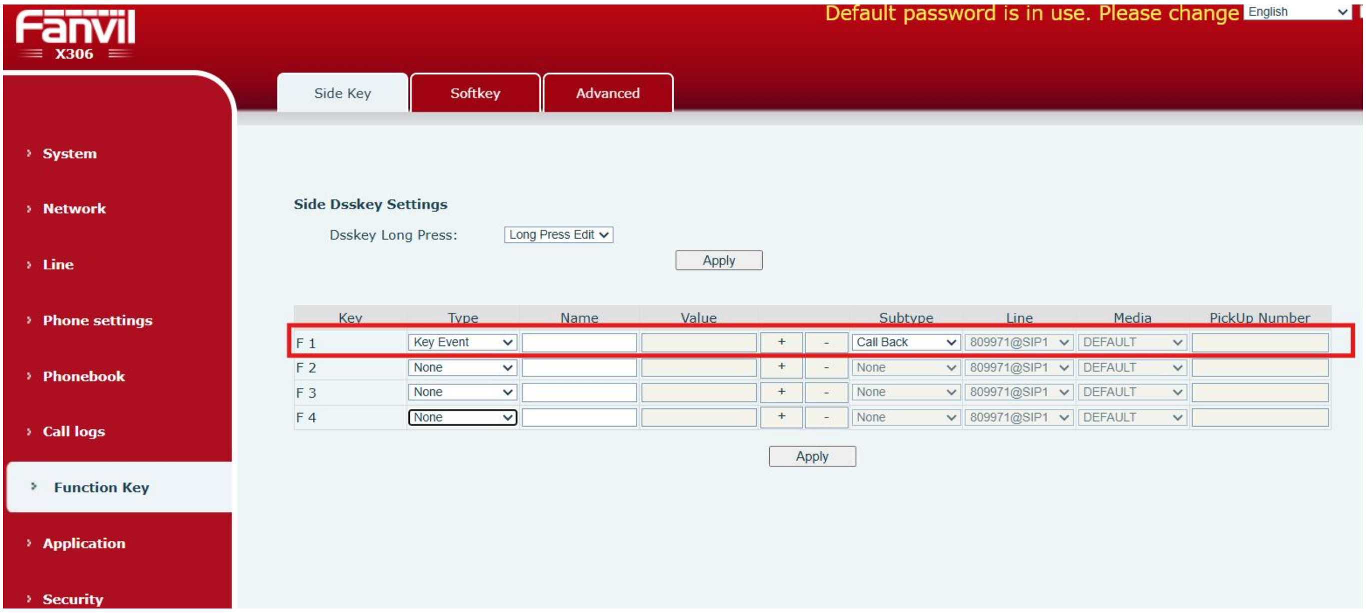
Picture 18 - Web page to start auto-answering

8.6 Callback

The user can dial back the number of the last call. If there is no call history, press the [**Callback**] button and the phone will say "can't process".

● Set the callback key through the web interface:

Log in the phone page, enter the [**Function Key**] >> [**Side Key**] page, select the function Key, set the type as the function Key, and set the subtype as the callback, as shown in the figure:



Picture 19 - Set the callback key on the web page

8.7 Mute

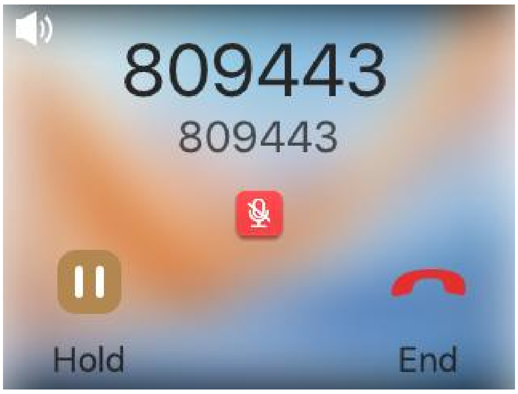
You can configure the softkey (select voice call on the display page, turn on microphone mute) to enable mute mode during the call, turn off the microphone of the phone, and prevent the other party from hearing local sound. Under normal circumstances, the silent mode automatically turns off as the call ends. You can also turn on the mute function on any interface (such as idle interface) to automatically mute the ringtone when a call comes in.

Silent mode can be turned on in all call modes (controller, headphones, or hands-free).

8.7.1 Mute the Call

- During the conversation, press the mute button: 

  Red mute icon is displayed in the call interface, as shown in the figure:



Picture 20 - Mute the call

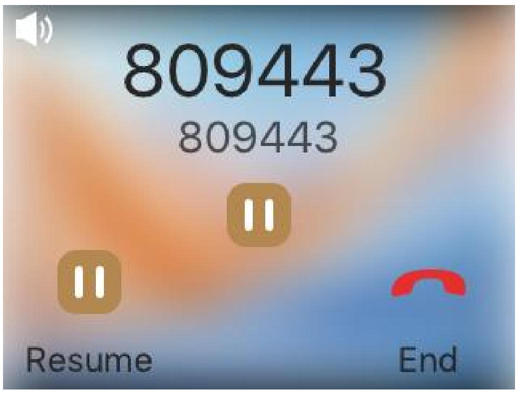
- Cancel mute: press cancel mute on the phone again. The mute icon is no longer displayed in the call screen.

8.7.2 Ringing Mute

- Silent ringtone for incoming calls: When the phone is in standby mode, press the mute button or use the volume down button on the phone to adjust the volume to 0. The ringtone mute icon will be displayed in the upper right corner of the phone. When there is an incoming call, the phone will display the incoming call interface but will not ring.
- Cancel ringtone mute: Press the mute button or call interface again, use the volume up button to mute the ringtone, and the mute icon will no longer be displayed in the upper right corner after cancellation.

8.8 Call Hold/Resume

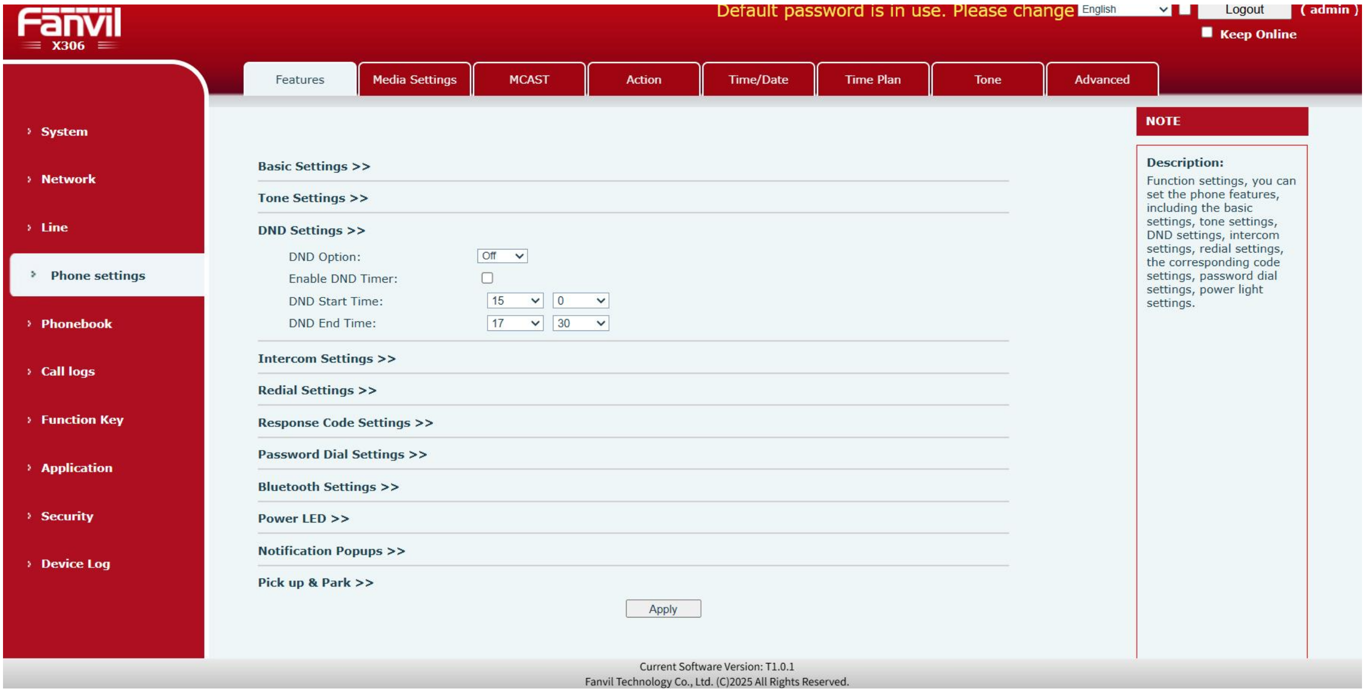
The user can press the [**Hold**] button to maintain the current call, and this button will become the [**Resume**] button, and the user can press the "resume" button to restore the call.



Picture 21 - Call hold interface

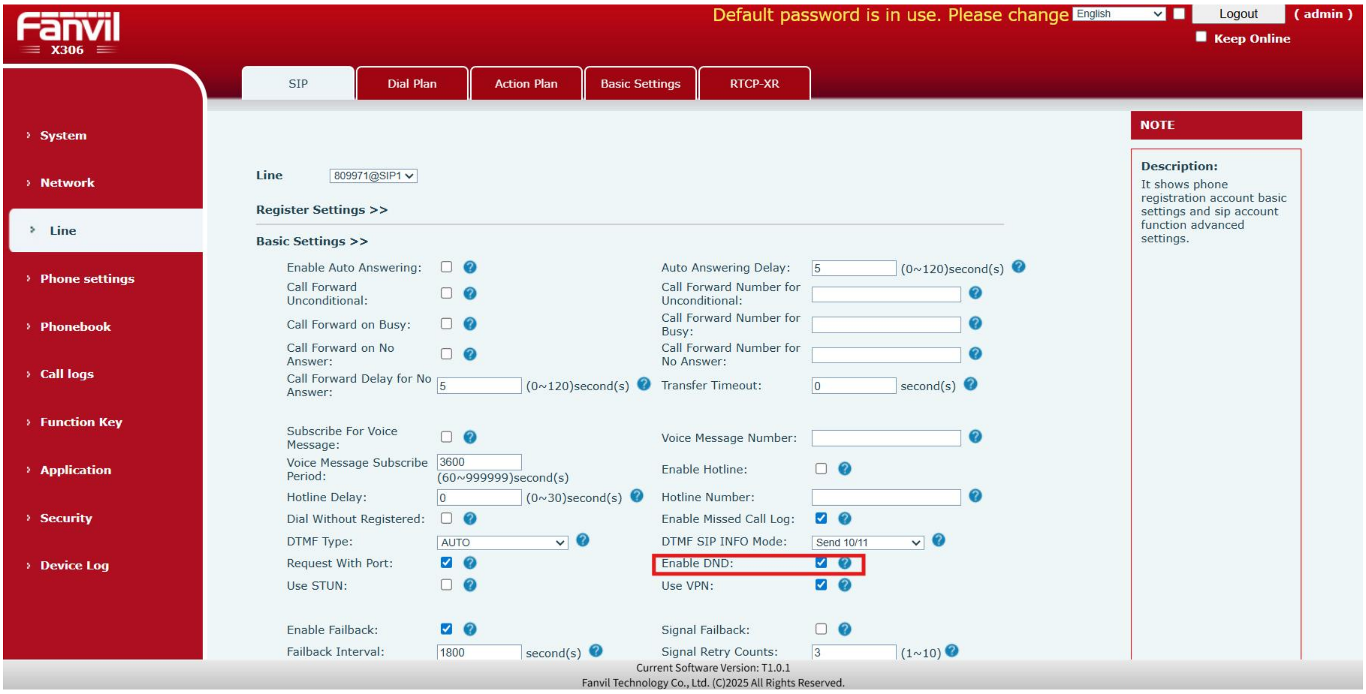
8.9 DND

- WEB interface: Enter [**Phone setting**] >> [**Features**] >> [**DND settings**]，set the DND type (off, phone, line), and DND timing function.



Picture 22 - DND Settings

The user turns on the DND for a specific route on the web page：Enter [**Line**] >> [**SIP**], select a [**Line**] >> [**Basic settings**], and enable DND.



Picture 23 - Line DND

8.10 Call Forward

Call forward is also known as ‘Call Divert’ which is to divert the incoming call to a specific number based on the conditions and configurations. User can configure the call forward settings of each line.

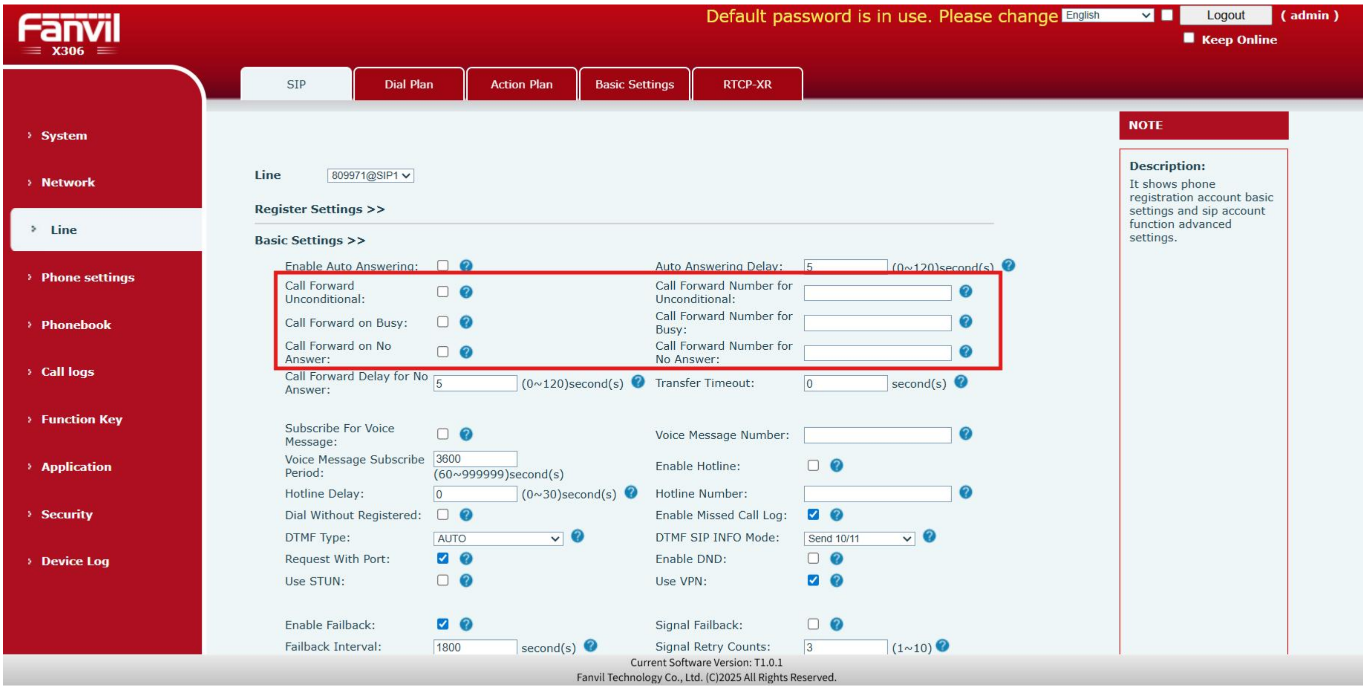
There are three types

① **Call Forward Unconditional –** Forward any incoming call to the configured number.

② **Call Forward on Busy –** When user is busy, the incoming call will be forwarded to the configured number.

③ **Call Forward on No Answer –** When user does not answer the incoming call after the configured delay time, the incoming call will be forwarded to the configured number.

- WEB interface: Enter [**Line**] >> [**SIP**], Select a [**Line**] >> [**Basic settings**], and set the type, number and time of forward forwarding.

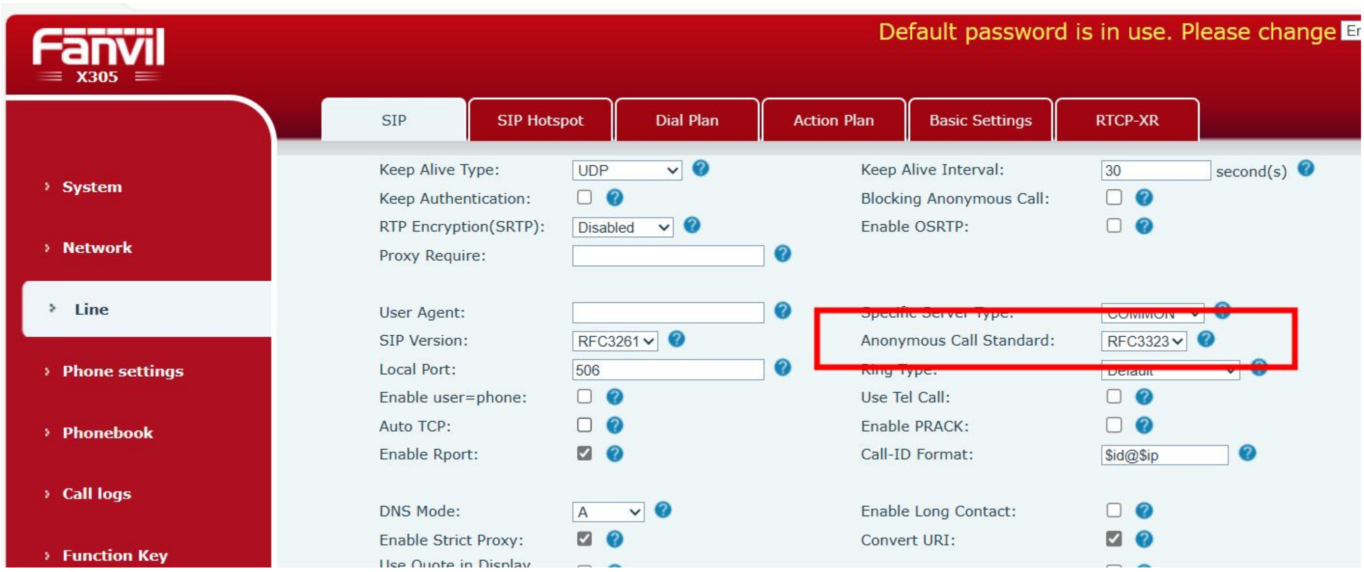


Picture 24 - Set call forward

8.11 Anonymous Call

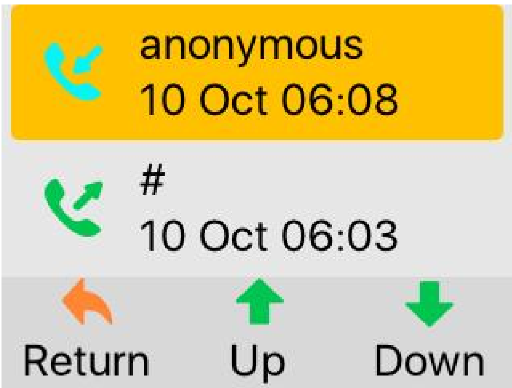
8.11.1 Anonymous Call

- On the web page [**Line**] >> [**SIP**] >> [**Advanced Settings**] can also open anonymous calls.
- Setting to enable anonymous calls also corresponds to the SIP line. That is, the setting under the SIP1 page can only take effect on the SIP1 line.



Picture 25 - Enable Anonymous web page call

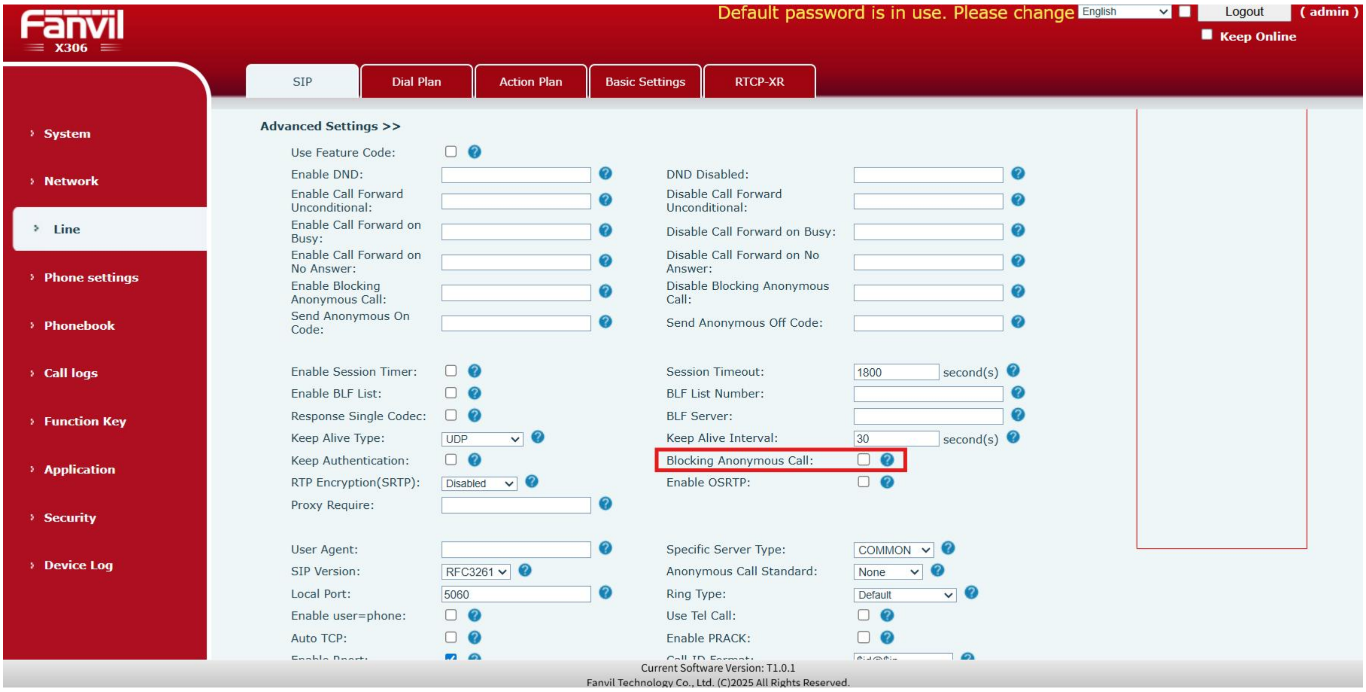
The following is a transcript of an anonymous call received by the phone.



Picture 26 - Anonymous call log

8.11.2 Ban Anonymous Call

- On the web page [**Line**] >> [**SIP**] >> [**Advanced Settings**], also can disable anonymous calls.
- The setup to disable anonymous calls also corresponds to the SIP line. That is, the setting under the SIP1 page can only take effect on the SIP1 line.



