



User Manual X210 & X210i

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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°C 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

The new Fanvil X210 and X210i IP Phone is a high-end enterprise desktop phone which comes with an intelligent DSS Key-mapping LCD to increase enterprise users' productivity at a cost-effective price.

The new DSS key design with 3 dynamic intelligent color displays can replace the traditional expansion board function. The main screen of the device smart display can dynamically display 10 Side DSS keys that can be customized by the user, and the two secondary screens can dynamically display 3 pages. Each page can display the setting contents of 32 DSS keys, and a total of 106 DSS key mappings that can be customized by the user. Every DSS key has a LED indication in green, red color to reflect the key state. Page turning shortcut allows users to quickly switch to the specified page. X210 and X210i is the most economic choice for SMB office and enterprise supervisors.

Evolved from Fanvil's X6 enterprise IP phones, X210 pushes its high-end cost-effective enterprise IP phone to another level. X210 inherits all enterprise features from Fanvil's X-Series enterprise phones, such as HD voice in handset, headset, and full-duplex speakerphone modes, PoE, Fast/Gigabit Ethernet, QoS, secure transmission, auto-provisioning, and more.

X210i is a visualization paging console phone for industry customer. It is equipped with a gooseneck microphone and supports HD hands-free calling. With intelligent programmable DSS buttons, you can set up a one-click call function to improve communication efficiency. It is compatible with the standard SIP protocol and can be used as a monitoring center or host for office manager with functions such as make calls for external & internal phones, two-way intercom, monitoring, and broadcasting. The X210i improves the management efficiency and emergency response capabilities.

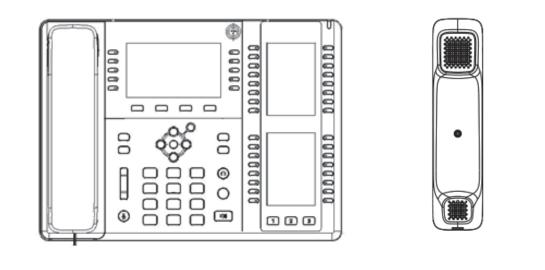
X210 and X210i is a great office productivity appliance for enterprise users. The old DSS key label is inconvenient and not environmental friendly. X210's intelligent DSS Key-mapping LCD provides users the flexibility to change DSS key definition and display through easy configuration. Meanwhile, with its intelligent design of the DSS key/LCD, it can be multiplied as expansion modules to save space and cost. X210 and X210i will provide the best user experience to advance enterprise users."

In order to help some users who are interested to read every detail of the product, this user manual is provided as a user's reference guide. Still, the document might not be up to date with the newly release software, so please kindly download updated user manual from Fanvil website, or contact with Fanvil support if you have any question using X210 and X210i.

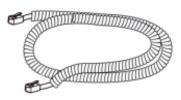


4.2 Packing Contents

4.2.1 Packing Contents



Phone



Receiver cable



Power adapter (Optional)



Stand



Handset

Network cable



Gooseneck Microphone (X210i only)



5 Install Guide

5.1 Use PoE or external Power Adapter

X210 and X210i, 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 power adapter will be used in priority and will switch to PoE power supply once it fails.

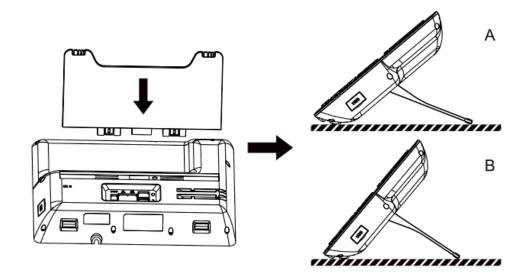
Please use the power adapter supplied by Fanvil and the PoE switch met the specifications to ensure the device work properly.



5.2 Desktop Installation

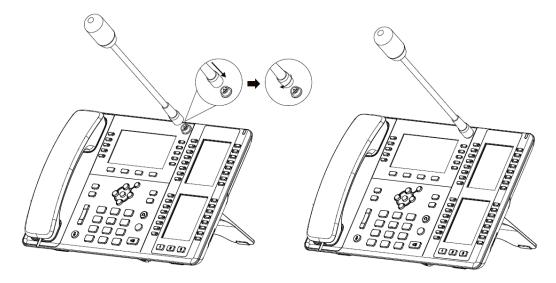
5.2.1 Desktop Installation

The device supports desktop use. If the phone is placed on the desktop, please follow the instructions in the picture below to install the phone.



Picture 1 - Device installation

After aligning the gooseneck microphone with the port, load it and tighten the nut.

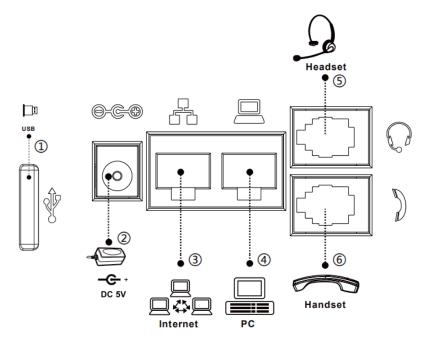


Picture 2 - Gooseneck MIC installation



Please connect power adapter, network, PC, handset, and headphone to the corresponding ports as described in below picture.

- ① USB port: connect USB device (U disk, WIFI adapter)
- ② Power port: connect the power adapter.
- ③ Network port: connecting local area network or internet.
- ④ PC port: the network port connect to the computer.
- (5) Headset port: connect headset.
- 6 Handset port: connect IP Phone handset.



Picture 3 - Connecting to the Device



6 Appendix Table

6.1 Appendix I - Icon

(-(Transfer
ø.	Hold
+	Volume up
_	Volume down
Æ	Mute Microphone (During Call)
Ð	Return
m	Contact
Ŋ	MWI
0	Handset
0	Redial
II))	Hands-free (HF) speaker

Table 1 - Keypad Icons

Table 2 - Status Prompt and Notification Icons

>>>>	Call out
(to	Call in
	Call Hold
"¥	Network Disconnected
控	Open VLAN
	Open VPN
溄	Keypad Locked
	Missed calls
	SMS
_	New voice message waiting



	Do-Not-Disturb activated on Phone
	Do-Not-Disturb inactivated on Phone
Ç	Call forward activated
A	Auto-answering activated
	Hands-free (HF) Mode
	Headphone (HP) Mode
2	Handset (HS) Mode
<u>v</u>	Mute Microphone
0	The Voice quality of calling
ô	The Voice encryption of calling
	Connecting WIFI
*	Open Bluetooth
(<u>1</u>))	Open SIP Hotline

Table 3 - DSSkey Icons

Function Icon	Sidekey Icon	Translate	Instruction
G	4ª	BLF/NEW CALL	The new call
Q	2	BLF/BXFER	Blind transfer
8	er er	BLF/AXFER	Attend transfer



		BLF/CONF	Conference
	۷.	BLF/DTMF	BLF/DTMF
0	4	Presence	Presence
9	9	MWI	Voice message
	J	Speed Dial	Speed Dial
Θ		Intercom	Intercom
0	C	Call Park	Call Park
G	¢.	Call forward	Call forward
0	ğ	Key Event	Function key
e	Ø	URL/Action URL	Network function key
		BLF List	BLF List
C	4	Multicast	Multicast
	Ι	Memory Key None	Memory Key subtype None
Ø	ð	None	Undefined DSS function key
	1	Line	SIP Line
	ili	DTMF	DTMF



6.2 Appendix II - Keyboard character query table

Mode Icon	Text Mode	Key Button	Characters Of Each Press
123	Numeric	1	1
120		2	2
		3	3
		4	4
		5	5
		6	6
		7	7
		8	8
		9	9
		0	0
		*	*
		#	#
	Lower Case	1	@:;()<>
abc	Alphabets	2	abc
		3	def
		4	ghi
		5	jkl
		6	m n o
		7	pqrs
		8	tuv
		9	w x y z
		0	(space)
		*	.,*/+-:_=
		#	# ^!&\$%

Table 4 - Look-up Table of Characters



[apo]	Upper Case	1	@:;()<>
ABC	Alphabets	2	ABC
		3	DEF
		4	GHI
		5	JKL
		6	MNO
		7	PQRS
		8	TUV
		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
		*	.,*/+-:_=
		#	# ^!&\$%



6.3 Appendix III – LED Definition

Туре	LED Light	State
Line Key	Off	Line inactive
	Green On	Line ready (Registered)
	Green Blinking	Ringing
	Red Blinking	Line is trying to register
	Red Blinking	Line error (Registration failure)
	Red On	Dialing/Line in use (Talking)
	Yellow Blinking	Call holding
BLF	Green On	Subscription number is idle.
	Red On	Subscription number is busy.
	Red On	Subscription number is dialing.
	Off	Subscription number is unavailable.
Presence	Green On	Subscription number is idle.
	Red On	Subscription number is busy.
	Red On	Subscription number is dialing.
	Off	Subscription number is unavailable.
DND	Red On	Enable DND
	Off	Disable DND
MWI	Green Blinking	New voice message waiting
	Off	No new voice message

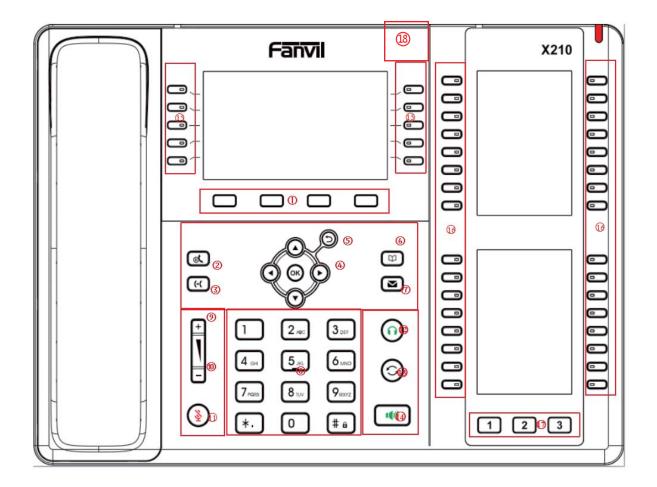
Table 5 - DSS KEY LED State



7 Introduction to the User

7.1 Instruction of Keypad

7.1.1 Instruction of Keypad



Picture 4 - Instruction of Keypad

The above picture shows the keypad layout of the device. Each key provides its own specific function. User should refer to the illustration in this section about the usage of each key and the description in this document about each function.

Table 6 - Instruction of Keypad

Number	The keypad	Instruction
Number	names	



1	Function	These four keys provide the corresponding menu function on
	Menu Key	the screen.
		Press the "Hold" key during the call, the user can hold the call,
2	Hold Key	and press it again to cancel the holding and restore the normal
		call state.
3	Transfer Key	By pressing the "Transfer" key, the user can transfer the current
		call to another number.
		The user can press the up/down navigation key to change the
		line or move the cursor in the screen list.On some Settings and
	Navigate and	text editing pages, the user can press the left/right navigation
(4)	OK Keys	key to change options or move the cursor in the screen list to
	OK Keys	the left/right.
		OK key:Default is equivalent to soft button confirmation, user
		can customize the function.
		The user can return to the previous menu by pressing the return key,
(5)	Return Key	and the function of dialing the phone or in the call is to reject or hang
		up.
	Contact Kov	Press the "Contact" key, the user can enter the address book
6	Contact Key	interface and select the contact person to call.
7	Voice Mail	Press the "voice mail" button, and the user enters the interface
	Key	of SMS and voice mail list.
		These 12 standard phone keys provide standard phone button
		functionality. At the same time, certain long key presses can be
8	DTMF Key	triggered to provide special functions.
		#- Long presses this key to open the keyboard lock
		configuration.
	Volume	In the standby state, ring and ring configuration interface, press
9	Down Key	this button to reduce the ring volume; Press this button to lower
		the volume on the call or volume adjustment screen.
	Volume Up	In the standby state, ring and ring configuration interface, press
10	Key	this button to increase the ring volume; Press this button to
	- ,	increase the volume on the call or volume adjustment screen.
11	Mute Key	During a call, the user can press this key to mute the microphone.
12	Headset Key	Users can press this key to open the headset channel



13	Redial Key	Press the Redial key to redial the last number dialed
(14)	Hands-free	The user can press this key to open the audio channel of the
	Key	speakerphone.
15	Side DSS	Long press the side DSS key to enter the function key setting
(5)	Key	interface and set the required functions
10	DSS	Long press the DSS shortcut key to enter the setting interface
	Shortcut Key	and set the required functions
17	Page Switch	Press the "page switch" key, the user can switch to the first,
	Key	second and third screen function key page.
	Gooseneck	
18	MIC Port	Connect gooseneck microphone
	(Only X210i)	

7.2 Using Handset / Hands-free Speaker / Headphone

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 headphone.

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.

Using Headphone

To use headphone, by default, user should headset button which is defined by DSS key to turn on the headphone. Same as handset and hands-free speaker, user can dial the number before or after headphone turned on.

Using Line Keys(Defined by DSS Key)

User can use line key to make or answer a call on specific line. If handset has been lifted, the audio channel will be opened in handset. Otherwise, the audio channel will be opened in hands-free speaker or headphone.



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, lock status, network connection status, etc.).

The lower half of the area is the function menu key, which is also the first layer of function menu keys, through which users can operate the phone.

Users can restore the phone to the default standby screen interface by picking up and dropping the handle.

The left and right part of the area shows default configuration of Side key, which dynamically displays the configuration of SIP information, message, headset, etc., which can be customized by users.

The icon description is described in 6.1 appendix I.

In some screens, there are many items or long text to be displayed which could not fit into the screen. They will be arranged in a list or multiple lines with a scroll bar. If user sees a scroll bar, user can use up/down navigator buttons to scroll the list. By long-pressed the navigator keys, user can scroll the list or items in a faster speed.



Features			15 : 34
1. Call Forwa	rd		
2. Auto Answ	er		
3. Call Waitin	g		
4. DND			
5. Intercom			
Return	Up	Down	ОК

Picture 6 - Scroll icon

7.4 Phone Status

The phone status includes the following information about the phone:

- Network Status:
 - VLAN ID

IPv4 or IPv6 status

- IP Address
- Network Mode
- The Phone Device Information:
 - Mac Address
 - Phone Mode
 - Hardware Version number
 - Software Version number
 - Phone Storage (RAM and ROM)
 - System Running Time
- SIP Account Information:
 SIP Account
 SIP Account Status (register / uncommitted / trying / time out)
- TR069 Connect Status (Displays only in the phone interface state)

The user can view the phone status through the phone interface and the web interface.

Phone interface : When the phone is in standby mode, press [Menu] >> [Status] and select the option to view the corresponding information, as shown in the figure:



 Network 	Phone	Account	TR069	
1. Vlan Id		None		
2. Mode		DHCP/IPv4		
3. IPv4		172.16.7.15	5	
Doturn				
Return				

Picture 7 - The Phone status

• WEB interface: Refer to <u>7.5 Web management</u> to log in the phone page, enter the [**System**] >> [Information] page, and check the phone status, as shown in the figure:

	Information	ccount Configurations	Upgrade	Auto Provision	Tools	Reboot
System						
etwork	System Information 💡					
LWOFK	Model:	X210				
	Hardware:	V1.0				
ne	Software:	1.8.5.5				
one settings	Uptime:	01:28	: 43			
	Network 🕜					
ionebook	WAN					
	Network mode:	DHCP				
Call logs	MAC:	MAC: 0c:38:3e:12				
	IPv4					
nction Key	IP:	172.16.	7.155			
	Subnet mask:	255.255	.255.0			
oplication	Default gateway:	172.16.	7.1			
curity	VQ status 🕜					
contry	Start time:		Stop time:			
evice Log	Local user:		Remote user			
INCE LOG	Local IP:		Remote IP:			
	Local Port:		Remote port	:		
	Local codec:		Remote code	ec:		
	Jitter:		JitterBufferM	ax:		
	Packets lost:		NetworkPack	etLossRate:		
	MOS-LQ:		MOS-CQ:			
	RoundTripDelay:		EndSystemD	elay:		

Picture 8 - 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 ¹⁸



check the IP address of the phone by pressing [Menu] >> [Status].

User:	
Password:	
Language:	English 🔻
	Logon

Picture 9 - 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 <u>11 Web configuration</u>.

7.6 Network Configurations

The device supports two kinds of network connection modes: wired network connection and wireless network connection. This section describes the wired network connection. For wireless network connection, refer to <u>10.5 wi-fi.</u>

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 find the phone function menu button [**Menu**] >> [**Advanced Settings**] >> [**Network**] >> [**Network**].

The default password for advanced Settings is "123".

NOTICE! If user saw a i '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 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 three 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, mask, gateway, and primary and secondary domains. This usually applies to some professional network user environments.

Please see 10.7.2.1 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 press the line key for a long time, or press the button in the function menu [Menu] >> [Advanced Settings] >> [Accounts] >> [Line 1] / [Line 2] / [Line 3] /.../ [Line 18] / [Line 19] / [Line 20] configuration, click ok 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.

SIP1				15:42			
1. Regis	tration	Enabl	Enabled (>				
2. Serve	er Address	192.1	68.7.1				
3. Auth.	User	6502	2				
4. Auth.	Password	****	****				
5. SIP U	ser	6502	2				
-		ľ					
Returr	ו Le	ft	Right	OK			

Picture 10 - Phone line SIP address and account information

SIP1		-	15:43
6. Display Name			
7. Server Port	5060)	
8. Proxy Address			
9. Proxy User			
10. Proxy Password			
Return 123	3	Delete	OK

Picture 11 - Phone display name and port

 WEB interface: After logging into the phone page, enter [Line] >> [SIP] and select SIP1/SIP2/SIP3/.../SIP18/SIP19/SIP20 for configuration, click apply to complete registration after configuration, as shown below:



	SIP SIP Hots	spot Dial Plan	Action Plan	Basic Settings	RTCP-XR	
› System						
> Network	Line 123456@S •					
	Register Settings >>					
> Line	Line Status:	Registered	Activ	ate:		
	Username:	123456	Authority	entication User:		
> Phone settings	Display name:		Authority	entication Password:		
	Realm:		Serve	er Name:		
> Phonebook						
	SIP Server 1:		SIP	Server 2:		
Call logs	Server Address:	172.16.1.2	Serve	er Address:		
	Server Port:	5060		er Port:	5060	
Function Key	Transport Protocol:	UDP V		sport Protocol:		
	Registration Expiration:	3600 second(s)		stration Expiration:		second(s)
Application			-			
	Proxy Server Address:		Back	up Proxy Server Address:		
Security	Proxy Server Port:	5060		up Proxy Server Port:	5060	
	Proxy User:		0			
Device Log	Proxy Password:		0			
	Basic Settings >>					
	Codecs Settings >> 🕜					
	Video Codecs >>					
	Advanced Settings >>					
	SIP Global Settings >>					
	3-11	Apply				

Picture 12 - Web SIP registration



8 Basic Function

8.1 Making Phone Calls

Default Line

The device provides twenty line services. If both lines are configured, user can make or receive phone calls on either line. If default line is configured by user, there will be a default line to be used for making outgoing call which is indicated on the top left corner. To change the default line, user can press left/right navigator buttons to switch between two lines. Enable or disable default line, user can press [Menu] >> [Features] >> [General] >> [Default Line] or configure from Web Interface (Web / PHONE / Features / Basic Settings).



