



# User Manual i10&i10V&i10D

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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 the 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 designed 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 the 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

The i10/i10V/i10D SIP mini intercom is designed for indoor scenes with IP54 waterproof and dustproof. Supports wall mounting installation. It combines security, audio/video intercom and broadcasting functionalities and offers a qualified communication solution for users.



# 5 Install Guide

## 5.1 Use POE or external Power Adapter

i10/i10V/i10D, called as 'the device' hereafter, supports two power supply modes, power supply from an external power adapter or over Ethernet (POE) complied switch.

POE power supply saves the space and cost of providing the device an 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 works properly.



# 5.2 Appendix Table

## 5.2.1 Common command mode

Action	Description		
IP Broadcast under	In standby mode, long press the speed dial button for 3 seconds,		
standby mode	there will be a toot sound will 5 seconds, please press the speed		
	dial button once within 5 seconds, the toot sound will stop		
	automatically reporting IP		
	In the standby mode, long-press the speed dial button for 3		
	seconds and the beep will last for 5 seconds. Within 5 seconds,		
	press the speed dial button three times quickly to switch to the		
	network mode.		
Quvitab patwork	If there is no IP at present, switch to the default static IP		
Switch network	(192.168.1.128).		
mode	Then switch to DHCP mode when it is the default static IP		
	(192.168.1.128)		
	When DHCP gets to IP, then do not switch and report the IP		
	directly.		
	Report the IP after the successful switch.		

#### Table 1 - Common command mode

# 5.2.2 Function key LED state

#### Table 2 - Function key LED state

Туре	LED	State
Speed dial	Normally on	Successfully registered
	Quick flashing	Registration failed/ network abnormal
	Slow flashing	In call



# 6 Basic Introduction

#### 6.1 Panel Overview

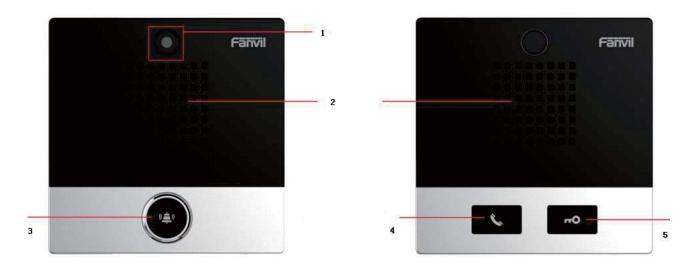


Figure 1 - Panel

Table 3 - Panel introduction

Number	Name	Description
1	IP Camera	Video signal acquisition and transmission
2	Speaker	Play sound
2	Speed diel butten	For speed dial, multicast, intercom, IP broadcast and
3	Speed dial button	other functions
4	Speed dial/Answer	For speed dial/answer button, multicast, intercom, IP
4	button	broadcast and other functions
5	Unlock	Unlock door

#### 6.2 Quick Setting

Before proceeding with this step, make sure your Internet broadband connection is working properly and complete the network hardware connection. The default factory mode is DHCP. IP address can be viewed by.

- In standby mode, long press the speed dial button for 3 seconds, there will be a toot sound will 5 seconds, please press the speed dial button once within 5 seconds (please do not operate within 30 seconds when power on), the toot sound will stop automatically reporting IP.
- Of the device or use the "IP scanning tool. exe"



software to find the IP address of the device.

(Download http://download.fanvil.com/tool/iDoorPhoneNetworkScanner.exe))

#	IP Address	Serial Number	MAC Address	SW Version	Description	
	192.168.1.128	i32V	00:a8:34:00:aa:60	T2.4.0.6274	i32V IP Door Phone	
						Refresh

Figure 2 - Quickly setting

- In the standby mode, long-press the speed dial button for 3 seconds and the beep will last for 5 seconds. Within 5 seconds, press the speed dial button three times quickly to switch to the network mode.
- Login to the device's WEB page for configuration according to the IP address
- Configure the account, user name, server address and other parameters required for registration provided by the service provider on the WEB configuration page;

## 6.3 WEB configuration

When the device and your computer are successfully connected to the network, enter the IP address of the device on the browser as http://xxx.xxx.xxx/ and you can see the login interface of the web page management.

User:	
Password:	
Language:	English 🔻
	Logon

Figure 3 - WEB Login

The username and password should be correct to log in to the web page. **The default username and password are "admin"**. For the specific details of the operation of the web page, please refer to <u>9 Web Configurations</u>



## 6.4 SIP Configurations

At least one SIP line should be configured properly to enable the 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 is stored in the configurations.

The SIP line configuration should be set via the WEB configuration page by entering the correct information such as phone number, authentication name/password, SIP server address, server port, etc. which are provided by the SIP server administrator.

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

	SIP SIP Hots	pot Basic Settings		
> System				
> Network	Line 21972@SIF   Register Settings >>			
> Line	Line Status:	Registered	Activate:	
Line	Username:	21972 Ø	Authentication User:	
> Intercom settings	Display name:	21972	Authentication Password:	
> Intercom settings	Realm:		Server Name:	
	Realm:		Server Name:	
› Call List				
	SIP Server 1:		SIP Server 2:	
Function Key	Server Address:	172.16.1.2	Server Address:	0
	Server Port:	5060	Server Port:	5060
> Security	Transport Protocol:	UDP V	Transport Protocol:	UDP V
	Registration Expiration:	3600 second(s) 🕐	Registration Expiration:	3600 second(s) @
> Device Log				
	Proxy Server Address:	0	Backup Proxy Server Address:	
> Security Settings	Proxy Server Port:	5060	Backup Proxy Server Port:	5060
7 Security Settings	Proxy User:		backup Froxy Server Fort.	5000
	Proxy Password:	0		
	PIOXy Password.	•		
	Basic Settings >>			
	A Province and the second			
	Codecs Settings >> 🕜			
	Advanced Settings >>			
	SIP Global Settings >>			
	STP Global Settings >>			
		Apply		

Figure 4 - SIP Line Configuration



# 7 Basic Function

## 7.1 Making Calls

After setting the function key to memory key and the subtype as speed dial and setting the number, press the function key to immediately call out the set number, as shown below:

Network	Function Key S	Settings >>							
	Key	Туре	Name	Value	Value2	Subtype		Line	
11207	DSS Key 1	Memory Key 🔻		21976		Speed Dial	۲	21972@SIP1	۳
Line	DSS Key 2	None 🔻				None	۳	AUTO	۳
and a second second	DSS Key 3	None 🔻				None	۳	AUTO	۳
Intercom settings Call List		• Key Settings 🕜 >>		Apply					
	Programmable	e key Settings 🐨 >>							
Function Key	Advanced Sett	ings >>							

#### Figure 5 - Function Setting

See detailed configuration instructions 9.23 Function Key

## 7.2 Answering Calls

After setting up the automatic answer and setting up the automatic answer time, it will hear the ringing bell within the set time and automatically answer the call after the timeout. Cancel automatic answering. When a call comes in, you will hear the ringing bell and will not answer the phone over time.

#### 7.3 End of the Call

When there is a call, you can press the speed dial button to hang up the call, the default setting is to end the call. See detailed configuration instructions <u>9.23 Function Key</u>.

#### 7.4 Auto-Answering

The user can turn off the auto-answer function (enabled by default) on the device webpage, and the ring tone will be heard after the shutdown, and the auto-answer will not time out.

#### • Enable auto answering on the line:

Web interface: enter [Line] >> [SIP], Enable auto answer, set mode and auto answer time and click submit.



> System						
> Network	Line 21972@SlF • Register Settings >>					
> Line	Basic Settings >>					
> Intercom settings	Enable Auto Answering:	<b>Ø</b>		Auto Answering Delay:	0	(0~120)second(s) 🧃
› Call List	Enable Hotline: Hotline Delay: Dial Without Registered:		(0~9)second(s) 🥝	Hotline Number:		0
> Function Key	Dial Without Registered: DTMF Type: Request With Port:		<b>v</b>	DTMF SIP INFO Mode:	Send 10/11	<b>v</b>
> Security	Use STUN:			Use VPN:	<b>I</b>	
› Device Log	Enable Failback: Failback Interval:		second(s) 😗	Signal Failback: Signal Retry Counts:	3	(1~10) 🕑
> Security Settings	Codecs Settings >> 💡					
	Advanced Settings >>					

Figure 6 - Enable Auto Answer

• Enable auto answering P2P:

Web interface: enter [**line**] >> [**Basic Settings**] >> [**SIP P2P Settings**], enable automatic answering, setting mode and automatic answering time, and click submit.

	SIP SIP Hotspot	Basic Settings
› System		
> Network	STUN Settings STUN NAT Traversal:	FALSE
> Line	Server Address: Server Port:	3478
› Intercom settings	Binding Period: SIP Waiting Time:	50 second(s) 800 millisecond
› Call List		Apply
> Function Key	SIP P2P Settings Enable Auto Answering	0
> Security	Auto Answering Delay: DTMF Type:	0 (0~120)second(s)
› Device Log	DTMF SIP INFO Mode:	Send 10/11 •
Security Settings		Apply

Figure 7 - Enable Auto Answer

• Auto Answer Timeout (0~60)

The range can be set to 0~60s, and the call will be answered automatically when the timeout is set.



# 7.5 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 signal will be prompted
- Enable call waiting tone: when you receive a new call on the line, the device will beep.

Users can enable/disable call waiting in the device interface and the web interface.

• Web interface: enter [Intercom Setting] >> [Features], enable/disable call waiting, enable/disable call waiting tone.

	Features Media Settings	MCAST	Action	Time/Date	Tone	
> System						
› Network	Basic Settings >> Enable Call Waiting:	<ul> <li>Ø</li> </ul>				
› Line	Enable Auto on Hook: Enable Silent Mode:	<ul> <li>?</li> <li>?</li> </ul>			3 (0~3)	0)second(s) 🕜
> Intercom settings	Ban Outgoing: Enable Restricted Incoming					
> Call List	List: Enable Restricted Outgoing List	• • • • • • • • • • • • • • • • • • •		Country Code:		
> Function Key	Country Code:		Area (	Code:		
> Security	Allow IP Call:		P2P IF	Prefix:		
> Device Log	Restrict Active URI Source IP: Line Display Format:	xxx@SIPn 🔻 🕜	🕜 Push )	(ML Server:		0
Security Settings	Call Number Filter:		Auto F	tesume Current:	?	
	Tone Settings >> Intecom Settings >>					
	Response Code Settings >>					
			Apply			_
	Features Media Setting	95 MCAST	Action	Time/Dat	e Tone	
> System						
> Network	Basic Settings >> Tone Settings >>					
> Line	Enable Holding Tone: Play Dialing DTMF Tone:	<ul> <li>?</li> <li>?&lt;</li></ul>		nable Call Waiting T ay Talking DTMF To		
> Intercom settings	Intecom Settings >>					
› Call List	Response Code Settings >>					
› Function Key			Apply			
› Security						
> Device Log						
Security Settings						

Figure 8 - Call Waiting



# 8 Advance Function

## 8.1 Intercom

The equipment can answer intercom calls automatically.