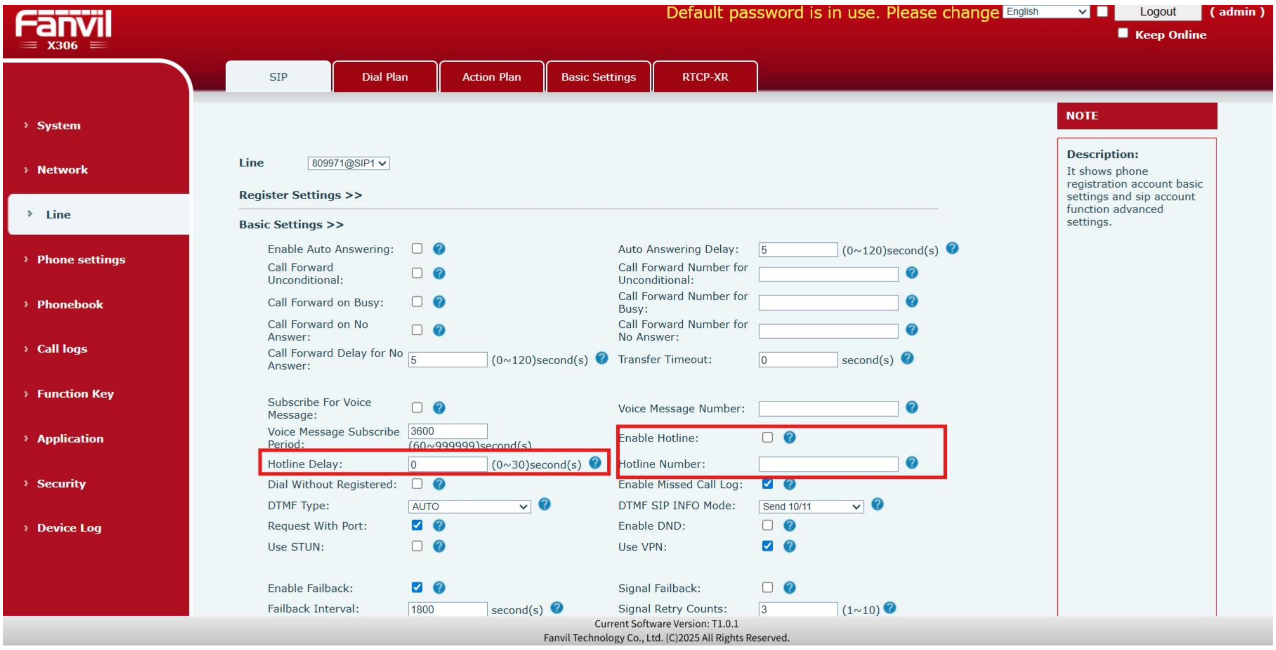
Picture 27 - Page Settings blocking anonymous call

8.12 Hotline

Support hotline dialing. After setting up hotline dialing, you can directly pick up the receiver, hands-free, etc., and the phone will automatically call according to the hotline delay time

● On the website [**Line**] >> [**SIP**] >> [**Basic Settings**], can also set up a hotline.

● The setup hotline also corresponds to the SIP line. That is, the hotline set in the SIP1 webpage can only be activated in the SIP1 line.



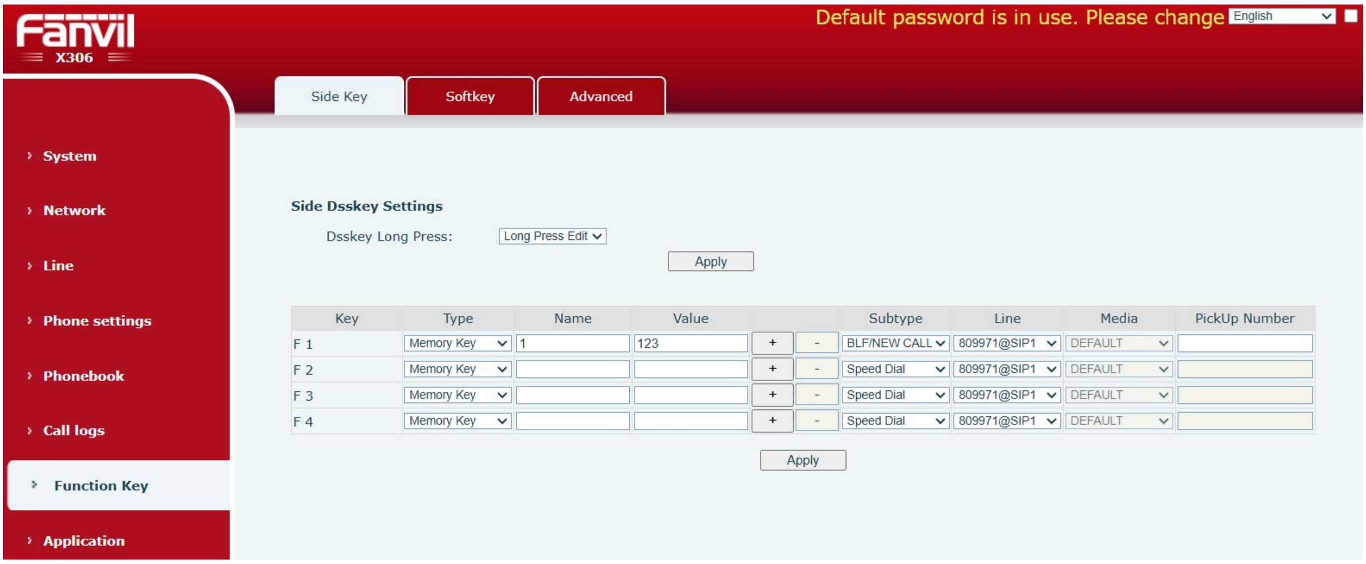
Picture 28 - Hotline set up on webpage

9 Advance Function

9.1 BLF (Busy Lamp Field)

9.1.1 Configure the BLF Functionality

- Page interface: log in the phone page, enter the [**Function key**] >> [**Side Key**] page, select a DSS key, set the function key type as memory key, choose subtype among BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF, set BLF/DTMF value as the number to be subscribed, set the corresponding SIP line.



Picture 29 - Web page configuration BLF function key

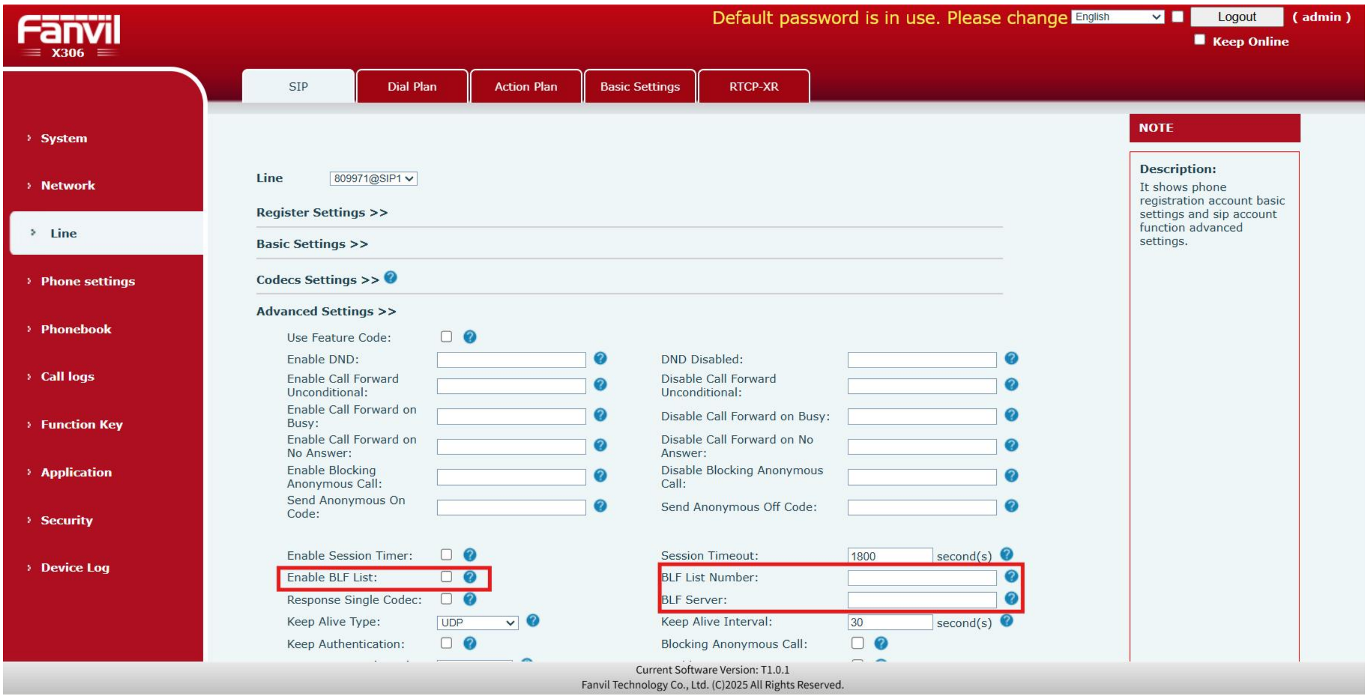
Table 6 - BLF Function key subtype parameter list

Subtype	**Standby is described**	**Calling is described**
BLF/NEW CALL	Pressing the BLF key while standby to dial the subscriber number.	When you press this BLF key while talking to another user, you create a new call along with the subscribed number.
BLF/DTMF	Pressing the BLF key while standby to dial the subscriber number.	When the BLF key is pressed while talking to another user, the phone automatically sends the DTMF corresponding to the BLF key number.

9.2 BLF List

BLF List Key is to put the number to be subscribed into a group on the server side, and the phone uses the URL of this group to make unified subscription. The specific information, number, name and status of each number can be resolved based on notify sent from the server. The unoccupied Memory Key is then set to the BLF List Key.

Configure BLF List function: log in the phone page, enter the [**Line**] >> [**SIP**] >> [**Advanced settings**] page, open the BLF List, and configure the BLF List number.



Picture 30 - Configure the BLF List functionality

Use the BLF List function: when the configuration is complete, the phone will automatically subscribe to the contents of the BLF List group. Users can monitor and call the corresponding number by pressing the BLF List key.

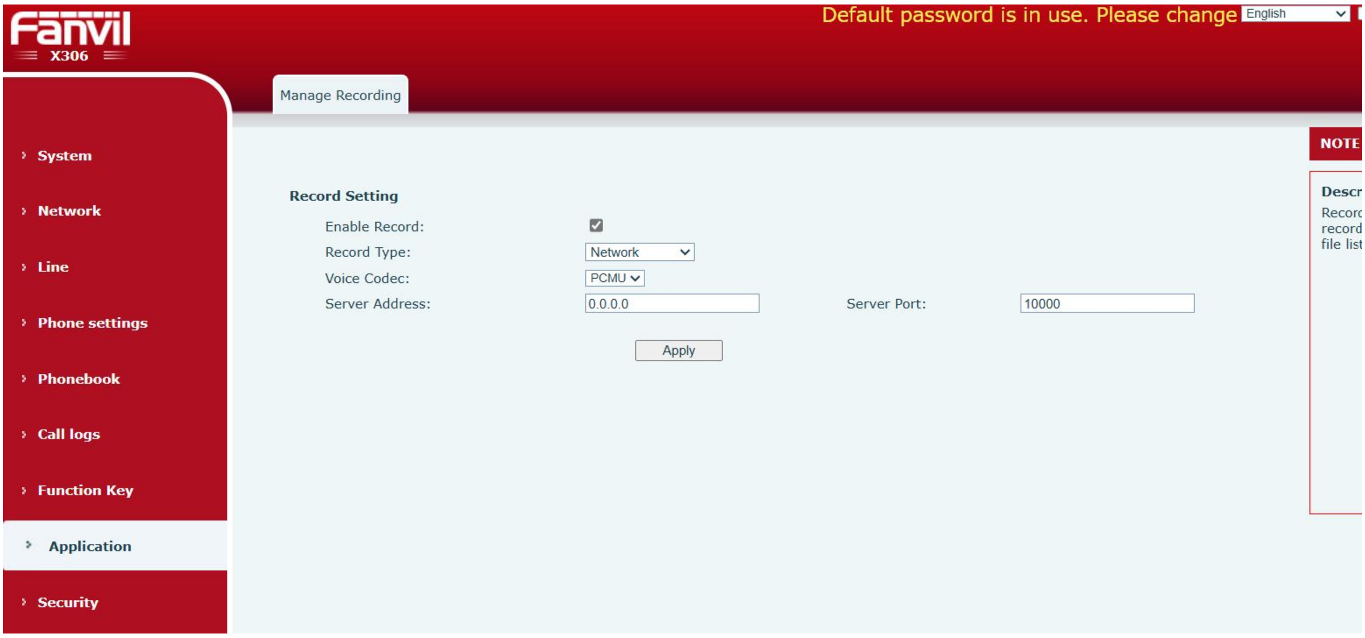
Picture 31 - BLF List number display

9.3 Record

The device supports recording during a call.

9.3.1 Online Record

When using the network server to record, it is necessary to open the recording in the phone web page [**Application**] >> [**Manage recording**]. The type is selected as network, and the address and port of the recording server are filled in and the voice coding is selected. The web is as follows:

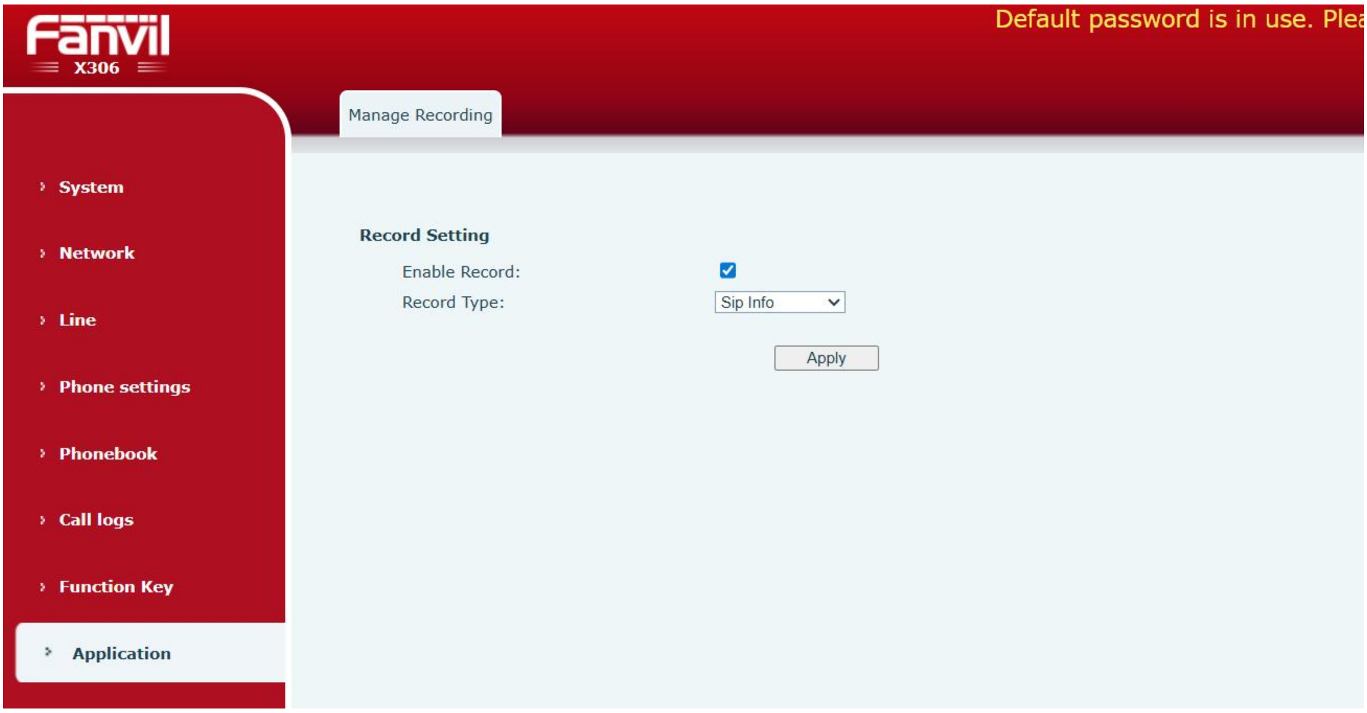


Picture 32 - Web server recording

Note: to be used with Fanvil recording software.

9.3.2 SIP INFO Record

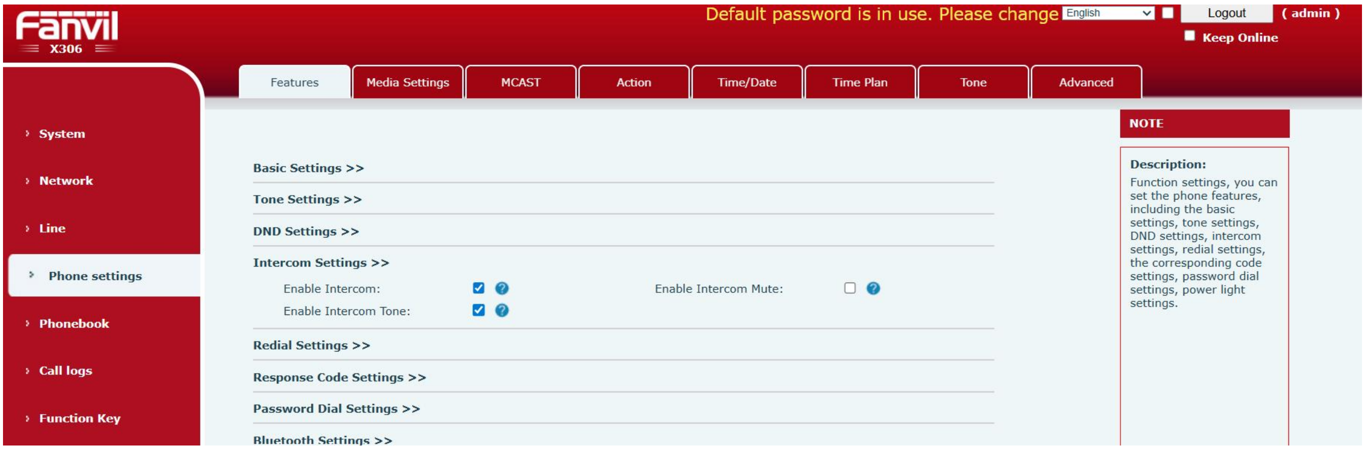
The phone is registered with a server that supports SIP INFO recording. After registering the account, check the recording module of [**Application**] >> [**Manage recording**] to open the recording, and the recording type is SIP INFO.



Picture 33 - Web SIP info recording

9.4 Intercom

When the Intercom is enabled, it can automatically receive calls from the intercom.



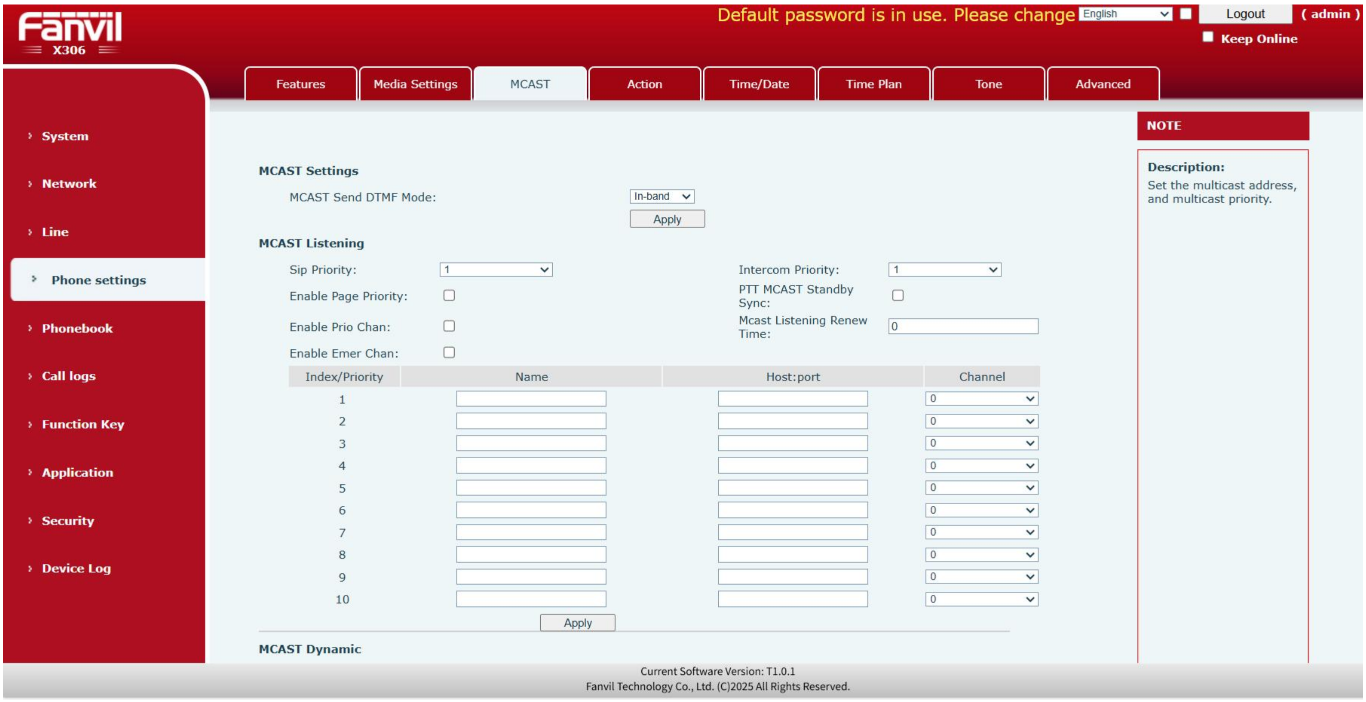
Picture 34 - Web Intercom configure

Table 7 - Intercom configure

Parameter	**Description**
Enable Intercom	When intercom is enabled, the device will accept the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after specific delay.
Enable Intercom Mute	Enable mute mode during the intercom call
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone

9.5 MCAST

This feature allows user to make some kind of broadcast call to people who are in multicast group. User can configure a multicast DSS Key on the phone, which allows user to send a Real Time Transport Protocol (RTP) stream to the pre-configured multicast address without involving SIP signaling. You can also configure the phone to receive an RTP stream from pre-configured multicast listening address without involving SIP signaling. You can specify up to 10 multicast listening addresses.



Picture 35 - Multicast Settings Page

Table 8 - MCAST Parameters on Web

Parameters	**Description**
Sip Priority	Define the priority of the active call, 1 is the highest priority, 10 is the lowest.
Enable Page Priority	The voice call in progress shall take precedence over all incoming paging calls.
Name	Listened multicast server name
Host:port	Listened multicast server's multicast IP address and port.

Multicast:

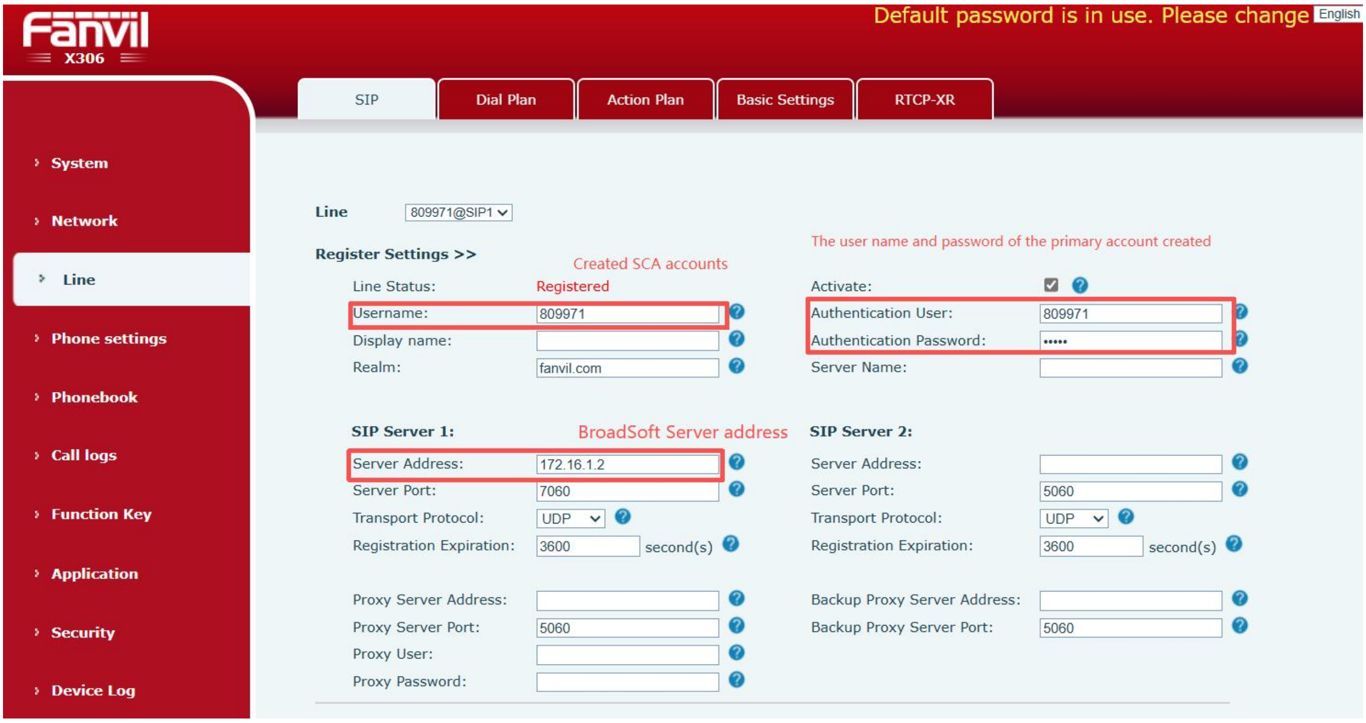
- Go to web page of [**Function Key**] >> [**Side Key**] , select the type to multicast, set the multicast address, and select the codec.
- Click Apply.
- Set up the name, host and port of the receiving multicast on the web page of [**Phone Settings**] >> [**MCAST**].
- Press the DSSKY of Multicast Key which you set.
- Receive end will receive multicast call and play multicast automatically.