Picture 13 - Default line

Dialing Methods

User can dial a number by,

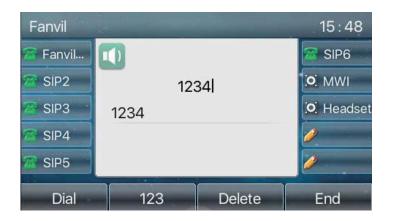
- Entering the number directly
- Selecting a phone number from phonebook contacts (Refer to <u>10.2.1 Local</u> <u>contacts</u>)
- Selecting a phone number from cloud phonebook contacts (Refer to <u>10.2.3</u> <u>Cloud Phone Book</u>)
- Selecting a phone number from call logs (Refer to <u>10.3 Call Log</u>)
- Redialing the last dialed number

Dialing Number then Opening Audio

To make a phone call, user can firstly dial a number by one of the above methods. When the dialed number is completed, user can press [**Dial**] button on the soft-menu, or press hand-free button to turn on the speaker or headphone, or lift the handset to call out with the current line, or user can press line key(Configured by DSS Keys) to call out with



specified line.



Picture 14 - Enable voice channel dialing

Opening Audio then Dialing the Number

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

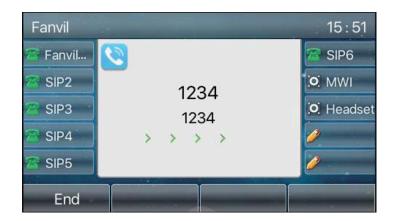


Picture 15 - Open the voice channel and dial the number



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 16 - 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 17 - Answering calls

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

Fanvil 🔕			_	16 : 01
🖀 Fanvil	0		HD	🕿 SIP6
🖀 SIP2	123	io: MWI		
🖉 SIP3		. Headset		
🖀 SIP4		356 @ :09 ©		2
🖀 SIP5				<i>i</i>
Hold	Xfer	Confer	ence	End

Picture 18 - Talking interface

Number	Name	Description		
1	Voice channel	The icon shows the voice channel mode being used.		
2	The current line	The line currently used by the phone.		
3 Calls to end		The name or number of the person on the other end of		
3		the call.		
(4)	Name on the other	Name on the other end		
4	end			
5	Call duration	The duration of a call after it has been established.		
6	Speech quality	Displays the current voice quality of the call.		
$\overline{\mathcal{O}}$	HD audio	Call using G.722 voice coding calls when displayed		
		HD voice icon.		

8.2.2 Make / Receive Second Call

The device can support up to two concurrent calls. When there is already a call established, user can still answer another incoming call on either lines or make a second call on either lines.

Second Incoming Call

When there is another incoming call during talking a phone call, this call will be waiting $^{\rm 26}$



for user to answer it. User will see the call message in the middle of current screen. The device will not be ringing but playing call waiting tone in the audio channel of the current call and the LED will be flashing in green. User can accept or reject the call as same as normal incoming call. When the waiting call is answered, the first call will be put on hold automatically.

Fanvil 🍘 Fanvil			16 : 14
SIP2	123	lH 356 00:3	O NAVA/I
🖀 SIP3 👘		356 00.3 80 (m	O Headse
SIP4		80 0	<u> </u>
CIP5			2
Xfer	Answer	Reject	End

Picture 19 - The second call interface

Second Outgoing Call

To make a second call, user may press [Xfer] / [Conf] button to make a new call on the default line or press the line key to make new call on specific line. Then dial the number the same way as making a phone call. Another alternative for making second call is to pressing DSS Keys dial out from the configured Keys (BLF/Speed Dial). When the user is making a second call with the above methods, the first call could be placed on hold manually first or will be put on hold automatically at second dial.

Switching between Two Calls

When there are two calls established, user will see a dual calls screen as the following picture.

🖀 Fanvil	Iu. 🚺	🖀 SIP6
SIP2	12356	MWI
SIP3	12356	Headse
SIP4	4380 4380 00:36	2
SIP5		2

Picture 20 - Two way calling



User can press up/down navigator buttons to switch screen page, and switch call focus by pressing [**Resume**] button.

Ending One Call

User may hang up the current talking call by closing the audio channel or press [End] button. The device will return to single call mode in holding state.

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, or put the receiver back and press the hands-free hook to 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.



	Features Media Settings MCAST Action Time/Date Tone Advanced						
› System							
> Network	Basic Settings >> Tone Settings >>						
› Line	DND Settings >>						
Phone settings	Intercom Settings >>						
> Phonebook	Redial Settings >> Redial Enter CallLog:						
> Call logs	Response Code Settings >> Password Dial Settings >>						
› Function Key	Power LED >>						
> Application	Notification Popups >> Apply						
> Security							
> Device Log							

Picture 21 - Redial set

8.5 Dial-up Query

Phone default to open the dial-up inquiry function, dial-out, enter two or more Numbers, dial the interface will automatically match call records, contacts in the number list, use the navigation key up and down keys can select the number, press the call out key or time out.

8.6 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 telephone interface or the webpage interface.

• Phone interface:

Press [Menu] >> [Features] >> [Auto Answer] button;

Press the button to select the line, use the left/right navigation key to turn on/off the auto answer option, and set the auto answer time to 5 seconds by default. After completion, press [**OK**] key to save;



The icon in the upper right corner of the screen A indicates that auto answer is enabled.

Auto Answer	16 : 1	9 Fanvil		16 : 19	
1. Fanvil		1. Auto Answer	Enabled	$\langle \rangle$	
2. SIP2		2. Auto Answer Delay	2. Auto Answer Delay 5		
3. SIP3					
4. SIP4					
5. SIP5					
Return Up	Down OK	Return Left	Right	ок	

Picture 22 - Line 1 enables auto-answering



Picture 23 - The line has enabled auto-answering

• 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.



		and and a		
	SIP SIP Hots	pot Dial Plan	Action Plan Basic Settings	RTCP-XR
› System				
› Network	Line Fanvil@SIF • Register Settings >>			
> Line	Basic Settings >>			
› Phone settings	Enable Auto Answering: Call Forward Unconditional:		Auto Answering Delay: Call Forward Number fo Unconditional:	5 (0~120)second(s) @
> Phonebook	Call Forward on Busy: Call Forward on No		Call Forward Number fo Busy: Call Forward Number fo	v
› Call logs	Answer: Call Forward Delay for No Answer:		No Answer: d(s) 🕜 Transfer Timeout:	0 second(s)
> Function Key	Conference Type:	Local V	Server Conference Number:	@
> Application	Subscribe For Voice Message: Voice Message Subscribe Period:	3600 (60~65535)se	Voice Message Number: cond(s) Enable Hotline:	0
> Security	Hotline Delay:	0 (0~9)second(s		
› Device Log	Dial Without Registered: DTMF Type: Request With Port: Use STUN:	AUTO V 0	Enable Missed Call Log: DTMF SIP INFO Mode: Enable DND: Use VPN:	 Ø Send 10/11 Ø Ø Ø
	Enable Failback: Failback Interval:	 	Signal Failback: Signal Retry Counts:	□ 2 3 (1~10) 2

Picture 24 - Web page to start auto-answering

8.7 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 phone interface: Under standby, press [Menu] >> [Basic Settings] >> [Keyboard Settings] >> [Function key] or [Keyboard Settings] >> [Soft function key] choose to set up the function keys, key type, type selection function name select callback function, input the callback key name, press [OK] key to save.



Picture 25 - Set the callback key on the phone



• Set the callback key through the web interface:

Log in the phone page, enter the [Function Key] >> [Side Key] or [Function Key] >> [Function 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:

	Funct	tion Key	Sid	e Key	Sof	tkey	Advanced					
System												
Network	Side I	Dsskey Sett	ings									
Line	Key	Туре		Name		Value	Subtype	i.	Line		PickUp Number	Icon Color
	F 1	Line	•				None	۲	210@SIP1	•		Default Green
Phone settings	F 2	Key Event	•				Call Back	*	AUTO	•		Default Green
r none sectings	F 3	Line	•				None	Ŧ	SIP3	•		Default Green
Phonebook	F 4	Line	T				None	۷	SIP4	Y		Default Green
Phonebook	F 5	Line	•				None	۷	SIP5	•		Default Green
	F 6	Line	•				None	۳	SIP6	Y		Default Green
Call logs	F 7	Key Event	۲				MWI	۳	AUTO	Y		Default Green
	F 8	Key Event	•				Headset	۳	AUTO	•		Default Green
Function Key	F 9	None	•				None	۲	AUTO	۲		Default Green
	F 10	None	•				None	۳	AUTO	•		Default Green
Application							Apply					
							Uhhia					
Security												

Picture 26 - Set the callback key on the web page

8.8 Mute

You can turn on mute mode during a call and turn off the microphone so that the local voice is not heard. Normally, mute mode is automatically turned off at the end of a call. You can also turn on mute on any screen (such as the free screen) and mute the ringtone automatically when there is an incoming call.

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

8.8.1 Mute the Call

During the conversation, press the mute button on the phone: the mute button on the phone will turn on the red light.
 Red mute icon is displayed in the call interface, as shown in the figure:



Fanvil			16 : 27
🖀 Fanvil		H	D 🖀 SIP6
O Redial	12	O MWI	
🖀 SIP3	12	. Headset	
🐻 SIP4		/	
🖀 SIP5		<u>₽</u>	Ø
Hold	Xfer	Conference	End

Picture 27 - Mute the call

• Cancel mute: press cancel mute on the phone again. The mute icon is no longer displayed in the call screen. The red light is off by mute button.

8.8.2 **Ringing Mute**

• Mute: press the mute button when the phone is in standby mode: Ψ

The top right corner of the phone shows the bell mute icon. Mute button red light is always on, when there is an incoming call, the phone will display the incoming call interface but will not ring.



Picture 28 - Ringing mute

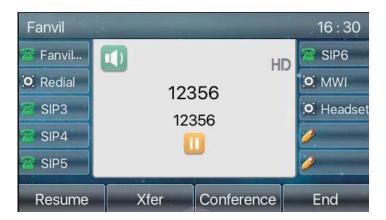
Cancel ring tone mute: On the standby or incoming call screen, press the mute button again or volume up transcel ring tone mute, no longer shows mute icon in

upper right corner after cancel The phone mute icon is off



8.9 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 29 - Call hold interface

8.10 DND

User may enable Do-Not-Disturb (DND) feature on the device to reject incoming calls (including call waiting). The DND can be enabled on line basis.

Enable/Disable phone all lines DND, Methods the following :

- Phone interface : Default standby mode ,
 - 1) Press [DND] button to enter the DND setting interface, select line or phone to

enable DND, the icon will become red $\fbox{}$.

2) Press [DND] button to enter the DND setting interface and disable DND, the

icon will be become blue



Fanvil	9 JAN	IWED	©‡⊗
🖀 Fanvil	16	: 31	🖀 SIP6
🖀 AUTO			. O MWI
🖀 SIP3	Fai		O Headset
🕾 SIP4		-	// · · · · ·
CIP5			* //
CallLog	Contact	DND	Menu

Picture 30 - Enable DND

If the user wishes to enable/disable the uninterrupted function on a specific line, the user can set the uninterrupted function on the page of configuring the line.

- 1) Press [Menu] >> [Features] >> [DND] button, Enter the [DND] editing interface.
- 2) Click the left/right navigation button to select the line to adjust the mode and state of "do not disturb", and then press the [**OK**] button to save.
- 3) The user will see the DND icon turn red, and the sip-line has enabled the mode of "DND".

DND			16 : 33				
1. DND Mode	Line	Line					
2. DND Timer	Disal	Disabled					
3. Line	SIP1	SIP1					
4. State	Disal	Disabled					
Return	Left	Right	ОК				

Picture 31 - DND setting interface

The user can also use the DND timer. After the setting, the DND function will automatically turn on and the DND icon will turn red in the time range.



DND	-			16 : 34		
1. DND Mode		Line		<		
2. DND Time	Enab	Enabled				
3. DND Start	15 : 00					
4. DND End Time		17 : 30				
5. Line	SIP1 🔹					
				1		
Return	Let	ft	Right	OK		

Picture 32 - DND timer

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

	1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 -		-	en ne		and and and a second	
	Features	Media Settings	MCAST	Action	Time/Date	Tone	Advanced
› System							
> Network	Basic Settings >						
› Line	DND Settings >						
Phone settings	DND Option Enable DND DND Start T	Timer:	Off • 15 • 0	•			
> Phonebook	DND End Tir	ne:	17 🔻 30	•			
> Call logs	Intercom Settin Redial Settings						
> Function Key	Response Code	Settings >>					
> Application	Password Dial S	ettings >>					
	Power LED >>						
> Security	Notification Pop	oups >>					
> Device Log				Apply			

Picture 33 - 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.



Line Fanvil@SIF▼ Register Settings >> Register Settings >> Basic Settings >> Auto Answering Delay: 5 0(0~12) Phone settings Call Forward on Busy: 0 Auto Answering Delay: 5 0(0~12) Phonebook Call Forward on Busy: 0 0 Call Forward Number for Busy: 0 <	SIP Hotspot Dial Plan Action Plan Basic Settings RTCP-XR	Dial Plan Acti	SIP SIP Hots	
Network Register Settings >> Basic Settings >> Basic Settings >> Basic Settings Auto Answering Delay: \$ (0~12 Phone settings Call Forward Question Phonebook Call Forward on Busy: @ Auto Answering Delay: \$ (0~12 Call Forward On Busy: @ Call Forward Number for Busy: Call Forward Number for Call Forward Number for Call Forward Number for No Answer: Call Forward Delay for No Call Forward Delay for No Answer: Call Forward Delay for No Server Conference Number: @ Server Conference Server Conference Number: @ Security Voice Message Subscribe Second(s) Enable Hotline: @ @ Security Dial Without Registered: @ (0~9)second(s) Hotline Number: @ @				System
> Line Basic Settings >> Phone settings Enable Auto Answering: Auto Answering Delay: Call Forward Number for Unconditional: Ourconditional: Call Forward on Busy: Call Forward Number for Unconditional: Call Forward Number for Busy: Call Forward Number for Busy: Call Forward Number for Call Forward Number for Call Forward Delay for No Call Forward Delay for No Call Forward Delay for No Server Conference Server Conference Subscribe For Voice Message: Voice Message Subscribe S6000 (60~65535)second(s) Enable Hotine: O Security 				> Network
Phone settings Call Forward Unconditional: Call Forward Number for Unconditional: Phonebook Call Forward on Nu Answer: Call Forward Number for Busy: Call Forward Number for Busy: Call Iogs Call Forward Delay for No Answer: Call Forward Delay for No Server Conference Call Forward Number for No Answer: Function Key Conference Type: Local Coll Conference Type: Local Coll Subscribe For Voice Message: Voice Message Subscribe Period: Coll Coll Coll Coll Coll Coll Coll Phonebook Coll Coll Coll Coll Coll				Line
Phonebook Call Forward on Busy: Image: Busy: Busy: Call Forward on No Call Forward on No Call Forward Number for No Answer: No Answer: Call Logs Call Forward Delay for No Image: Call Forward Delay for No Image: Call Forward Number for No Image: Call Forward Number for No Function Key Conference Type: Local Image: Call Image: Call Forward Number for No Image: Call Forward Number for No Image: Call Forward Number for No Application Subscribe For Voice Message Subscribe Period: Image: Call Image: Call Forward Number for No Image: Call Forward Number for No Security Hotline Delay: Image: Call Image: Call Forward Number for No Image: Call Forward Number for No Image: Call Forward Number for No Dial Without Registered: Image: Call Forward Number for No Image: Call Forward Number for No Image: Call Forward Number for No Dial Without Registered: Image: Call Forward Number for No Image: Call Forward Number for No Image: Call Forward Number for No Dial Without Registered: Image: Call Forward Number for No Image: Call Forward Number for No Image: Call Forward Number for No Dial Without Registered: Image: Call Forward Number for No Image: Call Forward Number for No Image: Call Forward Number for No	ard Call Forward Number for	0	Call Forward	Phone settings
Call logs Call Forward Delay for No 5 (0~120)second(s) @ Transfer Timeout: 0 second Function Key Conference Type: Local • @ Server Conference Number: Application Subscribe For Voice @ Voice Message Number: @ Voice Message Subscribe 9600 (60~65535)second(s) Enable Hotline: @ Security Hotline Delay: @ (0~9)second(s) @ Hotline Number: @	ard on Busy: Busy: V Busy: V Call Forward Number for A	Ĭ	Call Forward on No	Phonebook
Function Key Contrence Type: Local Number: Application Subscribe For Voice Message: Voice Message Subscribe Period: Voice Message Number: Security Hotline Delay: 0 (60~65535)second(s) Enable Hotline: 0 Dial Without Registered: 0 (0~9)second(s) Hotline Number: 0	ard Delay for No 5 (0~120)second(s) Transfer Timeout: 0 second(s) 0	(0~120)second(s)	Call Forward Delay for No	Call logs
Application Message: Image: Voice Message Number: Voice Message Subscribe 3600 (60~65535)second(s) Enable Hotline: Image: Security Hotline Delay: Image: Image: Image: Image: Image: Dial Without Registered: Image: I		al 🔻 🥝	Conference Type:	Function Key
Security Hotline Delay: 0 (0~9)second(s) Hotline Number: Dial Without Registered: Image: Comparison of the	Voice Message Number:		Message: Voice Message Subscribe	Application
			Hotline Delay:	Security
Device Log DTMF Type: AUTO Image: Comparison of the state of	e: AUTO • 0 DTMF SIP INFO Mode: Send 10/11 • 0 Vith Port: 0 0 Enable DND: 0 0	· · · · · · · · · · · · · · · · · · ·	DTMF Type: Request With Port:	Device Log

Picture 34 - Line DND

8.11 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,

- Unconditional Call Forward Forward any incoming call to the configured number.
- Call Forward on Busy When user is busy, the incoming call will be forwarded to the configured number.
- 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.
- Phone interface : Default standby mode
 - Press [Menu] >> [Features] >> [Call Forward] button, select the line by up/down navigation key, press [OK] button to set call forward..



Call Forward			16 : 38
1. Fanvil			
2. SIP2			
3. SIP3			
4. SIP4			
5. SIP5			
Return	Up	Down	ОК

Picture 35 - Select the line to set up call forwarding

 Select the call forward type by pressing the up/down navigation button. Click [OK] to configure call forwarding and delay time.

F	anvil			16 : 39
	1. Unconditi	ional		
	2. Busy Forv	vard		
	3. No Answe	er		
	Return	Up	Down	ОК

Picture 36 - Select call forward type

3) Select enable/disable by pressing the left/right navigation button.

Unconditiona	al		16 : 40
1. Unconditio	nal Enab	bled	()
2. Forward to	1234	1	
3. On Code			
4. Off Code			
Return	123	Delete	ОК

Picture 37 - Enable call forwarding and configure the call forwarding number

4) Browse the parameters set by the up/down navigation key and enter the required information. When finished, press the [**OK**] button to save the changes.



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

	SIP	SIP Hotspot	Dial Plan	Action Plan	Basic Settings	RTCP-:	XR
System							
Network	Line Fanvil	@SIF V					
	Register Settings	>>					
Line	Basic Settings >>	•					
	Enable Auto A	nswering: 🔲 🝘		Auto A	nswering Delay:	5	(0~120)secon
Phone settings	Call Forward				rward Number for		0
	Unconditional				ditional: rward Number for		
Phonebook	Call Forward o			Busy:	rward Number for		
	Call Forward of Answer:			No Ans			0
Call logs	Call Forward I Answer:	Delay for No 5	(0~120)secon	d(s) 🕜 Transf	er Timeout:	0	second(s) 🕜
	Conference Ty	pe: Local	. 0	Server	Conference		0
Function Key				NUMD	261		
	Subscribe For	Voice 🛛 🕜		Voice	lessage Number:		0
Application	Message: Voice Message	C. In all					
	Period:	3600	(60~65535)se		Hotline:		
Gecurity	Hotline Delay		(0~9)second(Number:		0
	Dial Without F	-			Missed Call Log:		
)evice Log	DTMF Type:	AUTO	• Ø		SIP INFO Mode:	Send 10/11	• 🕜
	Request With	Port: 🔲 🕜		Enable	DND:		
	Use STUN:	. 0		Use VF	'N:		
	Enable Failbac	k: 📝 🙆		Signal	Failback:		
	Failback Inter	val: 1800	second(s) 🕜	Signal	Retry Counts:	3	(1~10) 🕜

Picture 38 - Set call forward

8.12 Call Transfer

When the user is talking with a remote party and wish to transfer the call to another remote party, there are three way to transfer the call, blind transfer, attended transfer and Semi-Attended transfer.