	Features Media Settin	gs MCAST	Action	Time/Date	Tone	
› System						
> Network	Basic Settings >>					
7 Network	Tone Settings >>					
› Line	Intecom Settings >>					
-	Enable Intercom:		Enable In	itercom Mute:		
> Intercom settings	Enable Intercom Tone:		Enable In	itercom Barge:	<b>@</b>	
> Call List	Response Code Settings >>					
			Apply			
Function Key						
200						
> Security						

#### Figure 9 - Intercom

#### Table 4 - Intercom

Parameters	Description			
	When intercom is enabled, the device will accept the			
Enchla Intercom	incoming call request with a SIP header of Alert-Info			
Enable Intercom	instruction to automatically answer the call after a specific			
	delay.			
Enable Intercom Mute	om Mute Enable mute during intercom mode			
Enable Intercom Tone	If the incoming call is intercom call, the device plays the			
Enable Intercom Tone	intercom tone.			
	If enable intercom barge, the device answers the intercom			
Enable Intercom Barge	call automatically while it is in a call. If the current call is			
	intercom call, the device will reject the second intercom call.			

#### 8.2 MCAST

This feature allows the user to make some kind of broadcast call to people who are in the multicast group. The user can configure a multicast DSS Key on the device, which allows the 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 device to receive an RTP stream from the pre-configured multicast listening address without involving SIP signaling. You



can specify up to 10 multicast listening addresses.

	Features Media Set	tings MCAST	Action	Time/Date	Tone	
› System						
> Network	MCAST Listening					
, Network	Priority:	1	T			
	Enable Page Priority:					
> Line	Enable Prio Chan:					
	Enable Emer Chan:					
> Intercom settings	Index/Priority	Name		Host:port		Channel
	1		]			0
> Call List	2		]			0
	3		]			0
Function Key	4		]			0
	5		]			0
Security	6		]			0
	7		]			0
Device Log	8		]			0
	9		]			0
> Security Settings	10		]			0
		Apply				

Figure 10 - MCAST

Table 5 - MCAST

Parameters	Description
Priority	Define the current call's priority, 1 means the highest priority and
	10 means the lowest.
Enable Page Priority	If enable page priority, the device will receive the multicast from
	address with higher priority, regardless of which of the two
	multicast groups sending the multicast first.
Enable Prio Chan	If enable this option, the only multicast with the same port and
	channel can be connected. Channel 24 has the higher priority, its
	priority is higher than 1-23; Set channel value to be 0, it means no
	channel is used.
Enable Emer Chan	When enabled, channel 25 has the highest priority
Name	Set the multicast server name.
Host:port	Set the multicast server's multicast IP address and port.
Channel	0-25 (24 priority channel,25 emergency channel).

#### Multicast:

Send multicast:

- Go to web page of [Function Key] >> [Function Key Settings], select the type to be multicast, set the multicast address, and select the codec.
- Click Apply.
- Press the DssKey of Multicast Key which you set.

Receive multicast:

• Set up the name, host and port of the receiving multicast on the web page of



[Intercom Settings] >> [MCAST].

• When the remote server sends the multicast, the device will receive multicast call and play multicast automatically.

## 8.3 Hotspot

SIP hotspot is a simple utility. Its configuration is simple, which can realize the function of group vibration and expand the number of SIP account.

Take one device A as the SIP hotspot and the other devices (B, C) as the SIP hotspot client. When someone calls device A, devices A, B, and C will ring, and if any of them answers, the other devices will stop ringing and not be able to answer at the same time. When A B or C device is called out, it is called out with A SIP number registered with device A.

Parameters	Description
Enable Hotspot	Set the enable hotspot option in the SIP hotspot configuration TAB to
	enabled
Mode	This device can only be used as a client
Monitor Type	The monitoring type can be broadcast or multicast. If you want to restrict
	broadcast packets in the network, you can choose multicast. The type of
	monitoring on the server side and the client side must be the same, for
	example, when the device on the client side is selected for multicast, the
	device on the SIP hotspot server side must also be set for multicast
Monitor	The multicast address used by the client and server when the monitoring
Address	type is multicast. If broadcasting is used, this address does not need to
	be configured, and the system will communicate by default using the
	broadcast address of the device's wan port IP
Remote Port	Fill in a custom hotspot communication port. The server and client ports
	need to be consistent
Name	Fill in the name of the SIP hotspot. This configuration is used to identify
	different hotspots on the network to avoid connection conflicts
Line Settings	Sets whether to enable the SIP hotspot function on the corresponding
	SIP line

#### Table 6 - SIP Hotspot

#### Client Settings :

As a SIP hotspot client, there is no need to set up a SIP account, which is automatically acquired and configured when the device is enabled. Just change the mode to "client" and the other options are set in the same way as the hotspot.



	SIP SIP Hotspot	Basic Settings	
> System			
> Network	No Registration		
	SIP Hotspot Settings		
> Line	Enable Hotspot:	Disabled <b>v</b>	0
	Mode:	Client •	0
> Intercom settings	Monitor Type:	Broadcast 🔻	0
	Monitor Address:	224.0.2.0	0
> Call List	Local Port:	16360	0
	Name:	SIP Hotspot	0
> Function Key	Line Settings		
	Line 1:	Enabled •	
> Security	Line 2:	Enabled V	
> Device Log		Apply	
> Device Log		Apply	

Figure 11 - SIP Hotspot

The device is the hotspot server, and the default extension is 0. The device ACTS as a client, and the extension number is increased from 1 (the extension number can be viewed through the [SIP hotspot] page of the webpage).

Calling internal extension:

- The hotspot server and client can dial each other through the extension number before
- Extension 1 dials extension 0



# 9 Web Configurations

## 9.1 Web Page Authentication

Users can login the device's webpage to manage and operate the device. User must provide the correct username and password to login. If the password is incorrect for three times, the webpage will be locked for 5 minutes and then the user can try to login again.

The details as following:

- If one IP logins more than the specified number of times with different username/passwords, web login will be locked.
- If a same user login more than a specified number of times from different IP addresses, web login will be locked too.

## 9.2 System >> Information

Users can get the following information in **System>>Information** page:

Basic system information:

- Model
- Hardware Version
- Software Version
- Uptime
- Last uptime
- MEMinfo

And summarization of network status:

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

Besides, the summarization of SIP account status:

- SIP User
- SIP account status (Registered/Inactive/Trying/Timeout )



# 9.3 System >> Account

		Information Account	Configurations Upgrade Auto Provision FDMS
	> System		
		Add New User	
,	Network	Username	0
		Web Authentication Password	
,	Line	Confirm Password	
		Privilege	Administrators 🔻 🕜
,	Intercom settings		Add
	Call List	User Accounts	
		User	Privilege
	Function Key	admin	Administrators
		guest	Users
,	Security		
	Security	User Management	
	Device Log	admin 🔻	Delete Modify
í	Device Log		

Figure 12 - WEB Account

On this page, user can change the webpage login password.

Administration user can also add or delete users, manage users, set permissions and passwords for new users.

# 9.4 System >> Configurations

	Information Account	nt Configurations	Upgrade	Auto Provision	FDMS	Tools
> System						
› Network	Export Configurations 🕜	Pight click ba	re to SAVE configurat	tions in 'tyt' format		
› Line		Right click her	re to SAVE nc configu	urations in 'txt' format. tions in 'xml' format.		
› Intercom settings	Import Configurations 🕜					
› Call List		Configuration	file:	Sele	Import	
Function Key	Clear Configuration >> 🥝	Click "Clear" b	button to reset the co	onfiguration files!		
	Content to	Кеер		Content to R	leset	
> Security	MMI BASIC NET SIP	WORK		DSS KEY TR069		
> Device Log	AUTOPRO	VISION	_→			
Security Settings						
		*				-
			Clear			
	Clear Tables >> 🕜					
	Reset Phone >> 🕐					

Figure 13 - System Setting



In this page, administration user can view, export, or import the configuration file, or restore the device to factory settings.

## Export Configurations

Right click to download the device's configuration file to your PC, the file format is ".txt". (Notice: only administrator user can export the configuration file.)

## Import Configurations

Import the configuration file of settings. The device will restart automatically after successful importation, and the configuration will take effect after a restart

## Clear Configurations

Select the module in the configuration file to clear.

SIP: SIP account configuration

AUTOPROVISION: Provision related 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, by default all the tables are selected.

#### Reset Phone

The device data will be cleared, including configurations and database tables.

# 9.5 System >> Upgrade

	Information	Account Configurat	ions Upgrade	Auto Provision	FDMS	Tools
> System						
> Network	Software upgrad	le 🕜 Current Software Version:	R0.2.0			
› Line		System Image File:		Select	Upgrade	
› Intercom settings		rver Address1:				
› Call List	Upgrade Se	rver Address2:	Ap	ply		
› Function Key	Firmware Inform	nation Current Software Version:	20.0.0			
› Security		Server Firmware Version:	R0.2.0 Error			
> Device Log		New Firmware Information:				
> Security Settings	Ring Upgrade 🕜					
		Load Server File:		Select	(*.wav) Uploa	ad
	Ring List 🕜					
		Index	File Name	1	File Size	
						Delete

Figure 14 - Upgrade



In this page, user can upgrade the software for the device. After the upgrade, the device will automatically restart and update to the new version.

Click select to select the software file from local PC and then click upgrade to start upgrading.

#### Online upgrade:

Online Firmware update is when a device sends an HTTP request to a server, the server replies with a corresponding description file or 404 or timeout. After device gets the reply, it analyzes the version description file and prompts the user whether to upgrade the new version or not.

Upgrade Serve	er			
Upgrade :	Server A <mark>ddress1:</mark>			]
Upgrade :	Server Address2:		Apply	
Firmware Info	rmation			
	Current Software Version:	R0.2.0		
	Server Firmware Version: Upgrade New Firmware Information:	Error		

Figure 15 - Online Upgrade

Description				
Fill in the available primary upgrade server (HTTP server)				
address.				
Fill in the available backup upgrade server (HTTP server)				
address, when the primary server is not available, device				
will send the request to backup server.				
Displays the current device software version information.				
Displays the server software version information.				
When there is a corresponding TXT file and firmware file on				
the server side, the "upgrade" button changes from gray to				
available state. Click "upgrade" to choose whether to				
upgrade or not.				
When the server side has the corresponding TXT file and				
firmware file, the new firmware information will display the				
version information in TXT.				

