





i12 User Manual

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3 Safety Instruction

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

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

4 Overview

i12 is a SIP voice intercom specially developed for the needs of industry users. The media stream transmission adopts the standard IP/RTP/RTSP protocol. It has inherited the advantages of good stability and carrier-grade sound quality of the azimuth phone. The product is a fully digital network-type intercom device. Its core part adopts a mature VoIP solution, and its performance is stable and reliable. ; The buttons feel comfortable, easy to install, generous in appearance, durable and low power consumption.

5 Install Guide

5.1 Use POE or external Power Adapter

i12 supports two power supply methods, external power adapter and Ethernet (POE) switch power supply mechanism

The POE power supply mode saves space and the cost of additional power sockets. i12 is connected to the POE switch through a network cable to play the role of power supply and data transmission. By connecting to the POE switch of the UPS system, the i12 can continue to work even if the power is cut off, just like a traditional PSTN phone powered by a telephone line.

Users who do not have POE equipment can also use traditional power adapters. If the i12 is connected to the POE switch and the power adapter at the same time, the POE power supply is preferred. If the POE power supply fails, it will be switched to the power adapter.

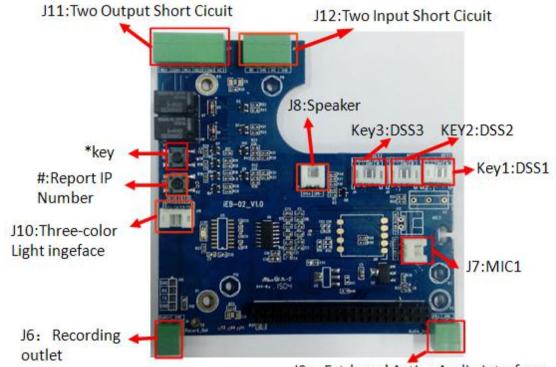
In order to ensure the normal operation of the equipment, please use the power adapter specified by Fanvil and the POE switch that meets the equipment standard.

5.2 Installation

Before you start using the device, please install the following:

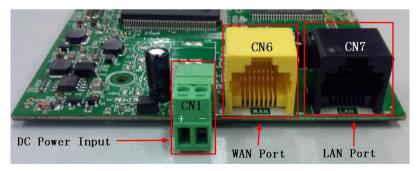
5.2.1 Interface description

• Expansion board interface



J9: Extdernal Active Audio Interface

• Motherboard interface



CN1	CN6	CN7
Power Supply	WAN Port	LAN Port
+9~+16V	WAN	LAN
CN1	CN6	CN7

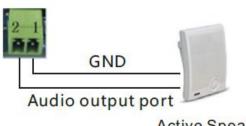
• Port description

Port	Description	Feature	Picture
CN1	DC Power Input port	Input Range:+9~+16V DC	CN1
CNI		(Notice: Plus-n-Minus connection of the Power)	
CN6	WAN port	10M/100M Adaptive Ethernet port, connected to the network	CN6
CN7	LAN Port	10M/100M Adaptive Ethernet port, connected to the computer	CN7
enz		(which can be configured to routing mode, or to bridge mode)	
19	External Active	One is the audio signal line, one is the GND line(Please connect to	
19	Speakers port	the GND line, otherwise there will be noise)	
	Audio Recording	By mixing equipment and remote call voice output.	
J6	output port	One is the audio signal line, one is the GND line(Please connect to	AA
		the GND line, otherwise there will be noise)	
Key1/key2/	DSS key port	Function keys. Can be defined hot keys, function keys(such as	
key3	(programmable keys)	hanging up, hands-free), multicast keys	LEDI-
J11	Short circuit output	Used to control electric locks, alarm lamp and so on	
J11	control Port		AAAAAA
J12	Short circuit Input	Used to connect to infrared detector, magnetic switch, vibration	
J12	detection Port	sensor and other input devices	AAAA
J10	Status indicator light	For an external status instructions (calling, ringing,	
10	port	network/registered)	

a) Port instructions

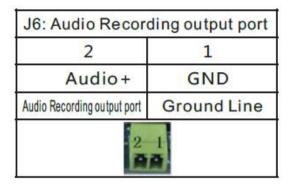
• External Active Speakers

J9: External Active Speakers Port		
2 1		
SPK+ GND		
Audio output port Ground Line		
2-1		



Active Speaker

• Audio Recording output port



GND GND

Audio Recording output PC Recording

• Two short circuit output port

- NO: Under the idle state is disconnected (normally open).
- COM: Contactor of the Relay (middle).
- > NC: Under the idle state is connected (normally close).