- Blind transfer: Do not need to negotiate with the other side, directly transfer the call to the other side.
- Semi-Attended transfer.: When you hear the ring back, transfer the call to the other party.
- Attended transfer: When the caller answers the call, transfer the call to the caller.

Note ! For more transfer Settings, please refer to <u>12.6 Line >> Dial Plan</u>.

8.12.1 Blind transfer

During the call, the user presses the function menu button [**Transfer**] or the transfer button on the phone , Enter the number to transfer or press the contact button or the history button to select the number, press the transfer key again or blind transfer to



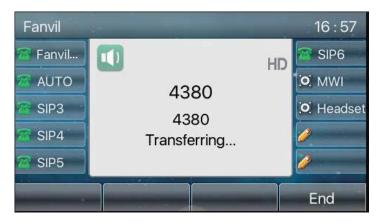
a third party. After the third party rings, the phone will show that the transfer is successful and hang up.



Picture 39 - Transfer interface

8.12.2 Semi-Attended transfer

During the call, the user presses the function menu button [transfer] or the transfer button on the phone to input the number to be transferred or press the contact button or the historical record button to select the number, and then press the call button. When the third party is not answered, press the transfer on the call interface to make the semi-attendance transfer or press the end button to cancel the semi-attendance transfer.



Picture 40 - Semi-Attended transfer

8.12.3 Attended transfer

Attendance transfer is also known as "courtesy mode", which is to transfer the call by calling the other party and waiting for the other party to answer the call.

Calling is the same procedure. In dual call mode, press the "transfer" button to transfer the first call to the second call.



Fanvil		17:05
🖉 Fanvil	UH 0.	🖀 SIP6
🕋 AUTO	4380 📶	O MWI
C SIP3		O Headset
🖀 SIP4	12356 00:04	2
🖀 SIP5		/
Hold	Xfer Conference	End

Picture 41 - Attended transfer

8.13 Call Waiting

• Enable call waiting: new calls can be accepted during a call.

• Disable call waiting: new calls will be automatically rejected and a busy tone will be prompted.

• Enable call waiting tone: when you receive a new call on the line, the tone will beep.

The user can enable/disable the call waiting function in the phone interface and the web interface.

Phone interface: Press [Menu] >> [Features] >> [Call waiting], the navigation key left/right button to enable/disable call waiting and call waiting tone. Press [Menu] >> [Features] >> [Call waiting], the navigation key left/right button to enable/disable call waiting and call waiting tone.

Call Waiting Set	tings		17:06
1. Call Waiting	Enab	led	\diamond
2. Waiting Tone	Enab	led	<>
i i i i i i i i i i i i i i i i i i i			
Return	Left	Right	OK

Picture 42 - Call waiting setting

• WEB interface: Enter [Phone Settings] >> [Features] >> [Basic Settings], enable/disable call waiting and call waiting tone.



	Features Media Settin	igs MCAST	Action	Time/Date	Tone	Advanced
› System						
> Network	Basic Settings >> Enable Call Waiting:	Ø	Er	nable Call Transfer:	@	
> Line	Semi-Attended Transfer: Enable Auto on Hook:	✓ Ø✓ Ø		nable 3-way Conference: uto HangUp Delay:	✓3	
> Phone settings	Ring From Headset:	Disabled V	Er	nable Auto Headset:	(0~30)second(s	:) 🕜
> Phonebook	Enable Silent Mode:	• •		isable Mute for Ring:	• •	
› Call logs	Enable Default Line: Default Ext Line:	Fanvil@SIP1 V	Ba	nable Auto Switch Line: an Outgoing:		
› Function Key	Default Ans Mode: Hide DTMF: Enable Restricted Incoming	Video	Er	efault Dial Mode: nable CallLog: nable Allowed Incoming	Video 🔻 🕜	
A CONTRACTOR AND A CONTRACT	List:	2 🕜		st:		

Picture 43 - Web call waiting setting

	Features	Media Settings	MCAST	Action	Time/Date	Tone	Advanced
› System							
› Network	Basic Settings >						
> Line	Tone Settings >: Enable Holdin	ng Tone:			Call Waiting Tone:		
> Phone settings	Play Dialing I		2	Play Ti	alking DTMF Tone:	2	
> Phonebook	Intercom Setting	gs >>					
> Call logs	Redial Settings						
> Function Key	Password Dial S						
	Power LED >>						
 Application 	Notification Pop	ups >>		Apply			
> Security							

Picture 44 - Web call waiting tone setting

8.14 Conference

8.14.1 Local Conference

To conduct local conference, the user needs to log in the webpage and enter [Line] >> [SIP] >> [Basic settings]. The meeting mode is set as local (the default is local mode), as shown in the figure:



	SIP SIP Hot	spot Dial Plan	Action Plan Basic Set	tings RTCP-XR	
› System					
> Network	Line Fanvil@SIF • Register Settings >>				
> Line	Basic Settings >>				
› Phone settings	Enable Auto Answering: Call Forward Unconditional:	• •	Auto Answering De Call Forward Numb Unconditional:	er for	-120)second(s) 🕜
> Phonebook	Call Forward on Busy: Call Forward on No		Call Forward Numb Busy: Call Forward Numb		0
> Call logs	Answer: Call Forward Delay for N Answer:	lo 5 (0~120)se	No Answer: cond(s) 🕐 Transfer Timeout:	0 sec	ond(s)
› Function Key	Conference Type:	Local 🔻 🥝	Server Conference Number:		0
> Application	Subscribe For Voice Message: Voice Message Subscrib		Voice Message Nur		0
> Security	Period: Hotline Delay: Dial Without Registered	0 (0~9)secor		□ 0 Log:	•
> Device Log	DTMF Type: Request With Port:	AUTO ▼ Ø	DTMF SIP INFO Mo Enable DND:		
	Use STUN:		Use VPN:		

Picture 45 - Local conference setting

Two ways to create a local conference:

 The device has two channels of communication. Press the conference button on the call interface. When selecting the conference number, select the other number that already exists.

Fanvil		17:19	Fanvil			17 : 20	Fanvil	-	_	17:21
Fanvil		🖀 SIP6	Tanvil	1. New Call		🖀 SIP6	🛜 Fanvil		00:04	C SIP6
AUTO	12356 📶	O MWI	😂 AUTO	2.12356		O. MWI	AUTO	-	4380	io: MWI
SIP3		O. Headset	SIP3			🗵 Headset	SIP3		4380	O Headset
SIP4	4380 00:07	1	SIP4			1	SIP4	0	12356	1
SIP5		0	SIP5			0	SIP5		12356	1
Hold	Xfer Conference	End	→ ок	Up	Down	Close	Hold		Split	End

Picture 46 - Local conference (1)

2) If the device has a call all the way, press the conference key in the call interface, enter the number to join the meeting and press the call; After the opposite end is answered, press the conference button again to set up the local tripartite conference:

Fanvil		17 : 17	Fanvil				17 : 18	Fanvil			17:18
Fanvil		at HD	🖀 Fanvil			.al HD	🖀 SIP6	🖀 Fanvil		00:02	SIP6
auto	4380	O MWI	C AUTO	0	4380		O MWI	🐴 AUTO		12356	IO MWI
SIP3	4380	• Headset	SIP3	0	4380 12356	00.00	O Headset	R SIP3		12356	O Headset
🕾 SIP4	00:57	A	🙈 SIP4		12356	00:03	2	SIP4	0	4380	A
🔛 SIP5		1	SIP5				<u>/</u>	🖀 SIP5		4380	A
Hold	Xfer Confe	rence End	\rightarrow Hold	Xfer	Confe	erence	End	Hold		Split	End

Picture 47 - Local conference (2)

Note: During the conference, press the split button to split the conference and press the



end button to end the call.

8.14.2 Network Conference

Users need server support for network conference.

Log in the web page, enter [Line] >> [SIP] >> [Basic settings], set the conference mode as server mode (default is local mode), set the server conference room number (please consult your system administrator), as shown in the figure:

Fa									
		SIP	SIP Hotspot	Dial Plan	Action Plan	Basic Settings	RTCP->	ĸĸ	
÷ 5)	ystem								
> N	etwork		il@SIF ▼						
•	Line	Register Setting Basic Settings >							
> Pl	hone settings	Enable Auto Call Forward Unconditiona			Call Fo	nswering Delay: orward Number for ditional:	5	(0~120)se	cond(s) 🕜
> Pl	honebook	Call Forward Call Forward Answer:			Busy:	orward Number for orward Number for			0
> Ci	all logs	Call Forward Answer:	Delay for No 5	(0~120)se		er Timeout:	0	second(s)	0
> Fi	unction Key	Conference 7	ype: Serve	r 🔻 🔞	Server Numb	Conference er:	1234		0
> A	pplication	Subscribe Fo Message: Voice Messag Period:				Message Number: • Hotline:			0
> S	ecurity	Hotline Dela	· · ·	(0~9)seco		e Number:			0
> D	evice Log	Dial Without DTMF Type:	Registered:			Missed Call Log: SIP INFO Mode:	Send 10/11	v 🕜	
		Request With Use STUN:			Enable Use VI	DND: PN:	 ○ ○		
		Enable Failback Inte		second(s)	-	Failback: Retry Counts:	3	(1~10) 🕜	

Picture 48 - Network conference

Method to join a network conference:

- Multi-party call number of network conference room and enter the password then all enter the conference room.
- The two phones have established common calls. Press the conference button to invite new members to the conference. Follow the voice prompt to operate.

Note: the upper limit of the number of participants in the network conference varies according to the server.

8.15 Call Park

Call park requires server support. Consult your system administrator for support. 44



When you are on the call, if it is not convenient to answer the phone at this time, you can press the configured park button to hold the call; After a successful park, you can resume the call by pressing the configured park button on other devices.

Set the call park button:

- Phone interface: long press a function key to enter the function key Settings interface, or through the [Menu] >> [Basic Settings] >> [Keyboard Settings] enter the function keys function key Settings interface, key function key type as memory and subtypes to call park, reside values for the server calls park number, set up corresponding SIP lines.
- WEB interface: log in the phone page, enter the [Function Key] >> [Function Key] page, select a DSSkey, set the function key type as memory key, the subtype as call park, and the value as the call park number of the server, and set the corresponding SIP line.

Dsskey			17 : 47				
1. Side Dsskey	1-1		<				
2. Type	Memo	Memory Key					
3. Line	Auto	Auto					
4. Subtype	Call Pa	Call Park 🔹					
5. Name							
	.eft	Right	ОК				

Picture 49 - Phone set call park

	Funct	ion Key	Side	Key	Softkey	Advance	4			
System										
Network	Side I	osskey Settir	ngs							
Line	Key	Туре		Name	Value	Subtyp	е	Line	PickUp Number	Icon Color
	F 1	Line	•			None	۲	Fanvil@SIP1 •		Default Green
Phone settings	F 2	Memory Key	•		1234	Call Park	۲	AUTO 🔻		Default Green
Those seconds	F 3	Line	•			None	٣	SIP3 V		Default Green
12200120000	F 4	Line	•			None	۳	SIP4 V		Default Green
Phonebook	F 5	Line	•			None	۳	SIP5 V		Default Green
	F 6	Line	•			None	۳	SIP6 T		Default Green
Call logs	F 7	Key Event	•			MWI	۲	AUTO 🔻		Default Green
	F 8	Key Event	•			Headset	۲	AUTO 🔻		Default Green
Function Key	F 9	None	•			None	۲	AUTO 🔻		Default Green
	F 10	None	•			None	-	AUTO V		Default Green

Picture 50 - WEB set call park



8.16 Pick Up

Pick up requires server support. Consult your system administrator for support.

You can use the Pick Up function to answer incoming calls from other users. The phone can pick up incoming calls by configuring DSSkey for BLF and setting the Pick Up code.

Phone interface: press [Menu] >> [Basic Settings] >> [Keyboard Settings] >> [DSS Key Settings], select the function key to set.

- Set the line, function key type as memory key, subtype as BLF/NEW CALL, set subscription number, and pick up code
 - Other phones call the subscription number, and the opposite end is in the incoming ring.
 - Press the DSS key to pick up the phone.
 - The caller picks up the call and speaks to it.
 WEB interface: Log in the phone webpage, enter the [Function Key] >>
 [Function Key] page, select a DSSkey, set the memory key type as memory key, the subtype as BLF/NEW CALL, and set the corresponding SIP line and pick up codes.

Dsskey			17 : 54
3. Line	SIP1		<>
4. Subtype	BLF/	New Call	<>
5. Name			
6. Tel			
7. Pickup Number	r *8		
-			
Return	123	Delete	ОК

Picture 51 - Phone pick up setting



	Fund	tion Key	Side Key	Softkey	Advanced			
› System								
> Network	Side	Dsskey Settin	gs					
Line	Key	Туре	Name	Value	Subtype	Line	PickUp Number	Icon Color
	F 1	Line	¥		None v	Fanvil@SIP1 v		Default Green
> Phone settings	F 2	Memory Key	T	1234	BLF/NEW CAI V	Fanvil@SIP1 •	*8	Default Green
r Filone settings	F 3	Line	¥		None v	SIP3 V		Default Green
	F 4	Line	T		None 🔻	SIP4 v		Default Green
> Phonebook	F 5	Line	T		None v	SIP5 V		Default Green
	F 6	Line	v		None 🔻	SIP6 v		Default Green
> Call logs	F 7	Key Event	•		MWI •	AUTO 🔻		Default Green
	F 8	Key Event	•		Headset •	AUTO 🔻		Default Green
Function Key	F 9	None	•		None 🔻	AUTO 🔻		Default Green
	F 10	None	•		None v	AUTO V		Default Green



8.17 Anonymous Call

8.17.1 Anonymous Call

The phone can set up anonymous calls to hide the calling number and the calling name.

- You can see anonymity in the context of [Menu] >> [Advanced Settings] >> [Accounts] >> [Advanced].
- The default is none, which is off, and RFC3323 and RFC3325 are optional.
- Select any one to open the anonymous call.



Picture 53 - Enable 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.



	SIP SIP Ho	tspot Dial Plan	Action Plan	Basic Settings	RTCP-XR	
		U		u		
System	SIP Encryption:	. 0	RTP E	ncryption(SRTP):	Disabled 🔻 🕜	
	Enable Session Timer:		Sessi	on Timeout:	0 se	cond(
Network	Enable BLF List:	. 0	BLF L	ist Number:		
	Response Single Codec	: 🔲 🕜	BLF S	erver:		
Line	Keep Alive Type:	UDP 🔻 🕜	Keep	Alive Interval:	30 se	cond(
	Keep Authentication:	. 🕜	Block	ng Anonymous Call:		
Phone settings	User Agent:		🕜 Speci	fic Server Type:		
	SIP Version:	RFC3261 T		mous Call Standard:	COMMON V	1
Phonebook	Local Port:	5060	Ring			
				el Call:		
Call logs	Enable user=phone: Auto TCP:			e PRACK:		

Picture 54 - Enable Anonymous web page call

4	All	In Out	Miss	Detail	18 : 02
	🔇 anonymous	anonymous	09 Jan 18:01	1. Number	anonymous
	X 12356	12356	09 Jan 18:00	2. Name	
	X 12356	12356	09 Jan 18:00	3. Line	1
	\$ 4380	4380	09 Jan 17:19	4. Time	09 Jan 18:01
	੯ 12356	12356	09 Jan 17:19	5. Duration	00:07
	Return Op	otion Delet	e Dial	Return EDia	l Option Dial

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

Picture 55 - Anonymous call log

8.17.2 Ban Anonymous Call

The device can be set to prohibit anonymous calls, that is anonymous calls to the number will be directly rejected.

- In the phone [Menu] >> [Features] >> [Ban anonymous call], click to enter and all SIP lines will be displayed.
- Click Softkey [Switch] or [<] [>] to switch the SIP line and enable anonymous call.



Ban Anonym	ious Call		18 : 05
1. Line	SIP1		<
2. State	Disal	oled	<>
Return	Enter	Switch	ок
Return	LING	Owner	OR

Picture 56 - Anonymous calls are not allowed on the phone

- 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.

	SIP SIP Hots	pot Dial Plan	Acti	on Plan Basic Settings	RTCP-XR	
› System	Enable Call Forward Unconditional:		0	Disable Call Forward Unconditional:		0
	Enable Call Forward on Busy:		0	Disable Call Forward on Busy	:	0
> Network	Enable Call Forward on No Answer:		0	Disable Call Forward on No Answer:		0
> Line	Enable Blocking Anonymous Call:		0	Disable Blocking Anonymous Call:		0
	Call Waiting On Code:		0	Call Waiting Off Code:		0
> Phone settings	Send Anonymous On Code:		0	Send Anonymous Off Code:		0
> Phonebook	SIP Encryption:	. 🕜		RTP Encryption(SRTP):	Disabled 🔻	
	Enable Session Timer:			Session Timeout:	0 s	econd(s) 🕜
> Call logs	Enable BLF List:	. 0		BLF List Number:		0
	Response Single Codec:	. 🕐		BLF Server:		0
> Function Key	Keep Alive Type:	UDP 🔻 🕜		Keep Alive Interval:	30 s	econd(s) 🕜
- Tuncton Key	Keep Authentication:			Blocking Anonymous Call:		

Picture 57 - Page Settings blocking anonymous call

8.18 Hotline

The device supports hotline dialing. After setting up the hotline dialing, directly pick up the handset, hands-free, earphone, etc., and the phone will automatically call according to the hotline delay time.

- In the phone [Menu] >> [Features] >> [Advanced] >> [Hotline], click to enter and all SIP lines will be displayed.
- Then set the hotline for each SIP line, which is off by default.



• Open the hotline, set the hotline number, set the delay time of the hotline.

Hot Line			18 : 11	Fanvil		-	18 : 11
1. Fanvil				1. Hot Line	Disab	led	$\langle \rangle$
2. SIP2				2. Number			
3. SIP3				3. Hot Line Delay	0		
4. SIP4							
5. SIP5			-				
Return	Up	Down	ОК	Return	Left	Right	OK

Picture 58 - Phone hotline setting interface

- 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.

	SIP SIP Hotsp	pot Dial Plan	Action Plan	Basic Settings	RTCP-XR	
> System						
Network	Line Fanvil@SIF • Register Settings >>					
Line	Basic Settings >>					
Phone settings	Enable Auto Answering: Call Forward Unconditional:	• •	Call Fo	nswering Delay: 5 rward Number for ditional:	(0~12	20)second(s
Phonebook	Call Forward on Busy: Call Forward on No Answer:	• •	Busy:	rward Number for		0
Call logs	Call Forward Delay for No Answer:	5 (0~120)		er Timeout: 0	secon	d(s) 🕜
Function Key	Conference Type:	Local 🔻 🕜	Server Numbe	Conference		0
Application	Subscribe For Voice Message:	• •	Voice N	lessage Number:		0
	Voice Message Subscribe Period:	3600 (60~655	35)second(s) Enable	Hotline:	0	
Security	Hotline Delay:	0 (0~9)sec	cond(s) 🕜 Hotline	Number:		0
	Dial Without Registered:	. 0	Enable	Missed Call Log:	2 🕜	
Device Log	DTMF Type:	AUTO 🔻 🕜	DTMF S	SIP INFO Mode:	Send 10/11 •	0
	Request With Port:	. 0	Enable	DND:	0	
	Use STUN:		Use VP	N:	2 🕜	
	Enable Failback:	?	Signal	Failback:	0	
	Failback Interval:	1800 second(s	s) 🕜 Signal	Retry Counts: 3	(1~10	0) 🕜

Picture 59 - Hotline set up on webpage

8.19 Emergency Call

The emergency call function is used to enable the keypad lock. Users can set the corresponding emergency call number on the phone. You can also call emergency



services when your phone is locked.