• The device requests TXT file to the server, the TXT file named with



vendor\_model\_hw1\_0.txt. Hw is followed by the hardware version information. All spaces in file names are changed to underlined.

- The URL requested by the device is HTTP:// server address /, and both the new version and the requested file are placed in the download directory of the HTTP server.
- The TXT file format must be UTF-8.
- Vendor\_model\_hw1\_0.txt file format is as following: Version=1.6.3 # software Version Firmware=xxx/xxx.z #xxx.z or http:// server IP: port/directory /xxx.z BuildTime = 2018.09.11 20:00 Info = TXT | XML Xxxxx Xxxxx Xxxxx Xxxxx Xxxxx

#### 9.6 System >> Auto Provision

Webpage: Login device's webpage and go to [System] >> [Auto provision].

	Information	Account	Configurations	Upgrade	Auto Provision	FDMS	Tools
> System							
> Network	Basic Settings						
> Network	CPE Serial Number: 00100400FV02001000000c383e396ab6						0
> Line	Authenticati	on Name:					0
, Line	Authenticati	on Password:					0
	Configuratio	on File Encryption Ke	ey:				0
> Intercom settings		figuration File Encr	yption Key:				0
> Call List		ail Check Times:		1			
> Call List		tact Interval:		720	(0,>=5)minute(s	5)	0
		Provision Informatio					0
Function Key		ommonConfig enab	led:				
	Enable Serv	er Digest:					0
> Security	DHCP Option >>	>					
> Device Log	DHCPv6 Option	>>					
	SIP Plug and Pl	ay (PnP) >>					
Security Settings	Static Provision	ing Server >>					
	Autoprovision N	low >>					
	TR069 >>						
			Apply				

#### Figure 16 - Auto Provision

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

PNP>DHCP>TR069> Static Provisioning



Transferring protocol: FTP/ TFTP/HTTP/HTTPS

More details, please refer to Fanvil Auto Provision.

## http://www.fanvil.com/Support/download/cid/14.html

#### Table 8 - Auto Provision

Parameters	Description		
Basic settings			
Current Configuration Version	Show the current config file's version. If the device confirm the downloaded .CFG configuration file is same with the one it uses, the device won't perform auto provision. Or if the device is matching the configuration file's context via Digest method, when the configuration file of server is modified or configuration file of device is different with server's, the device will perform provision.		
General Configuration Version	Show the common config file's version. If the device confirm the downloaded .CFG configuration file is same with the one it uses, the device won't perform auto provision. Or if the device is matching the configuration file's context via Digest method, when the configuration file of server is modified or configuration file of device is different with server's, the device will perform provision.		
CPE Serial Number	Serial number of the equipment		
Authentication Name	Configure FTP server's username, TFTP server does not require this option. When use FTP server, device uses anonymous as authentication name if user leave this option blank.		
Authentication Password	Corresponding password for FTP server.		
Configuration File Encryption Key	Encryption key for the encrypted configuration file.		
General Configuration File Encryption Key	Encryption key for encrypted common configuration file.		
Save Auto Provision Information	Configure whether to save the auto provision information or not.		
Download Fail Check Times	The default value is 5. When device fails to download configuration file, it will retry until it counts to fail check times.		
Enable Server Digest	When the feature is enabled, if the configuration file of server is changed, or device's configuration is different from server's, the		



	device will download and update.				
DHCP Option					
	The equipment supports configuration from Option 43, Option 66, or a				
Option Value	Custom DHCP option. User can select any of the three method to				
	perform auto provision, by default, the option is disabled.				
Custom Option	The custom option value should be same with the one of server, it can				
Value	be any number from 128 to 254.				
Enable DHCP	Sat the SIP server address through DHCP aption 120				
Option 120	Set the SIP server address through DHCP option 120.				
SIP Plug and Play	(PnP)				
	Whether enable PnP or not. If PnP is enable, i10 series device will				
Enable SIP PnP	send a SIP SUBSCRIBE message with broadcast method. Any server				
Enable SIF FIIF	which can support the feature will respond and send a Notify with URL				
	to phone. The device could get the configuration file with the URL.				
Server Address	Input SIP PnP server address.				
Server Port	Input SIP PnP server port.				
Transport Protocol	Select SIP PnP protocol, TCP or UDP.				
Update Interval	Configure SIP PnP message interval.				
Static Provisioning	J Server				
	Set FTP/TFTP/HTTP server IP address for auto update. The address				
	can be an IP address or Domain name, for example <u>ftp.domain.com</u> .				
Server Address	And the device supports to access server subdirectory,				
	192.168.1.1/ftp/config or ftp.domain.com/ftp/config, it means server				
	address is 192.168.1.1 or <u>ftp.domain.com</u> , file path is /ftp/config/.				
Configuration File	Input the configuration file name. If it is empty, i10 series device will				
Name	request the file which is named as its MAC address.				
Drate cel Torre	Select transportation protocol type, i10 series support				
Protocol Type	FTP/TFTP/HTTP and HTTPS				
	Set configuration file update interval time. As default it is 1, which				
Update Interval	means i10 series will check the update every 1 hour.				
	Select Provision Mode:				
11. I.C. M. I.	1. Disabled.				
Update Mode	2. Update after reboot.				
	3. Update at a time interval.				
TR069					
Enable TR069	Select it to enable TR069.				
Enable TR069	If TR069 is enabled, there will be a prompt tone when connecting to				
Warning Tone	TR069 server successfully.				



ACS Server Type	There are 4 kinds of ACS server: China Unicom, China Telecom,
ACC Server Type	common and esight.
ACS Server URL	Input ACS server address.
ACS User	Input ACS server username.
ACS Password	Input ACS server password.
	Enable/Disable TR069 Auto Login. If TR069 auto login is enabled,
TROGO Auto Login	every time user reboot the device, it will use the previous correct
TR069 Auto Login	username or password to connect ACS server instead of asking
	TR069 username or password.
STUN	Enter the STUN address
server address	
Enable the STUN	Select it to enable STUN.

# 9.7 System >> FDMS

	Information	Account	Configurations	Upgrade	Auto Provision	FDMS	Tools	Reboot Phone
> System								
> Network	FDMS Info Setti Community	-						
> Line	Building Nur Room Numb							
› Intercom settings				Ap	pply			
› Call List								

# Figure 17 - FDMS

Table 9 - FDMS

FDMS information Settings					
Community Name Name of equipment installation community.					
Building Number	Name of equipment installation building.				
Room Number	Name of equipment installation room.				

# 9.8 System >> Tools

This page provides users the tools to check the problems.



	Information	Account	Configurations	Upgrade	Auto Provision	FDMS	Tools
> System							
<ul> <li>Network</li> <li>Line</li> <li>Intercom settings</li> </ul>	Syslog Enable Sysle Server Addr Server Port: APP Log Lev Export Log:	ess: el:	0.0.0.0 514 Information				0 0 0
› Call List	Web Capture 😵		Apply				
› Function Key	Start		stop	]			
> Security	Watch Dog Enable Wato	h Dog:					
> Device Log			Apply	]			

Figure 18 - Tools

**Syslog:** When the user open syslog and set syslog server address, the log information of the device will be recorded in the syslog server during operation. If there is any problem, send the logs to Fanvil support team to analyze.

For other details, please refer to <u>10 Trouble Shooting</u>.

## 9.9 Network >> Basic

	Basic Service Por	: VPN	Advanced	
> System				
> Network	Network Mode 🎯 Network Mode:	IPv4 Only	T	
> Line	IPv4 Network Status			
› Intercom settings	IP: Subnet mask:	172.16.7.16		
› Call List	Default gateway: MAC:	172.16.7.1 0c:38:3e:39	9:6a:b6	
> Function Key	IPv4 Settings Static IP	DHCP	0	PPPoE
> Security	Enable Vendor Identifier: Vendor Identifier:	Disabled  Fanvil i10		
› Device Log	DNS Server Configured by: Primary DNS Server:	DHCP 172.16.1.1	<b>*</b>	
Security Settings	Secondary DNS Server : DNS Domain:			
		Apply		

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

Figure 19 - Network Basic Settings



Parameters	Description					
Network Mode	IPv4 only、IPv6 only、IPv4&IPv6					
Network Status						
IP	The current IP address of the equipment.					
Subnet	The current Subnet Mask of the device.					
mask	The current Subhet Mask of the device.					
Default	The current Cotowov ID address					
gateway	The current Gateway IP address.					
MAC	The MAC address of the equipment.					
Settings						
Select the app	propriate network mode. The equipment supports three kinds of network					
mode:						
Static IP	Network parameters must be entered manually and will not change. All parameters are provided by the ISP. Please contact ISP or network administrator for help if you do not know these information.					
DHCP	Network parameters are provided automatically by a DHCP server.					
PPPoE	Account and Password must be input manually. These are provided by your ISP.					
Enable	When enabled you will see the yender identifier information in the					
Vendor	When enabled, you will see the vendor identifier information in the DHCP option60 field					
Identifier						
Vendor	Support for customization. When vendor identity is enabled, you will					
Identifier	see the vendor identifier information in the DHCP option60 field					
DNS Server						
Configured	Select the Configured mode of the DNS Server.					
by						
Primary DNS Server	Enter the server address of the Primary DNS.					
Secondary DNS Server	Enter the server address of the Secondary DNS.					

#### Table 10 - Network Basic Setting

1 ) After set the parameters, click [Apply] to make settings take effect.

2) If you change the IP address, the current webpage will no longer respond, user should enter new IP address in URL to re-connect and re-login to the device's webpage.



# 9.10 Network >> Service Port

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

Veb Server Type:	HTTP V		0
Veb Logon Timeout:	15	(10~30)Minute	0
veb auto login:			
ITTP Port:	80		0
ITTPS Port:	443		0
RTP Port Range Start:	10000		0
RTP Port Quantity :	1000		0
		Apply	

# Figure 20 - Service Port

Table 11 - Service port

Parameter	Description	
Web Server Type	Reboot the device to make settings take effective. i10 series	
	supports two kinds of web login: HTTP and HTTPS.	
Web Logon Timeout	Default value is 15 minutes, when login time expires, web login will	
	exit automatically, user need to login again.	
Web auto login	If enable web auto login, after web login exits, refresh the webpage	
	to login, user does not need to input username and password.	
HTTP Port	The default value is 80. If you want more secure system	
	management, you can set other value.	
	Such as :8080, webpage login URL is: HTTP://ip:8080	
HTTPS Port	The default is 443, using method is similar to HTTP port.	
RTP Port Range Start	The value range is 1025 to 65535. The value of RTP port starts	
	from the initial value, each call, the value of voice and video port will	
	added 2.	
RTP Port Quantity	Number of calls.	