9.6 SCA (Shared Call Appearance)

To use an SCA account, server-side support is required. Please refer to the document Broadsoft's SCA server and terminal configuration instructions for specific details.

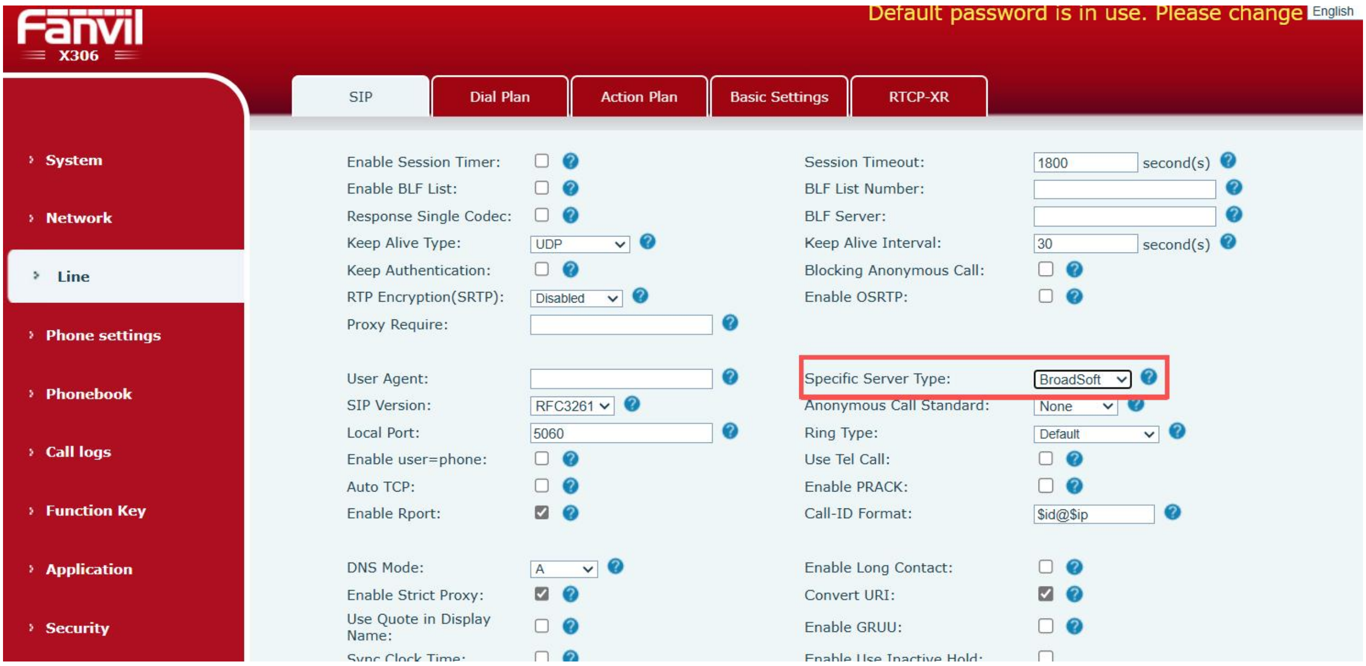
1） Configure on Phone

- When registering with the BroadSoft server, a Fanvil Phone can register the account created previously on multiple terminals.



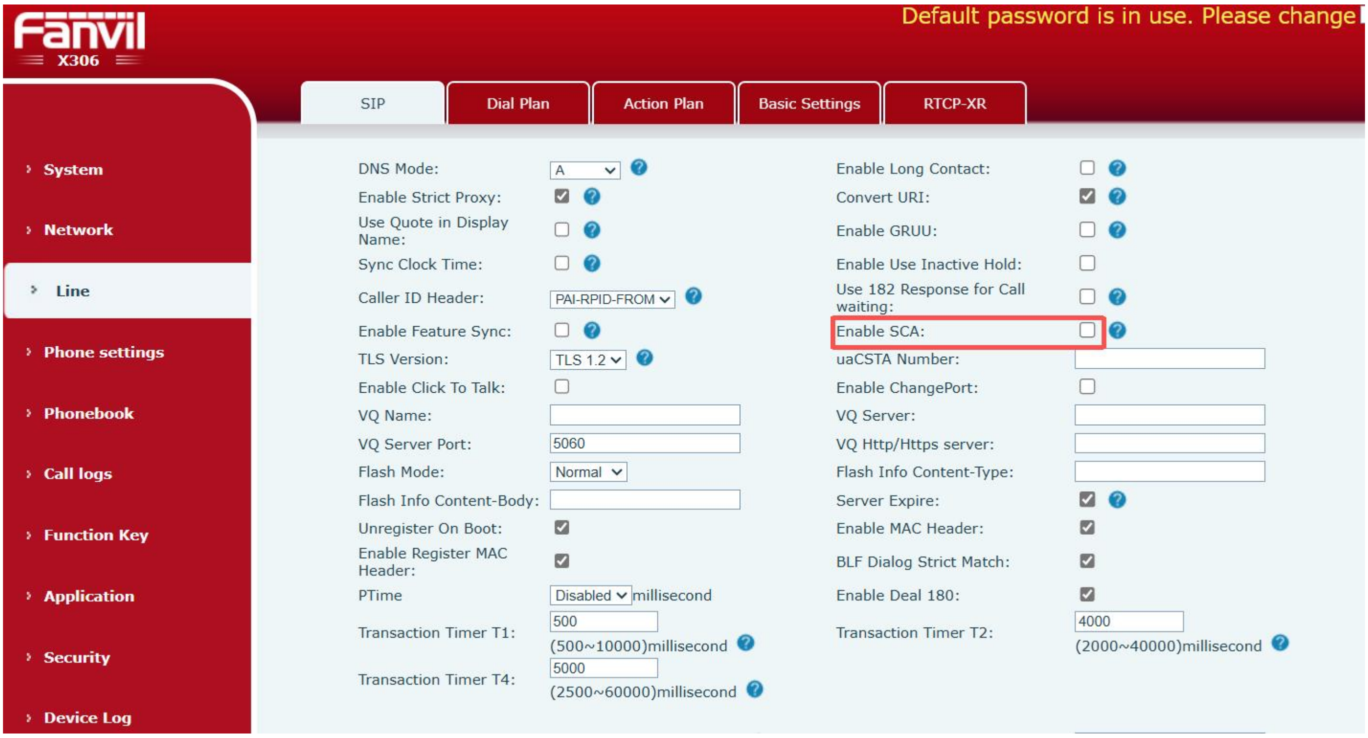
Picture 36 - Register BroadSoft account

- After the phone set registers with the BroadSoft server, a server type needs to be set. Specifically, log in to the webpage of the phone set, choose [**Line**] >> [**SIP**] >> [**Advanced Settings**] and set Specific Server Type to BroadSoft, as shown in the following figure.



Picture 37 - Set BroadSoft server

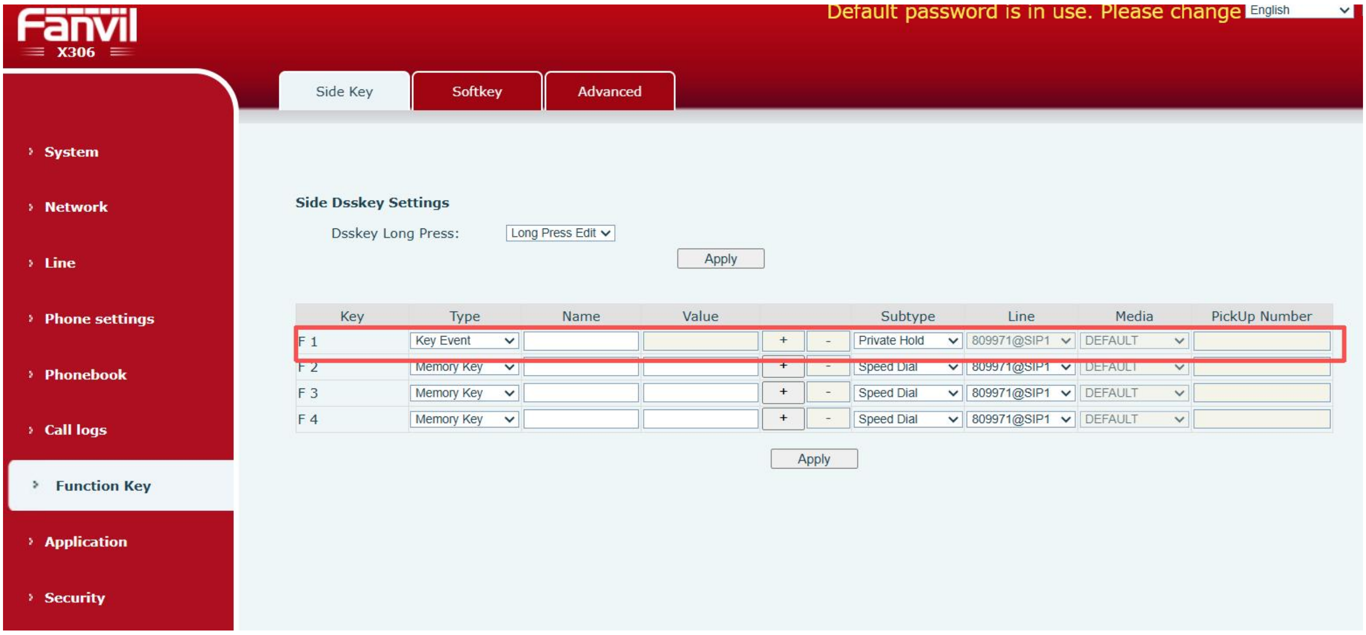
- If a Fanvil phone set needs to use the SCA function, enable it for the phone set. Specifically, log in to the webpage of the phone set, choose [**Line**] >> [**SIP**] >> [**Advanced Settings**], and select Enable SCA. If SCA is not enabled, the registered line is private line.



Picture 38 - Enable SCA

After an account is configured and successfully registered, you can configure lines whose DSS Key is Shared Call Appearance on the Function Key page to facilitate viewing the call status of the group. Each line key represents a call appearance. Understand the call status by referring to 6.3 Appendix III –LED .

To facilitate private hold, configure keys whose DSS Key is Private Hold on the Function Key page. Pay attention that the public hold key is the softkey-hold key during a call.



Picture 39 - Set Private Hold Function Key

After each phone set registered with the BroadSoft server is configured as above, the SCA function can be used.

3） Shared Call Appearance(SCA)

The following lists a couple of instances to facilitate understanding.

In the following scenarios, the manager and secretary register the same SCA account and the account is configured based on the preceding steps.

Scenario 1: When this account receives an incoming call, the phone sets of both the manager and the secretary will receive the call and ring. If the manager is busy, the manager can reject the call and the manager's phone set stops ringing but the secretary's phone set keeps ringing until the secretary rejects/answers the call or the call times out.

Scenario 2: When this account receives an incoming call, if the secretary answers the call first and the manager is required to answer the call, the secretary can press the Public Hold key to hold this call and notify the manager. The manager can press the line key corresponding to the SCA to answer the call.

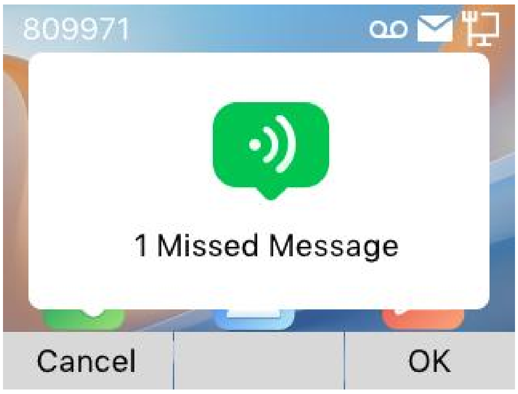
Scenario 3: The manager is in an important call with a customer and needs to leave for a while. If the manager does not want others to retrieve this call, the manager can press the Private Hold key.

Scenario 4: The manager is in a call with a customer and requires the secretary to join the call to make records. The secretary can press the corresponding SCA line key to barge in this call.

9.7 Message

9.7.1 SMS

If the service of the line supports the function of the short message, when the other end sends a text message to the number, the user will receive the notification of the short message.



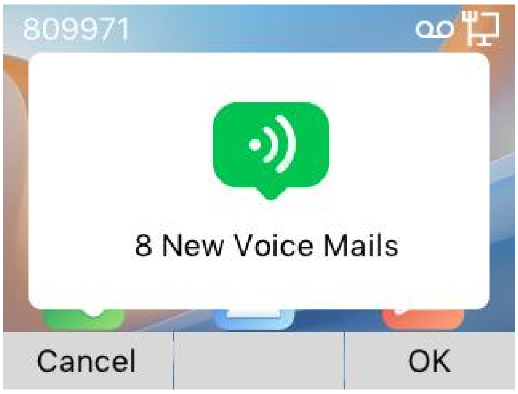
Picture 40 - SMS icon

View SMS:

- Enter the “#*107” to enter the [**Menu**] , and select [message]
- After selecting, press the [**OK**] to enter the SMS inbox interface.
- Select the unread message and press [**OK**] to read the unread message.

9.7.2 MWI (Message Waiting Indicator)

If the service of the lines supports voice message feature, when the user is not available to answer the call, the caller can leave a voice message on the server to the user. User will receive voice message notification from the server and device will prompt a voice message waiting icon on the standby screen.



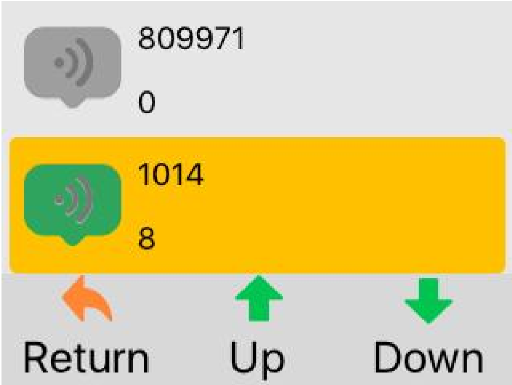
Picture 41 - New Voice Message Notification

Voice message icon

To listen to voice messages, users must first configure a voice mailbox number. After configuring a voice mailbox number, users can retrieve voice messages for the default line.

When the phone is in default standby mode,

- The sub screen is preset with a voice message shortcut key - the 'Voice Mail' button
- The "1" before the parentheses of SIP1 (1/1) line represents unread voice messages, and "1" represents the total number of voice messages.



Picture 42 - Voice message interface

10 Phone Settings

10.1 Basic Settings

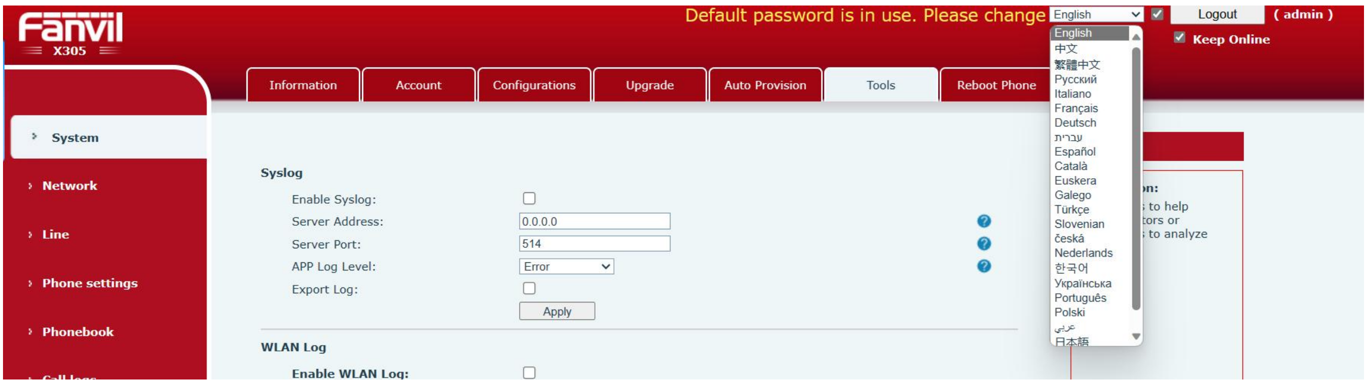
10.1.1 Language

The user can set the phone language through the phone interface and web interface.

- Phone end: After resetting the factory settings, the user needs to set the language; when setting the language during standby, Enter the “#*107” to enter the [**Menu**] >> [**Basic**] >> [**Language**] Settings, as shown in the figure.

Picture 43 - Phone language setting

- Web interface: Log in to the phone webpage and set the language in the drop-down box at the top right corner of the page, as shown in the figure:



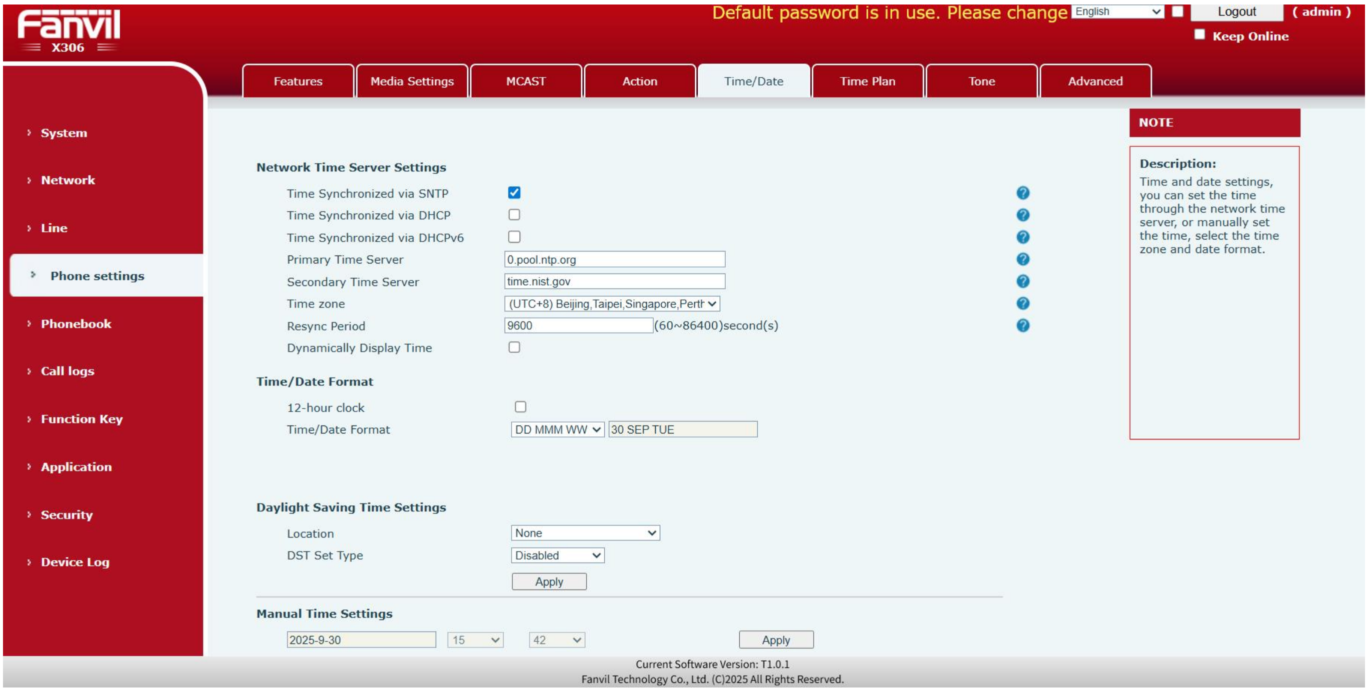
Picture 44 - Language setting on Web page

- The function box on the right side of the web interface language setting box is “Synchronize language to phone”; if selected, the phone language will be synchronized with the webpage language. If it is not selected, it will not be synchronized.

10.1.2 Time & Date

Users can set the phone time through the web interface.

- Web end: Log in to the phone webpage and enter [**Phone Settings**] >> [**Time/Date**] , as shown in the figure:



Picture 45 - Set time & date on webpage

Table 9 - Time Settings Parameters

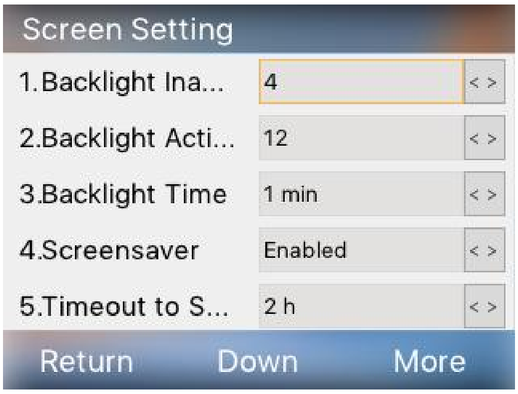
Parameters	**Description**
Mode	Auto/Manual Auto: Enable network time synchronization via SNTP protocol, default enabled. Manual: User can modify data manually.
SNTP Server	SNTP server address
Time zone	Select the time zone
Time/Date Format	Select time format from one of the followings: ■ DD-MM-YY ■ DD-MM-YYYY ■ MM-DD-YY ■ MM-DD-YYYY ■ YY-MM-DD ■ YYYY-MM-DD ■ DD-MMM-WWW ■ MMM-DD-WW ■ WW-DD-MMM ■ WW-MMM-DD
Separator	Choose the separator between year and moth and day
12-Hour Clock	Display the clock in 12-hour format
Daylight Saving Time	Enable or Disable the Daylight Saving Time

Settings	

10.1.3 Screen

The user can set the phone screen parameters through both of the phone interface and web interface.