	J11: Sh	ort circ	uit outp	out Port	
Output Port1(OUT2)		Output Port1(OUT1)			
6	5	4	3	2	1
NC2	COM2	NO2	NC1	COM1	NO1
	Common terminal		Normal close	Common terminal	
		6 5 4 6 6 6	3 2-1 888		• P • ·

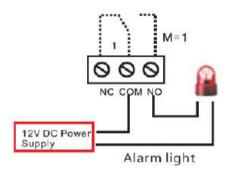
• Two short circuit input port

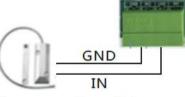
J1:	2: Short cir	cuit Input F	Port
Input Port2(IN2)		Input Po	ort1(IN1)
4	3	2	1
GND	IN2	GND	IN1
Input Port2	Input Port2	Input Port1	Input Port1



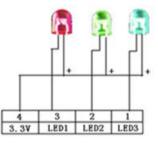
• Status lamp interface

4	3	2	1
3.3V	LED1	LED2	LED3
Power supply	Network	Call	Ringing



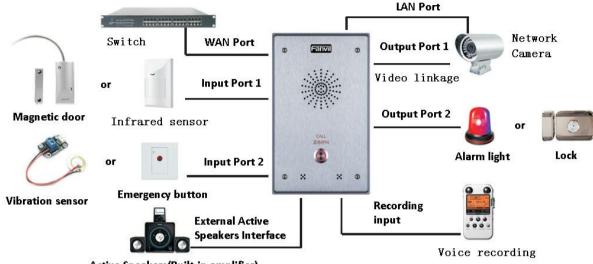


Door magnetic switch



5.2.2 Device connection confirmation

Check whether the power cord and network cable of the device are connected, and whether they are normal after 30 seconds after power-on. (Check the network indicator status)



Active Speakers(Built-in amplifier)



5.3 Appendix Table

5.3.1 Common command mode

Table 1 - C	Common con	nmand mode
-------------	------------	------------

Action		Description
IP	Broadcast	In standby, long press the speed dial key 10, there will be a beep and
under	standby	the indicator will flash quickly for 5 seconds, 5 seconds
mode		Press the speed dial key once inside, the beep sound stops and the
		IP is automatically reported
		In the standby mode, press and hold the speed dial button for 10
		seconds, there will be a beep and the indicator will flash quickly for 5
		seconds.
Switch	network	Within 5 seconds, quickly press the speed dial key three times to
mode		switch the network mode. Network status is static or
		PPPoE mode will be switched to DHCP mode; when the network is
		DHCP mode, it will be switched to static IP
		192.168.1.128, report IP after successful switching

5.3.2 Function key LED state

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

Table 2 - Function key LED state

6 User Getting Started

6.1 Quick setting

Before proceeding with this step, please confirm that your Internet broadband connection can work normally and complete the connection of the network hardware. The default network mode of this product when it leaves the factory is fixed IP address mode, and the default is 192.168.1.128.

- Long press the speed dial button for 10 seconds, wait for the horn to beep quickly, and then press the volume up button three times quickly to stop the beeping. Wait for 10 seconds, the system will automatically announce the IP address by voice after successfully switching to the dynamic IP acquisition. Switching again will become a fixed IP address.
- Log in to the WEB page of the device to configure according to the IP address;
- Configure the account, user name, server address and other required parameters for registration on the SIP page;

13	iDoorPhone Networ	k Scanner(V 1.0)				×
#	IP Address	Serial Number	MAC Address	SW Version	Description	
1	192. 168. 1. 128	i12	00:a8:34:68:23:a3	2.1.1.2834	i12	
						<u>R</u> efresh

Figure 2 - Quickly setting

6.2 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:	admin
Password:	••••
Language:	English 💌

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</u> <u>Web Configurations</u>