## 9.11 Network >> VPN

	Basic Service Port	VPN	Advanced			
› System						
> Network	Virtual Private Network (VPN) Sta VPN IP Address:	tus 0.0.0.0				
› Line	VPN Mode					
	Enable VPN:					0
> Intercom settings	Enable NAT:					
	L2TP: O	OpenVPN: 🔘				
› Call List	Open VPN mode:	tun 🔻				0
> Function Key	Layer 2 Tunneling Protocol (L2TP)					
	L2TP Server Address:	0.0.0				0
Security	Authentication Name:					0
	Authentication Password:					0
> Device Log						
			Apply			
Security Settings	OpenVPN Files 🕜					
	File Type	File Name	File Size			
	OpenVPN Configuration file:	client.ovpn	N/A	Select	Upload	Dele
	CA Root Certification:	ca.crt	N/A	Select	Upload	Dele
	Client Certification:	client.crt	N/A	Select	Upload	Dele

Figure 21 - VPN

Virtual Private Network (VPN) is a technology to allow the 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 activating 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 webpage.

#### L2TP

NOTICE: The device only supports non-encrypted basic authentication and non-encrypted data tunneling. For users who need data encryption, please use OpenVPN.

To establish a L2TP connection, the user should log in to the device webpage, go to page [**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 corresponding option. Press "Apply" to save changes and 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 option. 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 server automatically every time it boots up, unless user disable VPN. Sometimes, if the VPN connection does not establish immediately, user may try to reboot the device and check again.

OpenVPN

To establish an OpenVPN connection, user should get the following authentication and configuration files from the OpenVPN service 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

Select OpenVPN files and then click upload to upload these files to the device in the webpage [Network] >> [VPN]. 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 system boots up unless the user disable it manually.

	SIP SIP Hots	spot Basic Settings		
System				
Network	Line 21972@SIF •			
	Register Settings >>			
Line	Line Status:	Registered	Activate:	
	Username:	21972	Authentication User:	
Intercom settings	Display name:		Authentication Password:	
	Realm:		Server Name:	
Call List				
	SIP Server 1:		SIP Server 2:	
Function Key	Server Address:	172.16.1.2	Server Address:	
	Server Port:	5060	Server Port:	5060
Security	Transport Protocol:	UDP V	Transport Protocol:	UDP V
	Registration Expiration:	3600 second(s)	Registration Expiration:	3600 second(s)
Device Log				in the second se
	Proxy Server Address:	0	Backup Proxy Server Address:	
Security Settings	Proxy Server Port:	5060	Backup Proxy Server Port:	5060
	Proxy User:			
	Proxy Password:	0		

## 9.12 Line >> SIP



Basic Settings >>			
Enable Auto Answering:		Auto Answering Delay:	0 (0~120)second(s) 🕐
Enable Addo Answering.		Auto Answering Delay.	
Enable Hotline:			
Hotline Delay:	0 (0~9)second(s) 🕐	Hotline Number:	0
Dial Without Registered:			•
DTMF Type:	AUTO V	DTMF SIP INFO Mode:	Send 10/11 🔹 🕜
Request With Port:			
Use STUN:		Use VPN:	2
000010111		000 1111	
Enable Failback:		Signal Failback:	
Failback Interval:	1800 second(s) 🖉	-	3 (1~10) 🖉
Tonback Incorrent		bighti Heary countai	5 (1010)
Use Feature Code:			
Enable Blocking	0	Disable Blocking Anonymous	
Anonymous Call:		Call:	
Call Waiting On Code: Send Anonymous On		Call Waiting Off Code:	
Code:	0	Send Anonymous Off Code:	<b>Ø</b>
SIP Encryption:		RTP Encryption(SRTP):	Disabled 🔻 🥝
Enable Session Timer:		Session Timeout:	0 second(s)
Response Single Codec:		BLF Server:	<b>@</b>
Keep Alive Type:		Keep Alive Interval:	30 second(s) 🕜
Keep Authentication:		Blocking Anonymous Call:	
Union America		Caralifa Carala Turan	
User Agent: SIP Version:		Specific Server Type:	COMMON V
Local Port:	RFC3261 V	Anonymous Call Standard: Ring Type:	None V
Enable user=phone:	5060	Use Tel Call:	Default V
Auto TCP:		Enable PRACK:	
Enable Rport:			
Endbio Aporti			
DNS Mode:	A 🔻 🕜	Enable Long Contact:	
Enable Strict Proxy:		Convert URI:	
Use Quote in Display		Enable GRUU:	
Name: Svnc Clock Time:		Enable Use Inactive Hold:	
_,		Use 182 Response for Call	
Caller ID Header:	PAI-RPID-F V	waiting:	
Enable Feature Sync:		Enable SCA:	
CallPark Number:		Server Expire:	
TLS Version:	TLS 1.0 V	uaCSTA Number:	
Enable Click To Talk:		Enable ChangePort:	
Intercom Number:		Enable MAC Headers	
Unregister On Boot: Enable Register MAC		Enable MAC Header:	
Header:			
PTime(ms):	Disabled V	Enable Deal 180:	
Strict Branch:		Enable Crown	
		Enable Group:	
Enable RFC4475:		Enable Strict UA Match	h: 🔲 🕜
Registration Failure Retry Time	e: 32 second	(S) Local SIP Port:	5060
Enable uaCSTA:			
	Apply		

Figure 22 - SIP



Table 111 - SIP

Parameters	Description	
Register Settings	·	
Line Status	Display the current line status. To get the latest line status, user	
	has to refresh the page manually.	
Activate	Whether to activate the line or not.	
Username	Enter the username of the service account.	
Authentication User	Enter the authentication user name of the service account.	
Display Name	Enter the display name which will be sent in a call request.	
Authentication Password	Enter the authentication password of the service account.	
Realm	Enter the SIP domain provided 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 transportation protocol: TCP or UDP or TLS.	
Registration Expiration	Set SIP registration expiration time.	
SIP Server 2		
Server Address	Enter the IP or FQDN address of the SIP server	
Server Port	Enter the SIP server port, default is 5060	
Transport Protocol	Set up the SIP transportation protocol: TCP or UDP or TLS.	
Registration Expiration	Set SIP registration expiration time.	
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 username.	
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 time for incoming call before the system	
	automatically answers it.	
Enable Hotline	Enable hotline configuration, the device will dial the specific	
	number immediately once audio channel is opened.	
Hotline Delay	Set the delay time for hotline before the call sends out.	
Hotline Number	Set the hotline dialing number	



Dial Without Registered	Enable call out without registration.	
DTMF Type	Set the DTMF type for the line	
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'	
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	The time interval of detecting the availability of the main Proxy	
	using Register message.	
Signal Failback	When there are multiple proxy, whether to allow the invite/register	
C C	request to execute failback or not.	
Signal Retry Counts	When there are multiple proxy, the attempt times that the SIP	
	Request considers proxy is unavailable.	
Codecs Settings	Set the priority and availability of the codecs by adding or	
	removing them from the list.	
Advanced Settings		
Use Feature Code	When this setting is enabled, the features in this section will not be	
	handled by the device itself but by the server instead. In order to	
	control the device, the device will send feature code to the server	
	by dialing the number specified in each feature code field.	
Enable Blocking Anonymous Call	Set the feature code to dial to the server.	
Disable Blocking Anonymous Call	Set the feature code to dial to the server.	
Call Waiting On Code	Set the feature code to dial to the server.	
Call Waiting Off 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	
SIP Encryption	Enable SIP encryption, and SIP transmission will be encrypted.	
RTP Encryption	Enable RTP encryption, and RTP transmission will be encrypted.	
Enable Session Timer	Set the line to enable call ending by session timer refreshment.	
	The call session will be ended if there is no new session timer	
	event update received after the timeout period.	
Session Timeout	Set the session timer timeout period.	
Response Single Codec	If enable this option, the device will use single codec to respond to	
	incoming call request.	
BLF Server	Input BLF server address. Ordinary BLF application is that device	
	sends subscription message to SIP server. If your SIP server does	
	not support subscription, please input BLF server address to	