 Configure the emergency call number: log in the phone page, enter the [Phone Settings] >> [Function Settings]>> [Basic Settings]page, set up the emergency call code, if you need to set up more than one emergency call code, please use ", "to separate.

	Features Media Set	tings MCAST	Action	Time/Date	Tone	Advanced
› System						
> Network	Basic Settings >> Enable Call Waiting:	 Ø 	Ena	able Call Transfer:		
› Line	Semi-Attended Transfer: Enable Auto on Hook:	 Ø Ø 		able 3-way Conference: to HangUp Delay:		(c) Ø
Phone settings	Ring From Headset: Enable Silent Mode:	Disabled 🔻 🥝		able Auto Headset: able Mute for Ring:		(3)
> Phonebook	Enable Default Line:		Ena	able Auto Switch Line:		
> Call logs	Default Ext Line: Default Ans Mode:	Fanvil@SIP1 V 0		n Outgoing: fault Dial Mode:	Video 🔻 🕜	
• Function Key	Hide DTMF: Enable Restricted Incomi List:	Disabled 🔻 🥝 ng 🕑 🍘		able CallLog: able Allowed Incoming	✓✓✓✓✓	
> Application	Enable Restricted Outgoi List:	ng 🗹 🕜	Ena	able Country Code:		
Security	Country Code: Enable Number Privacy: Start Position:	0	Ma	a Code: tch Direction le Digits:	From left to righ	it 0~
› Device Log	Allow IP Call:	 Ø 		P IP Prefix:		
	Caller Name Priority:	LocalContact-NetContact-S	SIP DisplayName TEm	ergency Call Number:	110	
	Search path:	LDAP	D LD	AP Search:	LDAP 1 🔻 🕜	0

Picture 60 - Set up an emergency call number

2) When the phone set the keyboard lock, you can call the emergency call number without unlocking, as shown in the figure:



Picture 61 - Dial the emergency number



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] >> [Function 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. The pickup number is provided by the server. The specific use of reference 8.16 Pick up.

	Function Key S	de Key Softkey	Advanced				
› System							
> Network	Function Key Settings Dsskey Transfer Mo	de 🛛 Make a New C 🔻	Dsskey Home I	Page: None ▼			
› Line	Page1 Page2	Page3	Apply				
> Phone settings	Кеу Туре	Name Value	Subtype	Line	Media	PickUp Number	Icon Color
	DSS Key Memory Key V	1234	BLF/NEW CAI 🔻	Fanvil@SIP1 •	DEFAULT •		Default Green 🔻
> Phonebook	DSS Key Memory Key V	1234	BLF/BXFER •	Fanvil@SIP1 •	DEFAULT •		Default Green 🔻
> Call logs	DSS Key Memory Key 🔻	1234	BLF/AXFER 🔻	Fanvil@SIP1 •	DEFAULT 🔻		Default Green 🔻
Function Key	3 DSS Key Memory Key V	1234	BLF/CONF V	Fanvil@SIP1 •	DEFAULT V		Default Green 🔻
Application	4 DSS Key Memory Key V	1234	BLF/DTMF V	Fanvil@SIP1 •	DEFAULT 🔻		Default Green 🔻
Security	S DSS Key None ▼		None 🔻	AUTO 🔻	DEFAULT V		Default Green 🔻

Picture 62 - Web page configuration BLF function key

Phone interface: long press a function key to enter the function key Settings interface, or go to the [Menu] >> [Basic Settings] >> [Keyboard Settings] to enter the function key [Soft function key] to set settings interface, key function key types of memory, a subtype of BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF, BLF/DTMF, the values to be subscription number, and set up corresponding SIP lines.



Soft DSS Key	18 : 23						
1. Softkey	1-1	1-1					
2. Type	e Memory Key						
3. Line	SIP1	SIP1					
4. Subtype	BLF/New Call						
5. Name	5. Name						
Return	Left	Right	ОК				

Picture 63 - Phone configuration BLF function key

Subtype	Standby is described	Calling is described
BLF/NEW	Pressing the BLF key while standby to	When you press this BLF key while
CALL	dial the subscriber number.	talking to another user, you create a
		new call along with the subscribed
		number.
BLF/BXFE	Pressing the BLF key while standby to	When you press this BLF key while
R	dial the subscriber number.	talking to another user, you blind
		transfer the call to the subscribed
		number.
BLF/AXFE	Pressing the BLF key while standby to	When you press this BLF key while
R	dial the subscriber number.	talking to another user, you attendance
		transfer the call to the subscribed
		number.
BLF/Confer	Pressing the BLF key while standby to	When you press this BLF key while
ence	dial the subscriber number.	talking to another user, you invite the
		subscriber number to join the meeting.
BLF/DTMF	Pressing the BLF key while standby to	When the BLF key is pressed while
	dial the subscriber number.	talking to another user, the phone
		automatically sends the DTMF
		corresponding to the BLF key number.

 Table 8 - BLF Function key subtype parameter list

9.1.2 Use the BLF Function

The BLF, also known as a "busy light field," notifies the user of the status of the subscribed object and is used by the server to pick up the call. BLF helps you monitor the



other person's status (idle, ringing, talking, off).

BLF function:

- Monitor the status of subscribed phones.
- Call the subscribed number.
- Transfer calls/calls to the subscribed number.
- Pickup incoming calls from subscribed number.
- 1) Monitors the status of subscribed phones.

Configuration BLF function keys, when the subscription of the number of the state (idle, ringing, talking) is changed, the function key state of LED lights will have corresponding change, see <u>appendix III 6.3</u> to get to know each other under different status leds.

2) Call the subscribed number.

When the phone is in standby mode, press the configured BLF key to call out the subscribed number.

3) Transfer calls/calls to the subscribed number.

Refer to <u>Table 9.1.1-blf function key</u> subtype parameter list, the BLF key can be used for blind rotation, attention-rotation and semi-attention-rotation of the current call, and also can invite the subscribed number to join the call and send DTMF, etc.

4) Pickup incoming calls from subscribed phones.

When configuring BLF function key, configure the pickup number.

When be subscription number telephone ringing, refer to <u>appendix III 6.3 BLF LED</u> will flash a red light at this time. At this point, press the BLF button to answer the incoming call from the subscribed 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. If the state of the subscription object changes later, the corresponding led light state will be changed.

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.



	SIP S	IP Hotspot	Dial Plan	Actio	n Plan	Basic Settings	RTCP-XR	
› System	Codecs Settings >>	0						
> Network	Video Codecs >>							
	Advanced Settings >	>						
> Line	Use Feature Code	. 0	ia.					
	Enable DND:			0	DND Dis	abled:		
Phone settings	Enable Call Forwa Unconditional:	rd		0	Disable Uncondit	Call Forward		
	Enable Call Forwa	rd on		0		Call Forward on Busy:		
Phonebook	Busy: Enable Call Forwa	rd on				Call Forward on No		
	No Answer: Enable Blocking			v	Answer:	Blocking Anonymous		
Call logs	Anonymous Call:			0	Call:	BIOCKING ANONYMOUS		
	Call Waiting On Co	ode:		0	Call Wait	ting Off Code:		
Function Key	Send Anonymous Code:	On		0	Send An	onymous Off Code:		
Application	SIP Encryption:		0		RTP Enc	ryption(SRTP):	Disabled	v
	Enable Session Ti	ner: 🗌 🕜	ġ.		Session	Timeout:	0	second(s
Security	Enable BLF List:	. 0			BLF List	Number:		-
	Response Single C	Codec: 🔲 🕜			BLF Serv	ver:		
Device Log	Keep Alive Type:	UDP	• 0		Keep Ali	ve Interval:	30	second(s
	Keep Authenticati	on: 🗌 🕐			Blocking	Anonymous Call:		

Picture 64 - 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, call and transfer the corresponding number by pressing the BLF List key.

	Fund	tion Key	Side Key	Softkey	Adva	inced						
› System												
> Network		tion Key Setti Dsskey Transfe		a New C 🔻	and the second s	1000000000	Page: None	Ŧ				
> Line		Page1 Page	e2 Page3		Apply							
> Phone settings	Key DSS	Туре	Name	Value	Subtyp		Line		Media		PickUp Number	Icon Color
> Phonebook	1 DSS Key	BLF List Key			None None		AUTO		DEFAULT	۲ ۲		Default Green V
> Call logs	2 DSS Key 3	None 🔻			None	v	AUTO	•	DEFAULT	Y		Default Green 🔻
> Function Key	DSS Key 4	None •			None	٣	AUTO	Ŧ	DEFAULT	Y		Default Green 🔻

Picture 65 - BLF List number display

9.3 Record

The device supports recording during a call.



9.3.1 Local Record (USB flash disk)

Local recording is supported when USB flash drive is mounted.

When using local recording, it is necessary to start recording on the phone page [**Application**] >> [**Manage recording**], select the local type and set the voice coding. The webpage is as follows:

	Manage Recording			
› System				
> Network	Record Setting Enable Record:	2		
› Line	Record Type: Voice Codec:	Local T G729 T		
› Phone settings		Apply		
> Phonebook	Recording List	dex	File Name	File Size
→ Call logs				Delete
> Function Key				
> Application				
> Security				
> Device Log				

Picture 66 - WEB local recording

Local recording steps:

- Plug the U disk into the USB port of the phone, open the recording on the web page, and set the recording type as local recording.
- Set DSSkey type as key event and type as record in the phone/web interface.
- Set up one line call and press the recording key (set DSSkey).
- End the recording. End the call.

View local recording:

- Enter [Menu] >> [Application] >> [USB].
- Enter [**USB**] to view the recording file.
- Or enter the webpage [**Application**] under the [**Manage recording**] to view the recording file.

Listen to the record:

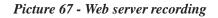


- Enter [Menu] >> [Application] >> [USB].
- Enter [**USB**] to view the recording file.
- Select the recording file that you want to listen to, and click the "play" button of Soft key to listen to the recording.

9.3.2 Server 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:

	Manage Recording			
› System				
› Network	Record Setting Enable Record:	ø		
› Line	Record Type: Voice Codec:	Network G729		
› Phone settings	Server Address:	172.16.7.39	Server Port:	10000
> Phonebook	Recording List	Apply		r
→ Call logs	Inc	dex	File Name	File Size
> Function Key				Delete
> Application				
> Security				
> Device Log				



Note: to be used with Fanvil recording software.

9.3.3 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.



	Manage Recording			
› System				
> Network	Record Setting Enable Record:	Ø		
> Line	Record Type:	Sip Info T		
> Phone settings	Recording List			
> Phonebook	Ind	ex	File Name	File Size
> Call logs				Delete
> Function Key				
> Application				
> Security				

Picture 68 - Web SIP info recording

9.4 Agent

Agent (Agent function) of the phone can be realized: when multiple people use a device for Agent services at different times, he or she can quickly register his or her SIP account on the same server. The Agent functions of the phone can be divided into Normal and Hotel Guest. The Hotel Guest mode requires server support.

Normal Mode:

Configure agent function: set a DSSkey as agent, press the function key or enter the [Menu] >> [Features] >> [Agent] to enter the agent page. The SIP server needs to be configured before the account can be configured.

Agent			18 : 31
1. Type	Norn	nal	
2. Number			
3. User			
4. Password			
5. Line	Line	1	<>
	1.0	D' Lu	
Return	Left	Right	Logon

Picture 69 - Configure the agent account in normal mode



Agent			18 : 31
1. Type	Hote	l Guest	<
2. Number			
3. Password			
4. Line	Line	1	\diamond
5. CallLog	Save	e All	\bigcirc
Return	123	Delete	Logon

Picture 70 - Configure the proxy account-hotel Guest mode

Parameter	Description
Normal mode	
Number	Set the proxy account number.
User	Set the proxy account number to verify the user name.
Password	Set the proxy account number to verify the password.
Line	Select the SIP line.
CallLog	Users can choose to save all types, or delete.
Hotel Guest mode	
Number	Set the proxy account number.
Password	Set the proxy account number to verify the password.
Line	Select the SIP line.
CallLog	Users can choose to save all types, or delete.
Statua	The user can select the status of the number, the optional
Status	status is: login, logout, invalid, valid, SMS.

Table	9	- Agency	mode
-------	---	----------	------

Using agent functions:

- When he phone has been configured on SIP server, fill in the correct number and user name password, click login and then the phone can be registered to the SIP server;
- 2) After registration, click logout and the phone can delete the user name and password, and log out of the SIP account.
- 3) Click Unregister and the phone retain the user name and password, and logs out of the SIP account.



Agent		18 : 32
1. Type	Normal	
2. Number	1234	
3. State	Logon	
4. CallLog	Save All	<>
		1
Return	Unregister	Logoff

Picture 71 - Agent logon page

9.5 Intercom

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

	Features Med	ia Settings	MCAST	Action	Time/Date	Tone	Advanced
› System							
> Network	Basic Settings >> Tone Settings >>						
> Line	DND Settings >>						
> Phone settings	Intercom Settings >> Enable Intercom:		0	Enable	Intercom Mute:		
> Phonebook	Enable Intercom To	one: 🕑	0	Enable	Intercom Barge:		
› Call logs	Redial Settings >> Response Code Settin	gs >>					
> Function Key	Password Dial Setting	5 >>					
> Application	Power LED >> Notification Popups >	>					
> Security				Apply			
› Device Log							

Picture 72 - Web Intercom configure

Table 10 - Intercom configure	Table	10 -	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	Enable mute mode during the intercom call



Mute			
Enable	Intercom	If the incoming call is intercom call, the phane plays the intercom tone	
Tone		If the incoming call is intercom call, the phone plays the intercom tone	
Enable	Intercom	Enable Intercom Barge by selecting it, the phone auto answers the intercom	
	Intercom	call during a call. If the current call is intercom call, the phone will reject the	
Barge		second intercom call	

9.6 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.

	Features Media	Settings MCAST	Action	Time/Date	Tone	Advanced
› System						
› Network	MCAST Settings Priority:	1	۲			
> Line	Enable Page Priority: Index/Priority		Name		Host:port	
> Phone settings	1 2	776			239.1.1.1:3366	
> Phonebook	3 4					
› Call logs	5 6 7					
› Function Key	7 8 9					
> Application	10	Ar	oply			
> Security						

Picture 73 - Multicast Settings Page

Parameters	Description	
Normal Call 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	



Multicast:

- Go to web page of [Function Key] >> [Function 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.

9.7 SCA (Shared Call Appearance)

Users need the support of server end to use SCA function. You can refer to http://www.fanvil.com/Uploads/Temp/download/20180920/5ba38181e4e4b.pdf

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

Fanvil						
	SIP SIP Hots	spot Dial Plan	Action Plan	Basic Settings	RTCP-XR	
› System						1.04
› Network		Created SCA ac		ne user name a imary account		d of the
> Line	Register Settings >> Line Status:	Registered		ivate:	Ø	
> Phone settings	Username: Display name: Realm:	1234 Fanvil	Ø Aut	hentication User: hentication Password: ver Name:	1234	0
> Phonebook	Broadsoft	Server address				
› Call logs	SIP Server 1: Server Address:	172.16.1.2	🕜 Ser	ver Address:		0
› Function Key	Server Port: Transport Protocol:	5060 UDP V 🔮	Tra	ver Port: nsport Protocol:	5060	
> Application	Registration Expiration: Proxy Server Address:	3600 second(s		istration Expiration: kup Proxy Server Addre	3600	second(s) 🕜
> Security	Proxy Server Address. Proxy Server Port: Proxy User:	5060		kup Proxy Server Addre	5060	0
> Device Log	Proxy Password:		0			
	Basic Settings >> Codecs Settings >> 🕢					

Picture 74 - 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.

	SIP SIP Hot	tspot Dial Plan	Action Plan	Basic Settings	RTCP-XR	
> System	SIP Encryption: Enable Session Timer:			P Encryption(SRTP): ssion Timeout:	Disabled v	0
> Network	Enable BLF List: Response Single Codec:	- 0 : 0		F List Number: F Server:		0
> Line	Keep Alive Type: Keep Authentication:	UDP v 🛛		ep Alive Interval: ocking Anonymous Call:	30 second(s)	0
Phone settings	User Agent:		g Sp	ecific Server Type:	BroadSoft V	
> Phonebook	SIP Version: Local Port:	RFC3261 V		onymous Call Standard: ng Type:	RFC3323 ▼ 🕜 Default ▼ 🕜	
› Call logs	Enable user=phone: Auto TCP:	• •	100	e Tel Call: able PRACK:		
	Enable Rport:					

Picture 75 - 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.

	SIP SIP Ho	tspot Dial Plan	Action Plan	Basic Settings	RTCP-XR	
› System	Enable Session Timer:	. 0		on Timeout:	0	second(s) 🕜
, system	Enable BLF List:			st Number:		•
THE REPORT OF	Response Single Codec			erver:		0
> Network	Keep Alive Type:	UDP 🔻 🕜	Keep	Alive Interval:	30	second(s) 🕜
(Contraction of the second sec	Keep Authentication:	. 🧭	Blocki	ng Anonymous Call:		
> Line						
	User Agent:		🕜 Specif	fic Server Type:	BroadSoft •	· 🕜
> Phone settings	SIP Version:	RFC3261 🔻 🕜	Anony	mous Call Standard:	RFC3323 V	0
	Local Port:	5060	🛛 🕜 🛛 Ring 1	Гуре:	Default 🔻	0
> Phonebook	Enable user=phone:	. 🧭	Use Te	el Call:		
	Auto TCP:	. 🧭	Enabl	e PRACK:		
> Call logs	Enable Rport:	Ø				
> Function Key	DNS Mode:	A v	Enable	e Long Contact:		
	Enable Strict Proxy:	Image:	Conve	ert URI:	I	
> Application	Use Quote in Display Name:		Enabl	e GRUU:	. 🧭	
	Sync Clock Time:	. 🧭	Enable	e Use Inactive Hold:		
> Security	Caller ID Header:	PAI-RPID-F 🔻 🕜	Use 1 waitin	82 Response for Call g:	. 🕜	
	Enable Feature Sync:	. 🧭	Enabl	e SCA:		
> Device Log	CallPark Number:		Ø Serve	r Expire:		
	TLS Version:	TLS 1.0 🔻 🕜	uaCST	FA Number:		
	Enable Click To Talk:		Enabl	e Chgport:		

Picture 76 - 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 <u>6.3 Appendix III - LED</u>.

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.