- Phone: When the phone is in the default standby state, Enter the “#*107” to enter the[**Menu**] >> [**Settings**] >> [**Screen**] to edit the screen parameters. After editing, click [**OK**] to save, as shown in the figure:

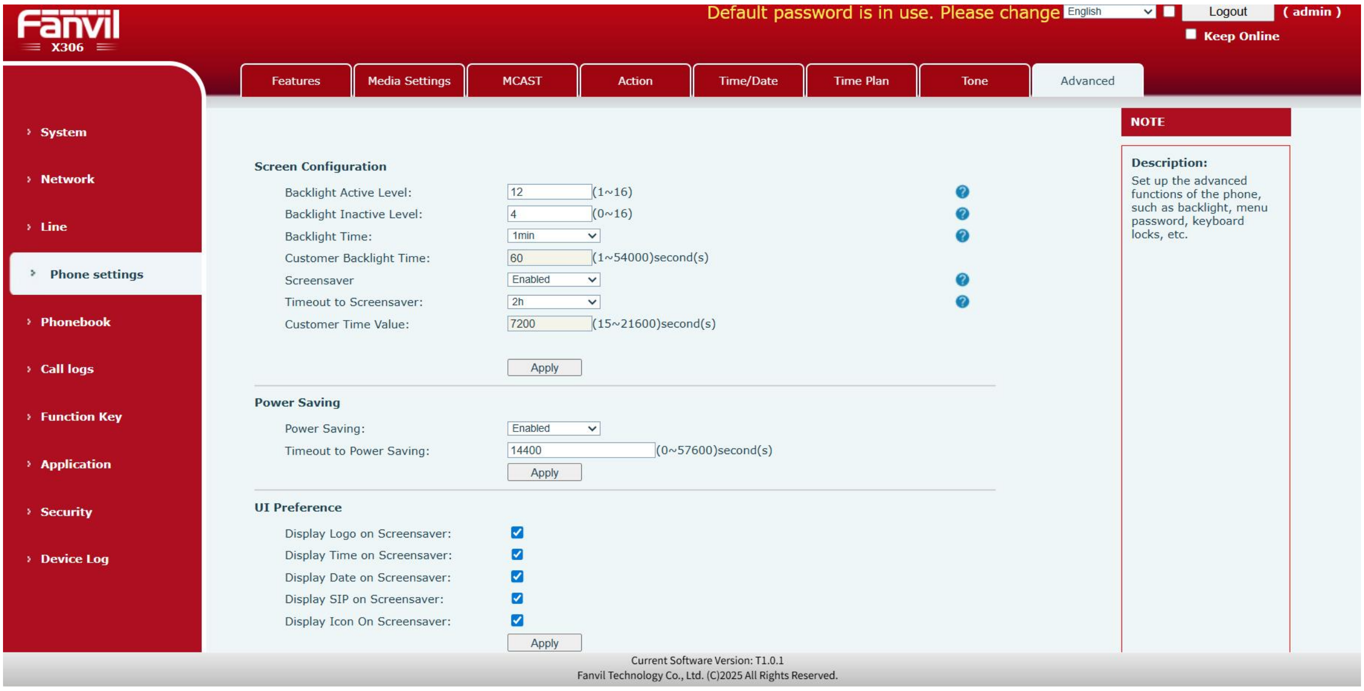


Picture 46 - Set screen parameters on phone

- Web : Go to [**Phone Settings**] >> [**Advanced**] Advanced, edit the screen parameters, and click Apply to save.

10.1.3.1 Brightness and backlight

- Set the brightness level in use from 1 to 16, [More]>>[Switch] switch brightness level.
- Set the brightness level in the energy-saving mode from 0 to 16, [More]>>[Switch] switch the brightness level.
- Set the backlight time to 1 min by default. You can turn it off or select 15 S/30 S/1 min/2 min/5 min/10 min/30 min/ 1 h /2h 3h /6h /15h。
- Web interface: enter [**Phone Settings**] >> [**Advanced**], edit screen parameters, and click submit to save.



Picture 47 - Page screen Settings

10.1.3.2 Screen Saver

- Press [**Screen Settings**] to find the [**Screen protection**] button, press the volume -(L)/+(R)buttons to open/close the screen protection, set the timeout time, the default is 2H, after completion, press [**OK**] key to save.
- After saving, return to standby mode and enter the screen saver after 2H, as follows:

Picture 48 - Phone screen saver

10.1.4 Ring

When the device is in the default standby mode,

- Enter * # 107 to enter the [Menu] and find the **[Settings]** button.
- Press the **[Settings]** button to find the [**Ring**] button.
- Press the [**Ring**] button to find the **[Hands Free Ring]** button, press **[More]**>>**[Switch]**, and then press **[OK]** to save.
- Press the [**Ring**] button to find the **[Ring Type]** button, press **[More]**>>**[Switch]** to edit the ringtone type,

and then press **[OK]** to save it.

10.1.5 Voice Volume

When the device is in the default standby mode,

- Enter * # 107 to enter the [**Menu**] and find the [**Settings**] button
- Press the 'Settings' button to find the' Voice Volume 'button
- Press the 'Voice Volume' button to find the 'Handle' and 'Hands Free' buttons.
- Press the **[Handset]** or [**Handsfree**] option, and use [**More**]>>[**Switch**] to edit the audio volume
- After completion, press [**OK**] to save.

10.1.6 Reboot

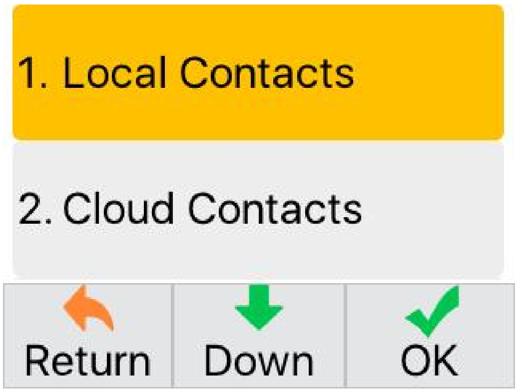
When the device is in the default standby mode,

- Enter * # 107 to enter the [**Menu**] and find the [**Settings**] button.
- Press the **[Settings]** button to find the **[Reboot System]** button
- Press [**OK**], and there will be a prompt message **[Reboot Now]** to prompt the user.
- After completion, press [**OK**] to restart the phone or press [**Cancel**] to exit.

10.2 Phone Book

10.2.1 Local Contact

User can save contacts' information in the phone book and dial the contact's phone number(s) from the phone book((By pressing the hands-free button or lifting the handle). To open the phone book, user can press soft-menu button [**Contact**] in the default standby screen or keypad.



Picture 49 - Phone book screen

NOTICE! The device can save up to total 1000 contact records.

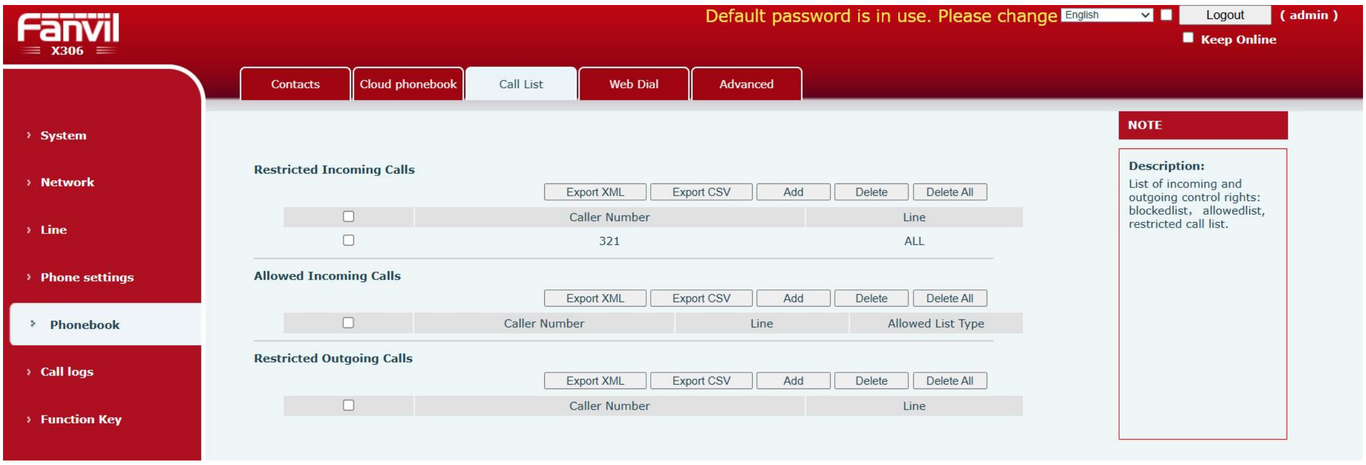
Picture 50 - Local Phone book

When there are contact records in the phone book, the contact records will be arranged in the alphabet order. User may browse the contacts with up/down keys. The record indicator tells user which contact is currently focused.

10.2.2 Blocked List

The device supports blocking the incoming call list. If a number is added to the blocking call list, the incoming call from that number will be directly rejected by the other end, and the local phone will display a missed call. (Numbers in the blocked call list can be called normally)

- Select any number in the phone book (both local and network) for configuration addition.
- Select any number in the call log for configuration addition.



Picture 51 - Web Blocked List

10.2.3 Cloud Phone Book

10.2.3.1 Configure Cloud Phone book

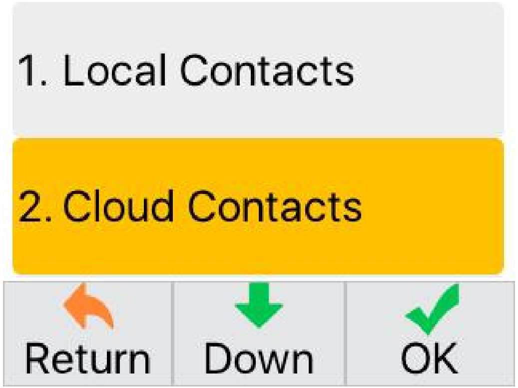
Cloud phonebook allows user to configure the device by downloading a phonebook from a cloud server. This is convenientfor office users to use the phonebook from a single source and save the effort to create and maintain the contact list individually. It is also a useful tool to synchronize his/her phonebook from a personal mobile phone to the device with Fanvil Cloud Phonebook Service and App which is to be provided publicly soon.

NOTICE! The cloud phonebook is ONLY temporarily downloaded to the device each time when it is

opened on the device to ensure the user get the latest phonebook. However, the downloading may take a couple seconds depending on the network condition. Therefore, it is highly recommended for the users to save important contacts from cloud to local phonebook for saving download time.

Open cloud phonebook list, press [**Menu**] >> [**PhoneBook**] >> [**Cloud Contacts**] in phonebook screen.

TIPS! The first configuration on cloud phone should be completed on Web page by selecting [PhoneBook] >> [Cloud Contacts]. The setting of addition/deletion on device could be done after the first setting on Web page.



Picture 52 - Cloud phone book list

10.2.3.2 Downloading Cloud Phone book

In cloud phone book screen, user can open a cloud phone book by pressing [**Enter**] button. The device will start downloading the phone book. The user will be prompted with a warning message if downloading failed. Once the cloud phone book is downloaded completely, the user can browse the contact list and dial the contact number same as in local phonebook.

Picture 53 - Downloading Cloud Phone book

10.3 Call Log

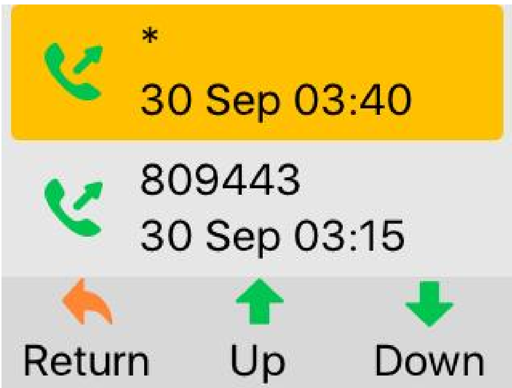
The device can store up to 1000 call log records and user can open the call logs to check all incoming, outgoing, and missed call records by pressing soft-menu button [**CallLog**] .

In the call logs screen, user may browse the call logs with up/down keys.

Each call log record is presented with 'call type' and 'call party number / name'.

The types of call ringtones and corresponding icons can be divided into the following categories:

- Missed Call Logs
- Incoming Call Logs
- Outgoing Call Logs
- Forward Call Logs

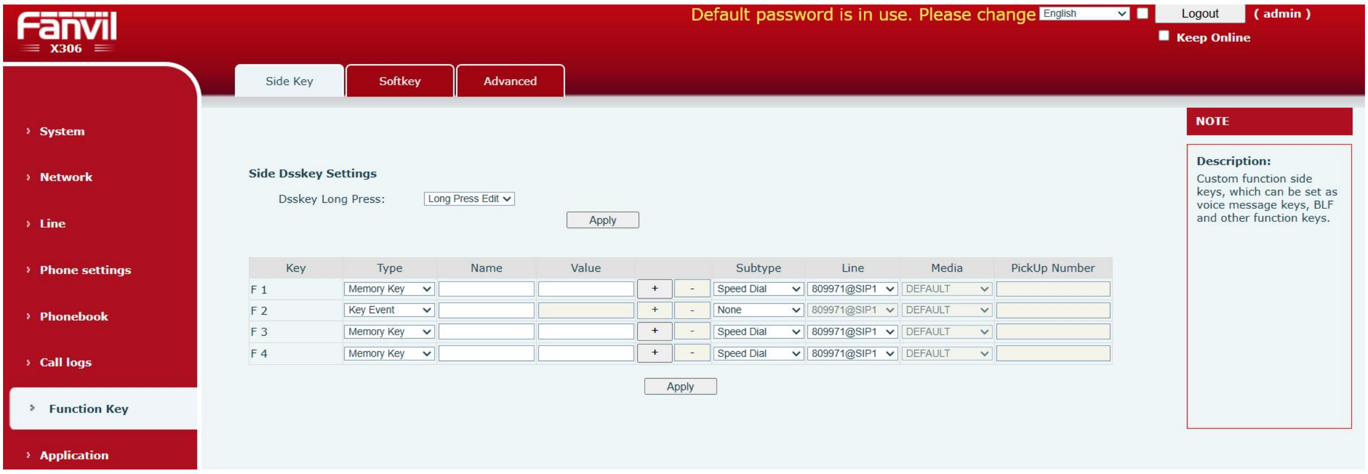


Picture 54 - Filter call record types

10.4 Function Key

Users can configure the speed dial key number and use it as a customizable function key. Users can modify the settings of the corresponding key by long pressing each shortcut key.

Webpage interface: [**Function key**] >> [**Side key**].



Picture 55 - DSS settings

Moreover, user also can add the user-defined title for the DSS Keys, which is configured as Memory Key / Line / URL / Multicast / Prefix.

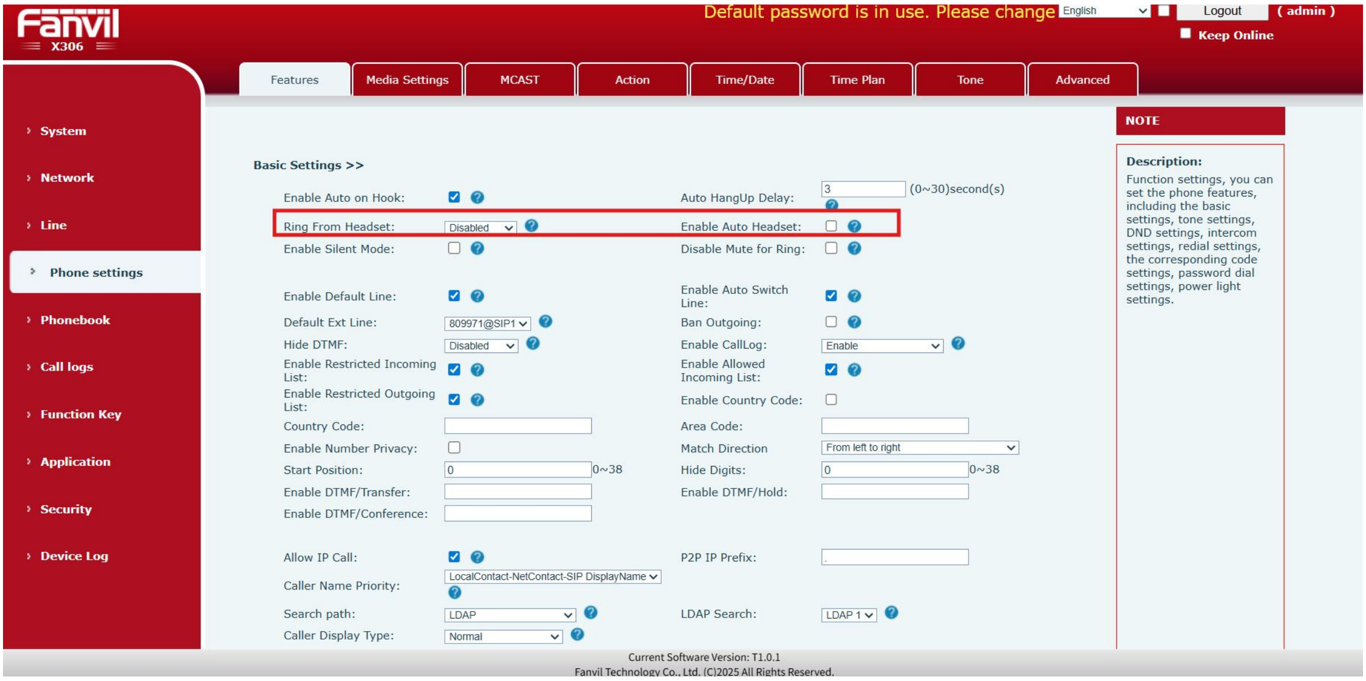
More detailed information *refers to* *12.23 Function Key* and 6.3 Appendix III –LED Definition .

10.5 Headset

The device supports Bluetooth headphones, enabling features such as incoming call prompts and headset calls. After connecting the headset to the phone, the phone's screen will display a Bluetooth connection icon

Picture 56 - Phone connected to Bluetooth

- On the webpage [**Phone settings**] >> [**Features**], you can set the headset answering function, and the ring tone for headset.



Picture 57 - Earphone function settings on the webpage

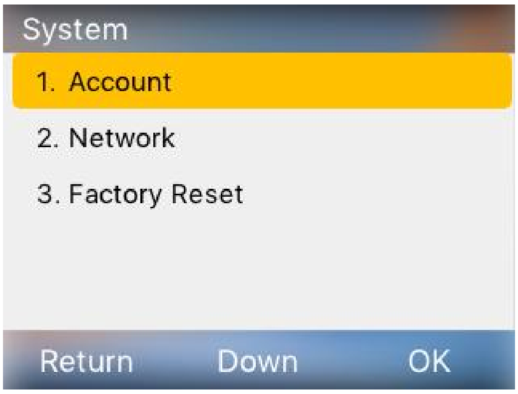
10.6 Advanced

10.6.1 Line Configurations

Picture 58 - SIP address and account information

Save the adjustment by pressing [**OK**] when done.

For users who want to configure more options, user should use web management portal to modify or Advanced Settings in accounts on the individual line to configure those options.



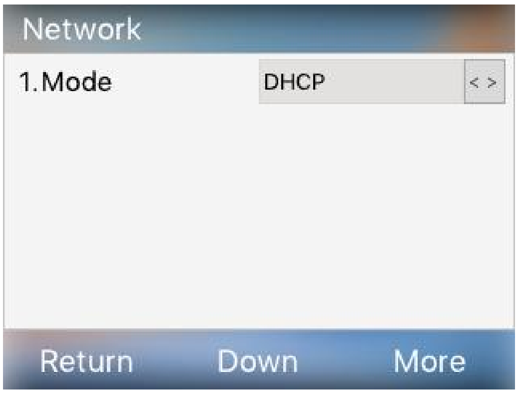
Picture 59 - Configure Advanced Line Options

10.6.2 Network Settings

10.6.2.1 Network Settings

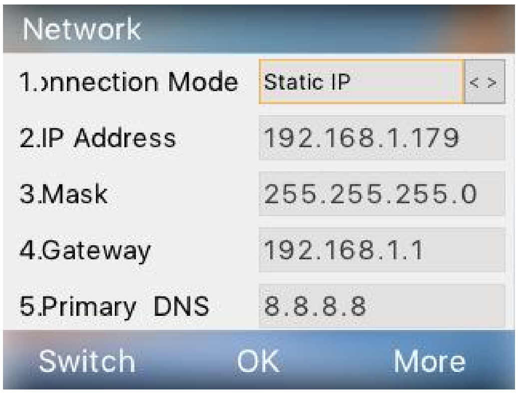
■ **IPv4**

In IPv4 mode, there are 2 connection mode options: DHCP and Static IP.



Picture 60 - DHCP network mode

When using DHCP mode, phone will get the IP address from DHCP server (router).



Picture 61 - Static IP mode

When using Static IP mode, user must configure the IP address manually.

- IP Address: Phone IP address.
- Mask: sub mask of your LAN.
- Gateway: The gateway IP address. Phone could access the other network via it.
- Primary DNS: Primary DNS address. The default is 8.8.8.8, Google DNS server address.
- Secondary DNS: Secondary DNS. When primary DNS is not available, it will work.

10.6.2.2 QoS & VLAN

■ **LLDP**

Link Layer Discovery Protocol. LLDP is a vendor independent link layer protocol used by network devices for advertising their identity, capabilities to neighbors on a LAN segment.

Phone could use LLDP to find the VLAN switch or other VLAN devices and use LLDP learn feature to apply the VLAN ID from VLAN switch to phone its self.

■ **CDP**

Cisco Discovery Protocol. CDP is a not-for-profit charity that runs the global disclosure system for investors, companies, cities, states and regions to manage their environmental impacts. According to the CDP, Cisco devices could share the OS version, IP address, hardware version and so on.

Table 10 - QoS & VLAN

Parameters	**Description**
LLDP setting	
Report	Enable LLDP
Packet Interval	LLDP requests interval time
Enable Learning Function	apply the learned VLAN ID to the phone configuration
QoS	
QoS Mode	configure SIP DSCP and audio DSCP

WAN VLAN	
WAN VLAN	WAN port VLAN configuration
LAN VLAN	
LAN VLAN	LAN port VLAN configuration
CDP	
CDP	CDP enable/disable ， CDP interval time

10.6.2.3 VPN

Virtual Private Network (VPN) is a technology to allow device to create a tunneling connection to a server and becomes part of the server's network. The network transmission of the device may be routed through the VPN server.

For some users, especially enterprise users, a VPN connection might be required to be established before activate a line registration. The device supports two VPN modes, Layer 2 Transportation Protocol (L2TP) and OpenVPN.

The VPN connection must be configured and started (or stopped) from the device web portal.

- **L2TP**

NOTICE! The device only supports non-encrypted basic authentication and non-encrypted data tunneling.

To establish a L2TP connection, users should log in to the device web portal, open webpage [**Network**] >> [**VPN**]. In VPN Mode, check the "Enable VPN" option and select "L2TP", then fill in the L2TP server address, Authentication Username, and Authentication Password in the L2TP section. Press "Apply" then the device will try to connect to the L2TP server.

When the VPN connection established, the VPN IP Address should be displayed in the VPN status. There may be some delay of the connection establishment. User may need to refresh the page to update the status. Once the VPN is configured, the device will try to connect with the VPN automatically when the device boots up every time until user disable it. Sometimes, if the VPN connection does not establish immediately, user may try to reboot the device and check if VPN connection established after reboot.

- **OpenVPN**

To establish an OpenVPN connection, user should get the following authentication and configuration files from the OpenVPN hosting provider and name them as the following,

OpenVPN Configuration file:	client.ovpn
CA Root Certification:	ca.crt
Client Certification:	client.crt
Client Key:	client.key

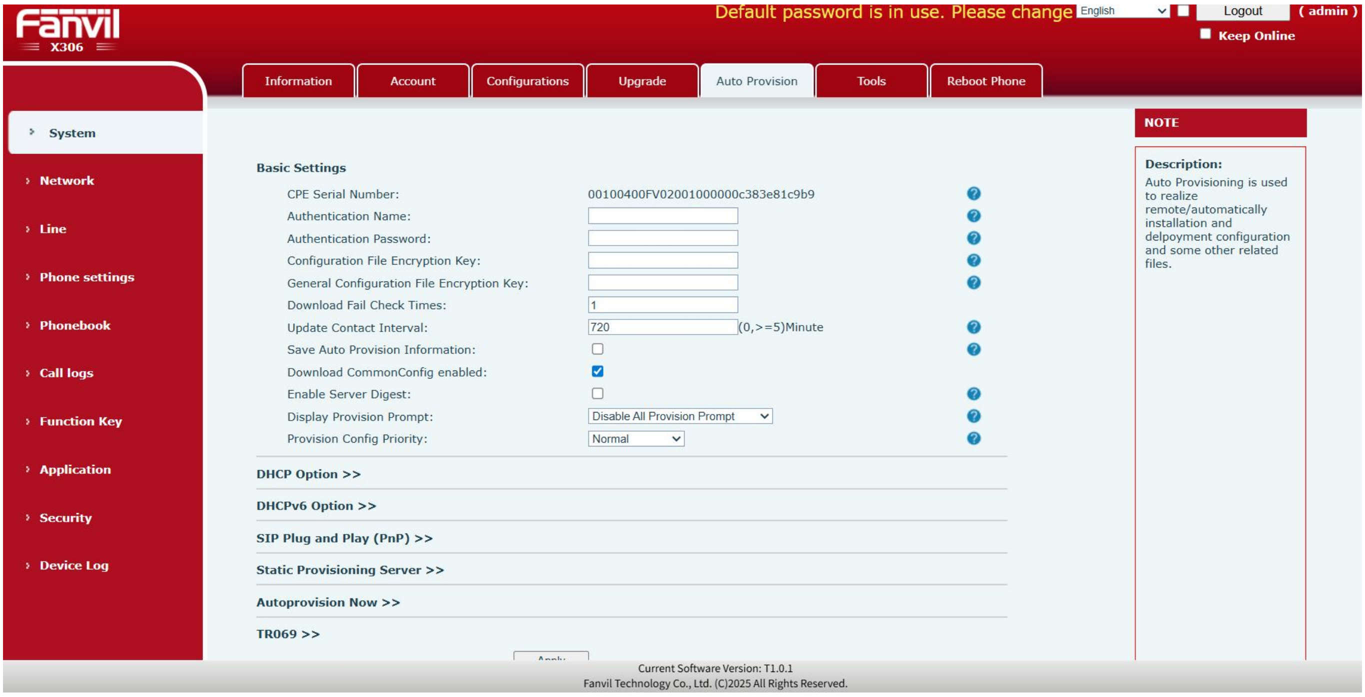
User then upload these files to the device in the web page [**Network**] >> [**VPN**], select OpenVPN Files. Then user should check "Enable VPN" and select "OpenVPN" in VPN Mode and click "Apply" to enable OpenVPN connection.