NAT opened.Keep Alive IntervalSet the keep alive packet transmitting interval.Keep AuthenticationKeep the previous authentication parameters.Blocking Anonymous CallReject any incoming call without presenting caller ID.User AgentSet the user agent, the default value is device model with software version.Specific Server TypeSet the line to collaborate with specific server type.SIP VersionSet the SIP version.Anonymous Call StandardSet the standard for anonymous call.Local PortSet the local port.Ring TypeSet the ring tone type for the line.Enable user=phoneIn SIP invite message, there is user=phone field.Use Tel CallEnable or disable use tel call.Auto TCPUsing TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes.Enable RportSet the line to add Rport in SIP headers.Enable PRACKSet the line to support PRACK SIP message.DNS ModeSelect DNS mode, options are A, SRV and NAPTR.Enable Long ContactEnable this will allow more parameters in contact field per RFC 3840. This option should work together with SEM server.			
NAT opened.Keep Alive IntervalSet the keep alive packet transmitting interval.Keep AuthenticationKeep the previous authentication parameters.Blocking Anonymous CallReject any incoming call without presenting caller ID.User AgentSet the user agent, the default value is device model with software version.Specific Server TypeSet the line to collaborate with specific server type.SIP VersionSet the SIP version.Anonymous Call StandardSet the local port.Ring TypeSet the local port.Ring TypeSet the ring tone type for the line.Enable user=phoneIn SIP invite message, there is user=phone field.Use Tel CallEnable or disable use tel call.Auto TCPUsing TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes.Enable RportSet the line to add Rport in SIP headers.Enable Rog ContactEnable of matching special server. When the 110 series receives packets from the server, it will reply with the source IP address, not the address in via field.Convert URIWhether to enable convert URI or not.Use Quote in Display NameWhether to add quote in display name, i.e. "Fanvil" vs Fanvil.Enable Inable Inactive HoldWith inactive hold enabled, you can see SDP is inactive in the SDP packet.Caller ID HeaderSet the Caller ID Header.Use 182 Response for CallSet the device to use 182 response code at call waiting.WaitingEnable Feature SyncEnable or disable SCA (Shared Call Appearance )			
Keep Alive Interval         Set the keep alive packet transmitting interval.           Keep Authentication         Keep the previous authentication parameters.           Blocking Anonymous Call         Reject any incoming call without presenting caller ID.           User Agent         Set the user agent, the default value is device 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 for anonymous call.           Local Port         Set the local port.           Ring Type         Set the ring tone type for the line.           Enable user=phone         In SIP invite message, there is user=phone field.           Use Tel Call         Enable or disable 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 Cong Contact         Enable this will allow more parameters in contact field per RFC 3840. This option should work together with SEM server.           Enable Strict Proxy         This is used for matching special server. When the 110 series receives packets from the server, it will reply with the source IP address, not the address in via field.           Convert URI         Whether to anable convert URI or not.	Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep	
Keep AuthenticationKeep the previous authentication parameters.Blocking Anonymous CallReject any incoming call without presenting caller ID.User AgentSet the user agent, the default value is device model with software version.Specific Server TypeSet the line to collaborate with specific server type.SIP VersionSet the SIP version.Anonymous Call StandardSet the standard for anonymous call.Local PortSet the local port.Ring TypeSet the ring tone type for the line.Enable user=phoneIn SIP invite message, there is user=phone field.Use Tel CallEnable or disable use tel call.Auto TCPUsing TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes.Enable RportSet the line to add Rport in SIP headers.Enable PRACKSet the line to support PRACK SIP message.DNS ModeSelect DNS mode, options are A, SRV and NAPTR.Enable Long ContactEnable this will allow more parameters in contact field per RFC 3840. This option should work together with SEM server.Enable Strict ProxyThis is used for matching special server. When the i10 series receives packets from the server, it will reply with the source IP address, not the address in via field.Convert URIWhether to enable Convert URI or not.Use Usot TimeTime Sync with server.Enable Inactive HoldWith inactive hold enabled, you can see SDP is inactive in the SDP packet.Caller ID HeaderSet the Caller ID Header.Use 182 Response for CallSet the Caller ID Header.Use 182 Response		NAT opened.	
Blocking Anonymous Call       Reject any incoming call without presenting caller ID.         User Agent       Set the user agent, the default value is device 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 for anonymous call.         Local Port       Set the local port.         Ring Type       Set the ing tone type for the line.         Enable user=phone       In SIP invite message, there is user=phone field.         Use Tel Call       Enable or disable 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 Long Contact       Enable this will allow more parameters in contact field per RFC 3840. This option should work together with SEM server.         Enable Strict Proxy       This is used for matching special server. When the i10 series receives packets from the server, it will reply with the source IP address, not the address in via field.         Convert URI       Whether to enable convert URI or not.         Use Quote in Display Name       Whether to add quote in display name, i.e. "Fanvil" vs Fanvil.         Enable GRUU       Enable Globally Routable User-Agent URI (GRUU) or not.         Sync Clock Time<	Keep Alive Interval	Set the keep alive packet transmitting interval.	
User Agent         Set the user agent, the default value is device 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 for anonymous call.           Local Port         Set the local port.           Ring Type         Set the ring tone type for the line.           Enable user=phone         In SIP invite message, there is user=phone field.           Use Tel Call         Enable or disable 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 support PRACK SIP message.           DNS Mode         Select DNS mode, options are A, SRV and NAPTR.           Enable Long Contact         Enable this will allow more parameters in contact field per RFC 3840. This option should work together with SEM server.           Enable Strict Proxy         This is used for matching special server. When the i10 series receives packets from the server, it will reply with the source IP address, not the address in via field.           Convert URI         Whether to add quote in display name, i.e. "Fanvil" vs Fanvil.           Enable Globally Routable User-Agent URI (GRUU) or not.         Sync Clock Time           Sync Clock Time         Time Sync with server.           En	Keep Authentication	Keep the previous authentication parameters.	
version.Specific Server TypeSet the line to collaborate with specific server type.SIP VersionSet the SIP version.Anonymous Call StandardSet the standard for anonymous call.Local PortSet the local port.Ring TypeSet the ring tone type for the line.Enable user=phoneIn SIP invite message, there is user=phone field.Use Tel CallEnable or disable use tel call.Auto TCPUsing TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes.Enable RportSet the line to add Rport in SIP headers.Enable PRACKSet the line to support PRACK SIP message.DNS ModeSelect DNS mode, options are A, SRV and NAPTR.Enable Long ContactEnable this will allow more parameters in contact field per RFC 3840. This option should work together with SEM server.Enable Strict ProxyThis is used for matching special server. When the 100 series receives packets from the server, it will reply with the source IP address, not the address in via field.Convert URIWhether to add quote in display name, i.e. "Fanvil" vs Fanvil.Enable GRUUEnable Globally Routable User-Agent URI (GRUU) or not.Sync Clock TimeTime Sync with server.Enable Inactive HoldWith inactive hold enabled, you can see SDP is inactive in the SDP packet.Caller ID HeaderSet the Caller ID Header.Use 182 Response for Call waitingSet the device to use 182 response code at call waiting.Enable Feature SyncEnable or disable Feature Sync with server.Enable Feature SyncEnable or disable Feat	Blocking Anonymous Call	Reject any incoming call without presenting caller ID.	
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SIP Version       Set the SIP version.         Anonymous Call Standard       Set the standard for anonymous call.         Local Port       Set the local port.         Ring Type       Set the ring tone type for the line.         Enable user=phone       In SIP invite message, there is user=phone field.         Use Tel Call       Enable or disable use tel call.         Auto TCP       Using TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes.         Enable PRACK       Set the line to add Rport in SIP headers.         Enable Long Contact       Enable this will allow more parameters in contact field per RFC 3840. This option should work together with SEM server.         Enable Strict Proxy       This is used for matching special server. When the i10 series receives packets from the server, it will reply with the source IP address, not the address in via field.         Convert URI       Whether to anable convert URI or not.         Use Quote in Display Name       Whether to add quote in display name, i.e. "Fanvil" vs Fanvil.         Enable Inactive Hold       With inactive hold enabled, you can see SDP is inactive in the SDP packet.         Caller ID Header       Set the Caller ID Header.         Use 182 Response for Call       Set the device to use 182 response code at call waiting.         waiting       Enable or disable Feature Sync with server.         Enable Feature Sync       Enable or disabl		version.	
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Enable user=phoneIn SIP invite message, there is user=phone field.Use Tel CallEnable or disable use tel call.Auto TCPUsing TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes.Enable RportSet the line to add Rport in SIP headers.Enable PRACKSet the line to support PRACK SIP message.DNS ModeSelect DNS mode, options are A, SRV and NAPTR.Enable Long ContactEnable this will allow more parameters in contact field per RFC 3840. This option should work together with SEM server.Enable Strict ProxyThis is used for matching special server. When the i10 series receives packets from the server, it will reply with the source IP address, not the address in via field.Convert URIWhether to enable convert URI or not.Use Quote in Display NameWhether to add quote in display name, i.e. "Fanvil" vs Fanvil.Enable Inactive HoldWith inactive hold enabled, you can see SDP is inactive in the SDP packet.Caller ID HeaderSet the Caller ID Header.Use 182 Response for Call waitingSet the device to use 182 response code at call waiting.Enable Feature SyncEnable or disable Feature Sync with server.Enable SCAEnable or disable SCA (Shared Call Appearance )	Local Port	Set the local port.	
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Caller ID HeaderSet the Caller ID Header.Use 182 Response for Call waitingSet the device to use 182 response code at call waiting.Enable Feature SyncEnable or disable Feature Sync with server.Enable SCAEnable/Disable SCA (Shared Call Appearance )	Enable Inactive Hold	With inactive hold enabled, you can see SDP is inactive in the	
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waitingEnable Feature SyncEnable SCAEnable/Disable SCA (Shared Call Appearance )	Caller ID Header	Set the Caller ID Header.	
Enable Feature SyncEnable or disable Feature Sync with server.Enable SCAEnable/Disable SCA (Shared Call Appearance )	Use 182 Response for Call	Set the device to use 182 response code at call waiting.	
Enable SCA Enable/Disable SCA (Shared Call Appearance )	waiting		
	Enable Feature Sync	Enable or disable Feature Sync with server.	
CallPark Number Set the CallPark number.	Enable SCA	Enable/Disable SCA (Shared Call Appearance )	
	CallPark Number	Set the CallPark number.	



Server Expire	Set the timeout of using the server.		
TLS Version	Choose TLS Version.		
uaCSTA Number	Set uaCSTA Number.		
Enable Click To Talk	This is used to match special server, click to call out directly after		
	enable this option.		
Enable Change port	Whether to enable change port or not.		
Intercom Number	Set intercom number.		
Unregister On Boot	Whether to enable logout function.		
Enable MAC Header	Whether to enable MAC header. When enable, there is MAC		
	information in SIP packet and user agent when register		
BLF Dialog Strict Match	Whether to enable accurate matching of BLF sessions.		
PTime(ms)	Set whether to bring ptime field, by default it is no.		
Enable Deal 180	Enable: after receives183+ SDP, device will play ivr; and after		
	receives180, device will play local tone.		
	Disable: after receives 183+ SDP, device will play ivr. After		
	receives180, device does not play local tone.		
SIP Global Settings			
Strict Branch	Enable or disable this to strictly match the Branch field.		
Enable Group	Enable or disable SIP group server function as server backup.		
Enable RFC4475	Enable or disable RFC4475.		
Enable Strict UA Match	Enable strict UA matching.		
Registration Failure Retry	Set the registration failure retry time.		
Time			
Local SIP Port	Modify the device SIP port.		
Enable uaCSTA	Set to enable the uaCSTA function.		

#### 9.13 Line >> SIP Hotspot

SIP hotspot is a simple and practical function. It is simple to configure, which can realize the function of group vibration and expand the number of SIP accounts. Please check **8.3 Hotspot** for more details.

#### 9.14 Line >> Basic Settings

STUN -Simple Traversal of UDP through NAT -A STUN server allows the device in a private network to know its public IP and port as well as the type of NAT being used. The equipment can use this information to register itself to a SIP server so that it can receive calls from public



network while it is in a private network.