6.3 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 as 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 Basic Settings	SIP Hotspot	Blacklist	Action Plan	Dial
› System					
> Network	Line SIP 1 -				
-	Basic Settings >>				
> Line	Line Status	Registered	SIP Proxy	/ Server Address	172.16.1.2
	Phone number	36	SIP Proxy	/ Server Port	5060
> Intercom settings	Display name		Backup P	roxy Server Address	
	Authentication Name		Backup P	roxy Server Port	5060
> LED	Authentication Password		Outboun	d proxy address	
	Activate	V	Outboun	d proxy port	
> Security settings			Realm		
	Codecs Settings >>				
Function Key					
	Advanced Settings >>				
		Apply			

Figure 4 - SIP Registration

7 Basic Function

7.1 Making Calls

After setting the shortcut key as Hot key and setting the number, press the shortcut key to immediately call out the set number, the settings are as follows:

Key	Type		Number 1	Number 2	Line	2	Subtype	
Dss Key 1	Hot Key	-	123		SIP1	-	Speed Dial	-
Dss Key 2	None				SIP1	¥	Speed Dial	Ŧ
Dss Key 3	None	-			SIP1	-	Speed Dial	-
Dss Key 4	None	-			SIP1	-	Speed Dial	+
	10000				Contract (-	A CONTRACTOR OF	
Dss Key 5	None				SIP1	Ŧ	Speed Dial	*
Dss Key 6	None				SIP1	Y	Speed Dial	*
Dss Key 6	None Cey to Answer	Enab	e 💌 Secondary 💌	Enable Speed Dial Ha	SIP1	Y		
Dss Key 6 nced Settings Jse Function K	None Cey to Answer ode Select	Enab		Enable Speed Dial Ha	SIP1	Y	Speed Dial	
Dss Key 6 nced Settings Jse Function K lot Key Dial M	None (ey to Answer ode Select fime	Enab Main-	Secondary 💌 (5~50)Second(s)	Enable Speed Dial Ha	SIP1	Y	Speed Dial	*

Figure 5 - Hot Key Setting

See detailed configuration instructions 9.26 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 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

Key	Туре	Number 1	Number 2	Line	Subtype	£
Dss Key 1	Key Event			SIP1 -	Release	
Dss Key 2	None 💌			SIP1 -	Speed Dial	v
Dss Key 3	None 🗨			SIP1 -	Speed Dial	V
Dss Key 4	None 💌			SIP1 -	Speed Dial	÷
Dss Key 5	None 💌			SIP1 +	Speed Dial	Ŧ
Dss Key 6	None 💽			SIP1 👻	Speed Dial	4

Figure 6 - Function key settings

You can hang up the call through the Release key (you can set the function key as the Release key) or

turn on the speed dial button to hang up the call. See detailed configuration instructions <u>9.26 Function</u> Key.

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.

Web interface: enter [Intercom Setting] >> [Features], Enable auto answer, set mode and auto answer time and click submit.

Limit Talk Duration	Disable 💌	Talk Duration	120	(20~600) Second(s)
DND Mode	Phone 💌	Ban Outgoing		
Enable Call Waiting		Enable Call Waiting Tone		
Enable Intercom		Enable Intercom Barge		
Enable Intercom Mute				
Enable Auto Dial Out		Auto Dial Out Time	5	(3~30)Second(s)
Enable Auto Answer	Lines and IP Call 💌	Auto Answer Timeout	0	(0~60)Second(s)
Dial Fixed Length to Send		Send length	4	
Voice Read IP	Enable 💌	System Language	English	-
Description	i12 IP Intercom Phone	Enable DND		
HangUp Delay	3 Second(s)(1~60)	Call Timeout	90	(1~3600)Second(s)
Dial Number Voice Play	Disable 💌	Ring Timeout	120	(1~3600)Second(s)
Hotline Number		Hotline Delay	0	(0~9)Second(s)

Figure 7 - Enable Auto Answer

- Auto Answer mode:
 - Disable: Turn off the automatic answer function, the device has a call, ring, will not time out to answer automatically.
 - Line1: Line 1 has an automatic call timeout.
 - Line2: Line 2 has an automatic call timeout.
 - Line1 and Line2: Line 1 and line 2 have an automatic call timeout.
 - Lines and IP Call: Line and IP direct dial call timeout automatically 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 DND

Users can turn on the do-not-disturb (DND) feature on the device's web page to reject incoming calls (including call waiting).Do not disturb can be set by the SIP line respectively on/off.

Turn on/off all lines of the device without interruption by the following methods:

Web interface: enter [Intercom Setting] >> [Features], set the DND Mode to phone and Enable DND.