Same as L2TP connection, the connection will be established every time when system rebooted until user disable it manually.

PH

10.6.3 Maintenance

Phone Webpage: Login and go to [**System**] >> [**Auto provision**].



Picture 62 - Page auto provision Settings

Fanvil devices support SIP PnP, DHCP options, Static provision, TR069. If all of the 4 methods are enabled, the priority from high to low as below:

PNP>DHCP>TR069> Static Provisioning

Transferring protocol: FTP, TFTP, HTTP, HTTPS

Details refer to **Fanvil Auto Provision in**

Table 11 - Auto Provision

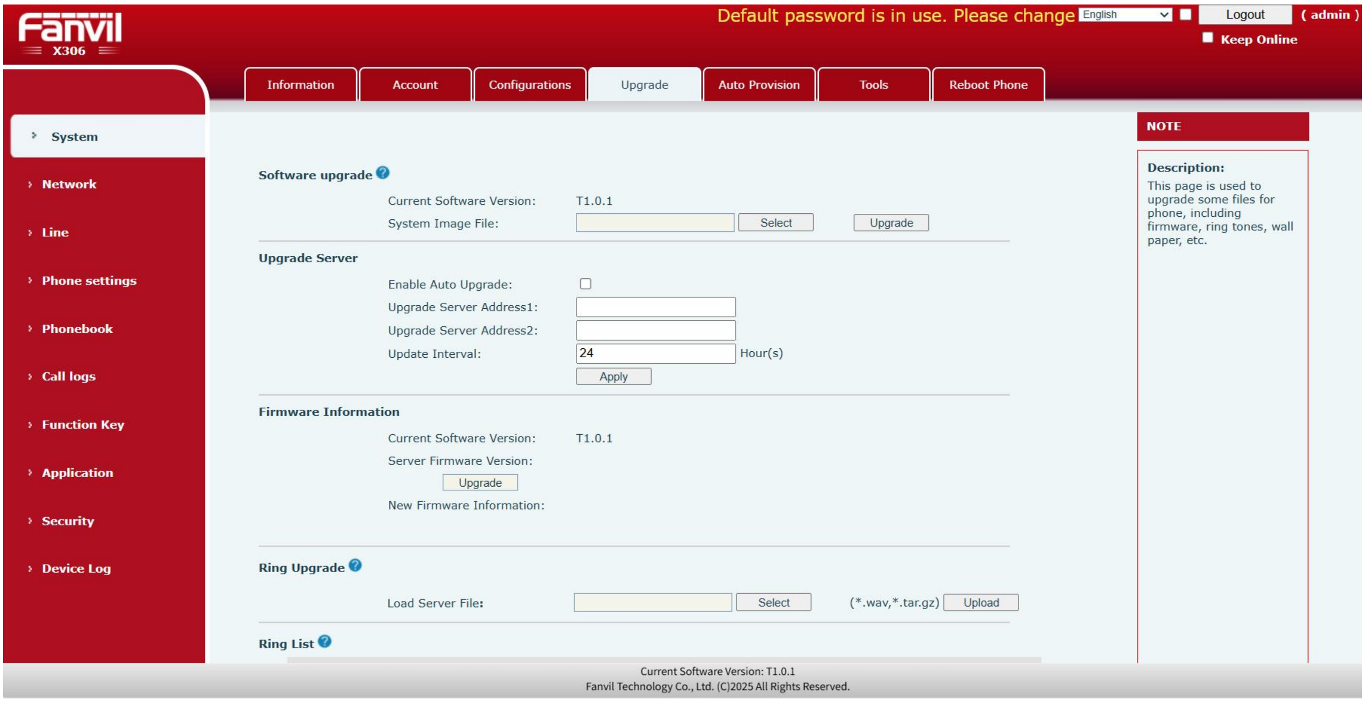
Parameters	**Description**
Basic settings	
CPE Serial Number	Display the device SN
Authentication Name	The user name of provision server
Authentication Password	The password of provision server
Configuration File Encryption Key	If the device configuration file is encrypted , user should add the encryption key here
General Configuration File Encryption Key	If the common configuration file is encrypted, user should add the encryption key here
Download Fail Check	If there download is failed, phone will retry with the configured times.

Times	
Update Contact Interval	Phone will update the phonebook with the configured interval time. If it is 0, the feature is disabled.
Save Auto Provision Information	Save the HTTP/HTTPS/FTP user name and password. If the provision URL is kept, the information will be kept.
Download CommonConfig enabled	Whether phone will download the common configuration file.
Enable Server Digest	When the feature is enable, if the configuration of server is changed, phone will download and update.
Display Provision Prompt	Configure if the phone display the provision prompt.
Provision Config Priority	During auto provision, the configuration file preferentially uses the local configuration of the phone or the configuration obtained by the server.
DHCP Option	
Option Value	Confiugre DHCP option, DHCP option supports DHCP custom option | DHCP option 66 | DHCP option 43, 3 methods to get the provision URL. The default is Option 66.
Custom Option Value	Custom Option value is allowed from 128 to 254. The option value must be same as server define.
Enable DHCP Option 120	Use Option120 to get the SIP server address from DHCP server.
DHCPv6 Option	
Option Value	Confiugre DHCPv6 option.
Custom Option Value	Custom Option value is allowed from 128 to 254. The option value must be same as server define.
SIP Plug and Play (PnP)	
Enable SIP PnP	Whether enable PnP or not. If PnP is enable, phone will send a SIP SUBSCRIBE message with broadcast method. Any server can support the feature will respond and send a Notify with URL to phone. Phone could get the configuration file with the URL.
Server Address	Broadcast address. As default, it is 224.0.0.0.
Server Port	PnP port
Transport Protocol	PnP protocol, TCP or UDP.
Update Interval	PnP message interval.
Static Provisioning Server	
Server Address	Provisioning server address. Support both IP address and domain address.
Configuration File Name	The configuration file name. If it is empty, phone will request the common file and device file which is named as its MAC address. The file name could be a common name, $mac.cfg, $input.cfg. The file

	format supports CFG/TXT/XML.
Protocol Type	Transferring protocol type ， supports FTP、TFTP、HTTP and HTTPS
Update Interval	Configuration file update interval time. As default it is 1, means phone will check the update every 1 hour.
Update Mode	Provision Mode. 1. Disabled. 2. Update After Reboot. 3. Update At Time Interval.
TR069	
Enable TR069	Enable TR069 after selection
ACS Server Type	There are 2 options Serve type, common and CTC.
ACS Server URL	ACS server address
ACS User	ACS server username (up to is 59 character)
ACS Password	ACS server password (up to is 59 character)
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting.
TLS Version	TLS version (TLS 1.0, TLS 1.1, TLS 1.2,TLS 1.3)
INFORM Sending Period	INFORM signal interval time. It ranges from 1s to 999999s
STUN Server Address	Configure STUN server address
STUN Enabled	To enable STUN server ~~for TR069~~

10.6.4 Firmware Upgrade

- Web page: Login phone web page, go to [**System**] >> [**Upgrade**].



Picture 63 - Online webpage upgrade page

Table 12 - Firmware upgrade

Parameter	Description
Upgrade server	
Enable Auto Upgrade	Allow automatic upgrade: Check the box to allow automatic upgrade, and the phone can detect TXT version information and available versions in the HTTP server.
Upgrade Server Address1	SFill in the available primary upgrade server (HTTP server) address.
Upgrade Server Address2	SFill in the available backup upgrade server (HTTP server) address, and when the primary server is unavailable, request the backup server.
Update Interval	Update cycle: The webpage enables automatic detection and upgrade, with a configured interval time. If the server has a new version, the phone will prompt for an upgrade at the interval time.
Firmware Information	
Current Software Version	It will show Current Software Version.
Server Firmware Version	It will show Server Firmware Version.
[Upgrade] button	If there is a new version txt and new software firmware on the server, the page will display version information and upgrade button will become available; Click [Upgrade] button to upgrade the new firmware.
New Firmware Information	When there is a corresponding TXT file and version on the server side, the TXT and version information will be displayed under the new version description information.

●The file requested by the phone to the server is a TXT file, with the file name being Vendor_model_ hw1_0. txt. After hw is the hardware version number, if hardware is not distinguished, it is written as hw1_0. Change all spaces in the file name to underscores.

●The URL requested by the phone is HTTP://server address/, and both the new version and the requested file need to be placed in the download directory of the HTTP server, as shown in the figure.

名称	类型
X306_Fanvil_hw1_1.txt	Text Document
X306_Fanvil_hw3_3.txt	Text Document
X306_Fanvil_hw2_2.txt	Text Document
X306-fanvil-release-9014--T1.0.1-UkrvT2025-09-26-11.50.54.z	WinRAR 压缩文件

- TXT file format must be UTF-8
- Vendor_model_ hw1_0. txt The file format is as follows：

  Version=0.0.4 #Firmware

  Firmware=xxx/xxx.z #URL，Relative paths are supported and absolute paths are possible, distinguished by the presence of protocol headers.

  BuildTime=2023.05.11 20:00

  Info=TXT|XML

  Xxxxx
  Xxxxx
  Xxxxx
  Xxxxx
- After the interval of update cycle arrives, if the server has available files and versions, Click to check the version information and upgrade.

10.6.5 Factory Reset

The phone is in default standby mode.

- Press #* 107 to enter the phone【Menu】>>【Advanced】, and press the【OK】key.
- Press【Advanced】 to enter the password (default password is 123) to enter the interface.
- Press the【Factory Reset】 button and select Clear All.
- After completion, press【OK】to clear. After clearing, the machine will automatically restart.

11Web Configurations

11.1 Web Page Authentication

The user can log into the web page of the phone to manage the user's phone information and operate the phone. Users must provide the correct user name and password to log in.

11.2 System >> Information

User can get the system information of the device in this page including,

- Model
- Hardware
- Software
- Uboot
- Uptime
- VLAN Speed
- MEMInfo
- System time
- SN

And summarization of network status,

- **WAN**
- Network mode
- Ethernet MAC
- Wi-Fi MAC
- Bluetooth MAC
- **IPv4**
- Ethernet IP
- Subnet mask
- Default gateway

And obtain the voice quality status of the phone,

- Start time
- Stop time
- Local user
- Remote user
- Local IP
- Remote IP
- Local Port
- Remote port
- Local codec

- Remote codec
- Jitter
- JitterBufferMax
- Packets lost
- NetworkPacketLossRate
- MOS-LQ
- MOS-CQ
- RoundTripDelay
- EndSystemDelay
- SymmOneWayDelay
- JitterBufferRate

Besides, summarization of SIP account status,

- Line 1 (Line Status: Registered/Not Submitted/Attempting/Timeout)
- Line 2 (Line Status: Registered/Not Submitted/Attempting/Timeout)

11.3 System >> Account

On this page the user can change the password for the login page.

Users with administrator rights can also add or delete users, manage users, and set permissions and passwords for new users.

11.4 System >> Configurations

On this page, users with administrator privileges can view, export, or import the phone configuration, or restore the phone to factory Settings.

- **Clear Configurations**

Select the module in the configuration file to clear.

SIP: account configuration.

AUTOPROVISION: automatically upgrades the configuration

TR069:TR069 related configuration

MMI: MMI module, including authentication user information, web access protocol, etc.

DSS Key: DSS Key configuration

- **Clear Userdata**

Select the local database table to be cleared. The default clear box is empty and not selected.

- **Reset Phone**

The phone data will be cleared, including configuration and database tables.

11.5 System >> Upgrade

Upgrade the phone software version, customized ringtone, background, etc., can also be upgraded to delete the file. Ring tone support “.wav” format.

11.6 System >> Auto Provision

The Auto Provision settings help IT manager or service provider to easily deploy and manage the devices in mass volume. For the detail of Auto Provision, please refer to this link Table11 Auto Provision.

11.7 System >> Tools

Tools provided in this page help users to identify issues at trouble shooting. Please refer to 13 Trouble Shooting for more detail.

11.8 System >> Reboot Phone

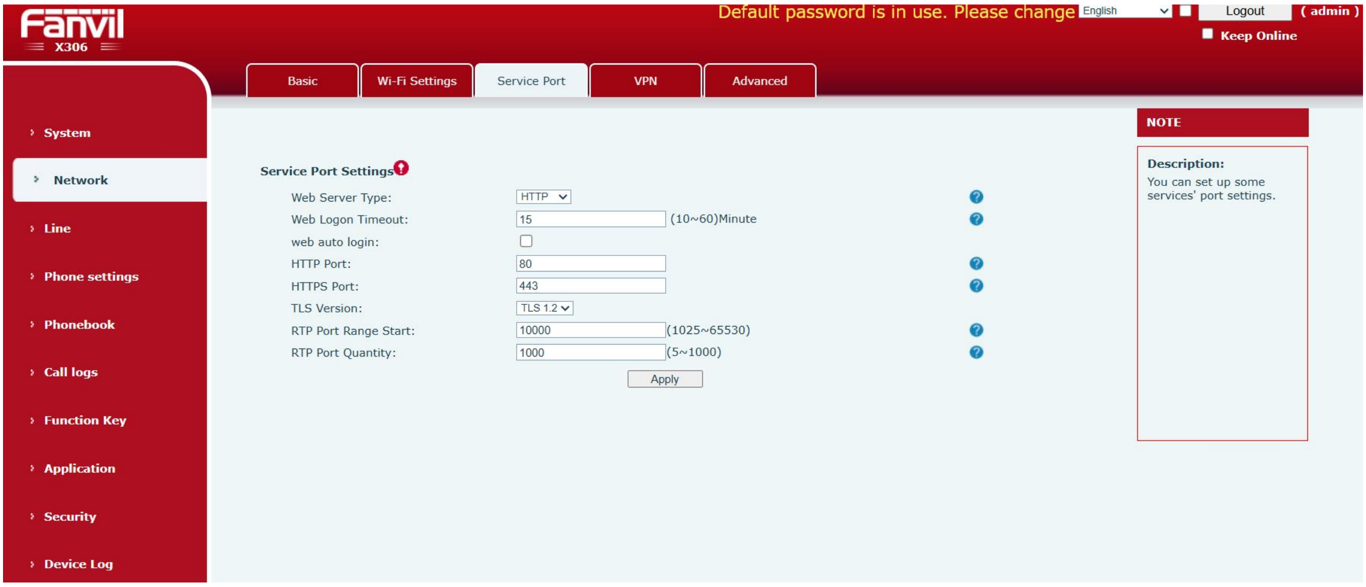
This page can restart the phone.

12 Network >> Basic

This page allows users to configure network connection types and parameters.

12.1 Network >> Service Port

This page provides settings for Web page login protocol, protocol port settings and RTP port.



Picture 64 - Service Port Settings

Table 13 - Service port

Parameters	Description
Web Server Type	Reboot to take effect after settings. Optionally, the web page login is HTTP/HTTPS.
Web Logon Timeout	Default as 15 minutes, the timeout will automatically exit the login page, need to login again.
Web auto login	After the timeout does not need to enter a user name password, will automatically login to the web page.
HTTP Port	The default is 80. If you want system security, you can set ports other than 80. Such as :8080, webpage login: HTTP://ip:8080
HTTPS Port	The default is 443, the same as the HTTP port.
TLS Version	Default TLS 1.2, optional: TLS 1.0, TLS 1.1, TLS 1.2, TLS 1.3
RTP Port Range Start	The value range is 1025 to 65530. The value of RTP port starts from the initial value set. For each call, the value of voice and video port is added 2.
RTP Port Quantity	The value range is 5 to 1000. Number of calls.

12.2 Network >> VPN

Users can configure a VPN connection on this page. See 10.6.2.3 VPN for more details.

12.3 Network >> Advanced

Advanced network Settings are typically configured by the IT administrator to improve the quality of the phone service. For configuration, query the 10.6 advanced Settings.

12.4 Line >> SIP

Configure the Line service configuration on this page.

Table 14 - Line configuration on the web page

Parameters	Description
Register Settings	
Line Status	Display the current line status at page loading. To get the up to date line status, user has to refresh the page manually.
Activate	Whether the service of the line is activated
Username	Enter the username of the service account.
Authentication User	Enter the authentication user of the service account
Display Name	Enter the display name to be sent in a call request.
Authentication Password	Enter the authentication password of the service account
Realm	Enter the SIP domain if requested by the service provider
Server Name	Input server name.
SIP Server 1	
Server Address	Enter the IP or FQDN address of the SIP server
Server Port	Enter the SIP server port, default is 5060
Transport Protocol	Set up the SIP transport line using TCP or UDP or TLS.
Registration Expiration	Set SIP expiration date.
SIP Server 2	
Server Address	Enter the IP or FQDN address of the SIP server
Server Port	Enter the SIP server port, default is 5060
Transport Protocol	Set up the SIP transport line using TCP or UDP or TLS.
Registration Expiration	Set SIP expiration date.
Proxy Server Address	Enter the IP or FQDN address of the SIP proxy server.
Proxy Server Port	Enter the SIP proxy server port, default is 5060.
Proxy User	Enter the SIP proxy user.
Proxy Password	Enter the SIP proxy password.

Backup Proxy Server Address	Enter the IP or FQDN address of the backup proxy server.
Backup Proxy Server Port	Enter the backup proxy server port, default is 5060.
Basic Settings	
Enable Auto Answering	Enable auto-answering, the incoming calls will be answered automatically after the delay time
Auto Answering Delay	Set the delay for incoming call before the system automatically answered it
Call Forward Unconditional	Enable unconditional call forward, all incoming calls will be forwarded to the number specified in the next field
Call Forward Number for Unconditional	Set the number of unconditional call forward
Call Forward on Busy	Enable call forward on busy, when the phone is busy, any incoming call will be forwarded to the number specified in the next field.
Call Forward Number for Busy	Set the number of call forward on busy .
Call Forward on No Answer	Enable call forward on no answer, when an incoming call is not answered within the configured delay time, the call will be forwarded to the number specified in the next field.
Call Forward Number for No Answer	Set the number of call forward on no answer.
Call Forward Delay for No Answer	Set the delay time of not answered call before being forwarded.
Transfer Timeout	Set call transfer timeout, wait for timeout to send BYE after transfer.
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
Voice Message Number	Set the number for retrieving voice message
Voice Message Subscribe Period	Set the interval of voice message notification subscription
Enable Hotline	Enable hotline configuration, the device will dial to the specific number immediately at audio channel opened by off-hook handset or turn on hands-free speaker or headset
Hotline Delay	Set the delay for hotline before the system automatically dialed it
Hotline Number	Set the hotline dialing number
Dial Without Registered	Set call out by proxy without registration
Enable Missed Call Log	If enabled, the phone will save missed calls into the call history record.
DTMF Type	Set the DTMF type to be used for the line
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'

Request With Port	Whether the URI carry port information
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected automatically
Use VPN	Set the line to use VPN restrict route
Use STUN	Set the line to use STUN for NAT traversal
Enable Failback	Whether to switch to the primary server when it is available.
Failback Interval	A Register message is used to periodically detect the time interval for the availability of the main Proxy.
Signal Failback	Multiple proxy cases, whether to allow the invite/register request to also execute failback.
Signal Retry Counts	The number of attempts that the SIP Request considers proxy unavailable under multiple proxy scenarios.
Codecs Settings	Set the priority and availability of the codecs by adding or remove them from the list.
Advanced Settings	
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send feature code to the server by dialing the number specified in each feature code field.
Enable DND	Set the feature code to dial to the server
DND Disabled	Set the feature code to dial to the server
Enable Call Forward Unconditional	Set the feature code to dial to the server
Disable Call Forward Unconditional	Set the feature code to dial to the server
Enable Call Forward on Busy	Set the feature code to dial to the server
Disable Call Forward on Busy	Set the feature code to dial to the server
Enable Call Forward on No Answer	Set the feature code to dial to the server
Disable Call Forward on No Answer	Set the feature code to dial to the server
Enable Blocking Anonymous Call	Set the feature code to dial to the server
Disable Blocking Anonymous Call	Set the feature code to dial to the server
Send Anonymous On Code	Set the feature code to dial to the server

Send Anonymous Off Code	Set the feature code to dial to the server
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session will be ended if there is not new session timer event update received after the timeout period
Session Timeout	Set the session timer timeout period
Enable BLF List	Enable/Disable BLF List
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
Response Single Codec	If setting enabled, the device will use single codec in response to an incoming call request
BLF Server	The registered server will receive the subscription package from ordinary application of BLF phone. Please enter the BLF server, if the sever does not support subscription package, the registered server and subscription server will be separated.
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT pinhole opened
Keep Alive Interval	Set the keep alive packet transmitting interval
Keep Authentication	Keep the authentication parameters from previous authentication
Blocking Anonymous Call	Reject any incoming call without presenting caller ID
RTP Encryption(SRTP)	enables RTP encryption, and RTP transmission will be encrypted.
Enable OSRTP	after selecting 'Enable OSRTP'.
Proxy Require	Set Proxy Request
User Agent	Set the user agent, the default is Model with Software Version.
Specific Server Type	Set the line to collaborate with specific server type
SIP Version	Set the SIP version
Anonymous Call Standard	Set the standard to be used for anonymous
Local Port	Set the local port
Ring Type	Set the ring tone type for the line
Enable user=phone	Sets user=phone in SIP messages.
Use Tel Call	Configure whether to enable phone calling
Auto TCP	Using TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes
Enable Rport	Set the line to add rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
Call-ID Format	Configure the format of the Call ID field for IP calls
DNS Mode	Select DNS mode, A, SRV, NAPTR
Enable Long Contact	Allow more parameters in contact field per RFC 3840

Enable Strict Proxy	Compatible with special servers (using the other party's source address when returning messages, no longer using the address in the via field)
Convert URI	Convert not digit and alphabet characters to %hh hex code
Use Quote in Display Name	Whether to add quote in display name, i.e. “Fanvil” vs Fanvil
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
Sync Clock Time	Time Sync with server
Enable Use Inactive Hold	With the post-call hold capture package enabled, you can see that in the INVITE package, SDP is inactive.
Caller ID Header	Set the Caller ID Header
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response
Enable Feature Sync	Feature Sync with server
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
TLS Version	Choose TLS Version.
uaCSTA Number	Set uaCSTA Number.
Enable Click To Talk	With the use of special server, click to call out directly after enabling.
Enable ChangePort	Whether enable port update
VQ Name	Setting VQ Name.
VQ server	setting VQ server address.
VQ Server Port	setting VQ port.
VQ Http/Https server	Settings VQ HTTP/HTTPS Server.
Flash mode	Chose Flash mode，normal or SIP info.
Flash Info Content-Type	Set the SIP info content type.
Flash Info Content-Body	Set the SIP info content body.
Server Expire	Set the timeout period for using the server
Unregister On Boot	Whether enable the logout function
Enable MAC Header	Whether enable SIP packets with or without MAC when registering with the user agent.
Enable Register MAC Header	Whether register for user agent with or without MAC.
BLF Dialog Strict Match	Whether to enable BLF session exact matching.
PTime	Set the packaging duration, selectable: disabled, 10, 20, 30, 40, 50, 60, in milliseconds
Enable Deal 180	Whether to enable “Enable Deal 180”
Transaction Timer T1	Configure the duration of SIP transaction timer T1, range: 500-10000, unit: milliseconds.