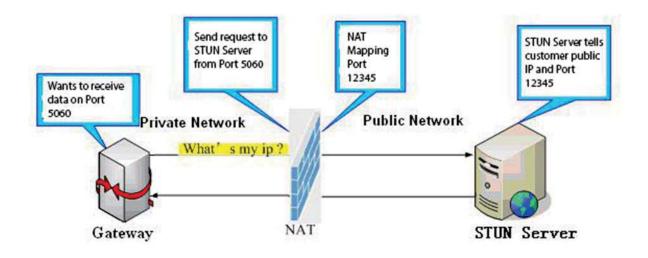


Figure 23 - Network Basic

SIP SIP Hotspot	Basic Settings	
STUN Settings		
STUN NAT Traversal:	FALSE	
Server Address:		
Server Port:	3478	
Binding Period:	50	second(s)
SIP Waiting Time:	800	millisecond
	Apply	
SIP P2P Settings		
Enable Auto Answering	I.	
Auto Answering Delay:	0	(0~120)second(s)
DTMF Type:	RFC2833 •	
DTMF SIP INFO Mode:	Send 10/11	

Figure 24 - Line Basic Setting

Table 12 - Line Basic Setting

Parameters	Description
STUN Settings	



Server Address	Input the STUN server address.
Server Port	Input the STUN server port, default is 3478.
Binding Period	Set the STUN binding period which can be used to keep the NAT
	open.
SIP Waiting Time	Set the timeout of STUN binding before sending SIP messages.
SIP P2P Settings	
Enable Auto Answering	Enable timeout to automatically answer IP calls
Auto Answering Delay	Automatic answer timeout setting
DTMF Type	Set the DTMF type of the line.
DTMF SIP INFO Mode	Set up SIP INFO mode to send '*' and '#' or '10' and '11'

# 9.15 Intercom Setting >> Features

	Features Media Settings	MCAST	Action Time/Date	Tone
› System				
› Network	Basic Settings >> Enable Call Waiting:	✓ Ø		
> Line	Enable Auto on Hook: Enable Silent Mode:	• •	Auto HangUp Delay: Disable Mute for Ring:	3 (0~30)second(s) 🕜
> Intercom settings	Ban Outgoing: Enable Restricted Incoming	•		
> Call List	List: Enable Restricted Outgoing List:	<ul><li>✓ ②</li><li>: ✓ ②</li></ul>	Enable Country Code:	
> Function Key	Country Code:		Area Code: P2P IP Prefix:	
> Security	Restrict Active URI Source IP:		Push XML Server:	
> Device Log	Line Display Format: Call Number Filter:	xxx@SIPn ▼	Auto Resume Current:	
Security Settings	Tone Settings >>			
	Intecom Settings >>			
	Response Code Settings >>		Apply	

## Figure 25 - Intercom Setting

## Table 13 - Intercom Setting

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow user to take second incoming call
	during an established call. By default it is enabled.
Enable Auto Onhook	Enable auto onhook or not. If enable, the device will hang up the
	call and return to the idle status automatically.



Auto Onhook Time	Specify Auto Onhook time, the device will hang up and return to
	the idle automatically after Auto onhook time.
Enable Silent Mode	When enabled, the device is muted, there is no ringing when calls,
	you can use the volume keys and mute key to unmute.
Disable Mute for Ring	Disable the mute mode, if this option is clicked, mute button on
	device does not take effect.
Ban Outgoing	Enable or disable ban outgoing, if enable, the device can not dial
	out any number.
Enable Restricted Incoming List	Whether to enable restricted incoming call list.
Enable Restricted Outgoing List	Whether to enable the restricted outgoing list.
Enable Country Code	Whether to enable the country code.
Country Code	Fill in the country code.
Area Code	Fill in the area code.
Allow IP Call	If enabled, user can dial out with IP address.
P2P IP Prefix	Set prefix for point-to-point IP calls.
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP
	address. Notice: this function is usually used to manage device.
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.
Line Display Format	Custom line format: SIPn or SIPn:xxx or xxx@SIPn
Call Number Filter	Configure a special ampersand, the called number is 78-9, the
	ampersand will be filtered when device sends the call out.
Auto Resume Current	Automatically break HOLD if current call changes.
Tone Settings	
Enable Holding Tone	Whether to enable call holding tone.
Enable Call Waiting Tone	Whether to enable call waiting tone.
Play Dialing DTMF Tone	Play DTMF tone on the device when user presses a digits wehen
	dial the call, by default it is enabled.
Play Talking DTMF Tone	Play DTMF tone on the device when user presses a phone digits
	during taking, by default it is enabled.
Intercom Settings	
Enable Intercom	When intercom is enabled, the device will accept the incoming call
	which requests with a SIP header of Alert-Info automatically.
Enable Intercom Mute	Enable mute mode during the intercom call.
Enable Intercom Tone	If the incoming call is intercom call, the device plays the intercom



	tone.
Enable Intercom Barge	While enable intercom barge, the device will auto answer the
	intercom call during a call. If the current call is intercom call, the
	device will reject the second intercom call.
Response Code Settings	
Busy Response Code	Set the SIP response code when line is busy.
Reject Response Code	Set the SIP response code when device reject one call.

# 9.16 Intercom Setting >> Audio

	Features Media Settings	MCAST	Action	Time/Date	Tone	
› System						
> Network	Codecs Settings >> 😢					
7 Network	Media Settings >>					
> Line	Default Ring Type:	1.wav 🔻 🕜				
_	Speakerphone Volume:	5	(1~9) 🕜			
> Intercom settings	Speakerphone Ring Volume:	3	(0~9) 🕜			
Sec.	G.723.1 Bit Rate:	6.3kb/s 🔻 🔮		AMR Payload Type:	108 (9	96~127) 🕜
> Call List	DTMF Payload Type:	101	(96~127) 🕜			
	OPUS Payload Type:	107	(96~127)	OPUS Sample Rate	OPUS-NB(I T	
> Function Key	ILBC Payload Type:	97	(96~127) 🕜	ILBC Payload Length	20ms 🔻 🕜	
	Enable VAD:					
> Security	RTP Control Protocol(RTCP) Setti	ings >>				
	RTP Settings >>					
> Device Log						
	Alert Info Ring Settings >>					
Security Settings			Apply			

# Figure 26 - Media Setting

# Table 14 - Media Setting

Parameter	Description		
Codecs Settings	Select enable or disable voice codecs:		
	G.711A/U, G.722, G.729AB, iLBC, opus.		
Media Settings			
Default Ring Type	Configure default ringtones. If no special ringtone is set, the default		
	ringtone will be used.		
Speakerphone Volume	Set the speaker volume, value can be 1~9.		
Speakerphone Ring Volume	Set the speaker ring volume, value can be 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 is 96~127.		
Opus playload type	Set Opus load type, range is 96~127.		



OPUS Sample Rate	Set Opus sampling rate, including opus-nb (8KHz) and opus-wb (16KHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length.
Enable VAD	Whether to enable voice activity detection.
RTP Control Protocol(RTC	P) Settings
CNAME user	Set CNAME user
CNAME host	Set CNAME host
RTP Settings	
RTP keep alive	Hold the call and send the packet every 30s.
Alert Info Ring Settings	
Value	Set the value to specify the ring type.
Ring Type	Select ring type.

# 9.17 Intercom Setting >> MCAST

It is easy and convenient to use multicast function to send notice to each member of the multicast group via setting the multicast key on the device and sending multicast RTP stream to pre-configured multicast address. By configuring monitoring multicast address on the device, the device will receive multicast from the configured monitoring multicast address.

	Features Media Settings	MCAST Act	tion Time/Date	Tone	
> System					
> Network	MCAST Listening	1			
> Line	Priority: Enable Page Priority: Enable Prio Chan:				
Intercom settings	Enable Emer Chan: Index/Priority	Name	Host:por		Channel
> Call List	1 2				T
> Function Key	3 4 5				۲
> Security	6				۲
> Device Log	8			0	
Security Settings	10	Apply		0	•

Figure 27 - MCAST



#### 9.18 Intercom Setting >> Action

#### **Action URL Event Settings**

Set URL for the device to report its action to server. These actions are recorded and sent as xml files to the server. Sample format is http://InternalServer /FileName.xml.

(Internal Server: The IP address of server; File Name: the device's xml file used to report action.)

#### Table 15 - action URL

# Notice: The operation URL is used by the IPPBX system to submit device events. Please refer to the details Fanvil Action URL.

http://www.fanvil.com/Uploads/Temp/download/20190122/5c46debfbde37.pdf

### 9.19 Intercom Setting >> Time/Date

Users can configure the device's time settings on this page.

	Features Media Settings	MCAST	Action	Time/Date	Tone	
› System						
> Network	Network Time Server Settings Time Synchronized via SNTP	•				٩
› Line	Time Synchronized via SNTP Time Synchronized via DHCP Time Synchronized via DHCPv6					0
Intercom settings	Primary Time Server Secondary Time Server	0.pool.ntp.org time.nist.gov				0
<ul> <li>Call List</li> </ul>	Time zone Resync Period	(UTC+8) Beijing,Sin	ngapore,Perth,Irkuts ▼ second(s)	]		0
Function Key	Time/Date Format	_				
> Security	12-hour clock Time/Date Format	DD MMM WW	▼ 30 JUL TUE			
> Device Log						
	Daylight Saving Time Settings					
› Security Settings	Location DST Set Type	None Disabled	<b>v</b>			
		Apply				
	Manual Time Settings					
	2019-7-30 9	▼ 11 <b>▼</b>		Apply		

Figure 28 - time/date

Table 16 - time/date

Parameter	Description			
Network Time Server Setting	IS			
Time Synchronized via SNTP	Enable time-sync through SNTP protocol.			
Time Synchronized via				
DHCP	Enable time-sync through DHCP protocol.			
Time Synchronized via	Enable time-sync through DHCPv6 protocol.			



DHCPv6	
Primary Time Server	Set primary time server address.
	Set secondary time server address, when primary server is not
Secondary Time Server	reachable, the device will try to connect to secondary time
	server to get time synchronization.
Time zone	Select the time zone.
Resync Period	Time interval of re-synchronization with time server
Time/Date Format	
12-hour clock	Enable or disable 12-hour clock.
Time/Date Format	Set time/date format.
Daylight Saving Time Setting	js
Location	Select the user's time zone specific area
DST Set Type	Select DST type, and set the DST rules.
Fixed Type	Select DST fixed type.
Offset	The DST offset time.
Month Start	The DST start month.
Week Start	The DST start week.
Weekday Start	The DST start weekday.
Hour Start	The DST start hour.
Month End	The DST end month.
Week End	The DST end week.
Weekday End	The DST end weekday.
Hour End	The DST end hour.
Manual Time Settings	
Manual Time Settings	Set time manually, please disable SNTP service first.

# 9.20 Intercom settings >> Tone

User can set device's tone in this page.

You can select the corresponding country and use the settings directly, or select custom and set the tone manually.