	Functi	an Kaw										
		оп кеу	Side Key	Softkey	Advanc	ed						
System												
Network		on Key Sett		e a New C 🔻	Dsskey Hon	ne F	Page: None	۲				
Line	F	Page1 Pag	ge2 Page3		Apply							
Phone settings	Key	Туре	Name	Value	Subtype		Line		Media		PickUp Number	Icon Color
		(ey Event	•		Private Hold	۳	AUTO	٣	DEFAULT	•		Default Green
Phonebook	1 DSS					_	J					
	Key N	lone	•		None	۲	AUTO	۷	DEFAULT	۲		Default Green
Call logs	2 DSS Key N	lone	•		None	۲	AUTO	۲	DEFAULT	•		Default Green
Function Key	3 DSS											
Function Key	Key N	lone	•		None	۲	AUTO	۲	DEFAULT	۲		Default Green
Application	DSS Key N	lone	•		None	•	AUTO	Ŧ	DEFAULT	•		Default Green
STREE AREA INTO A	5 DSS							10				
Security		lone	•		None	۲	AUTO	۲	DEFAULT	•		Default Green

Picture 77 - Set Private Hold Function Key

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

To facilitate viewing the call status of a group, configure lines whose DSS Key is SCA. The following table describes the LEDs of lines in different states.

State&Direction	Local	Remote
Idle	Off	Off
Seized	Steady green	Steady red
Progressing (outgoing call)	Steady green	Steady red
Alerting (incoming call)	Fast blinking green	Fast blinking green
Active	Steady green	Steady red
Public Held (hold)	Slow blinking green	Slow blinking red
Held-private (private hold)	Slow blinking yellow	Steady red
Bridge-active (Barge-in)	Steady green	Steady red
Bridge-held	Steady green	Steady red

Table 12 - LED Status of SCA

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.

9.8 Message

9.8.1 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 and display the icon of the new SMS on the standby screen interface.



Picture 78 - SMS icon



Send messages:

- Go to [Menu] >> [Message] >> [SMS].
- Users can create new messages, select lines and send numbers.
- After editing is complete, click Send.

View SMS:

- Use the navigation keys to select the standby icon [message
- After selecting, press the navigation key [OK] to enter the SMS inbox interface.
- Select the unread message and press [OK] to read the unread message.

Reply to SMS:

- Use the navigation keys to select the standby icon [Message].
- After selecting, press the navigation key [**OK**] to enter the SMS inbox interface.
- Select the message you want to reply to, select Softkey's [Reply], edit it, and click Send.

9.8.2 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 79 - New Voice Message Notification

Voice message icon

To listen to a voice message, the user must first configure the voicemail number. After



the voicemail number is configured, the user can retrieve the voicemail of the default line.

When the phone is in the default standby state,

- The Side Key is pre-installed with a voice message shortcut key [MWI] key.
- Press [MWI] to open the voice message configuration interface, and select the line to be configured by pressing the up/down navigation buttons.
- Press the [Edit] button to edit the voice message number. When finished, press the [OK] button to save the configuration.
- In the following picture, "17" in front of Fanvil line brackets represents unread voice messages, and "17" represents the total number of voice messages.

Voice Messa	age	09:40			
1. Fanvil (17/17)					
2. SIP2 (0/0)					
3. SIP3 (0/0)					
4. SIP4 (0/0)					
5. SIP5 (0/0					
Return	Edit	Play			

Picture 80 - Voice message interface

Fanvil			09:42
1. Voice Mail	Enab	led	<
2. Number	*97		
		_	
Return	123	Delete	ОК

Picture 81 - Configure voicemail number

9.9 SIP Hotspot

SIP hotspot is a simple but practical function. With simple configurations, the SIP hotspot function can implement group ringing. SIP accounts can be expanded.



Phone set functions as a SIP hotspot and other phone sets (B and C) function as SIP hotspot clients. When somebody calls phone set A, phone sets A, B, and C all ring. When any phone set answers the call, other phone sets stop ringing. The call can be answered by only one phone set. When B or C initiates a call, the SIP number registered by phone set A is the calling number.

	SIP SIP Hots	pot Dial Plan	Acti	on Plan Basic Settings	RTCP-XR	
› System						
> Network	Line Fanvil@SIF •					
> Line	Register Settings >> Line Status:	Registered		Activate:	Ø	
	Username:	901	0	Authentication User:	901	0
> Phone settings	Display name: Realm:	Fanvil	0	Authentication Password: Server Name:	•••••	0
> Phonebook						-
	SIP Server 1:			SIP Server 2:		
> Call logs	Server Address:	172.16.1.4	0	Server Address:		0
	Server Port:	5060	0	Server Port:	5060	0
Function Key	Transport Protocol:	UDP 🔻 🕜		Transport Protocol:	UDP 🔻 🕜	
> Application	Registration Expiration:	3600 second(s)	0	Registration Expiration:	3600 second(s)) 🕜
	Proxy Server Address:		0	Backup Proxy Server Address:		0
> Security	Proxy Server Port:	5060	0	Backup Proxy Server Port:	5060	0
	Proxy User:		0		Laction	-
> Device Log	Proxy Password:		0			

To set a SIP hotspot, register at least one SIP account.

Picture	82 -	Register	SIP	account
---------	------	----------	-----	---------

Table	13	- SIP	hotspot	Parameters
-------	----	-------	---------	-------------------

Parameters	Description
	If your phone is set to "SIP hotspot server",
	Device Table will display as Client Device Table
Device Table	which connected to your phone. If your phone is set to "SIP hotspot client",
	Device Table will display as Server Device Table
	which you can connect to.
SIP hotspot	
Enable hotspot	Set it to be Enable to enable the feature.
	Choose hotspot, phone will be a "SIP hotspot
Mode	server"; Choose Client, phone will be a "SIP
	hotspot Client"
Monitor Type	Either the Multicast or Broadcast is ok. If you want to limit the broadcast packets, you'd better use broadcast. But, if client choose broadcast,



	the SIP hotspot phone must be broadcast.
Monitor Address	The address of broadcast, hotspot server and
Monitor Address	hotspot client must be same.
Remote Port	Type the Remote port number.

Configure SIP hotspot server:

		- 12		
	SIP SIP Hotspot	Dial Plan Action Plan	Basic Settings R	TCP-XR
› System				
	Client Table			
> Network	IP	MAC		Alias Line
> Line	172.16.7.181	0c:38:3e:30:10:f6		1 1
> Line	SIP Hotspot Settings			
> Phone settings	Enable Hotspot:	Enabled T		0
	Mode:	Hotspot 🔻		0
> Phonebook	Monitor Type:	Broadcast 🔻		0
	Monitor Address:	224.0.2.0		0
> Call logs	Local Port:	16360		0
	Name:	SIP Hotspot		0
> Function Key	Line Settings			
	Line 1:	Enabled V		
Application	Line 2:	Enabled V		
	Line 3:	Enabled v		
> Security	Line 4:	Enabled v		
	Line 5:	Enabled v		
Device Log	Line 6:	Enabled v		

Picture 83 - SIP hotspot server configuration

Configure SIP hotspot client:

To set as a SIP hotspot client, no SIP account needs to be set. The Phone set will automatically obtain and configure a SIP account. On the SIP Hotspot tab page, set Mode to Client. The values of other options are the same as those of the hotspot.

	SIP SIP	Hotspot Dial Plan	Action Plan	Basic Settings	R	RTCP-XR	
› System							
	Hotspot Table						
> Network	IP	Server name	Online Status	Connection Status	Alias	Line	
> Line	172.16.7.181	SIP Hotspot	OnLine	Connected	1	0	Disconnect
	SIP Hotspot Settings						
› Phone settings	Enable Hotspot:	Enable	d 🔻				0
	Mode:	Client	•				0
> Phonebook	Monitor Type:	Broadd	ast 🔻				0
	Monitor Address:	224.0.2	.0				0
> Call logs	Local Port:	16360					0
	Name:	SIP Hot	spot				0
› Function Key	Line Settings						
	Line 1:	Enable	ed 🔻				
Application	Line 2:	Enable	ed 🔻				
	Line 3:	Enable	ed 🔻				
> Security	Line 4:	Enable	ed 🔻				

Picture 84 - SIP hotspot client configuration



As the hotspot server, the default extension number is 0. When the phone is used as the client, the extension number is increased from 1, you can view the extension number through the [**SIP Hotspot**] page.

Call extension number:

- The hotspot server and the client can dial each other through the extension number.
- For example, extension 1 dials extension 0.



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, go to [Menu] >> [Basic] >> [Language] Settings, as shown in the figure.

Langu	lage		-	15 : 51
	English			
0	简体中文			
0	繁體中文			
0	Русский			
0	Italiano			
Retu	urn	Up	Down	ОК

Picture 85 - 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:

Fanvil							English English 中文 繁建中文
	Information Account	Configurations Upgrade	Auto Provision	Tools	Reboot Phone		Pycoută Italiano Deutsch
> System						NOTE	Français גירית Español
Network	System Information 🕖 Model:	X210				Description: It shows some basic	Català Euskera Galego
> Line	Hardware: Software:	V1.0 1.8.5.5				information of the phone, including model, hardware and software	Türkçe Slovenian česká Nederlands
› Phone settings	Uptime:	25:48:39				version, running time, network status, account registration status, etc.	한국어 Українська
> Phonebook	Network 🥑 WAN						
• Call logs	Network mode: MAC:	DHCP 0c:38:3e:12:c8:96					

Picture 86 - 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 phone interface and web interface.

Phone end: When the phone is in the default standby state, press the [Menu] >>
 [Basic] >> [Time & Date], use the up/down navigation button to edit parameters,
 press the [OK] to save after completion, as shown in the figure:

16:04
0
ngapore, <
$\langle \rangle$
<>
ОК

Picture 87 - Set time & date on phone

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

	Features	Media Settings	MCAST	Action	Time/Date	Tone	Advanced
ystem							
	Network Time	Server Settings					
letwork	Time Sync	hronized via SNTP					
	Time Sync	hronized via DHCP					
ine	Time Sync	hronized via DHCPv6					
	Primary Ti	me Server	0.pool.ntp.org				
Phone settings	Secondary	Time Server	time.nist.gov				
	Time zone		(UTC+8) Beijing,Si	ngapore,Perth,Irkuts	T		
Phonebook	Resync Per	riod	60	second(s)			
0. XVR	Time/Date For	rmat					
Call logs	12-hour cl						
volumento como	Time/Date		DD MMM WW	TIO JAN TI	HU		
unction Key							
Application	D. P. L. C. J.						
10.0		g Time Settings					
Security	Location		None	•			
	DST Set Ty	/pe	Disabled	•			
Device Log			Apply				
	Manual Time S	ottings					

Picture 88 - Set time & date on webpage



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 format	Select time format from one of the followings:			
	■ 1 JAN, MON			
	1 January, Monday			
	JAN 1, MON			
	January 1, Monday			
	MON, 1 JAN			
	Monday, 1 January			
	MON, JAN 1			
	Monday, January 1			
	DD-MM-YY			
	DD-MM-YYYY			
	MM-DD-YY			
	MM-DD-YYYY			
	■ YY-MM-DD			
	■ YYYY-MM-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			

Table 14 - Time Settings Parameters

10.1.3 Screen

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

Phone end: When the phone is in the default standby state, go to [Menu] >>
 [Basic] >> [Screen Settings] to edit the screen parameters. After editing, click [OK]
 to save, as shown in the figure:



Screen Setting		16 : 11
1. Backlight Active Lev	12	<>
2. Backlight Inactive	4	$\langle \rangle$
3. Backlight Time	45	$\langle \rangle$
4. Screensaver	Disabled	$\langle \rangle$
Return Left	Right	ОК

Picture 89 - Set screen parameters on phone

• Web end: 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, [<] or [>] switch brightness level.
- Set the brightness level in the energy-saving mode from 0 to 16, [<] or [>] switch the brightness level.
- Set the backlight time to 30 seconds by default. You can turn it off or select 15 seconds /30 seconds /45 seconds /60 seconds /90 seconds /120 seconds.
- The screen saver can be turned on or off by default.
- Web interface: enter [Phone Settings] >> [Advanced], edit screen parameters, and click submit to save.

Backlight Active Level:	12	(1~16)	
Backlight Inactive Level:	4	(0~16)	
Backlight Time:	45	(0~120)second(s)	
Screensaver	Enabled	T	
Timeout to Screensaver:	5	(0~120)second(s)	

Picture 90 - Page screen Settings

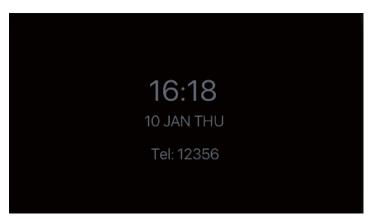
10.1.3.2 Screen Saver

 Press [Screen Settings] to find the [Screen protection] button, press [left] / [right] button to open/close the screen protection, set the timeout time, the default is 15S,



after completion, press [OK] button to save.

• After saving, return to standby mode and enter the screen saver after 15s, as follows:



Picture 91 - Phone screen saver

10.1.4 Ring

When the device is in the default standby mode,

- Press soft-button [Menu] till you find the [Basic] item.
- Enter [Basic] item till you find [Ring] item.
- Enter [**Ring**] item and you will find [**Headset**] or [**Handsfree**] item, press left / right navigator keys to adjust the ring volume, save the adjustment by pressing [**OK**] when done.
- Enter [**Ring type**] item, press left / right navigator keys to change the ring type, save the adjustment by pressing [**OK**] when done.

10.1.5 Voice Volume

When the device is in the default standby mode,

- Press soft-button [Menu] till you find the [Basic] item.
- Enter [Basic] item till you find [Voice Volume] item.
- Enter [Voice Volume] item and you will find [Headset], [Handsfree] and [Headset] item.
- Enter [Headset] or [Handsfree] or [Headset] item, press Left / Right navigator keys to adjust the audio volume for different mode.
- Save the adjustment by pressing [**OK**] when done.



10.1.6 Greeting Words

When the device is in the default standby mode,

- Press soft-button [Menu] till you find the [Basic] item.
- Enter [Basic] item till you find [Greeting Words] item.
- Press [OK] to enter the setting interface to edit the Greetings Words.
- Save the adjustment by pressing [**OK**] when done.

NOTICE! The welcome message can only be displayed in the upper left corner of standby mode when the default option is disabled.

10.1.7 Reboot

When the device is in the default standby mode,

- Press soft-button [Menu] till you find the [Basic] item.
- Enter [Basic] item till you find [Reboot] item.
- Press [OK] a prompt message, "restart now," prompts the user.
- Press [OK] to restart the phone or [Cancel].
 The phone is in standby mode,
- The configurable [**OK**] key is the restart key. **Press** [**OK**], a prompt message, "restart now" prompts the user.
- Press [OK] to restart the phone or [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. To open the phone book, user can press soft-menu button [**Contact**] in the default standby screen or keypad.

By default the phone book is empty, user may add contact(s) into the phone book manually or from call logs.



Contact			16 : 22
1. Local Cont	tacts		
2. Black list			
3. White List			
4. Cloud Cor	ntacts		
5. LDAP			
Return	Up	Down	ОК

Picture 92 - Phone book screen

NOTICE! The device can save	up to total 2000 contact records.
-----------------------------	-----------------------------------

All Contacts		-	
🦲 Jack		1234	
Mouse		5678	
🚺 Tom		4567	
Return	Option	Add	Dial

Picture 93 - 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 navigator keys. The record indicator tells user which contact is currently focused. User may check the contact's information by pressing [**OK**] button.

10.2.1.1 Add / Edit / Delete Contact

To add a new contact, user should press [Add] button to open Add Contact screen and enter the contact information of the followings,

- Contact Name
- Tel. Number
- Mobile Number



- Other Number
- Line
- Ring Tone
- Contact Group
- Photo



Picture 94 - Add New Contact

User can edit a contact by pressing [**Option**] >> [**Edit**] button.

To delete a contact, user should move the record indicator to the position of the contact to be deleted, press [**Option**] >> [**Delete**] button and confirm with [**OK**].

10.2.1.2 Add / Edit / Delete Group

By default, the group list is blank. User can create his/her own groups, edit the group name, add or remove contacts in the group, and delete a group.

- To add a group, press [Add Group] button.
- To delete a group, press [**Option**] >> [**Delete**] button.
- To edit a group, press [Edit] button.

The Number behind the group name means the total contacts number of selected groups.



Local Conta	cts	-	16 : 28
1. All Conta	cts (8)		
2. PE (2)			
3. QA (2)			
Return	Option	Add Group	OK

Picture 95 - Group List

10.2.1.3 Browse and Add / Remove Contacts in Group

User can browse contacts in a group by opening the group in group list with [OK] button.

All Contacts	PE	QA	
9 A		456	
<u>о</u> р		5643	
Return	Option	Add	Dial

Picture 96 - Browsing Contacts in a Group

When user is browsing contacts of a group, user can also add contacts in that group by pressing [Add] button to enter the group contacts management screen, then press [OK] button to save the contact. The contact will also be added in local phonebook. User can delete contact from group by [Option] >> [Delete].



Add Contact	ts		16 : 30
1. Name			
2. Office Nur	mber		
3. Mobile			
4. Other Nun	nber		
5. Line	Auto	Ì	$\langle \rangle$
Return	Abc	Delete	ОК

Picture 97 - Add Contacts in a Group

10.2.2 Black list

X210 Support blacklist, such as the number added to the blacklist, the number of calls directly refused to the end, the end of the phone shows no incoming calls. (Blacklisted Numbers can be called out normally)

- There are multiple ways to add a number to Blacklist on X210 device. It can be added directly on [Menu] >> [Contact] >> [Blacklist].
- Select any number in the phone book (both local and network) for configuration addition.
- Select any number in the call log for configuration addition.

Black list			16 : 35	Add Black List			16 : 36
1. 4321				1. Number			
2. 6543				2. Line	All		<>
				3. Number/Prefix	Num	lber	<>
					100		01
Return	Option	Add	Dial	Return	123	Delete	ОК

Picture 98 - Add Blacklist

- There are various ways to add number to the blacklist on web page, which can be added in the [Phone book] >> [Call list] >> [Restricted Incoming Calls].
- Select any number in the phone book (both local and network) for configuration addition.
- Select any number in the call log for configuration addition.



	Add	Delete	Delete Al
Caller Number		L	ine
4321			ALL
6543			ALL

Picture 99 - Web Blacklist

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 100 - 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 [**OK**] / [**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.

Cloud Cont	acts	16 : 42
1. Phonebo	ook	
	0	
	Downloading	J
Return	Option	OK

Picture 101 - Downloading Cloud Phone book

Cloud Conta	icts	-	16:43
1. FAE-Grou	qu		
2. PM-Grou	р		
3. HW-Grou	ıp		
4. MT-Grou	р		
5. Manage-	Group		
Return	Search	Option	Dial

Picture 102 - Browsing Contacts in 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 navigator keys. Each call log record is presented with 'call type' and 'call party number / name'. User can check further call log detail by pressing [**OK**] button and dial the number with [**Dial**] button, or add the call log number to phonebook with pressing [**Option**] >> [**Add to Contact**].

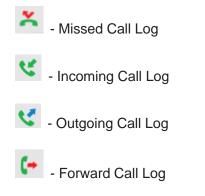


User can delete a call log by pressing [**Delete**] button and can clear all call logs by pressing [**Delete All**] button.

All	In	Out	Miss
(+ 4380	4380) 10) Jan 16:50
👗 4380	4380) 10) Jan 16:49
(+ 12356	1235	6 10) Jan 16:49
× 4380	4380) 10) Jan 16:47
12356	1235	6 10) Jan 16:47
Return	Option	Delete	Dial

Picture 103 - CallLog

Users can also filter the call records of specific call types to narrow down the scope of search records, and select a call record type by left and right navigation keys.