Transaction Timer T2	Configure the duration of SIP transaction timer T2, range: 2000-40000, unit: milliseconds
Transaction Timer T4	Configure the duration of SIP transaction timer T4, range: 2500-60000, unit: milliseconds
CallPark Number	Keep the call to the configured number, record the voice broadcast number, and use another terminal to call the recorded number to retrieve the call.
PickUp Number	Set pickup number
Retrieve Number	Set retrieve number
Intercom Number	Set intercom number
SIP Global Settings	
Strict Branch	Set up to strictly match the Branch field.
Enable Group	Set open group.
Enable RFC4475	Set to enable RFC4475.
Enable Strict UA Match	Enable strict UA matching.
Registration Failure Retry Time	Set the registration failure retry time.
Local SIP Port	Modify the phone SIP port.
Enable uaCSTA	Set to enable uaCSTA functionality

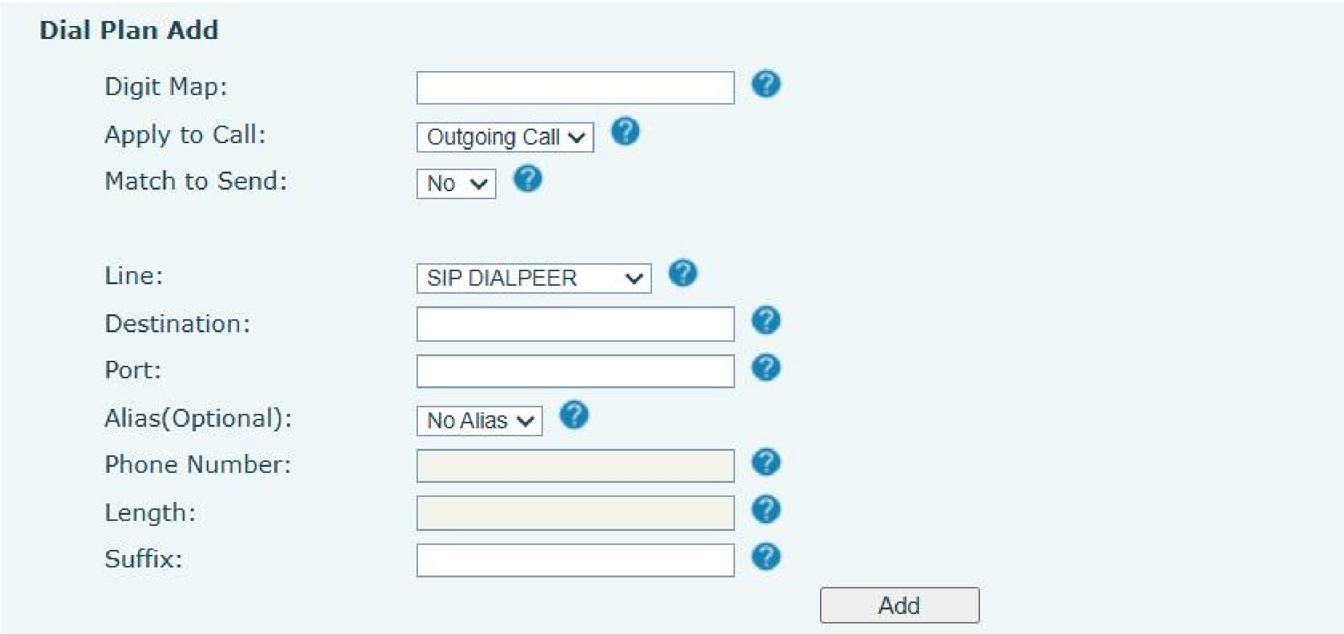
12.5 Line >> Dial Plan

Picture 65 - Dial plan settings

Table 15 - Phone 4 dialing methods

Parameters	**Description**
Press # to invoke dialing	The user dials the other party's number and then adds the # number to dial out;
Dial Fixed Length	The number entered by the user is automatically dialed out when it reaches a fixed length
Timeout dial	The system dials automatically after timeout
Enable E.164	Please refer to e. 164 standard specification

Add dialing rules:



Picture 66 - Custom setting of dial - up rules

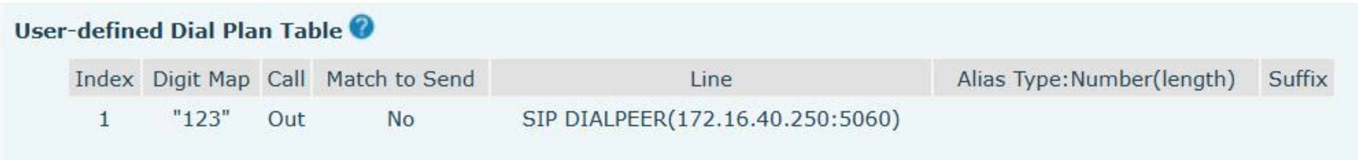
Table 16 - Dial - up rule configuration table

<table>
<tr><th>Parameters</th><th>Description</th></tr>
<tr><td>Dial rule</td><td>There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules.
In prefix matching, only part of the number is entered followed by T. The mapping with then take place whenever these digits are dialed. Prefix mode supports a maximum of 30 digits.</td></tr>
<tr><td colspan="2">Note: Two different special characters are used.
■ x -- Matches any single digit that is dialed.
■ [ ] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.</td></tr>
<tr><td>Destination</td><td>Set Destination address. This is for IP direct.</td></tr>
<tr><td>Port</td><td>Set the Signal port, and the default is 5060 for SIP.</td></tr>
<tr><td>Alias(Optional)</td><td>Set the Alias. This is the text to be added, replaced or deleted. It is an optional item.</td></tr>
<tr><td colspan="2">Note: There are four types of aliases.
■ all: xxx – xxx will replace the phone number.
■ add: xxx – xxx will be dialed before any phone number.
■ del –The characters will be deleted from the phone number.
■ rep: xxx – xxx will be substituted for the specified characters.</td></tr>
</table>

Suffix	Characters to be added at the end of the phone number. It is an optional item.
Length	Set the number of characters to be deleted. For example, if this is set to 3, the phone will delete the first 3 digits of the phone number. It is an optional item.
Phone Number	Set Phone number.

This feature allows the user to create rules to make dialing easier. There are several different options for dial rules. The examples below will show how this can be used.

Example 1: All Substitution -- Assume that it is desired to place a direct IP call to IP address 172.16.40.250. Using this feature, 123 can be substituted for 172.16.40.250.

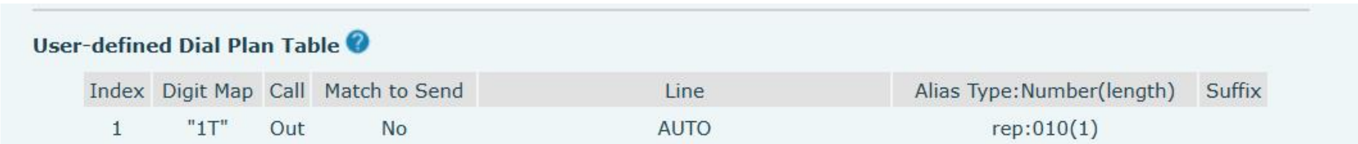


User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix
1	"123"	Out	No	SIP DIALPEER(172.16.40.250:5060)		

Picture 67 - Dial rules table (1)

Example 2: Partial Substitution -- To dial a long distance call to Beijing requires dialing area code 010 before the local phone number. Using this feature 1 can be substituted for 010. For example, to call 62213123 would only require dialing 162213123 instead of 01062213123.



User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix
1	"1T"	Out	No	AUTO	rep:010(1)	

Picture 68 - Dial rules table (2)

Example 3: Addition -- Two examples are shown. In the first case, it is assumed that 0 must be dialed before any 11 digit number beginning with 13. In the second case, it is assumed that 0 must be dialed before any 11 digit number beginning with 135, 136, 137, 138, or 139. Two different special characters are used.

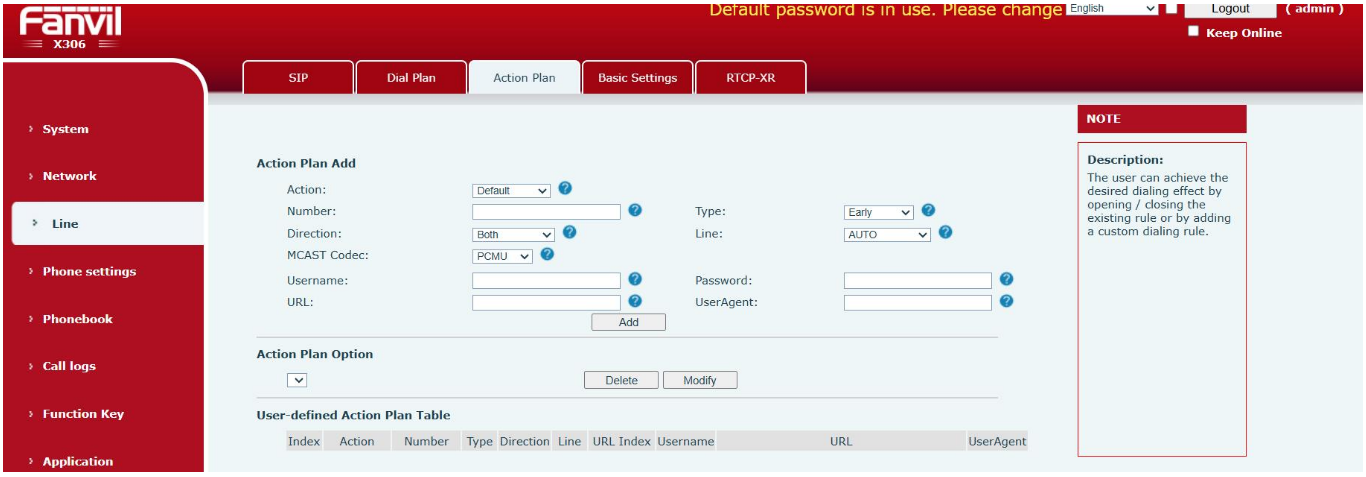
x -- Matches any single digit that is dialed.

[] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.

12.6 Line >> Action Plan

Action plan: a technical implementation defined and designed by fanvil for remote control and behavior linkage between fanvil terminal equipment and other equipment. That is, when an event occurs on the fanvil terminal, the terminal can execute an action, which is completed according to a plan rule.

Log in to the phone web, visit [**Line**] >[**Action plan**], and configure action plan rules.



Picture 69 - Action Plan

Table 17 - Action Plan

Parameter	Description
Action	Default: when the rule is triggered, the phone displays video or converts multicast according to the RTSP URL or multicast address port set by the website. MCAST-Xfer: when the rule is triggered, the phone converts the incoming call or multicast into multicast and sends it to the set multicast address port. Record: the phone automatically turns on the recording function when the rule is triggered. Mute: the phone will mute automatically when the rule is triggered. Answer: when the rule is triggered, the phone automatically answers the incoming call.
Number	Auxiliary phone number
Type	Early: trigger execution before call establishment. Connected: trigger execution after call establishment.
Direction	For call mode, incoming/outgoing call
Line	Set up outgoing lines.
Username	Bind the user name of the IP camera.
Password	Bind IP camera password.
URL	Video streaming information or MCAST IP

	address.
User Agent	Set user agent information
MCAST	It is the multicast encoding sent when the multicast conversion rule is triggered

12.7 Line >> Basic Settings

Set up the register global configuration.

Table 18 - Set the line global configuration on the web page

Parameters	Description
STUN Settings	
STUN NAT Travelsal	Display whether STUN penetration is successful.
Server Address	Set the STUN server address.
Server Port	Set the STUN server port to 3478 by default.
Binding Period	Set the STUN binding cycle to ensure NAT traversal is enabled.
SIP Waiting Time	Set the timeout period for STUN binding before sending SIP messages.
SIP P2P Settings	
Enable Auto Answering	Set whether to enable automatic answering.
Auto Answering Delay	Set automatic answering delay time, range: 0-120 seconds.
DTMF Type	DTMF modes, including In band, RFC2833, SIP-IN, AUTO, RFC2833+SIP-IN.
DTMF SIP INFO Mode	When dtmfSendType is SIPInfo, the representation of #/* includes: send 10/11, send */#.
Use VPN	Set whether to use VPN.
Call-ID Format	Set the format of the Call ID field in SIP messages, valid formats: $id, $id @ $ip.
Display name	Set display name.

12.8 Line>>RTCP-XR

Table 19- RTCP-XR

Parameters	Description
VQ RTCP-XR Settings	
VQ RTCP-XR Session Report:	Set STUN server address, divided into disabled and enabled, default is enabled

VQ RTCP-XR Interval Report	Set STUN server address, divided into disabled and enabled, default is enabled
Period for Interval Report	Valid values range from 5 to 99,the defualt value is 60.
Warning threshold for Moslq	Valid values range from 15 to 40,the defualt value is 40.
Critical threshold for Moslq	Valid values range from 15 to 40,the defualt value is 25.
Warning threshold for Delay	Valid values range from 10 to 2000,the defualt value is 150.
Critical threshold for Delay(10~2000)	Valid values range from 10 to 2000,the defualt value is 200.
Display Report options on phone	It is enabled by default.
Display Report options on Web	It is enabled by default.

12.9 Phone settings >> Features

Configuration phone features.

Table 20 - General function Settings

Parameters	**Description**
Basic Settings	
Enable Auto Onhook	Configure whether to enable automatic hang up, end call, and return to standby mode automatically.
Auto HangUp Delay	Configure the automatic hang up time. If it is in hands-free mode, the phone will automatically return to standby mode after exceeding the auto handdown time. If it is in controller mode, the dial tone will be automatically played after exceeding the auto handdown time.
Ring From Headset	Configure ringing through headset.
Enable Auto Headset	Enable this feature. If the user has headphones plugged into their phone, pressing the answer or line button can use the headphones to answer calls, and enabling the automatic answering function is also the same.
Enable Silent Mode	After turning on, the phone is in a silent state. When there is an incoming call, it will not ring. You can use the volume and mute keys to release this state.
Disable Mute for Ring	The mute button on the phone does not work when turned on.

Enable Default Line	If enabled, the phone will be assigned a default line instead of SIP1.
Enable Auto Switch Line	Enable phone to select an available SIP line as default automatically.
Default Ext Line	Select the default route for outgoing calls.
Ban Outgoing	Prohibit outgoing calls. When enabled, dialing off the phone will send a busy tone and prompt to hang up.
Hide DTMF	Configure the form of hidden DTMF.
Enable CallLog	Choose whether to save call logs.
Enable Restricted Incoming List	Whether to enable the restricted call list.
Enable Allowed Incoming List	Whether to enable the allowed call list.
Enable Restricted Outgoing List	Whether to enable the restricted outgoing call list.
Enable Country Code	Whether to enable the country code
Country Code	Fill in the country code.
Area Code	Fill in the area code.
Enable Number Privacy	Whether to enable number privacy.
Match Direction	Matching direction, there are two kinds of rules from right to left and from left to right.
Start Position	The starting position to hide after enabling number privacy.
Hide Digits	The number of digits to hide after enabling number privacy.
Enable DTMF/Transfer	Set the DTMF value received by the server after a transfer operation.
Enable DTMF/Hold	Set the DTMF value received by the server after a hold operation.
Enable DTMF/Conference	Set the DTMF value received by the server after a conference operation.
Allow IP Call	If enabled, the phone allows direct IP calls.
P2P IP Prefix	Set the prefix for point-to-point IP calls.
Caller Name Priority	Select the priority for displaying the caller's name.
Search path	Select the search range.
LDAP Search	If the search range is set to LDAP, this needs to be selected.
Call Display Type	Set the display method for the calling party information.
Restrict Active URI Source IP	Set the valid URI commands that the phone accepts from a specific IP address. Note: This function is usually used for device management.
Push XML Server	Configure the XML server. When the phone receives a request, it will determine.
Enable Pre-Dial	whether to display the corresponding content sent by the server on the specified phone.
Enable Multi Line	If enabled, the phone can handle up to 10 concurrent calls. If disabled, it

	can handle up to 2 concurrent calls.
Line Display Format	Customize the line format: SIPn/SIPn: xxx/xxx@SIPn.
Contact As Allowed List Type	Contact person allowed incoming call types: NONE/BOTH/DND Allowed List/FWD Allowed List.
Block XML When Call	Whether to disable XML push during a call.
SIP Notify	After enabling, when the phone receives content with relevant notify information, it will display the corresponding message.
Call Number Filter	Set the call numbers to be filtered.
Auto Resume Current	Set whether to enable automatic resumption of the current call. If the current path changes, automatically cancel Hold.
Call Timeout	Set the call timeout time. If the remote phone does not answer within this time, the local end will automatically hang up.
Ring Timeout	Set the ringback time for incoming calls.
Ring Priority	Set the priority for handling incoming calls.
Enable Display To Info	Set whether to display the information in the "to" field.
Enable Push XML Auth	Set whether to push XML authentication.
Tone Settings	
Enable Holding Tone	After activation, a prompt tone will be played during the call.
Play Dialing DTMF Tone	When the user presses the number keys while dialing, a DTMF prompt tone will be heard. The phone is set to be on by default.
Play Talking DTMF Tone	When the user presses the number keys during the call, a DTMF prompt tone will be heard. The phone is set to be on by default.
Auto Answer Tone	After activation, an "beep" prompt sound will be heard during automatic answering.
Ring Back Tone	Customize the outgoing call prompt tone.
Busy Tone	Customize the prompt tone when hanging up.
DND Settings	
DND Option	Can set to off, phone, or line.
Enable DND Timer	Whether to enable timed automatic activation of Do Not Disturb. After activation, it will automatically turn on Do Not Disturb from the start time to the end time.
DND Start Time	Set the start time.
DND End Time	Set the end time.
Intercom Settings	
Enable Intercom	When the intercom system is enabled, the device will automatically answer incoming calls by accepting the SIP header Alert-Info instruction of the call request.

Enable Intercom Mute	Enable the mute function during the intercom mode call.
Enable Intercom Tone	When there is an incoming call in the intercom mode, you will hear a prompt tone.
Redial setting	
Enable Call Completion	When the configuration item is enabled, after the other party of the call initiates a response code 486 to the initiating party, and the other party's phone ends the current call, this end will prompt "Call completed", and after clicking "Confirm", it will redial the other party's number.
Enable Auto Redial	Configure whether to enable the automatic redial function. After the call ends, when the phone at the other end answers with 486 (the default busy response code), the phone will automatically redial according to the set number and time interval.
Auto Redial Interval	Set the time interval for automatic redial.
Auto Redial Times	Set the number of automatic redials.
Redial Enter CallLog	Configure whether to redial the most recently called number when pressing the redial key, or enter the call history.
Response Code Settings	
DND Response Code	Set the SIP response code for do not disturb.
Busy Response Code	Set the SIP response code when the phone is busy.
Reject Response Code	Set the SIP response code when the call is rejected.
Password Dial Settings	
Enable Password Dial	Enable password dialing. When the input quantity starts with the password prefix, the following quantity N * prefix will hide the password. N represents the value of the password length field you input. For example: if the prefix is 3 and the password is set, and the password length is 2, then when you enter 34567, it will be displayed as 3 * * 67.
Password Dial Prefix	Configure the prefix of the password dialing number
Encryption Number Length	Configure the length of the hidden number
Bluetooth Settings	
Enable Bluetooth	Enable Bluetooth settings
Bluetooth Name	Bluetooth display name
Power LED	
Talking LED(priority level from high to low)	
Ringing	The power light status when the power light is on when there is an incoming ringtone, including off/on/slow flashing/fast flashing. The default is fast flashing.
Hold/Held	The power light status when maintaining/keeping, including off/on/slow flashing/fast flashing. The default is off.

Mute	The power light status when in mute, including off/on/slow flashing/fast flashing. The default is off.
Talk/Dial	The power light status when in call/dialing, off is off, on is red constantly on, default is off
Common LED (priority level from high to low)	
Missed call	The power indicator status when there are missed calls is set, including off/on/slow flashing/fast flashing, with the default being slow flashing.
SMS/Voice Mail	The power indicator status when there are unread voice messages or short messages is set, including off/on/slow flashing/fast flashing, with the default being slow flashing.
Registration Failed	The power indicator status when registration fails is set, including off/on/slow flashing/fast flashing, with the default being slow flashing.
Phone Silent	The power indicator status when the phone is muted is set, including off/on/slow flashing/fast flashing, with the default being slow flashing.
Common	The default power indicator status is set, including off/on, with the default being off.
Power Saving	Whether to enable the energy-saving mode is set, including off/on, with the default being on.
Notification Popups	
Display Missed Call Popup	After activation, there is a pop-up notification for missed calls. When deactivated, there is no such pop-up notification. It is set to be activated by default.
Display Voice Mail Popup	After activation, there is a pop-up notification for unattended voice messages. When deactivated, there is no such pop-up notification. It is set to be activated by default.
Display SMS Popup	After activation, there is a pop-up notification for unread messages. When deactivated, there is no such pop-up notification. It is set to be activated by default.
Display Other Popup	After activation, when there are other exceptions such as the handset not being hung up, registration failure, IP acquisition failure, Tr069 connection failure, etc., there is a pop-up notification. When deactivated, there is no such exception notification. It is set to be activated by default.
Display Device Connect Popup	After activation, there is a pop-up notification for external devices. When deactivated, there is no pop-up notification. It is set to be activated by default.
Pick up & Park	
Display BLF PickUp popup	When there is a call to the subscribed BLF number, does it display the interrupting prompt interface.

Play BLF PickUp Tone	When there is a call to the subscribed BLF number, does it play the interrupting interception prompt sound.
Ring Type For BLF PickUp	Set the type of interrupting ringtone.
Display Call Park Popup	Set whether to display the call residence prompt.
Play Call Park Tone	Set whether to play the call residence prompt sound.
Ring Type For Call Park	Set the type of call residence prompt sound.

12.10 Phone settings >> Media Settings

Change voice Settings.

Table 21 - Voice settings

Parameters	Description
Codecs Settings	Select enable or disable voice encoding: G.711A/U, G.722, G.729AB, G726, ILBC, opus,G723.1
Audio Settings	
Handset Volume	Set the Handset volume, the value must be 1~9.
Default Ring Type	Configure default ringtones. If no special ringtone is set for the phone number, the default ringtone will be used.
Speakerphone Volume	Set the hands-free volume to 1-9.
Headset Ring Volume	Set the volume of the Headset ringtone to 0~9.
Headset Volume	Set the volume of the headset to 1~9.
Speakerphone Ring Volume	Set the volume of hands-free ringtone to 0~9.
Handset SignalTone Volume	Handle signal sound volume, ranging from 0 to 9.
Headset SignalTone Volume	Earphone signal sound volume, ranging from 0 to 9.
Speakerphone SignalTone Volume	Hands free signal sound volume, ranging from 0 to 9.
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
Headset Mic Gain	Set the Headset's radio volume gain to fit different models of Headsets.
Controller Mic Gain	Set the receiver volume gain of the joystick.
Hands free Mic Gain	Set the volume gain for hands-free radio.
Opus playload type	Set Opus load type, range 96~127.
OPUS Sample Rate	Set Opus sampling rate, including opus-nb (8KHz) and opus-wb (16KHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length.
Enable voice activity detection	Whether enable voice activity detection.
Enable voice message dialing tone	When there is a new voice message, the phone will activate a special dial tone.