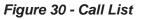
	Features Media Settings	MCAST Action Time/Date Tone	
› System			
> Network	Tone Settings		
	Select Your Tone:	Australia	<b>v</b> 🕜
› Line	Dial Tone:	350+440/0	0
, Line	Ring Back Tone:	440+480/2000,0/4000	0
	Busy Tone:	480+620/500,0/500	0
Intercom settings	Congestion Tone:		0
	Call waiting Tone:	440/300,0/10000,440/300,0/10000,0/0	0
› Call List	Holding Tone:		0
	Error Tone:		0
> Function Key	Stutter Tone:		0
	Information Tone:		0
> Security	Dial Recall Tone:	350+440/100,0/100,350+440/100,0/100,350+440/100,0/100,350+440/0	0
	Measage Tone:		0
› Device Log	Howler Tone:		0
-	Number Unobtainable Tone:	400/500,0/6000	0
› Security Settings	Warning Tone:	1400/500,0/0	0
- Security Settings	Auto Answer Tone:		0
		Apply	

Figure 29 - Tone

## 9.21 Call List >> Call List

Call List Web Dia > System Restricted Incoming Calls Network Add Delete Delete All Caller Number Line > Line 513 ALL Restricted Outgoing Calls Intercom settings Add Delete Delete All > Call List Caller Number Line 511 ALL Function Key Security > Device Log Security Settings

User can set restricted incoming calls list and restricted outgoing calls list in this page.



#### Restricted Incoming Calls:

The function is same with blacklist. Add the numbers in restricted incoming calls list, the device will reject all the calls from these blacklist numbers, unless user deletes the numbers from the list.

User can add both numbers and prefix in the restricted incoming calls list, the i10 series device will reject all the calls from the blacklist numbers or calls from numbers with blacklist prefix.

Restricted Outgoing Calls:

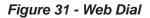


Add numbers to restricted outgoing calls list, the i10 series device will end the calling when user dial these numbers, unless user removes the numbers from the list.

#### 9.22 Web Dial

In this page, user can make calls, answer the calls or hang up the calls.

	Call List Web Dial
› System	
› Network	Web Dial Settings
> Line	Dial Answer Hang-up
› Intercom settings	
> Call List	
> Function Key	
> Security	
> Device Log	
› Security Settings	



# 9.23Function Key

> System									
(1993) - A	Function Key S	ettings >>							
> Network	Key	Туре	Name	Value	Value2	Subtype	3	Line	
194 - C	DSS Key 1	Memory Key 🔻	callc600	513		Speed Dial	۲	512@SIP1	
Line	DSS Key 2	Memory Key 🔻	calli20s	511		Speed Dial	٣	512@SIP1	,
	DSS Key 3	None 🔻				None	¥	AUTO	
Intercom settings				Apply					
Call List	Programmable	Key Settings 🔞 >>	•						
	Advanced Setti	ings >>							
Function Key									
<ul> <li>Security</li> <li>Device Log</li> </ul>									
<ul> <li>Security Settings</li> </ul>									
Advanced Cettinger S.S.									
Advanced Settings >>									
Dial Mode Select	Main-Se	econdan 🔻							
Call Switched Time	16	(5~50)second	(s)						
First Number Start Time	06:00	(00:00~23:59	)) Fi	rst Number End 1	Fime 18:	00 (00:	00^	23:59)	
			Apply						



Key	Desktop	Ringing	Talking	Desktop Long Pressed
Key1	Dsskey1(callc600) V	Answer 🔻	End 🔻	Main Menu 🔻
Key2	Dsskey2(calli20s) T	End 🔻	Volume Down	None 🔻
Key3	None 🔻	Answer 🔻	End 🔻	None 🔻

# Figure 32 - Function Key

Table 17 - Function Key

Parameters	Description					
Function Key Settings						
	Speed dial: User can set one number or IP address in Value					
	option. It is convenient to use this function to make calls to					
Memory Key	specified numbers/IP address used continually.					
	Intercom: The intercom function make the operator or secretary to					
	answer calls directly, which is popular in office.					
Kov Event	Use key event function to activate one application directly.					
Key Event	Example: None/Handfree.					
DTMF	Send the DTMF directly with the corresponding settings.					
MCAST Paging	Set paging IP address and voice codec, user can initiate paging					
MCAST Paging	directly by pressing the button.					
Action URL	User can use specified URL to make calls or open the doors.					
MCASTListoping	When device is idle, use the MCAST listening key to monitor the					
MCAST Listening	MCAST from the paging IP address user set.					
Programmable Key Setting	s					
	None: None.					
Dealstan	Dsskey1: Call out or pick up calls according to dsskey1's settings.					
Desktop	Dsskey2: Call out or pick up calls according to dsskey2's settings.					
	Dsskey3: Call out or pick up calls according to dsskey3's settings.					
	Answer: When there is one incoming call and auto answer is					
Ringing	disabled, use this key to pick up the call.					
	End: When there is one incoming call, use this key to end the call.					
	End: Press the key to end the call when device is in one call.					
	Volume Up: Press the key to increase volume when the device is					
	in one call.					
Talking	Volume Down: Press the key to decrease volume when the device					
	is in one call.					
	Dsskey1: Call out or pick up calls according to dsskey1's settings.					
	Dsskey2: Call out or pick up calls according to dsskey2's settings.					



	Dsskey3: Call out or pick up calls according to dsskey3's settings.
	None: None.
Dealstan Lang proceed	Main Menu: Long press the key to make device go to command
Desktop Long pressed	mode, details please check chapter 5.2.1 Common command
	mode.
Advanced Settings	
	Set the dial mode between calling 1st number and calling 2nd
	number.
	Main-Secondary: If the 1 <sup>st</sup> number does not pick up the call in
Dial Mode Select	specified time, then the device will call 2 <sup>nd</sup> number.
	Time Period: Device check the system time and send the call to $1^{st}$
	number in 1 <sup>st</sup> number's time period, or the device will send call to
	the 2 <sup>nd</sup> number.
Call Switched Time	Set the switched time between 1 <sup>st</sup> number and 2 <sup>nd</sup> number when
	device calls out, by default the value is 16 seconds.
First Number Start Time	Set 1 <sup>st</sup> called number's start time, by default it is 06:00am.
First Number End Time	Set 1 <sup>st</sup> called number's end time, by default it is 18:00pm.

## > Key Event

The speed dial key type could be set as Key Event.

Key	Туре	Name	Value	Value2	Subtype	2	Line	
DSS Key 1	Key Event 🔹				None	۳	512@SIP1	۷
DSS Key 2	DTMF .	calli20s	511		None Handfree		512@SIP1	٣
DSS Key 3	Key Event 🔻	1			None	•	AUTO	٣

Figure 33 - Function Key Settings

Table 18 - Function Key Settings

Туре	Subtype	Usage
	None	No responding
Key Event	Handfree	Handfree

### > Memory Key

When the speed dial key set as Memory Key, the device would dial preset telephone number. This button can also be used to set the IP address: you can press the speed dial button to directly make an IP call.



Key	Туре	Name	Value	Value2	Subtype	Line
DSS Key 1	Memory Key 🔻	callc600	513	172.18.60.142	Speed Dial	512@SIP1
DSS Key 2	Memory Key 🔻	calli20s	511		None Speed Dial	512@SIP1
DSS Key 3	Key Event 🔻				Intercom	AUTO

Figure 34 - Memory Key Settings

Table 19 - Memory Key Settings

Туре	Number	Line	Subtype	Usage
		TI OID	Speed	Set speed dial, press the key to call out
	Fill the called		Dial	the number.
Hot Key	party's SIP account or IP address	account correspondi ng lines	Intercom	In Intercom mode, if the caller's IP phone supports Intercom feature, the device can automatically answer the
				Intercom calls

#### > Multicast

Multicast function is to deliver voice streams to configured multicast address; all equipment monitored the multicast address can receive and play the broadcasting. Using multicast functionality would make deliver voice one to multiple which are in the multicast group simply and conveniently.

The DSS Key multicast web configuration for calling party is as follow:

Key	Туре	Name	Value	Value2	Subtype	Э	Line	
DSS Key 1	Memory Key 🔻	callc600	513	172.18.60.142	Speed Dial	۳	512@SIP1	۲
DSS Key 2	MCAST Paging V	calli20s	224.0.0.5:3356		G.711U	٣	512@SIP1	۲
DSS Key 3	Key Event V				G.711U G.711A		AUTO	٧
			Apply		G.729AB			
			Apply		opus G.722			

Figure 35 - Multicast Settings

Table 20 - Multicast Settings

Туре	Number	Subtype
		G.711U
Multicast		G.711A
	Set the host IP address and port number, they must be separated by a	G729AB
	colon (The IP address range is 224.0.0.0 to 239.255.255.255, and the	iLBC
	port number is preferably set between 1024 and 65535).	opus
		G.722



## 9.24 Security >> Web Filter

	Web Filter Trust Certificates Device Certi	ficates Firewall	
› System			
> Network	Web Filter Table 🥝 Start IP Address	End IP Address	Option
› Line	192.168.1.1	192.168.254.254	Modify Delete
› Intercom settings	Web Filter Table Settings		
› Call List		End IP Address	Add
› Function Key	Web Filter Setting 🕑		
> Security	Enable Web Filter	Apply	

In this page, user can set the IP address segment which is allowed to access the device.

#### Figure 36 - Multicast Settings

Add or delete the allowed IP address segment. Please input start IP address in Start IP address option, and input end IP address in End IP address option and click Apply to save the settings. Click Delete to remove the corresponding IP address segment.

Enable Web Filter: Enable or disable web filter, select it and click apply to save the settings. Notice: Please remove your PC's IP address from the filter IP address list, or your PC is not able to access the device's webpage.

#### 9.25 Security >> Trusted Certificates

User can upload or delete the trusted certificates in this page.



	Web Filter Trust Cert	ificates Device Certificates	Firewall		
>> System					
> Network	Permission Certificate				
→ Line	Permission Certificate Common Name Validatio	Disabled Disabled	• Ø • Ø		
> Intercom settings	Certificate mode	All Certificates	v 😢		
→ Call List	Import Certificates 💜				
> Function Key	Load Server File		Select	Upload	
Security	Certificates List				
› Device Log	Index File Nam	e Issued To	Is	sued By Expi	iration File Size Delete
> Security Settings					

Figure 37 - Trusted Certificates

## 9.26Security >> Device Certificates

Select the device certificate to use default certificates or custom certificate. You can upload and delete uploaded certificates.