All	In	Out Miss		All	In	Out	Miss
🤇 anonym	ous anonym	ous 09 Jan 18:01		12356	12356	10	Jan 16:47
4380	4380	09 Jan 17:19		7000	7000	10	Jan 09:31
12356	12356	09 Jan 17:19		12356	12356	10	Jan 09:27
\\$ 12356	12356	09 Jan 16:29		12356	12356	09	Jan 17:17
V 12356	12356	09 Jan 16:27		4380	4380	09	Jan 17:16
Return	Option	Delete Dial	Re	turn	Option	Delete	Dial
All							
	l In	Out Miss		In	Out	Miss	Forward
<mark>×</mark> 4380	In 4380	Out Miss 10 Jan 16:49		In 4380	Out		Forward Jan 16:50
× 4380						10	
	4380	10 Jan 16:49		4380	4380	10) Jan 16:50
× 4380	4380 4380	10 Jan 16:49 10 Jan 16:47	•	4380	4380	10) Jan 16:50
× 4380 × 1	4380 4380 1	10 Jan 16:49 10 Jan 16:47 10 Jan 15:36		4380	4380	10) Jan 16:50

Picture 104 - Filter call record types



10.4 Function Key

Line/DSS/BLF is supported on every page of the secondary screen. There are 3 pages in total. Users can customize and configure each DSS key on each page.

Users can use the page switch key to switch DSS display pages quickly. In addition, the user can also long press each DSS key to modify the corresponding key Settings.



Picture 105 - DSS LCD key Page Configuration Screen

The DSS Key could be configured as followings,

- Memory Key
 - Speed Dial/Intercom/BLF/Presence/Call Park/Call Forward (to someone)
- Line
- Key Event
 - MWI/DND/Hold/Transfer/Phonebook/Redial/Pickup/Call Forward (to specified line)/Headset/ SMS/Release
- DTMF
- Action URL
- BLF List Key
- Multicast
- Action URL
- XML Browser

Each DSS key can set the DSS Theme. The Settings of the phone interface and webpage interface are as follows:

Phone interface: log press the DSS key to enter the following.



Dsskey			16 : 55
1. Dsskey	1-1		0
2. Type	None	9	$\langle \rangle$
3. Dss Them	e Gree	n	$\langle \rangle$
Return	Left	Right	OK

Picture 106 - DSS LCD Screen Configuration

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

	22		Side Key	Softkey	Advanced				
System									
Network	Fun	ction Key S Dsskey Tra		Make a New C 🔻		Page: None 🔻			
Line		Page1	Page2 Page3]	Apply				
Phone settings	Key	Туре	Name	e Value	Subtype	Line	Media	PickUp Number	Icon Color
anone-secungs	DSS								
		None			None	AUTO			Default Green
	Key 1	None	•		None 🔻	AUTO 🔻	DEFAULT Y		Default Green • Default Green
Phonebook	Key 1 DSS								Default Green Default Blue
Phonebook	Key 1 DSS Key		•		None None	AUTO •	DEFAULT V		Default Green Default Blue Default Yellow
	Key 1 DSS Key 2	None							Default Green Default Blue Default Yellow Default Red
	Key 1 DSS Key	None			None 🔻		DEFAULT		Default Green Default Blue Default Yellow Default Red Default Purple
	Key 1 DSS Key 2 DSS Key 3	None	•		None 🔻	AUTO 🔻	DEFAULT		Default Green Default Blue Default Yellow Default Red
Call logs	Key 1 DSS Key 2 DSS Key 3 DSS	None	▼ [None V	AUTO T	DEFAULT		Default Green Default Blue Default Yellow Default Red Default Purple Custom
Call logs	Key 1 DSS Key 2 DSS Key 3 DSS Key	None	•		None 🔻	AUTO •	DEFAULT V		Default Green Default Blue Default Yellow Default Red Default Purple
Call logs	Key 1 DSS Key 2 DSS Key 3 DSS Key 4	None None None	▼ [None V	AUTO T	DEFAULT		Default Green Default Blue Default Yellow Default Red Default Purple Custom
Call logs Function Key	Key 1 DSS Key 2 DSS Key 3 DSS Key 4 DSS	None None	▼ [None None None None	AUTO T	DEFAULT		Default Green Default Blue Default Yellow Default Red Default Purple Custom
Call logs Function Key	Key 1 DSS Key 2 S Key 3 DSS Key 4 DSS Key 5	None None None	*		None None None None None None None None None None None	AUTO T	DEFAULT		Default Green Default Blue Default Yellow Default Red Default Purple Custom
Phonebook Call logs Function Key Application	Key 1 DSS Key 2 DSS Key 3 DSS Key 4 DSS Key	None None None None	*		None V None V None V None V	AUTO •	DEFAULT		Default Green Default Blue Default Yellow Default Red Default Purple Custom

Picture 107 - 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.

NOTICE! User-defined title is up to 10 characters.

More detailed information *refers to* <u>12.23 Function Key</u> and <u>6.3 Appendix III - LED</u> <u>Definition</u>.



10.5 Wi-Fi

X210 supports wireless Internet access and requires the use of a specified USB WIFI dongle.

When the device is in the default standby mode,

- Press soft-button [Menu] till you find the [Basic] item.
- Enter [Basic] item till you find [WIFI] item.
- Press [WIFI] to enter the setting interface.
- Select the wireless network and use the left and right keys to activate it. Enable the X210 to search the current wireless network automatically.
- Select the available network, enter the user name and password to connect successfully.

Tip: if no wireless USB dongle is inserted, the prompt "wireless adapter has been removed" will appear.

If a USB dongle is plugged in, the wireless network will be priority network even if the network cable is plugged in.

Network	Phone Account TR069	Fanvil	10 JAN THU	?
1. Vlan Id	None	🖀 Fanvil	17 04	🖀 SIP6
2. Mode	DHCP/IPv4	🖀 SIP2		O MWI
3. IPv4	172.16.130.65	🖉 SIP3	Fanvil	O Headset
		🖀 SIP4		<i></i>
		🖀 SIP5		>
Return		CallLog	Contact DND	Menu

Picture 108 - WIFI settings

10.6 Headset

10.6.1 Wired Headset

- X210 supports wired earphone with RJ9 interface, which can play incoming call sound and talk with earphone.
- After the phone is connected to the headset, the default DSS key of headset will be green light which indicating that the headset can be used normally.
- On the webpage [**Phone settings**] >> [**Features**], you can set the headset answering function, and the ring tone for headset.



	Features Media Set	tings MCAST	Action	Time/Date	Tone	Advance
System						
o de la composición d	Basic Settings >>					
Network	Enable Call Waiting:	I		Enable Call Transfer:		
	Semi-Attended Transfer:			Enable 3-way Conference:	I	
Line	Enable Auto on Hook:			Auto HangUp Delay:	3 (0~30)second	(c) 🙆
Phone settings	Ring From Headset:	Disabled 🔻 🕜		Enable Auto Headset:		
	Enable Silent Mode:	0		Disable Mute for Ring:		
Phonebook						
	Enable Default Line:			Enable Auto Switch Line:		
Call logs	Default Ext Line:	Fanvil@SIP1 🔻 🕜		Ban Outgoing:		
	Default Ans Mode:	Video 🔻 🕜		Default Dial Mode:	Video 🔻 🕜	
Function Key	Hide DTMF:	Disabled 🔻 🕜		Enable CallLog:		
T diredon Rey	Enable Restricted Incomi List:	ng 🗷 🕜		Enable Allowed Incoming List:	@	
Application	Enable Restricted Outgoi List:	ng 🗷 🕜		Enable Country Code:		
	Country Code:			Area Code:		
Security	Enable Number Privacy:			Match Direction	From left to righ	t

Picture 109 - Headset function settings

10.6.2 Bluetooth Headset

X210 supports Bluetooth headset, compatible with CSR 4.0 chip Bluetooth headset, no need to use USB dongle. The phone has built-in Bluetooth and Bluetooth antenna. When the device is in the default standby mode,

- Press soft-button [Menu] till you find the [Basic] item.
- Enter [Basic] item till you find [Bluetooth] item.
- Press [Bluetooth] to enter the setup interface.
- Select Bluetooth, and use the left and right keys to enable Bluetooth. Select Paired Device. If No paired is displayed, press [Scan] key to search, the select the scanned device to connect.

Bluetooth		17:09	Searching		17 : 10	
1. Bluetooth	Enabled 😯		1. 🖵 Fanvil X7C	0C:38:3E:31:97:9F		
2. Paired Device	No Paired		2. 🖵 00:A8:59:00:11:3B	00:A8:5	00:A8:59:00:11:3B	
3. My Dev Name	Fanvil X210		3. 🗌 nubia Z17mini	DC:F0:9	DC:F0:90:16:A0:7	
4. My Dev Mac	0C:38:3E:12:C8:97		4. 🗌 NX529J	90:C7:D8:1C:0F:2		
			5. 🖵 Fanvil X210i	00:A8:5	59:A1:B2:C5	
Return C	lear Scan	ОК	Return	Connect	Cancel	

Picture 110 - Bluetooth Settings Screen

The use of Bluetooth headset can be divided into three types: call answering; Hang up;



Bluetooth redial.

• call answering

When the Bluetooth headset is connected to the phone, the incoming call can be answered by pressing the Bluetooth answer button.

• Hang up

1) When talking with Bluetooth headset, you can hang up the phone by pressing the button on Bluetooth headset.

2) When there is an incoming call, double-click the answer button to reject the call.

3) When the caller is in the ringing state, press the answer button of the headset to cancel the call.

• Bluetooth redial

When the Bluetooth headset is connected, double-click the answer button to redial the number dialed last time.

NOTICE! some models do not support double - click redial function. Whether this function is supported or not, you can check the instruction of the headset, or connect the Bluetooth headset to the phone, and double-click the answer button to see whether it will redial.

10.6.3 EHS Headset

Phone into [Menu] >> [Function] >> [Advanced], Select [EHS Headset], can open EHS Headset (default closed EHS Headset).



Picture 111 - EHS Headset setting



10.7 Advanced

10.7.1 Line Configurations

Fanvil		17 : 15	Fanvil		-	17 : 16
1. Registration	Enabled	<>	7. Server Port	506	0	
2. Server Address	172.16.1.2		8. Proxy Addre	ess		
3. Auth. User			9. Proxy User			
4. Auth. Password			10. Proxy Pass	word		
5. SIP User	123456		11. Proxy Port	506	ol	
					r	1
Return Le	eft Right	OK	Return	123	Delete	OK

Picture 112 - 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.

Fanvil		17 : 17	Fanvil		17 : 17
1. Basic			1. Domain Realm	I	
2. Advanced			2. Dial Without Regist	. Disabled	<>
			3. Anonymous	RFC3323	<>
			4. DTMF Mode	AUTO	<>
			5. Use STUN	Disabled	0
Return Up	Down	ОК	Return abo	: Delete	ОК
Fanvil		17 : 18	Fanvil		17:18
6. Sync Clock Time	Disabled	<>	11. Park Number		
7. Local Port	5060		12. Join Call Number	ſ	
8. Ring Type	Default	<>	13. Missed Call Logs	Enabled	<>
9. MWI Number			14.Feature Sync	Disabled	<>
10. Pickup Number			15. SCA	Disabled	<
Return 123	B Delete	ОК	Return Left	t Right	ОК

Picture 113 - Configure Advanced Line Options



10.7.2 Network Settings

10.7.2.1 Network Settings

■ IP Mode

There are 3 network protocol mode options, IPv4, IPv6 and IPv4 & IPv6.

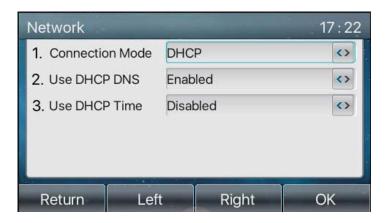
User could select available mode via "<" or ">". The selected IP mode will be activated after pressing [OK] button.

WAN Port			17 : 21
1. IP Mode			
2. IPv4			
3. IPv6			
Return	Up	Down	ОК

Picture 114 - Network mode Settings

IPv4

In IPv4 mode, there are 3 connection mode options: DHCP, PPPoE and Static IP.

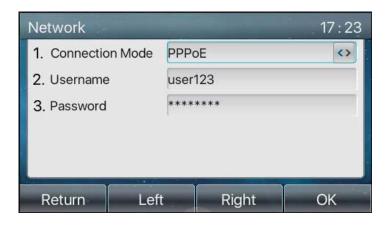


Picture 115 - DHCP network mode



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

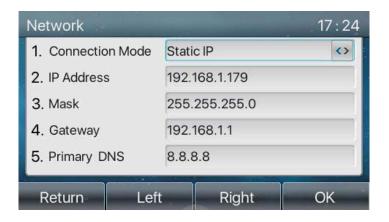
- Use DHCP DNS: It is enabled as default. "Enable" means phone will get DNS address from DHCP server and "disable" means not.
- Use DHCP time: It is disabled as default. "Enable" to manage the time of get DNS address from DHCP server and "disable" means not.



Picture 116 - PPPoE network mode

When using PPPoE, phone will get the IP address from PPPoE server.

- Username: PPPoE user name.
- Password: PPPoE password.



Picture 117 - Static IP network 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.

IPv6

In IPv6, there are 2 connection mode options, DHCP and Static IP.

- DHCP configuration refers to IPv4 introduction in last page.
- Static IP configuration is almost same as IPv4's, except the IPv6 Prefix.
- IPv6 Prefix: IPv6 prefix, it is similar with mask of IPv4.

Network		17 : 25
1. Connection Mode	Static IP	<>
2. IP Address		
3. IPv6 Prefix		
4. Gateway		
5. Primary DNS		
3	Y	1
Return Lef	t Right	ОК

Picture 118 - IPv6 Static IP network mode

10.7.2.2 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.

Parameters	Description
LLDP setting	
Report	Enable LLDP
Interval	LLDP requests interval time

Table	15 -	QoS	&	VLAN
-------	------	-----	---	------



Learning	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

Note: QoS & VLAN details refer to

http://www.fanvil.com/Uploads/Temp/download/20180920/5ba383b56c3ef.pdf

10.7.2.3 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. For users who need data encryption, please use OpenVPN instead.

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.

http://www.fanvil.com/Uploads/Temp/download/20180920/5ba38303bfcf0.pdf

10.7.2.4 Web Server Type

Configure the Web Server mode to be HTTP or HTTPS and will be activated after the reboot. Then user could use http/https protocol to access pone web page.

Network			17:27	Web Server	Туре		17 : 27
1. Network				1. Protocol	НТТ	Р	<
2. QoS & VI	LAN						
3. VPN							
4. Web Ser	ver Type						
Return	Up	Down	ОК	Return	Left	Right	ОК

Picture 119 - The phone configures the web server type

10.7.3 Set The Secret Key

When the device is in the default standby mode,



- Select [Menu] >> [Advanced setting], and enter it via [Confirm] or [OK] button.
- As default, the Advance setting password is 123.
- User will see the follow page after menu Advanced setting Security.

Security	17:34	Menu Passwo	rd		17:34
1. Menu Password		1. Current pass	sword		
2. Keyboard Password		2. New passwo	ord		
		3. Confirm pas	sword		
Return Up Down	OK	Return	123	Delete	ОК

Picture 120 - Set the Menu password

Menu password is the permission for accessing the advanced setting.

- [Current password] is the password user configured before. If no configuration before, the default password is 123.
- [New password] is the new password user to use.
- After configuring the menu password, it will work immediately.

Security			17 : 17
1. Menu Pas	sword		
2. Keyboard	Password		
Return	Up	Down	ОК

Picture 121 - Keypad lock password

Security	17:36	Keyboard Passwo	l 17:37		
1. Menu Password 2. Keyboa 🕡 Enter Password		1. Keyboard Status	Enabled	O	
Return 123 Delete	ОК	Return	eft Right	ОК	

Keyboard password is used to unlock the phone once it's locked.

Picture 122 - Set the keypad lock password

User could only set to enable or disable the keyboard password in LCD screen.



- Enter [Keyboard password] setting by pressing [confirm] or [OK] button after password entered. If no menu password configuration before, it is 123 as default.
- If the menu password is correct, phone will go to keyboard password interface. As default, the keyboard password is disabled. When it is enabled, the keyboard will be locked after timeout.
- If user does not configure the keyboard lock time, (it is 0 as default). Long pressing "#" will lock the phone. There will be a lock icon in the top of LCD. Phone will reminder "Enter Password" after pressing any keys.



Picture 123 - Phone keypad lock password input interface

	Features Media Settin	gs MCAST	Action	Time/Date	Tone	Advan
System						
Network	Screen Configuration					
Network	Backlight Active Level:	12	(1~16)			0
> Line	Backlight Inactive Level:	4	(0~16)			0
Line	Backlight Time:	45	(0~120)second(s)			0
> Phone settings	Screensaver	Enabled •				0
Phone settings	Timeout to Screensaver:	5	(0~120)second(s)			0
Phonebook		Apply]			
Call logs	LCD Menu Password Settings					
Cuiriogs	Menu Password:					0
Function Key		Apply]			
	Keyboard Lock Settings					
Application	Keyboard Password:					0
	Keyboard Time:	0				
Security	Enable Keyboard Lock:					0
		Apply]			
Device Log	Greeting Words					
	Greeting Words:	VOIP PHONE	(0-12 cł	naracter(s))		
		Apply]			

Picture 124 - Web keyboard lock password Settings

10.7.4 Maintenance

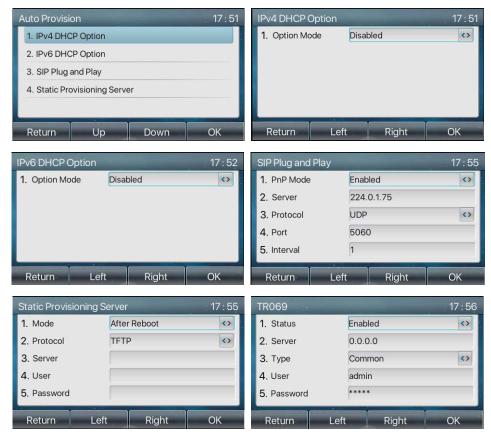
Phone Webpage: Login and go to [System] >> [Auto provision].



	Information	Account	Configurations	Upgrade	Auto Provision	Tools Reboot Phi	
System							
2004 - S201 T	Basic Settings						
letwork	CPE Serial N	CPE Serial Number:		00100400FV02001	0		
	Authenticati	on Name:				0	
ine	Authenticati	on Password:				0	
	Configuratio	n File Encryption K	ey:	í.		0	
one settings	General Con	figuration File Encr	yption Key:			0	
	Download Fa	ail Check Times:		1			
onebook	Update Cont	act Interval:		720	(0,>=5)minute(s)	0	
	Save Auto P	rovision Informatio	in:	0		0	
logs	Download C	ommonConfig enab	pled:				
188 are to	Enable Serv	er Digest:		0		0	
ion Key	DHCP Option >3	DHCP Option >>					
ication	DHCPv6 Option	>>					
And and a second	SIP Plug and Pla	ay (PnP) >>					
urity	Static Provision	Static Provisioning Server >>					
vice Log	Autoprovision N	low >>					
	TR069 >>						
	Enable TR06	9:		0		0	
	ACS Server	Type:		Common •		0	
	ACS Server	URL:		0.0.0.0		0	
	ACS User:			admin		0	
	ACS Passwo	rd:				0	
	Enable TR06	9 Warning Tone:				0	
	TLS Version:			TLS 1.0 *		0	
	INFORM Ser	ding Period:		3600	(1~9999)second(s)	0	
	STUN Serve	r Address:				0	

Picture 125 - Page auto provision Settings

LCD: [Menu] >> [Advanced setting] >> [Maintenance] >> [Auto Provision].