Fork spring reaction time	Configure the minimum response time, default is 200ms.
Enable the fork spring to generate Flash	Whether enable the fork spring to generate Flash.
EHS earphones	After activation, EHS earphones can be used for communication.
RTP Control Protocol(RTCP) Settings	
CNAME user	Set CNAME user.
CNAME host	Set CNAME host.
RTP Settings	
RTP keep alive	Hold the call and send the packet after 30s.
RTP Relay	Set up RTP relay.
Alert Info Ring Settings	
The values from notification information 1 to notification information 10	Set the value to specify the ringtone type.
Line	Set the corresponding line for incoming calls.
Ring Type	Type1-Type9.

12.11 Phone settings >> MCAST

The multicast function can be used to easily and conveniently send announcements to every member of the multicast. By setting the multicast key on the phone, multicast RTP streams can be sent to pre configured multicast addresses. By configuring a monitoring multicast address on the phone, listen and play the RTP stream sent by that multicast address.

Table 22 - Multicast parameters

Parameters	Description
Normal Call Priority	Define the priority of the active call, 1 is the highest priority, 10 is the lowest.
Enable Page Priority	Regardless of who calls in first, the call will be answered first by the multicast with higher priority
Name	Listened multicast server name
Host: port	Listened multicast server's multicast IP address and port.

12.12 Phone settings >> Action

Action URL

Note! Action urls are used for IPPBX systems to submit phone events.

12.13 Phone settings >> Time/Date

The user can configure the time Settings of the phone on this page.

Table 23 - Time&Date settings

Parameters	**Description**
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol.
Time Synchronized via DHCP	Enable time-sync through DHCP protocol.
Synchronize time using DHCPv6	Enable time synchronization using DHCPv6 protocol.
Primary Time Server	Set primary time server address.
Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device will try to connect to secondary time server to get time synchronization.
Time Zone	Select the time zone.
Resync Period	Set time synchronization period.
Dynamically Display Time	Time of re-synchronization with time server.
Date format	
12-Hour Clock	Set the time display in 12-hour mode.
Date Format	Select the time/date display format.
Summer Time Settings	
Location	Choose your own location.
DST Set Type	Set DST type.
Manual time setting	Manually set the current time.

12.14 Phone settings >> Time Management

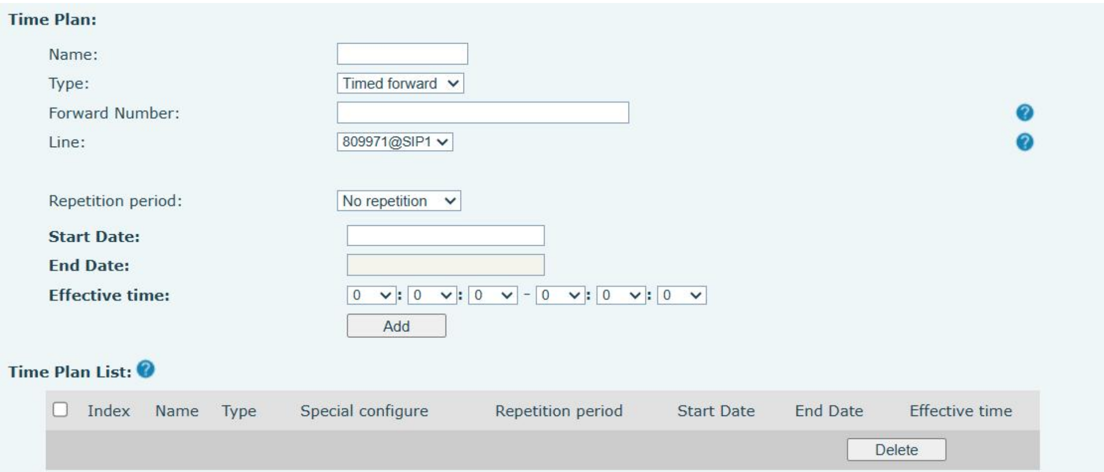
Time Plan settings can be used to set a time point or time period. The time point is to execute a certain action at a certain time, and the time period is to execute a certain action during a certain period of time.

Table 24 - Time Management

configuration	value	description
Time management type	1: Scheduled restart 2: Timed upgrade 3: Timed Forward Rotation 4: Regularly modify configurations	Type, action executed at time point/time period
repetition cycle	0: Do not repeat 1: Daily	repetition type

	2: Weekly 3: Monthly	
Start date	Select calendar date	Select the start date for time management
End Date	Select calendar date	Select the end date for time management
Effective time	xx:xx:xx-xx:xx:xx	The effective time period for configuring time management function

When the Time Plan type is selected as timed forwarding, the webpage will prompt for the forwarding number and forwarding route, as shown in the figure.



Picture 70 - Time Management(1)

Forward number: Configure the forward number and forward it to the set time period.

Route: Forward to the designated route. When the route is set to a certain route, it only applies to that route.

1. Timed pre transfer rule:

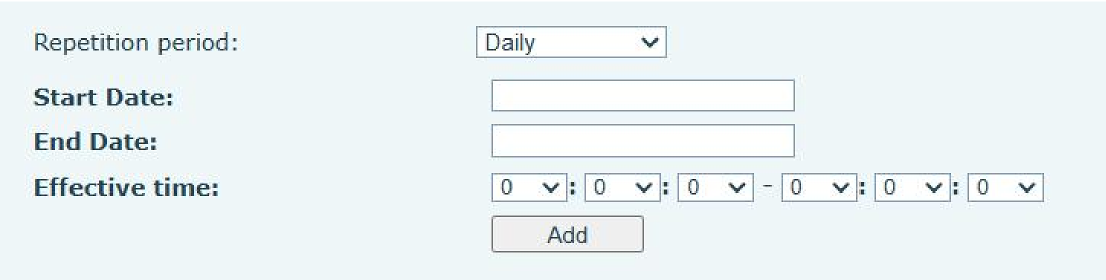
- When there is a forward turn below the line, use the forward turn number below the line; When there is no forwarding number in SIP line, if there is an incoming call within the time period set for scheduled forwarding, the phone will forward to the designated number for scheduled forwarding; When outside the time frame, do not move forward. Priority Line>Time Plan.
- The type of timed forward rotation is unconditional forward rotation.

12.14.1 Repeat cycle selection daily

Select daily for the repetition period, and enter the effective time in the input box in the date format of 00:00-23:59 at any time.

The first and third input boxes only allow the input of any integer from 00 to 23, and automatically add 0 before entering integers less than 10.

The second and fourth input boxes only allow the input of any integer from 00 to 59, and automatically add 0 before entering integers less than 10.

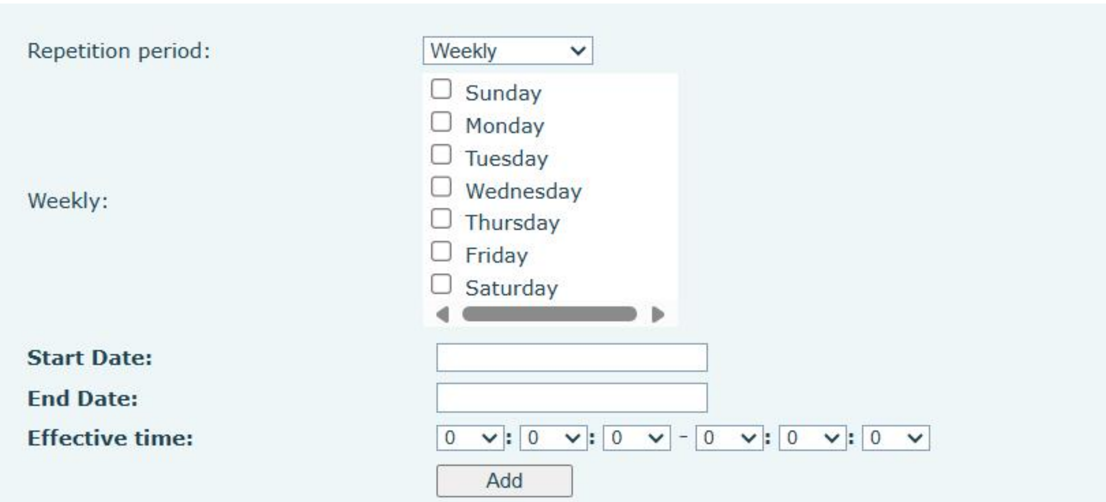


Picture 71 - Time Management(2)

12.14.2 Repeat cycle selection per week

Select the day of the week box, check it to take effect.

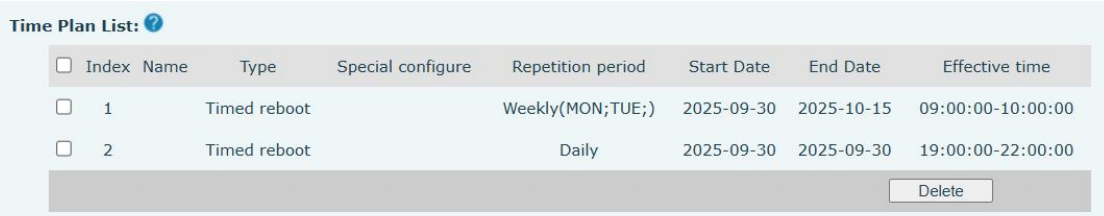
The final effective time is the combination of the day of the week and the set time.



Picture 72 - Time Management(3)

12.14.3 Time Management List

After submitting the configuration, a list of all submitted configurations will be displayed, sorted by week (day, Monday, Tuesday, etc.), and sorted by time if the week is the same (time from small to large). Function sequence: restart first and then upgrade.

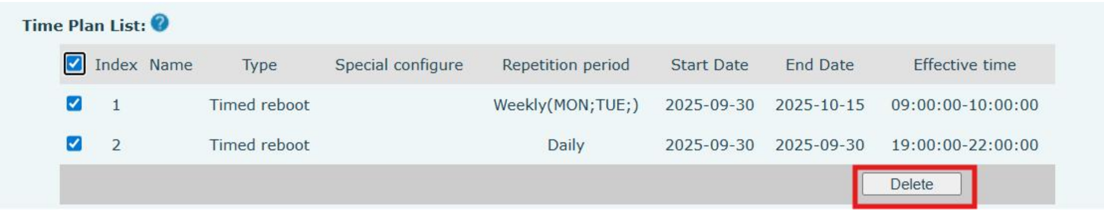


Picture 73 - Time Management(4)

12.14.4 Delete

Check the box before the serial number, and click to select all configuration items in the list.

Click 'delete' to select 'delete configuration' in the configuration list, which will become invalid after deletion.

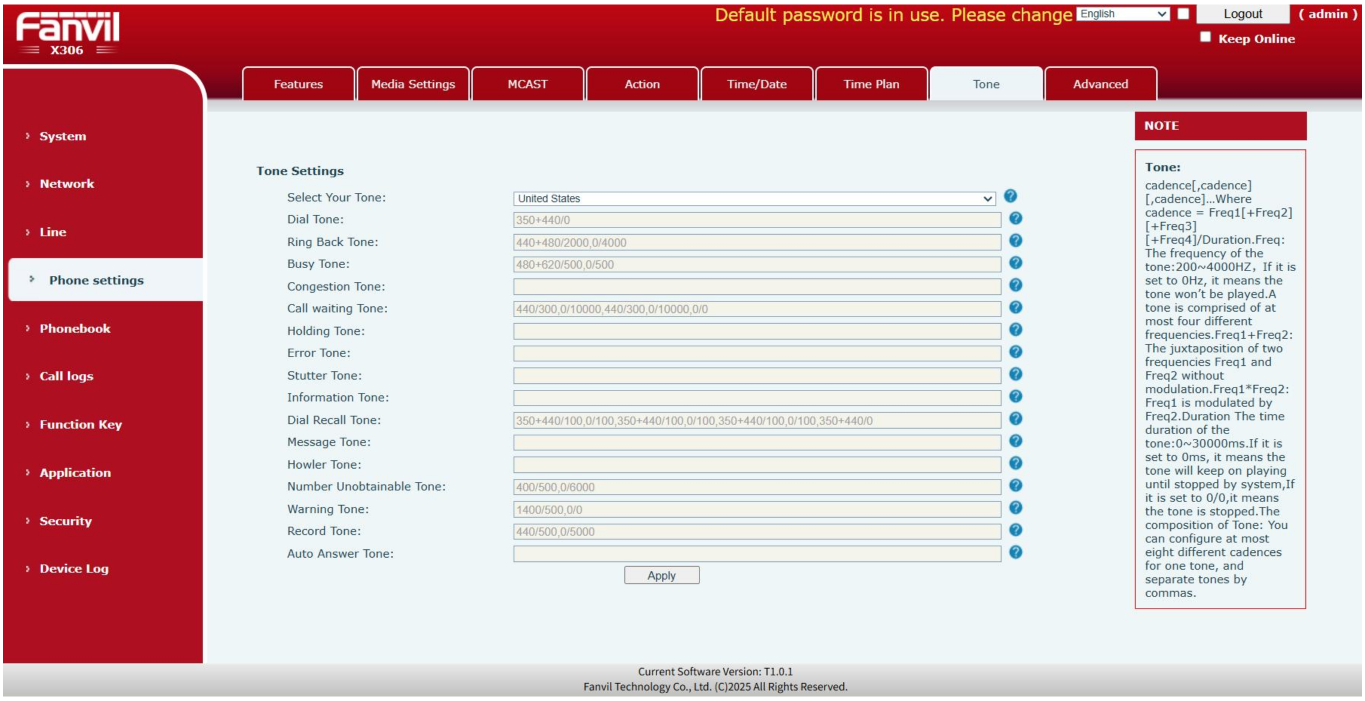


Picture 74 - Time Management(5)

12.15 Phone settings >> Tone

This page allows users to configure a phone prompt.

You can either select the country area or customize the area. If the area is selected, it will bring out the following information directly. If you choose to customize the area, you can modify the button tone, call back tone and other information.



Picture 75 - Tone settings on the web

12.16 Phone settings >> Advanced

User can configure the advanced configuration settings in this page.

- Screen Configuration.
- Enable Energy Saving
- UI display
- LCD Menu Password Settings.

The password is 123 by default.

- Keyboard Lock Settings.
- Configure Greeting Words

The greeting message will display on the top left corner of the LCD when the device is idle, which is limited to 12 characters. The default chars are 'VoIP Phone'.

12.17 Phonebook >> Contact

User can add, delete, or edit contacts in the phonebook in this page. User can browse the phonebook and sorting it by name, phones, or filter them out by group.

To add a new contact, user should enter contact's information and press "Add" button to add it.

To edit a contact, click on the checkbox in front of the contact, the contact information will be copied to the contact edit boxes, press "Modify" button after finished editing.

To delete one or multiple contacts, check on the checkbox in front of the contacts wished to be deleted and click the "Delete" button, or click the "Clear" button with selecting any contacts to clear the phonebook.

User can also add multiple contacts into a group by selecting the group in the dropdown options in front of "Add to Group" button at the bottom of the contact list, selecting contacts with checkbox and click "Add to Group" to add selected contacts into the group.

Similarly, user can select multiple users and add them into Blocked List by click "Add to Blocked List" button.

12.18 Phonebook >> Cloud phonebook

Cloud Phonebook

User can configure up to 4 cloud phonebooks. Each cloud phonebook must be configured with an URL where an XML phonebook is stored. The URL may be based on HTTP/HTTPs or FTP protocol with or without authentication. If authentication is required, user must configure the username and password.

To configure a cloud phonebook, the following information should be entered,

■Phonebook name (must)

■Phonebook URL (must)

■Access username (optional)

■Access password (optional)

LDAP Settings

The cloud phonebook allows user to retrieve contact list from a LDAP Server through LDAP protocols.

User must configure the LDAP Server information and Search Base to be able to use it on the device. If the LDAP server requests an authentication, user should also provide username and password.

To configure a LDAP phonebook, the following information should be entered,

■Display Title (must)

■LDAP Server Address (must)

■LDAP Server Port (must)

■Search Base (must)

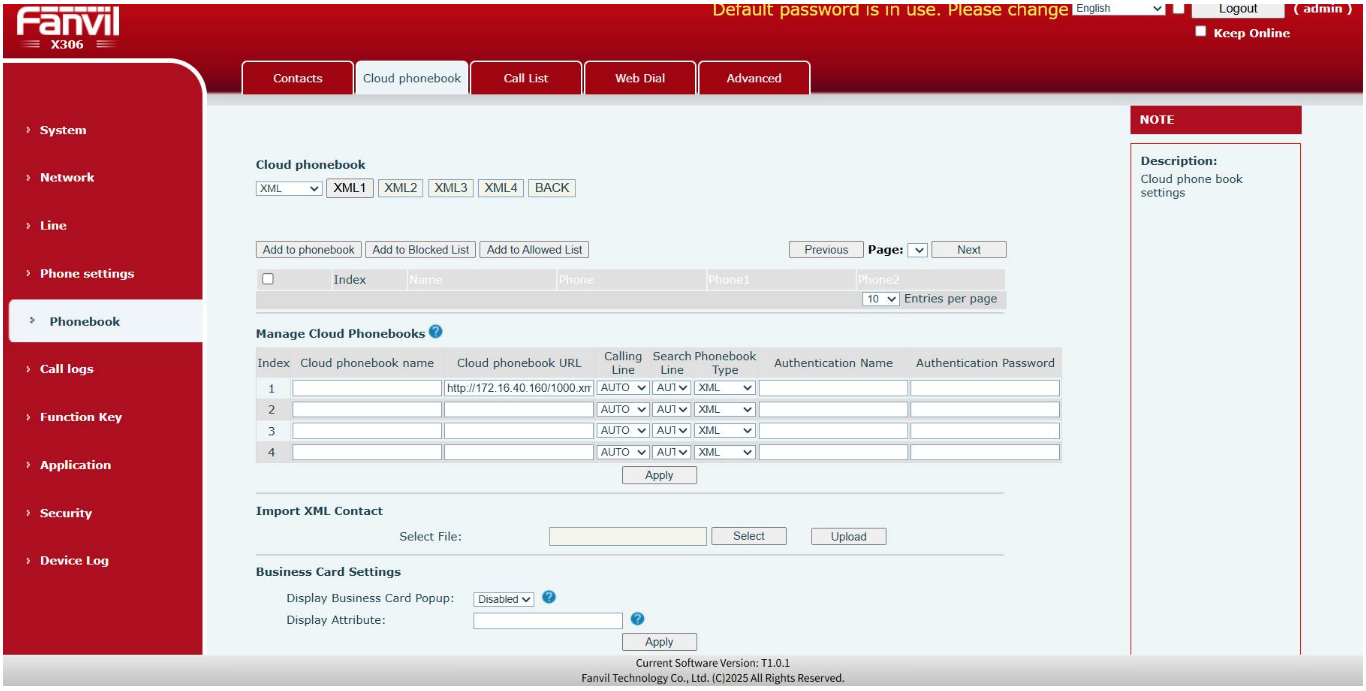
■Access username (optional)

■Access password (optional)

Web page preview

Phone page supports preview of Internet phone directory and contacts

- After setting up the XML Voip directory or LDAP,
- Select [**Phone book**] >> [**Cloud phone book**] >> [**Cloud phone book**] to select the type.
- Click the set XML/LDAP to download the contact for browsing.



Picture 76 - Web cloud phone book Settings

12.19 Phonebook >> Call List

■ Restricted Incoming Calls:

It is similar like a Blocked List. Add the number to the Blocked List, and the user will no longer receive calls from the stored number until the user removes it from the list.

Users can add specific Numbers to the Blocked List or add specific prefixes to the Blocked List to block calls with all Numbers with this prefix.

■ Allowed Incoming Calls:

When DND is enabled, the incoming call number can still be called.

- Restricted Outgoing Calls:

Adds a number that restricts outgoing calls and cannot be called until the number is removed from the table.

12.20 Phonebook >> Web Dial

Use web pages for call, reply, and hang up operations.

12.21 Phonebook >> Advanced

Users can export the local phone book in XML, CSV, and VCF format and save it on the local computer.

Users can also import contacts into the phone book in XML, CSV, and VCF formats.

Attention! If the user imports the same phone book repeatedly, the same contact will be ignored. If the name is the same but the number is different, the contact is created again.

Users can delete groups or add new groups on this page. Deleting a contact group does not delete contacts in that group.

12.22 CallLog

The user can browse the complete call record in this page. The call record can be sorted by time, call number, contact name or line, and the call record can be screened by call record type (incoming call, outgoing call, missed call, forward call).

The user can also save the number in the call record to his/her phone book or add it to the Blocked List/Allowed List

Users can also dial the web page by clicking on the number in the call log.

Users can also download call records conditionally and save them locally.

12.23 Function Key >> Side Key

Dsskey Long Press: Disable、Edit

The device provides 4 user-defined shortcuts that users can configure on a web page.

Table 25 - Function Key configuration

Parameters	Description
Memory Key	**BLF (NEW CALL/BXFE /AXFER)**: It is used to prompt user the state of the subscribe extension, and it can also pick up the subscribed number, which help user monitor the state of subscribe extension (idle, ringing, a call). There are 3 types for one-touch BLF transfer method. p.s. User should enter the pick-up number for specific BLF key to fulfill the pick-up operation. **Presence**: Compared to BLF, the Presence is also able to view whether the user is online.

	Note: You cannot subscribe the same number for BLF and Presence at the same time **Speed Dial**: You can call the number directly which you set. This feature is convenient for you to dial the number which you frequently dialed. **Intercom**: This feature allows the operator or the secretary to connect the phone quickly; it is widely used in office environments.
Line	It can be configured as a Line Key. User is able to make a call by pressing Line Key.
Key Event	User can select a key event as a shortcut to trigger. For example: MWI / DND / Release / Headset / Hold / etc.
DTMF	Allow users to easily dial or edit numbers
URL	Directly open a specific URL
BLF List key	
MCAST Paging	Configure multicast address and speech encoding. The user can initiate multicast by pressing this key
Action URL	Users can use specific URLs to perform basic call operations on the phone
MCAST Listening	Configure group listening address, user presses this button to perform multicast listening

12.24 Function Key >> Softkey

The User Settings mode and display style, display page.