	Web Filter Trust Certificat	tes Device Certificates	Firewall		
> System					
> Network	Device Certificates 🥹				
) Line	Device Certificates	Default Certificates	(existence)		
› Intercom settings	Import Certificates 💡				
› Call List	Load Server File		Select Upload		
Function Key	Certification File 💈				
> Security	File Name	Issued To	Issued By	Expiration	File Size Delete
> Device Log					
<ul> <li>Security Settings</li> </ul>					

Figure 38 - Device Certificates



## 9.27Security >> Firewall

	Web Filter Trust Certificates Device Certificates Firewall
› System	
> Network	Firewall Type 📀
› Line	Apply
› Intercom settings	Firewall Input Rule Table 🥝
› Call List	Index Deny/Permit Protocol Src Address Src Mask Src Port Range Dst Address Dst Mask Dst Port Range
› Function Key	Firewall Output Rule Table 🥝 Index Deny/Permit Protocol Src Address Src Mask Src Port Range Dst Address Dst Mask Dst Port Range
> Security	Firewall Settings 🔞
> Device Log	Input/Output     Input •     Src Address       Deny/Permit     Deny •     Src Mask         Dst Mask     Add
› Security Settings	Protocol UDP   Src Port Range Dst Port Range
	Input/Output     Input     Index To Be Deleted     Delete

#### Figure 39 - Firewall

In this page, user can select whether to enable input or output firewall, and set the detailed rules. These settings are used to prevent illegal network access, limit the internal user to access Internet sources, enhance the security.

Table	21	- Fi	rewall
-------	----	------	--------

Parameters	Description					
Firewall Type						
Enable Input Rules	Enable or disable input rules.					
Enable Output Rules	Enable or disable output rules.					
Input/Output	Set the rule to be input rule or output rule.					
Deny/Permit	Set the rule to be denying rule or permitting rule.					
Protocol	Select the firewall protocol, options are UDP, TCP and ICMP.					
Src Address	Input source address. The source IP can be one host address or					
	network address, you can input 0.0.0.0 to represent all the IF					
	addresses, or input one *.*.*.0 network IP, like 192.168.1.0.					
Src Mask	Input source mask. If user configure source mask to be					
	255.255.255.255, the source IP should be one detailed IP address; if					
	user configure source mask to be 255.255.255.0, the source IP					
	contains a segment of IP addresses.					
Src Port Range	Input source port range.					
Dst Address	Input destination address. The destination IP can be one specified IP					



	address, or 0.0.0.0 which represents all the IP addresses, or network				
	IP address *.*.*.0, like 192.168.1.0.				
Dst Mask	Input destination mask. If user configure destination mask to be				
	255.255.255.255, the destination IP should be one detailed IP				
	address; if user configure source mask to be 255.255.255.0, the				
	destination IP contains a segment of IP addresses.				
Dst Port Range	Input destination port range.				

Input the parameters and click Add, the new rules will be added to the firewall list, for example:

Fire	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	icmp	192.168.1.14	255.255.255.0	1-1023	192.168.1.118	255.255.255.0	2-1024

Select Input/Output rule, and enter the index of the rule in Index to Be deleted option, click delete then the correponding rule will be removed.

Rule Delete Option 🕜				
Input/Output	Input 🔻	Index To Be Deleted	1	Delete

### 9.28 Device Log

In this page, user can get the device's logs. When device works abnormally, user can get the logs and send to Fanvil support team. Details please check chapter <u>10.5 Get Log Information</u>.

## 9.29 Security Settings

asic Settings				
Ringtone Duration:			5 (1~600)s	
Input & Tamper Ser	ver Address:			0
Message:Alarm_Inf	o:Description=;SI	P User=512;Mac=0c:38	:3e:3a:06:65;IP=172.18.6	0.192;port=Input
			Apply	
nput Settings >>				
	Low Level Trigger/C	lose Triager)		
			Triggered Rig	ngtone: 2 way
			Apply	
	Ringtone Duration: Input & Tamper Ser	Ringtone Duration: Input & Tamper Server Address: Message:Alarm_Info:Description=;SI nput Settings >> Input1: Triggered By: Low Level Trigger(C	Ringtone Duration: Input & Tamper Server Address: Message:Alarm_Info:Description=;SIP User=512;Mac=0c:38 mput Settings >> Input Settings >> Input : Triggered By: Low Level Trigger(Close Trigger) •	Ringtone Duration:       5 (1~600)s         Input & Tamper Server Address:



tput Settings >>		
Triggered By DTMF Rin	gTone:	2.wav T
Triggered By URI Ringt	one:	2.wav T
Triggered By SMS Ring	tone:	2.wav
Triggered By Dsskey Ri	ingtone:	None
Output1:		
Standard Status:	NC:closed V	Output Duration: 5 (1~600)s
Output Trigger Mode:	Trigger By DTMF	DTMF Trigger Code: 1234
		DTMF Reset Code: 4321
		Reset By: By Duration <b>v</b>
	Trigger By Active URI	Trigger Message: OUT1_SOS
		Reset Message: OUT1_CLR
	Trigger By SMS	Trigger Message: ALERT=OUT1_SOS
		Reset Message: ALERT=OUT1_CLR
	Trigger By Input:	Input1
	Trigger By Call State:	Talking Ringing Calling
	Trigger By DssKey:	None 🔻
		Apply
mper Alarm Setti	ings >>	
Enable Tamp	er Alarm	
Alerm command	Tamp	per_Alarm
Reset command	Tamp	per_Reset Reset
Alerm Ringtone	NON	NE V
		Apply
		بالمطب

Figure 40 - Security Settings

Table 22 - Security Settings

Security Settings			
Parameters	Description		
Basic Settings			
Ringtone Duration	Set the ringtone duration, default value is 5 seconds.		
	Set remote server address. The device will send message to the		
Input & Tamper	server when the alarm is triggered. The message format is :		
Server Address	Alarm_Info:Description=i10;SIP User=;Mac=0c:38:3e:3a:06:65;IP=;		
	port=Input.		
Input settings			
Input Detect	Enable or disable Input Detect		
	When choosing the low level trigger (closed trigger), detect the input		
Triggered by	port (low level) closed trigger.		
	When choosing the high level trigger (disconnect trigger), detect the		
	input port (high level) disconnected trigger.		
	Send SMS: Set the alert message send to server if selected.		
Triggered Action	Dss Key: The device will perform corresponding Dss Key		
	configurations if any key is selected, by default the value is none.		
	Triggered Ringtone: Select triggered ring tone.		



Output Settings		
Output Response	Enable or disable Output Response	
Triggered by DTMF	Select the DTMF trigger ring tone.	
Ring tone		
Triggered by URI	Select the URI trigger ring tone.	
Ringtone		
Triggered By SMS	Select the SMS trigger ring tone.	
Ringtone		
Triggered By Dsskey	Select the Dsskey trigger ring tone.	
Ringtone		
	When choosing the low level trigger (NO: normally open), when meet	
Standard Status	the trigger condition, trigger the NO port disconnected.	
	When choosing the high level trigger (NC: normally close), when meet	
	the trigger condition, trigger the NC port close.	
Output Duration	Set the output change duration time, the default is 5 seconds.	
	Enable or disable trigger by DTMF. The device will check the received	
Trigger by DTMF	DTMF sent by remote device, if it matches the DTMF trigger code, the	
	device will trigger corresponding output port.	
DTMF Trigger Code	Input the DTMF trigger code, default value is 1234.	
DTMF Reset Code	Input the DTMF reset code, default value is 4321.	
	Reset the output port mode by duration or state.	
Reset By	By duration: Reset the output port status when output duration occurs.	
i i coor by	By state: Reset the output port status when device's call state	
	changes.	
	Enable or disable trigger by URI.	
Trigger by URI	User can send commands from remote device or server to i10 series	
	device, if the command is correct, then device will trigger	
	corresponding output port.	
Trigger Message	Input trigger message for trigger by URI mode.	
Rest Message	Input reset message for trigger by URI mode.	
Trigger by SMS	Enable or disable trigger by SMS.	
	User can send ALERT command to i10 series device, if the command	
	is correct, then device will trigger corresponding output port.	
Trigger SMS	Input trigger message for trigger by SMS mode.	
Reset SMS	Input reset message for trigger by SMS mode.	
	Select the input port, when the input port meets the trigger condition,	
Trigger by Input	the output port will be triggered (The Port level time change, By <	
	Output Duration > control)	



	Select call state to trigger the output port, options are:		
	Talking: When the device's talking status changes, trigger the output		
	port.		
Trigger By Call state	Ringing: When the device's ringing status changes, trigger the output		
	port.		
	Calling: When the device's calling status changes, trigger the output		
	port.		
	Enable or disable trigger by dsskey. If any of the dsskey is selected,		
Trigger By DssKey	when the dsskey application performs, the output port will be		
	triggered.		



## **10 Trouble Shooting**

When the device doesn't work properly, user can try the following methods to restore the device to normal operation or collect relevant information to send a problem report to the Fanvil technical support mailbox.

#### 10.1 Get device system information

User can obtain information through the [**System**] >> [**Information**] option on device's webpage. The following information will be provided:

Device information (model, software and hardware version), network Information and SIP Accounts Information etc.

#### 10.2 Reboot device

User can restart the device through the webpage, click [**System**] >> [**Reboot Phone**] and click [**Reboot**] button, or directly unplug the power to restart the device.

When the device has problems and user can't access the web page, you can disassemble the surface shell and press the "**RESET**" button. The device will restart and the configuration will not change.

#### **10.3 Device factory reset**

Restoring the factory settings will delete all configurations, database and configuration files on the device and the device will be restored to factory default state.

To restore the factory settings, please go to [**System**] >> [**Configuration**] >> [**Reset Phone**] page, and click [**Reset**] button, the device will return to the factory default state.

#### **10.4 Network Packets Capture**

Sometimes, when the device has problems, the data packet is very helpful. In order to obtain the data packet of the device, please log in the device's webpage, and go to [**System**] >> [**Tools**] page, and click the [**Start**] option in the "Web Capture". A message will inform user that capturing starts and at this time, user can perform related operations, such as starting/deactivating the line or making a call, please click the [**Stop**] button on the webpage after complete. Network packets are saved in a file, users can analyze the packet or send it to Fanvil Technical Support team.



## **10.5 Get Log Information**

Log information is helpful when encountering an abnormal problem. In order to get the log, the user can login device's webpage, and go to page [**Device Log**], click the [**Start**] option, and perform device until the problem appears, click [**Save**] to save the logs to local PC, user can analyze the logs or send the log file to the technician to check the problem.

## **10.6 Common Trouble Cases**

Trouble Case	Solution	
Device could not boot up	1. Please check device's power connection and confirm the powe	
	adapter or PoE switch is in Fanvil list.	
	2. If the device go to "POST mode" (the LED flashes slowly), it means	
	the device system is damaged. Please contact Fanvil technical	
	support to help you restore.	
Device could not register to a	1. Please check network cable connection and confirm the device is	
service provider	connected to Internet well.	
	2. If the network connection is good, please check your SIP line	
	configuration again. If all configurations are correct, please contact	
	your service provider for support, or obtain a registration network	
	packet according to the instructions in "10.4 Network Data Capture",	
	and send the packet to Fanvil support team to help analyze the issue.	

### Table 23 - Common Trouble Cases