Picture 126 - Phone 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

http://www.fanvil.com/Uploads/Temp/download/20180920/5ba3816f8d5f0.pdf

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	If the device configuration file is encrypted, user should add		
Encryption Key	the encryption key here		
General Configuration File	If the common configuration file is encrypted, user should add		
Encryption Key	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	Save the HTTP/HTTPS/FTP user name and password. If the		
Information	provision URL is kept, the information will be kept.		
Download Common	Whether phone will download the common configuration file.		
Config enabled			
Enable Server Digest	When the feature is enable, if the configuration of server is		
Enable Gerver Digest	changed, phone will download and update.		
DHCP Option			
	Confiugre DHCP option, DHCP option supports DHCP custom		
Option Value	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.		
SIP Plug and Play (PnP)	r		
Enable SIP PnP	Whether enable PnP or not. If PnP is enable, phone will send		
	a SIP SUBSCRIBE message with broadcast method. Any		

Table 16 - Auto Provision



	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 Serve	r		
	Provisioning server address. Support both IP address and		
Server Address	domain address.		
	The configuration file name. If it is empty, phone will request		
	the common file and device file which is named as its MAC		
Configuration File Name	address.		
	The file name could be a common name, \$mac.cfg, \$input.cfg.		
	The file format supports CFG/TXT/XML.		
	Transferring protocol type , supports FTP、TFTP、HTTP and		
Protocol Type	HTTPS		
	Configuration file update interval time. As default it is 1, means		
Update Interval	phone will check the update every 1 hour.		
	Provision Mode.		
Lindota Mada	1. Disabled.		
Update Mode	2. Update after reboot.		
	3. Update after 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	If TR069 is enabled, there will be a prompt tone when		
Tone	connecting.		
TLS Version	TLS version (TLS 1.0, TLS 1.1, TLS 1.2)		
INFORM Sending Period	INFORM signal interval time. It ranges from 1s to 999s		
STUN Server Address	Configure STUN server address		
STUN Enable	To enable STUN server for TR069		



Firmware Upgrade 10.7.5

Fanvil							
	Information	Account	Configurations	Upgrade	Auto Provision	Tools	Reb
> System							

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

	Information	Account	Configuration	s Upgrade	Auto Provision	Tools	Reboot Phone
> System							
› Network	Software upgra						
		Current Softwa	are Version:	1.8.5.5			
> Line		System Image	File:		Select	Upgrade	
	Upgrade Server						
> Phone settings		Enable Auto U	pgrade:				
		Upgrade Serve	Address1:				
> Phonebook		Upgrade Serve	r Address2:				
		Update Interva	al:	24	hour		
> Call logs				Apply			
	Firmware Infor	mation					
Function Key		Current Softwa	are Version:	1.8.5.5			
		Server Firmwa		10.010			
Application		Up	ograde				
		New Firmware	Information:				
> Security							

Picture 127 - Web page firmware upgrade

LCD interface: go to [Menu] >> [Advanced setting] >> [Firmware Upgrade].

Firmware Upgrade 18 : 00
Current Version
Server Version
Return

Picture 128 - Firmware upgrade information display

Parameter	Description
Upgrade server	
	Enable automatic upgrade, If there is a new version txt
Enable Auto Upgrade	and new software firmware on the server, phone will
	show a prompt upgrade message after Update Interval.
Upgrade Server Address1	Set available upgrade server address.

Table 17 - Firmware upgrade



Upgrade Server Address2	Set available upgrade server address.		
Update Interval	Set Update Interval.		
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 version description 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 from the server is a TXT file called vendor_model_hw10.txt.Hw followed by the hardware version number, it will be written as hw10 if no difference on hardware. All Spaces in the filename are replaced by underline.
- The URL requested by the phone is HTTP:// server address/vendor_Model_hw10
 .txt: The new version and the requested file should be placed in the download directory of the HTTP server, as shown in the figure:

名称	修改日期	类型	大小
fanvil_x6_hwv1_0.txt	2018/9/11 17:57	文本文档	1 KB
fanvil_x6_hwv1_1.txt	2018/9/11 17:57	文本文档	1 KB
fanvil_x6_hwv1_2.txt	2018/9/11 17:57	文本文档	1 KB
🗎 fanvil x6 hwv1 3.txt	2018/9/11 17:57	文本文档	1 KB
📜 x6-6904-P0.12.12-1.6.3-2502T2018-0	2018/8/21 19:52	WinRAR 压缩文	35,847 KB

- TXT file format must be UTF-8
- vendor_model_hw10.TXT The file format is as follows:
 - Version=1.6.3 #Firmware

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

BuildTime=2018.09.11 20:00

Info=TXT|XML

Xxxxx Xxxxx Xxxxx Xxxxx

• After the interval of update cycle arrives, if the server has available files and



versions, the phone will prompt as shown below. Click [view] to check the version information and upgrade.



Picture 129 - Firmware upgrade

10.7.6 Factory Reset

The phone is in default standby mode.

- Press [Menu] to find [Advanced Settings], and press [OK].
- Press [Advanced Settings] to enter the password (default password is 123) to enter the interface.
- Press the [Restore factory Settings] button to select the file to be cleared.

Press **[OK]** to clear after completion. When you select clear configuration file and clear all, the phone will restart automatically after clearing.



11 Web 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 Version
- Software Version
- Uptime

And summarization of network status,

- Network Mode
- MAC Address
- IP
- Subnet Mask
- Default Gateway

Besides, summarization of SIP account status,

- SIP User
- SIP account status (Registered / Unapplied / Trying / 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 Tables

Select the local data table to be cleared, all selected by default.

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, DSS Key icon, 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 Auto Provision Description.

http://www.fanvil.com/Uploads/Temp/download/20180920/5ba3816f8d5f0.pdf

11.7 System >> Tools

Tools provided in this page help users to identify issues at trouble shooting. Please refer to <u>13 Trouble Shooting</u> 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.

Web Server Type:	HTTP V	C C
Web Logon Timeout:	15 (10~30)Minute	6
web auto login:		
HTTP Port:	80	6
HTTPS Port:	443	6
RTP Port Range Start:	10000	6
RTP Port Quantity :	1000	6
	Apply	

Picture 130 - Service Port Settings

Parameter	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.	
RTP Port Range Start	The value range is 1025 to 65535. 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	Number of calls.	

Table 18 - Service port



12.2 Network >> VPN

Users can configure a VPN connection on this page. See <u>10.7.2.3 VPN</u> 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 <u>10.7 advanced</u> Settings.

12.4 Line >> SIP

Configure the Line service configuration on this 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	

Table 19 - Line configuration	n on the web page
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Transport Protocol	Set up the SIP transport line using TCP or UDP or TLS.
Registration Expiration	Set SIP expiration date.
SIP 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 the timeout of call transfer process.
Conference Type	Set the type of call conference, Local=set up call conference by the device itself, maximum supports two remote parties, Server=set up call conference by dialing to a conference room on the server
Server Conference Number	Set the conference room number when conference type is set to be Server
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	Set the interval of voice message notification subscription				
Period					
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 headphone				
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'				
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected				
	automatically				
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				
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.				
Video Codecs	Select video code to preview video.				
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				
Disable DND	Set the feature code to dial to the server				
Enable Call Forward	Set the feature code to dial to the server				
	1				



Unconditional Enable Call Forward on Busy Set the feature code to dial to the server Disable Call Forward on No Set the feature code to dial to the server Answer Disable Call Forward on No Disable Call Forward on No Set the feature code to dial to the server Answer Disable Call Forward on No Asswer Set the feature code to dial to the server Call Disable Blocking Anonymous Set the feature code to dial to the server Call Disable Blocking Anonymous Set the feature code to dial to the server Call Set the feature code to dial to the server Call Waiting On Code 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 Send Anonymous Off Code Set the feature code to dial to the server SP Encryption Enable SIP encryption such that SIP transmission will be encrypted Enable Session Timer Set the line to enable call ending by session timer refreshment. The call session timer timeout period Set the Ine to enable Call Endure code to in status of a group. Multiple BLF List allows one BLF Key to monitor the status of a group. Multiple BLF	Unconditional	
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Keep Alive Interval Set the keep alive packet transmitting interval	Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep
		NAT pinhole opened
Keep Authentication Keep the authentication parameters from previous authentication	Keep Alive Interval	Set the keep alive packet transmitting interval
resp realistication provide duffetilidation	Keep Authentication	Keep the authentication parameters from previous authentication



Blocking Anonymous Call	Reject any incoming call without presenting caller ID
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	Set use tel call
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
DNS Mode	Select DNS mode, A, SRV, NAPTR
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets
	from the server, it will use the source IP address, not the address in
	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 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	Set the device to use 182 response code at call waiting response
waiting	
Enable Feature Sync	Feature Sync with server
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
CallPark Number	Set the CallPark number.
Server Expire	Set the timeout to use the server.
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 Chgport	Whether port updates are enabled.
VQ Name	Open the VQ name for VQ RTCP-XR.



VQ Server	Open VQ server address for VQ RTCP-XR.
VQ Port	Open VQ port for VQ RTCP-XR.
VQ HTTP/HTTPS Server	Enable VQ server selection for VQ RTCP-XR.
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.
PickUp Number	Set the scramble number when the Pickup is enabled.
JoinCall Number	Set JoinCall Number.
Intercom Number	Set Intercom Number.
Unregister On Boot	Whether to enable logout function.
Enable MAC Header	Whether to open the registration of SIP package with user agent
	with MAC or not.
Enable Register MAC Header	Whether to open the registration is user agent with MAC or not.
BLF Dialog Strict Match	Whether to enable accurate matching of BLF sessions.
PTime(ms)	Set whether to bring ptime field, default no.
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	Set the registration failure retry time.
Time	
Local SIP Port	Modify the phone SIP port.
Enable uaCSTA	Set to enable the uaCSTA function.

12.5 Line >> SIP Hotspot

Please refer to <u>9.9 SIP Hotspot.</u>



12.6 Line >> Dial Plan

Basic Settings		
	Press # to invoke dialing	?
	Dial Fixed Length11 to Send	?
	Send after10 second(s)(3~30)	?
	Press # to Do Blind Transfer	?
	Blind Transfer on Onhook	0
	Attended Transfer on Onhook	0
	Attended Transfer on Conference Onhook	?
	Enable E.164	?
	Apply	

Picture 131 - Dial plan settings

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
Press # to Do Blind Transfer	The user enters the number to be transferred and then
	presses the "#" key to transfer the current call to a third
	party
Blind Transfer on Onhook	After the user enters the number, hang up the handle
	or turn off the hands-free function to transfer the
	current call to a third party.
Attended Transfer on Onhook	Hang up the handle or press the hands-free button to
	realize the function of attention
	-transfer, which can transfer the current call to a third
	party.
Attended Transfer on Conference Onhook	During a three-way call, hang up the handle and the
	remaining two parties remain on the call.
Enable E.164	Please refer to e. 164 standard specification

Table 20 - Phone 7 dialing methods



Add dialing rules:

Digit Map: Apply to C Line: Alias(Optic Suffix:		PEER	• •	Match Send: Destin Phone Numbe	ation:] 0	0	Media: Port: Length:	0	0
Plan Optic	on 🕜				Add	Modify				
		e 🕜								
r-defined I	Jiai Pian Tabi									

Picture 132 - Custom setting of dial - up rules

Parameters	Description			
Dial rule	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.			
Note: Two differen	t special characters are used.			
x Matches a	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.			
Destination	Set Destination address. This is for IP direct.			
Port	Set the Signal port, and the default is 5060 for SIP.			
Alias	Set the Alias. This is the text to be added, replaced or deleted. It is an optional			
	item.			
Note: There are fo	bur types of aliases.			
 all: xxx – xxx will replace the phone number. 				

Table 21 - Dial - u	ip rule configuration t	able
---------------------	-------------------------	------

■ 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.

	Suffix	Characters to be added at the end of the phone number. It is an optional item.
ſ	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.

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 172.168.2.208. Using this feature, 123 can be substituted for 172.168.2.208.

User	User-defined Dial Plan Table 🎯									
	Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix	Media		
	1	"123"	Out	No	SIP DIALPEER(172.16.1.15:5560)			Default		

Picture 133 - 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 Ta	ble 🕜					
	Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix	Media
	1	"1T"	Out	No	Fanvil@SIP1	rep:010(1)		Default

Picture 134 - 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.



12.7 Line >> Action Plan

When calling to a phone, the bounded IP camera synchronously transmits video to the opposite phone (video support).

Parameter	Description
Number	Auxiliary phone number (support video)
Туре	Support video display on call.
Direction	Support call display video for call mode, call/call
	display video.
Line	Set up outgoing lines.
Username	Bind the user name of the IP camera.
Password	Bind IP camera password.
URL	Video streaming information.
User Agent	Set user agent information

Table 22 - IP camera

12.8 Line >> Basic Settings

Set up the register global configuration.

Table 23 - Set	the line globa	l configuration	on the web page
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Parameters	Description
STUN Settings	
Server Address	Set the STUN server address
Server Port	Set the STUN server port, default is 3478
Binding Period	Set the STUN binding period which can be used to keep the NAT
	pinhole opened.
SIP Waiting Time	Set the timeout of STUN binding before sending SIP messages
TLS Certification File	Upload or delete the TLS certification file used for encrypted SIP
	transmission.
Parameters	Description



12.9 Line >> RTCP-XR

RTCP-XR mode is based on RFC3611 (RTP Control Extended Report), which can measure and evaluate network packet loss, delay and voice quality by sending RTCP-XR packets.

Parameters	Description
VQ RTCP-XR Settings	
VQ RTCP-XR Session Report	VQ report on whether session mode is enabled or not.
VQ RTCP-XR Interval Report	Whether to turn on Interval mode for VQ report sending.
Period for Interval Report(5~99)	The time interval at which VQ reports are sent
	periodically.
Warning threshold for Moslq(15~40)	When the phone calculated the Moslq value x10 below
	the set threshold, a warning was issued.
Critical threshold for Moslq(15~40)	When the phone calculates the Moslq value x10 below
	the set threshold, the critical report is issued.
Warning Threshold for Delay(10~2000)	When the one-way delay of the phone is greater than
	the set threshold, warning is issued.
Critical Threshold for Delay(10~2000)	When the phone computes that the one-way delay is
	greater than the set threshold, the critical report is
	issued.
Display Report Options on web	Whether to display the VQ report data for the last call
	through the web page.

Table 24 - VQ RTCP-XR Settings

12.10 Phone settings >> Features

Configuration phone features.

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow user to take second incoming call
	during an established call. Default enabled.
Enable Call Transfer	Enable Call Transfer.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it
Enable 3-Way Conference	Enable 3-way conference by selecting it



Enable Auto Onhook The phone will hang up and return to the idle automatically at hands-free mode Auto Onhook Time Specify Auto Onhook time, the phone will hang up and return to the idle automatically after Auto Hand down time at hands-free mode, and play dial tone Auto Onhook time at handset mode, and play dial tone Auto Onhook time at handset mode. Ring for Headset Enable Ring for Handset by selecting it, the phone plays ring tone from handset. Auto Headset Enable this feature, headset plugged in the phone, user press 'answer' key or line key to answer a call with the headset automatically. Enable Silent Mode When enabled, the phone is muted, there is no ringing when calls, you can use the volume keys and mute key to unmute. Disable Mute for Ring When it is enabled, you can't mute the phone Enable Default Line If enabled, user can assign default SIP line for dialing out rather than SIP1. Enable Auto Switch Line Enable phone to select an available SIP line as default automatically number. Hide DTMF Configure the hide DTMF mode. Enable CallLog Select whether to save the call log. Enable Restricted Incoming List Whether to enable restricted all list. Enable Country Code Fill in the country code is enabled. Country Code Fill in the country code. Area Code Fill in the area code. Enable Number Privacy		hands-free mode
Auto Onhook Time Specify Auto Onhook time, the phone will hang up and return to the idle automatically after Auto Hand down time at hands-free mode, and play dial tone Auto Onhook time at handset mode Ring for Headset Enable Ring for Handset by selecting it, the phone plays ring tone from handset. Auto Headset Enable this feature, headset plugged in the phone, user press 'answer' key or line key to answer a call with the headset automatically. Enable Silent Mode When enabled, the phone is muted, there is no ringing when calls, you can use the volume keys and mute key to unmute. Disable Mute for Ring When it is enabled, you can't mute the phone Enable Default Line If enabled, user can assign default SIP line for dialing out rather than SIP1. Enable Auto Switch Line Enable phone to select an available SIP line as default automatically Default Ext Line Select the default line to use for outgoing calls If you select Ban Outgoing to enable it, and you cannot dial out any number. Hide DTMF Configure the hide DTMF mode. Enable Restricted Incoming List Whether to enable restricted call list. Enable Restricted Outgoing List Whether to enable the allowed call list. Enable Country Code Fill in the country code is enabled. Country Code Fill in the area code. Enable Roumber Privacy	Auto Onhook Time	
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mode, and play dial tone Auto Onhook time at handset modeRing for HeadsetEnable Ring for Handset by selecting it, the phone plays ring tone from handset.Auto HeadsetEnable this feature, headset plugged in the phone, user press 'answer' key or line key to answer a call with the headset automatically.Enable Silent ModeWhen enabled, the phone is muted, there is no ringing when calls, you can use the volume keys and mute key to unmute.Disable Mute for RingWhen it is enabled, you can't mute the phoneEnable Default LineIf enabled, user can assign default SIP line for dialing out rather than SIP1.Enable Auto Switch LineEnable phone to select an available SIP line as default automaticallyDefault Ext LineSelect the default line to use for outgoing callsBan OutgoingIf you select Ban Outgoing to enable it, and you cannot dial out any number.Hide DTMFConfigure the hide DTMF mode.Enable Restricted Incoming ListWhether to enable restricted call list.Enable Restricted Outgoing ListWhether to enable the allowed call list.Enable Routry CodeFill in the country code is enabled.Country CodeFill in the country code.Area CodeFill in the area code.Enable PrivacyWhether to enable number privacy.Match DirectionMatching direction, there are two kinds of rules from right to left and from left to right.Start PositionOpen number privacy after the start of the hidden location.		
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Hide DTMFConfigure the hide DTMF mode.Enable CallLogSelect whether to save the call log.Enable Restricted Incoming ListWhether to enable restricted call list.Enable Allowed Incoming ListWhether to enable the allowed call list.Enable Restricted Outgoing ListWhether to enable the restricted allocation list.Enable Country CodeWhether the country code is enabled.Country CodeFill in the country code.Area CodeFill in the area code.Enable Number PrivacyWhether to enable number privacy.Match DirectionMatching direction, there are two kinds of rules from right to left and from left to right.Start PositionOpen number privacy to hide the number of digits.	Ban Outgoing	If you select Ban Outgoing to enable it, and you cannot dial out
Enable CallLogSelect whether to save the call log.Enable Restricted Incoming ListWhether to enable restricted call list.Enable Allowed Incoming ListWhether to enable the allowed call list.Enable Restricted Outgoing ListWhether to enable the restricted allocation list.Enable Country CodeWhether the country code is enabled.Country CodeFill in the country code.Area CodeFill in the area code.Enable Number PrivacyWhether to enable number privacy.Match DirectionMatching direction, there are two kinds of rules from right to left and from left to right.Start PositionOpen number privacy to hide the number of digits.		any number.
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Match DirectionMatching direction, there are two kinds of rules from right to left and from left to right.Start PositionOpen number privacy after the start of the hidden location.Hide DigitsTurn on number privacy to hide the number of digits.	Area Code	Fill in the area code.
Match Directionand from left to right.Start PositionOpen number privacy after the start of the hidden location.Hide DigitsTurn on number privacy to hide the number of digits.	Enable Number Privacy	Whether to enable number privacy.
and from left to right.Start PositionOpen number privacy after the start of the hidden location.Hide DigitsTurn on number privacy to hide the number of digits.	Match Direction	Matching direction, there are two kinds of rules from right to left
Hide Digits Turn on number privacy to hide the number of digits.		and from left to right.
	Start Position	Open number privacy after the start of the hidden location.
Allow IP Call If enabled, user can dial out with IP address		Turn on number privacy to hide the number of digits.
	Hide Digits	
P2P IP Prefix Prefix a point-to-point IP call.		
Caller Name Priority Change caller ID display priority.	Allow IP Call	If enabled, user can dial out with IP address