Table 26 - Softkey configuration

Parameter	Description
Softkey Mode	
Softkey mode	Disabled and More，Default is Disabled
Softkey Style	
Softkey display style	Softkey Exit on Left or Right
Screen	
Call Dialer	Redial/2aB/Delete/Exit/Call Back/Dial/MWI/Local Contacts/Pickup/CallLog/Missed/Clear/In/Dialed/Pause/Headset/Audio/Remote XML/DSS Key
Conference	Hold/Split/End/Release/Mute/DSS Key/Headset
Desktop	CallLog/Menu/Local Contacts/DND/Prev Account/Next Account/Blocked List/Call Back/CallForward/Locked/Memo/ Missed/MWI/Dialed/Reboot/Redial/Remote XML/SMS/ Headset/Status/DSS Key/In
Divert Dialed	Redial/2aB/Delete/Exit/Forward/Local Contacts/CallLog /Clear/Missed/Dialed/Headset/Video/Audio/Remote XML

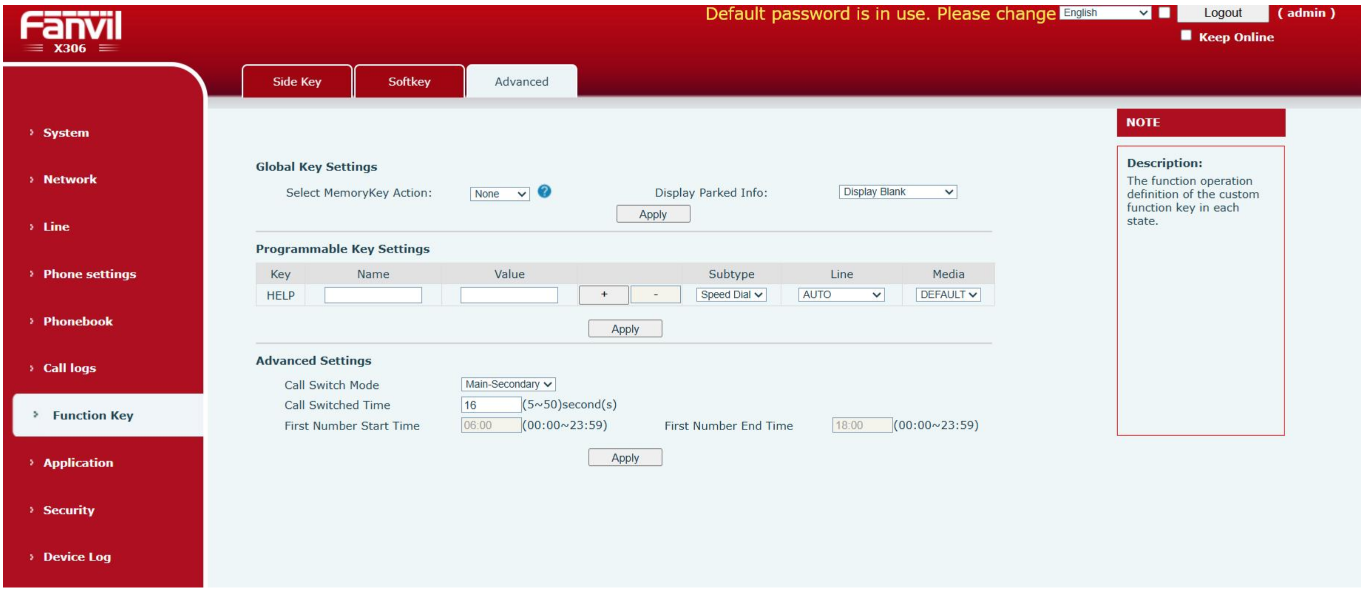
	/DSS Key
Ending	Redial/End/Headset/Release/DSS Key
Predictive Dialer	Dial/2aB/Delete/Exit/Call Back/Local Contacts/Redial /Pickup/MWI/Join/CallLog/Release/Missed/Pause/Dialed/ Headset/Video/Audio/Remote XML/DSS Key/In
Ringing	Answer/Forward/Reject/Mute/Release/Headset/Video/Audio/DSS key
Talking	Hold/End/Mute/Release/Local Contacts/Listen/CallLog Private/Headset/Audio/DSS Key
Trying	End/Release/Headset/DSS Key
Waiting	Hold/End/Answer/Forward/Mute/New cal/Reject/Release/Headset/Listen/Audio/DSS Key

12.25 Function Key >> Advanced

One key transfer: for example, set the memory key 4370. Press the memory key when talking with 4374 to decide whether to call 4370 or transfer 4374 to 4370.

Select memory key function: for example, the phone set the memory key value to 4370. When 4370 calls, press this key to hold the call or hang up.

- **Global Key Settings**



Picture 77 - Global Key Settings

- **Programmable key Settings**

The HELP button is a speed dial button that can be configured with a speed dial number name [name] and a number [value]. After pressing it, the corresponding number can be quickly called out

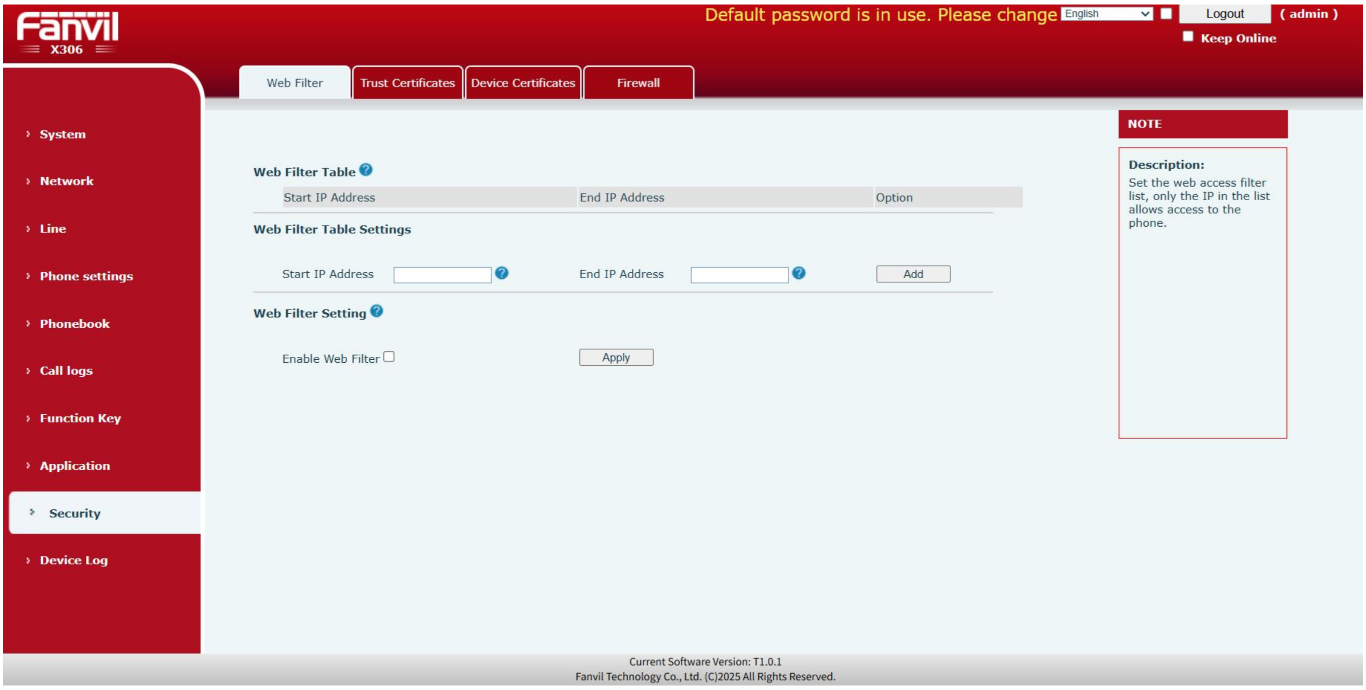
Note: When a one-way call is in progress, pressing the HELP button will hang up the current call and call *out the configuration number*

12.26 Application >> Manage Recording

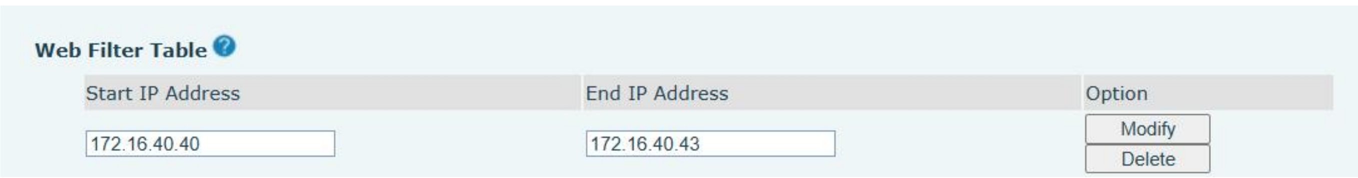
See [9.3 Record](#) for details of recording.

12.27 Security >> Web Filter

The user can set up a configuration management phone that allows only machines with a certain network segment IP access.



Picture 78 - Web Filter settings



Picture 79 - Web Filter Table

Add and remove IP segments that are accessible; Configure the starting IP address within the start IP, end the IP address within the end IP, and click [**Add**] to submit to take effect. A large network segment can be set, or it can be divided into several network segments to add. When deleting, select the initial IP of the network segment to be deleted from the drop-down menu, and then click [**Delete**] to take effect.

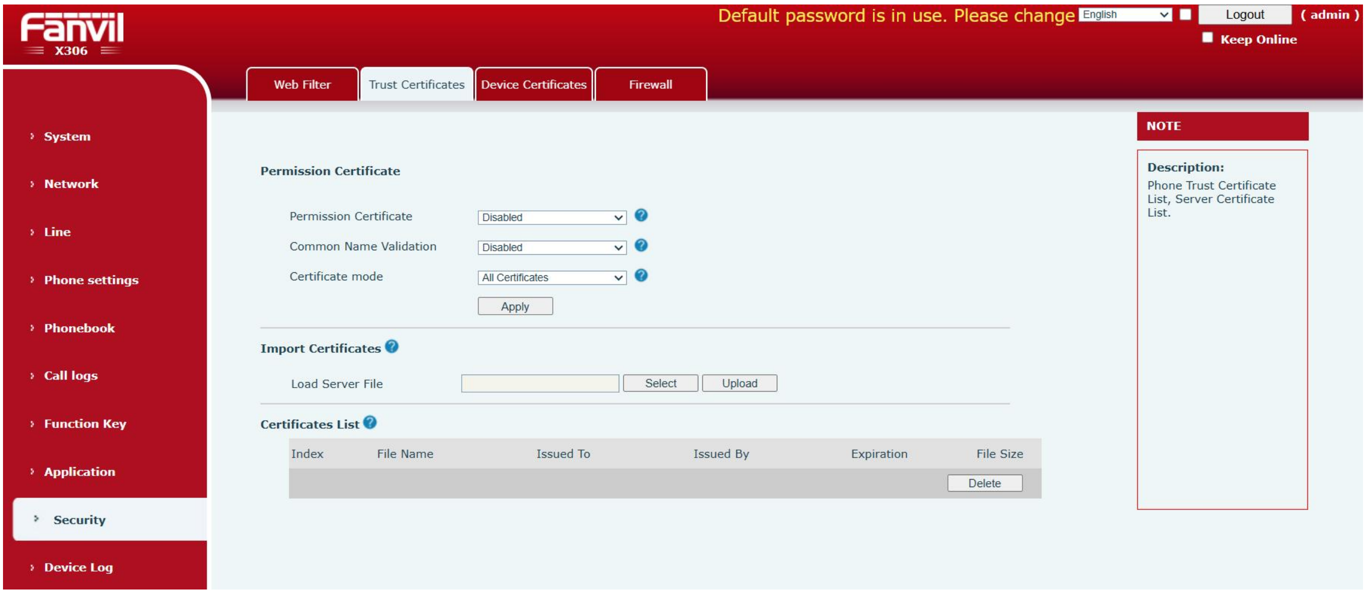
Enable web page filtering: configure enable/disable web page access filtering; Click the "apply" button to take effect.

Note: if the device you are accessing is in the same network segment as the phone, please do not configure the filter segment of the web page to be outside your own network segment, otherwise you will not be able to log in the web page.

12.28 Security >> Trust Certificates

Set whether to open license certificate and general name validation, select certificate module.

You can upload and delete uploaded certificates.

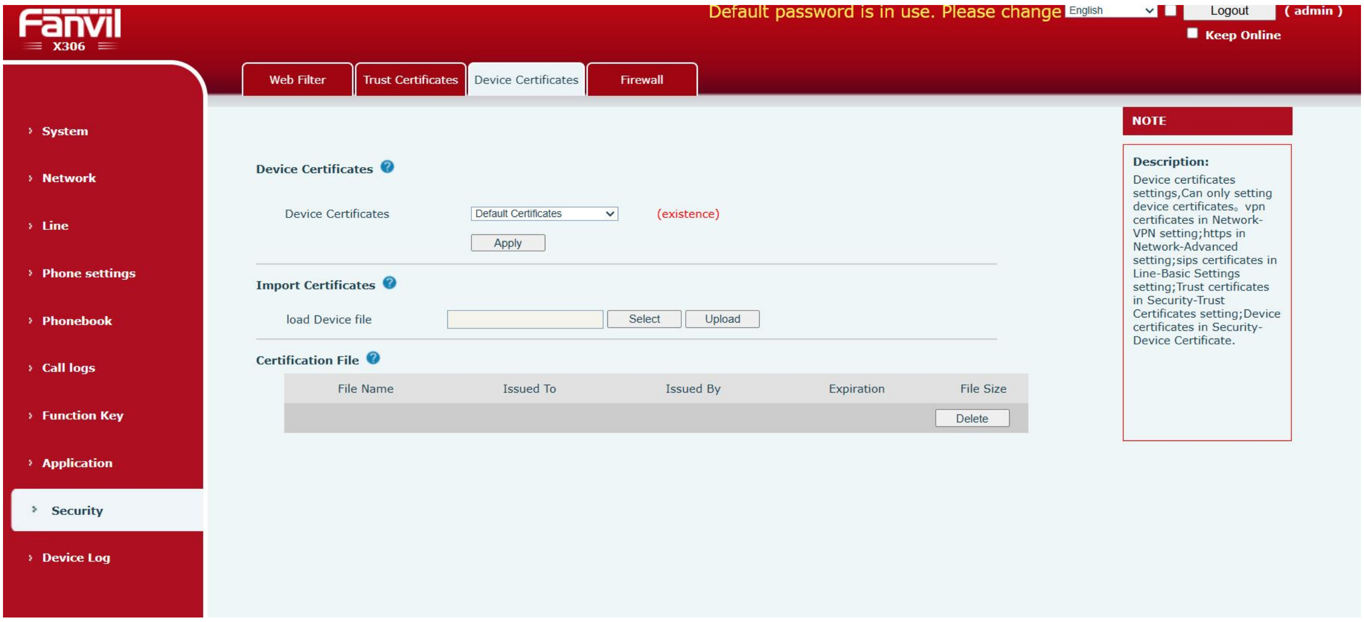


Picture 80 - Certificate of settings

12.29 Security >> Device Certificates

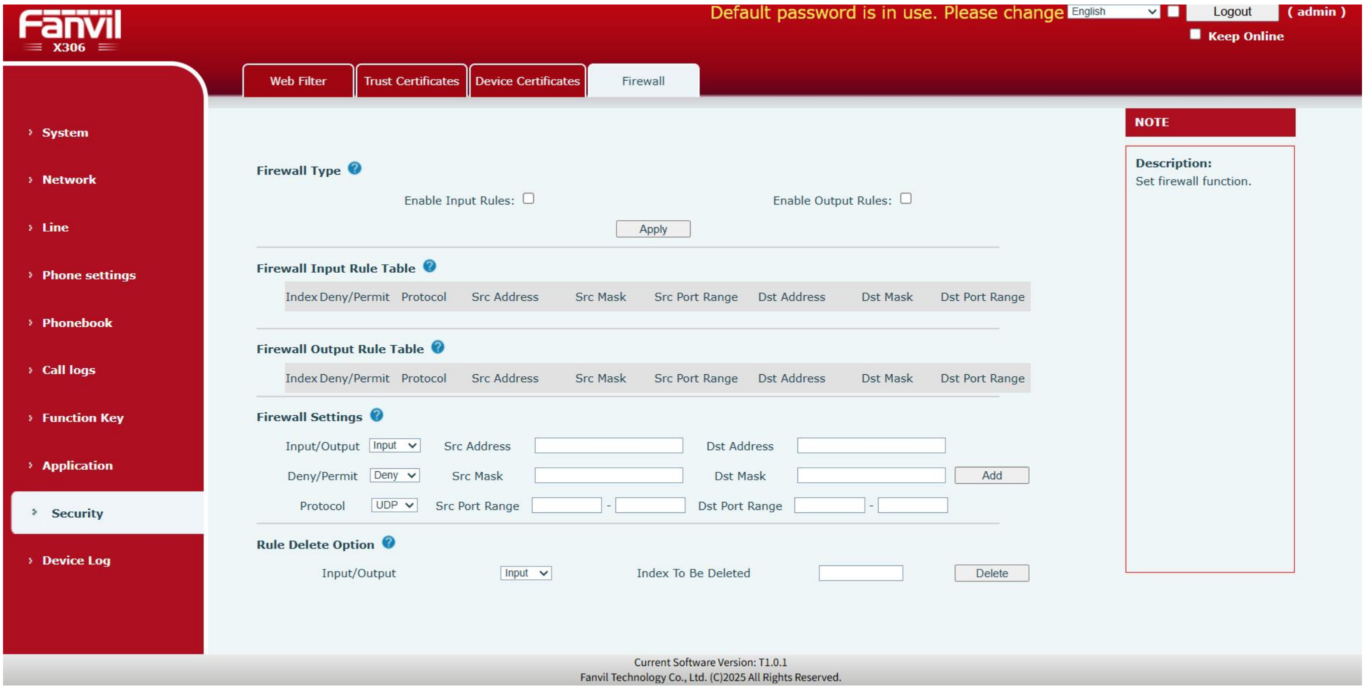
Select the device certificate as the default and custom certificate.

You can upload and delete uploaded certificates.



Picture 81 - Device certificate setting

12.30 Security >> Firewall



Picture 82 - Network firewall Settings

Through this page can set whether to enable the input, output firewall, at the same time can set the firewall input and output rules, using these Settings can prevent some malicious network access, or restrict internal users access to some resources of the external network, improve security.

Firewall rule set is a simple firewall module. This feature supports two types of rules: input rules and output rules. Each rule is assigned an ordinal number, allowing up to 10 for each rule.

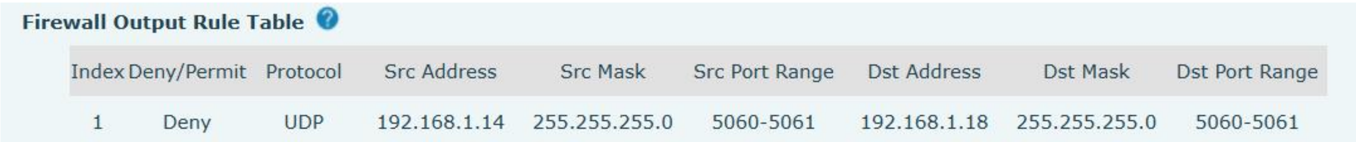
Considering the complexity of firewall Settings, the following is an example to illustrate:

Table 27 - Network Firewall

Parameter	Description
Enable Input Rules	Indicates that the input rule application is enabled.
Enable Output Rules	Indicates that the output rule application is enabled.
Input/Output	To select whether the currently added rule is an input or output rule.
Deny/Permit	To select whether the current rule configuration is disabled or allowed;
Protocol	There are four types of filtering protocols: TCP | UDP | ICMP | IP.
Src Port Range	Filter port range
Src Address	Source address can be host address, network address, or all addresses 0.0.0.0; It can also be a network address similar to *.*.*.0, such as: 192.168.1.0.
Dst Address	The destination address can be either the specific IP address or the full address 0.0.0.0; It can also be a network address similar to *.*.*.0, such as:

	192.168.1.0.
Src Mask	Is the source address mask. When configured as 255.255.255.255, it means that the host is specific. When set as 255.255.255.0, it means that a network segment is filtered.
Dst Mask	Is the destination address mask. When configured as 255.255.255.255, it means the specific host. When set as 255.255.255.0, it means that a network segment is filtered.

After setting, click [**Add**] and a new item will be added in the firewall input rule, as shown in the figure below:



Firewall Output Rule Table

Index	Deny/Permit	Protocol	Src Address	Src Mask	Src Port Range	Dst Address	Dst Mask	Dst Port Range
1	Deny	UDP	192.168.1.14	255.255.255.0	5060-5061	192.168.1.18	255.255.255.0	5060-5061

Picture 83 - Firewall Input rule table

Then select and click the button [Apply].

In this way, when the device is running: ping 192.168.1.18, the packet cannot be sent to 192.168.1.18 because the output rule is forbidden. However, other IP of the ping 192.168.1.0 network segment can still receive the response packet from the destination host normally.

Picture 84 - Delete firewall rules

Select the list you want to delete and click [**Delete**] to delete the selected list.

12.31 Device Log >> Device Log

You can grab the device log, and when you encounter an abnormal problem, you can send the log to the technician to locate the problem. See 13.4.1 Get log information.

13 Trouble Shooting

When the phone is not in normal use, the user can try the following methods to restore normal operation of the phone or collect relevant information and send a problem report to Fanvil technical support mailbox.

13.1 Get Device System Information

Users can get information by pressing the [**Menu**] >> [**Status**] option in the phone.The following information will be provided:
The network information
Equipment information (model, software and hardware version), etc.

13.2 Reboot Device

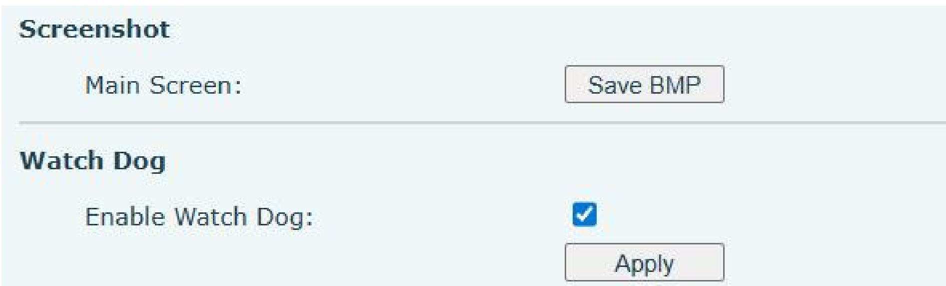
Users can reboot the device from soft-menu, [**Menu**] >> [**Basic**] >> [**Reboot System**], and confirm the action by [**OK**]. Or, simply remove the power supply and restore it again.

13.3 Reset Device to Factory Default

Reset Device to Factory Default will erase all user’s configuration, preference, database and profiles on the device and restore the device back to the state as factory default.
To perform a factory default reset, user should press [**Menu**] >> [**Advanced**] , and then input the password to enter the interface. Then choose [**Factory Reset**] and press [**Enter**], and confirm the action by [**OK**]. The device will be rebooted into a clean factory default state.

13.4 Screenshot

If there is a problem with the phone, the screenshot can help the technician locate the function and identify the problem. In order to obtain screen shots, log in the phone webpage [**System**] >> [**Tools**], and you can capture the pictures of the main screen and the secondary screen (you can capture them in the interface with problems).



Picture 85 - Screenshot

13.5 Network Packets Capture

Sometimes it is helpful to dump the network packets of the device for issue identification. To get the packets dump of the device, user needs to log in the device web portal, open page [**System**] >> [**Tools**] and click [**Start**] in “Network Packets Capture” section. A pop-up message will be prompt to ask user to save the capture file. User then should perform relevant operations such as activate/deactivate line or making phone calls and click [**Stop**] button in the web page when operation finished. The network packets of the device during the period have been dumped to the saved file.

Picture 86 - Web capture

User may examine the packets with a packet analyzer or send it to Fanvil support mailbox.

13.6 Get Log Information

Log information is helpful when encountering an exception problem. In order to get the log information of the phone, the user can log in the phone web page, open the page [**Device log**], click the [**Start**] button, follow the steps of the problem until the problem appears, and then click the [**End**] button, [**Save**] to local analysis or send the log to the technician to locate the problem.

13.7 Common Trouble Cases

Table 28 - Trouble Cases

Trouble Case	Solution
Device could not boot up	1. The device is powered by external power supply via power adapter or PoE switch. Please use standard power adapter provided by Fanvil or PoE switch met with the specification requirements and check if device is well connected to power source. 2. If you saw “POST MODE” on the device screen, the device system image has been damaged. Please contact location technical support to help you restore the phone system.
Device could not register to a service provider	1. Please check if device is well connected to the network. The network Ethernet cable should be connected to the [Network] port NOT the [PC] port. If the cable is not well connected to the network

	icon [WAN disconnected] will be flashing in the middle of the screen. 2. Please check if the device has an IP address. Check the system information, if the IP displays "Negotiating…", the device does not have an IP address. Please check if the network configurations is correct. 3. If network connection is fine, please check again your line configurations. If all configurations are correct, please kindly contact your service provider to get support, or follow the instructions in "13.5 Network Packet Capture" to get the network packet capture of registration process and send it to Fanvil support to analyze the issue.
No Audio or Poor Audio in Handset	1. Please check if Handset is connected to the correct Handset ( ) port NOT headset ( ) port. 2. The network bandwidth and delay may be not suitable for audio call at the moment.
Poor Audio or Low Volume in headset	1. There are two headset wire sequence in the market. Please use the headset provided by Fanvil, or consult Fanvil the wire sequence if you wish to use a third-party headset. 2. The network bandwidth and delay may be not suitable for audio call at the moment.
Audio is chopping at far-end in Hands-free speaker mode	This is usually due to loud volume feedback from speaker to microphone. Please lower down the speaker volume a little bit, the chopping will be gone.