E 0 1111 1	
Emergency Call Number	
Search path	Select the search path.
LDAP Search	Select from with one LDAP for search
Emergency Call Number	Configure the Emergency Call Number. Despite the keyboard is
	locked, you can dial the emergency call number
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP
	address. More details please refer to this link
	http://www.fanvil.com/Uploads/Temp/download/20180920/5ba3
	641fe81a5.pdf。
Push XML Server	Configure the Push XML Server, when phone receives request, it
	will determine whether to display corresponding content on the
	phone which sent by the specified server or not.
Enable Pre-Dial	Disable this feature, user enter number will open audio
	channel automatically.
	Enable the feature, user enter the number without opening audio
	channel.
Enclose Multi-Line	If enabled, up to 10 simultaneous calls can exist on the phone,
Enable Multi Line	and if disabled, up to 2 simultaneous calls can exist on the phone.
Line Display Format	Custom line format: SIPn/SIPn: xxx/xxx@SIPn
Contact As White List Type	NONE/BOTH/DND White List/FWD White List
Block XML When Call	Disable XML push on call.
OID notify	When enabled, the phone displays the information when it
SIP notify	receives the relevant notify content.
Tone Settings	
Enable Holding Tone	When turned on, a tone plays when the call is held
Enable Call Waiting Tone	When turned on, a tone plays when call waiting
Play Dialing DTMF Tone	Play DTMF tone on the device when user pressed a phone digits
	at dialing, default enabled.
Play Talking DTMF Tone	Play DTMF tone on the device when user pressed a phone digits
	during taking, default enabled.
DND Settings	•
DND Option	Select to take effect on the line or on the phone or close.
Enable DND Timer	Enable DND Timer, If enabled, the DND is automatically turned
	on from the start time to the off time.
DND Start Time	Set DND Start Time
DND End Time	Set DND End Time
Intercom Settings	1
Enable Intercom	When intercom is enabled, the device will accept the incoming
118	,



	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
Enable Intercom Barge	Enable Intercom Barge by selecting it, the phone auto answers
	the intercom call during a call. If the current call is intercom call,
	the phone will reject the second intercom call
Response Code Settings	
DND Response Code	Set the SIP response code on call rejection on DND
Busy Response Code	Set the SIP response code on line busy
Reject Response Code	Set the SIP response code on call rejection
Password Dial Settings	
Enable Password Dial	Enable Password Dial by selecting it, When number entered is
	beginning with the password prefix, the following N numbers after
	the password prefix will be hidden as *, N stand for the value
	which you enter in the Password Length field. For example: you
	set the password prefix is 3, enter the Password Length is 2, then
	you enter the number 34567, it will display 3**67 on the phone.
Encryption Number Length	Configure the Encryption Number length
Password Dial Prefix	Configure the prefix of the password call number
Power LED	
Common	Standby power lamp state, off when off, open is always bright red.
Common	Off by default.
	The status of power lamp when there is unread short
SMS/MWI	message/voice message, including off/on/slow flash/quick flash,
	default slow flash.
Missed	The state of the power lamp when there is a missed call,
Missed	including off/on/slow flash/quick flash, the default slow flash.
	In the talk/dial state, the power lamp state, off is off, on is always
Talk/Dial	red bright, the default is off.
Dinging	Power lamp status when there is an incoming call, including
Ringing	off/on/slow flash/quick flash, default flash.
Muto	Power lamp status in mute mode, including off/on/slow
Mute	flash/quick flash, off by default.
	The power lamp state, including off/on/slow flash/quick flash, is
Hold/Held	turned off by defeative an left/retained
	turned off by default when left/retained.



Diaplay Missad Call Danun	No incoming call popup prompt after opening, no popup prompt
Display Missed Call Popup	when closing, open by default.
	Voice message popup prompt is not answered after opening, and
Display MWI Popup	it is opened by default if there is no popup prompt when closing.
	There is a popup prompt when the WIFI adapter is connected.
Display Device Connect Popup	There is no popup prompt when the WIFI adapter is closed. It is
	on by default.
Display SMS Popup	There is popup prompt for unread messages after opening, and
Display Sivis Popup	there is no popup prompt when closing. It is opened by default.
	When the handle is not hung back after opening, registration fails,
	IP acquisition fails, Tr069 connection fails and other
Display Other Popup	abnormalities, there will be popup prompt when it is opened;
	otherwise, there will be no prompt when it is closed, and it will be
	opened by default.

12.11 Phone settings >> Media Settings

Change voice Settings.

Parameter	Description
Codecs Settings	Select enable or disable voice encoding:
	G.711A/U,G.722,G.723,G.729,
	G.726-16,G726-24,G726-32,G.726-40,
	ILBC,AMR,AMR-WB, Opus
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 earphone ringtone to 1~9.
Headset Volume	Set the volume of the headset to 1~9.
Speakerphone Ring Volume	Set the volume of hands-free ringtone to 1~9.
G.723.1 Bit Rate	5.3kb/s or 6.3kb/s is available.
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
AMR Payload Type	Set AMR load type, range 96~127.
Headset Mic Gain	Set the earphone's radio volume gain to fit different models of



	earphones.
Opus playload type	Set Opus load type, range 96~127.
	Set Opus sampling rate, including opus-nb (8KHz) and opus-wb
OPUS Sample Rate	(16KHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length
Enable MWI Tone	When there is a new voice message message, the phone will start
	a special dial tone.
Enable VAD	Whether voice activity detection is enabled.
Onhook Time	Configure a minimum response time, which defaults to 200ms
EHS Type	EHS headset is available after enabling.
RTP Control Protocol(RTC	CP) 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
Alert Info Ring Settings	
Value	Set the value to specify the ring type.
Ring Type	Туре1-Туре9

12.12 Phone settings >> 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.

Parameters	Description
Normal Call 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

Table 27 - Multicast parameters



12.13 Phone settings >> Action

Action URL

Note! Action urls are used for IPPBX systems to submit phone events. Please refer to Fanvil Action URL for details. http://www.fanvil.com/Uploads/Temp/download/20180920/5ba3641fe81a5.pdf

12.14 Phone settings >> Time/Date

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

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
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	Time of re-synchronization with time server
12-Hour Clock	Set the time display in 12-hour mode
Date Format	Select the time/date display format
Daylight Saving Time Settings	
Local	Choose your local, phone will set daylight saving time
	automatically based on the local
DST Set Type	Choose DST Set Type, if Manual, you need to set the start time
	and end time.
Fixed Type	Daylight saving time rules are based on specific dates or
	relative rule dates for conversion. Display in read-only mode in
	automatic mode.
Offset	The offset minutes when DST started
Month Start	The DST start month
Week Start	The DST start week

Table 28 - Time&Date settings



Weekday Start	The DST start weekday
Hour Start	The DST start hour
Minute Start	The DST start minute
Month End	The DST end month
Week End	The DST end week
Weekday End	The DST end weekday
Hour End	The DST end hour
Minute End	The DST end minute
Manual Time Settings	You can set your time manually

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.

	Features Media Settings	MCAST	Action	Time/Date	Tone	Advan
System						
Network	Tone Settings					
Network	Select Your Tone:	United States	5			•
	Dial Tone:	350+440/0				
Line	Ring Back Tone:	440+480/200	0,0/4000			
	Busy Tone:	480+620/500,	0/500			
Phone settings	Congestion Tone:					
	Call waiting Tone:	440/300,0/10	000,440/300,0/10000,0	0/0		
Phonebook	Holding Tone:					
	Error Tone:					
Call logs	Stutter Tone:					
	Information Tone:					
Function Key	Dial Recall Tone:	350+440/100,	0/100,350+440/100,0/	100,350+440/100,0/100,3	50+440/0	
	Measage Tone:					
Application	Howler Tone:					
	Number Unobtainable Tone:	400/500,0/60	00			
Security	Warning Tone:	1400/500,0/0				
	Record Tone:	440/500,0/500	00			
Device Log	Auto Answer Tone:					

Picture 135 - 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
- Backlight Time
- 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 16 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 blacklist by click "Add to Blacklist" button.

12.18 Phonebook >> Cloud phonebook

Cloud Phonebook

User can configure up to 8 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)

Note! Refer to the LDAP technical documentation before creating the LDAP phonebook and phonebook server.

http://www.fanvil.com/Uploads/Temp/download/20180920/5ba382eb399eb.pdf

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.



	Contacts	Cloud phonebook	Call List	Web D	ial 🛛 🗚	dvanced		
System								
Network			(ML3 XML4 BA	ICK				
Line	XML LDAP BroadSoft Add to phonebook	Add to Blacklist A	dd to Whitelist			Previou	s Page: V Ne	ext
Phone settings	Index	Name	Phone			TOTICA	Phone2	
Phonebook	Manage Cloud Ph	onebooks 🖗					10 • Entries per	page
Call logs	Index Cloud phon		phonebook URL	Calling Line	Search A	uthentication Name	e Authentication Pas	swo
	1 Phonebook	tftp://17	2.16.7.39/51.xml	AUTO 🔻	AUTO 🔻			
Function Key	2][AUTO 🔻	AUTO 🔻			_
	3				AUTO V			
Application	4				AUTO V			
Security	LDAP Settings							
Device Log	LDAP	[LDAP 1	Ŧ				
	Display Title:	Γ		0	Version:		Version 3 🔻 🕜	
	Server Addres	ss:		0	Server Port:		389	
	LDAP TLS Mo	de:	LDAP 🔻		Calling Line:		AUTO 🔻 🕜	
	Authenticatio	n: [Simple 🔻 🕜		Search Line:		AUTO 🔻 🕜	
	Username:		admin	0	Password:		•••••	
	Search Base:			0	Max Hits:		50	
	Telephone:	Įt	elephoneNumber	0	Mobile:		mobile	
	Other:	6	other	0	Name Attr:		cn sn ou	
	Sort Attr:	6	n		Display name	:	cn	

Picture 136 - Web cloud phone book Settings

12.19 Phonebook >> Call List

Restricted Incoming Calls:

It is similar like a blacklist. Add the number to the blacklist, 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 blacklist or add specific prefixes to the blacklist 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 Call Logs

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 blacklist/whitelist.

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 >> Function Key

One-key transfer Settings: establish new call, blind transfer, attention-transfer, one-key three-party, Play DTMF.

DSS Key home page: None/Page1/Page2/Page3

The device provides 96 user-defined shortcuts that users can configure on a web page.



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
	Кеу.
Key Event	User can select a key event as a shortcut to trigger.
	For example: MWI / DND / Release / Headset / Hold / etc.
DTMF	It allows user to dial or edit dial number easily.
URL	Open the specific URL directly.
Multicast	Configure the multicast address and audio codec. User presses the key to
	initiate the multicast.
Action URL	The user can use a specific URL to make basic calls to the phone.
XML browser	Users can set the DSS Key for specific URL download and other operations.

Table 29 - Function Key configuration

12.24 Function Key >> Side Key

Side Key function and settings please refer to 12.23 Function Key.

12.25 Function Key >> Softkey

The User Settings mode and display style, display page.

Table 30 - 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						
	Redial/2aB/Delete/Exit/Call Back/Dial/Join/MWI/Local					
Call Dialer	Contacts/Pickup/CallLog/Missed/Clear/In/Dialed/Pause/Next					
	line/Prev line/Headset/Audio/Video/Remote XML/DSS Key					
Conference	Hold/Split/End/Release/Mute/DSS Key/Headset					
	CallLog/Menu/Local Contacts/DND/Prev Account/Next					
Dealstan	Account/Blacklist/Call Back/CallForward/Locked/Memo/					
Desktop	Missed/MWI/Dialed/Reboot/Redial/Remote XML/SMS/					
	Headset/Status/DSS Key/In					
	Redial/2aB/Delete/Exit/Forward/Local Contacts/CallLog					
Divert Dialed	/Clear/Missed/Dialed/Headset/Video/Audio/Remote XML					
	/DSS Key					
Ending	Redial/End/Headset/Release/DSS Key					
	Dial/2aB/Delete/Exit/Call Back/Local Contacts/Redial					
Dradiative Dialas	/Pickup/MWI/Join/CallLog/Release/Missed/Pause/Dialed/					
Predictive Dialer	Headset/Video/Audio/Remote XML/DSS Key/In/Next line					
	/Prev line					
Dinging	Answer/Forward/Reject/Mute/Release/Headset/Video/Audio/					
Ringing	DSS key					
	Hold/Transfer/Conference/End/Mute/Release/New Call/					
Talking	Local Contacts/Listen/CallLog/Next call/Prev call/					
	Private/Headset/Video/Audio/DSS Key					
Transfer Alerting	End/Transfer/Headset/Release/DSS Key					
	Redial/Delete/Exit/2aB/Dial/Local Contacts/Transfer/					
Transfer Dialer	CallLog/Clear/Missed/Dialed/Pause/Headset/Video/Audio/R					
	emote XML/DSS Key					
Trying	End/Release/Headset/DSS Key					
	Hold/Transfer/Conference/End/Answer/Forward/Mute/Next					
Waiting	call/New call/Prev call/Reject/Release/Headset/Listen/					
	Video/Audio/DSS Key					



12.26 Function Key >> Advanced

Programmable key Settings

Please refer to the Table 30 Softkey configuration

IP Camera List

Index	IP Camera	Username	Password	Preview	Dsskey
		Refres	h Apply		

Picture 137 - IP Camera List

12.27 Application >> Manage Recording

See <u>9.3 Record</u> for details of recording.

12.28 Security >> Web Filter

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

	Web Filter	Trust Certificates	Device Certificates	Firewall	
› System					
> Network	Web Filter Table				
› Line	Start IP Addre			End IP Address	Option
, Line	Web Filter Table	Settings			
> Phone settings	Start IP Addre	ess	0	End IP Address	 Add
> Phonebook	Web Filter Settin	g 🕜			
› Call logs	Enable Web Fi	ilter 🔲	ļ	Apply	
> Function Key					
> Application					
> Security					
> Device Log					

Picture 138 - Web Filter settings



Web	9 Filter Table 🕜		
	Start IP Address	End IP Address	Option
	192.168.1.1	192.168.254.254	Modify Delete

Picture 139 - 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.

12.29 Security >> Trust Certificates

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

You can upload and delete uploaded certificates.

	n Certificate	Disabled 🔻	0		
Common	Name Validation	Disabled 🔻	0		
Certificat	e mode	All Certificates 🔹	0		
		Apply			
mport Certil			Select Upload		
ertificates L	.ist 🕜				
		Issued To	Issued By	Expiration	File Siz
Index	File Name	155060 10		a second a second second second	

Picture 140 - Certificate of settings



12.30 Security >> Device Certificates

Select the device certificate as the default and custom certificate. You can upload and delete uploaded certificates.

Device Certificates 🕜				
Device Certificates	Default Certificates	(existence)		
mport Certificates 🥝				
Load Server File		Select Upload		
Certification File 🕜				
certification File 🕜	Issued To	Issued By	Expiration	File Size

Picture 141 - Device certificate setting

12.31 Security >> Firewall

	Web Filter Trust C	Certificates Device Certificate	Firewall		
System					
Network	Firewall Type 💡	Enable Input Rules: 🗐		Enable Output Rules: 🔲	
Line		Enable Input Rules.	Apply	Enable Output Rules.	
Phone settings	Firewall Input Rule Tab	ile 🕜			
Phonebook	Index Deny/Permit	Protocol Src Address	Src Mask Src Port Range	Dst Address Dst Mask	Dst Port Ran
	Firewall Output Rule Ta	ble 🕜			
Call logs	Index Deny/Permit	Protocol Src Address	Src Mask Src Port Range	Dst Address Dst Mask	Dst Port Ran
unction Key	Firewall Settings 🕜				
pplication	Input/Output Input Deny/Permit Deny			ddress	Add
Security	Protocol UDP	▼ Src Port Range	- Dst Por	t Range	
Device Log	Rule Delete Option 🥝				
	Input/Output	Input 🔻	Index To Be Delet	ed	Delete

Picture 142 - 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:

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.				
Dopy/Dormit	To select whether the current rule configuration is disabled				
Deny/Permit	or allowed;				
Drotocol	There are four types of filtering protocols: TCP UDP				
Protocol	ICMP IP.				
Src Port Range	Filter port range				
	Source address can be host address, network address, or				
Src Address	all addresses 0.0.0.0; It can also be a network address				
	similar to *.*.*.0, such as: 192.168.1.0.				
	The destination address can be either the specific IP				
Dst Address	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.				
	Is the source address mask. When configured as				
Src Mask	255.255.255.255, it means that the host is specific. When				
SICINASK	set as 255.255.255.0, it means that a network segment is				
	filtered.				
	Is the destination address mask. When configured as				
Dst Mask	255.255.255.255, it means the specific host. When set as				
Del IVIder	255.255.255.0, it means that a network segment is				
	filtered.				

Table 31 - Network Firewall

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



Firewall Input Rule Table 💡									
	Index D	eny/Permit	Protocol	Src Address	Src Mask	Src Port Range	Dst Address	Dst Mask	Dst Port Range
	1	deny	udp	192.168.1.0	192.168.1.154	0-9	255.255.255.0	255.255.255.0	0-9

Picture 143 - Firewall Input rule table

Then select and click the button [Apply].

In this way, when the device is running: ping 192.168.1.118, the packet cannot be sent to 192.168.1.118 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.

Input/Output Index To Be Deleted	
input · Index to be beleed	Delete

Picture 144 - Delete firewall rules

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

12.32 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 <u>13.6 Get log information</u>.



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).



Scree	enshot	
ſ	Main Screen:	Save BMP
5	Sub Screen:	Save BMP

Picture 145 - 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.

〕 无标题 - Google Chrome	-		pbx 使用 🗙 💽 🤅	t T16871 [bug/20181	1° 🗙 😧 🗄 T17849 X	210的DVT样 ×	🔁 🗄 T17751 [bug/201
() 172.16.7.203/cgi-bin/WebCapture	?type=Start						
			Configurations	Upgrade	Auto Provision	Tools	Reboot Phone
			514				Ø
			Information	¥			0
> Phone Setungs	Export Log:		Apply]			
> Phonebook	Web Capture 🕜						
> Call logs	Start		stop				
› Function Key	Screenshot		0.000 0000	7			
	Main Screen: Sub Screen:		Save BMP Save BMP				
> Application	Watch Dog		Ouve bin				
> Security	Enable Watch Dog	:	Apply]			

Picture 146 - 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

Trouble Case	So	lution
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	1.	Please check if device is well connected to the network. The
service provider		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 " <u>13.5 Network Packet Capture</u> " 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	1.	Please check if Handset is connected to the correct Handset (
Handset		port NOT Headphone (🎧) port.
	2.	The network bandwidth and delay may be not suitable for audio
		call at the moment.
Poor Audio or Low Volume in	1.	There are two Headphone wire sequence in the market. Please
Headphone		use the Headphone provided by Fanvil, or consult Fanvil the wire
		sequence if you wish to use a third-party headphone.

Table 32 - Trouble Cases



	2. The network bandwidth and delay may be not suitable for audio
	call at the moment.
Audio is chopping at far-end	This is usually due to loud volume feedback from speaker to
in Hands-free speaker mode	microphone. Please lower down the speaker volume a little bit, the
	chopping will be gone.