OND Mode	Phone 💌	Ban Outgoing		
Enable Call Waiting	V	Enable Call Waiting Tone		
Enable Intercom		Enable Intercom Barge		
Enable Intercom Mute				
Enable Auto Dial Out		Auto Dial Out Time	5	(3~30)Second(s)
Enable Auto Answer	Lines and IP Call 💌	Auto Answer Timeout	0	(0~60)Second(s)
Dial Fixed Length to Send		Send length	4	
/oice Read IP	Enable 💌	System Language	English	(
Description	i12 IP Intercom Phone	Enable DND		
langUp Delay	3 Second(s)(1~60)	Call Timeout	90	(1~3600)Second(s)
Dial Number Voice Play	Disable 💌	Ring Timeout	120	(1~3600)Second(s)
Hotline Number		Hotline Delay	0	(0~9)Second(s)

Figure 8 - Set DND Option

Turn on/off the DND of a specific line of the device, as follows:

• enter [Line] >> [SIP], choose a Line and enter [Line] >> [Advanced settings], Enable DND.

Line SIP 1 Basic Settings >>				
Codecs Settings >>				
Advanced Settings >>				
Enable Hotline				
Hotline Delay	0	(0~9)Second(s)	Hotline Number	
Enable DND			Ring Type	Default 💌
Blocking Anonymous Call			Conference Type	Local 💌
Use 182 Response for Call waiting			Server Conference Number	
Anonymous Call Standard	None 💌]	Transfer Timeout	0 Second(s)
Dial Without Registered			Enable Long Contact	
Click To Talk			Enable Use Inactive Hold	
User Agent			Use Quote in Display Name	
Response Single Codec			TLS Version	TLS 1.2 💌

Figure 9 - Enable do not disturb on a certain line

7.6 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 during a call, the device will sound a beep-beep tone.

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.

ND Mode	Phone 💌	Ban Outgoing		
Enable Call Waiting		Enable Call Waiting Tone]
Enable Intercom		Enable Intercom Barge		
Enable Intercom Mute				
Enable Auto Dial Out		Auto Dial Out Time	5	(3~30)Second(s)
Enable Auto Answer	Lines and IP Call 💌	Auto Answer Timeout	0	(0~60)Second(s)
Dial Fixed Length to Send		Send length	4	
Voice Read IP	Enable 💌	System Language	English	•
Description	i12 IP Intercom Phone	Enable DND		
HangUp Delay	3 Second(s)(1~60)	Call Timeout	90	(1~3600)Second(s)
Dial Number Voice Play	Disable 💌	Ring Timeout	120	(1~3600)Second(s)
Hotline Number		Hotline Delay	0	(0~9)Second(s)

Figure 10 - Web page setting call waiting

8 Advance Function

8.1 Intercom

When there is an intercom call, the device can answer it automatically.

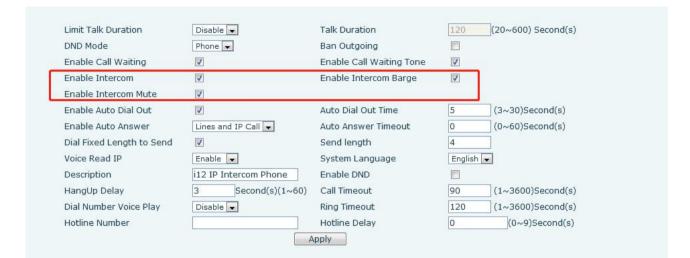


Figure 11 - WEB Intercom

Table 3 - Intercom

Parameters	Description
	When the intercom system is enabled, the device will accept the
Enable intercom	SIP header Call-Info of the incoming call request
	Instruction to answer the phone automatically
Enable intercom	Automatically answer the call in intercom mode during the call, if
	the current call is in intercom mode
barge	Mode, refuse to answer the new intercom mode
Enable intercom	Turn on the mute function during an intercom mode coll
Mute	Turn on the mute function during an intercom mode call

8.2 MCAST

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

Enable Auto Mcast Sip Priority Enable Page Priority	Image: Auto Mcast Timeo 0 Intercom Priority	ut Delete Time 10 (5~10s)
Index/Priority	Name	Host:port
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		

Figure 12 - MCAST

Table 4 - MCAST

Parameters	Description
Enable Auto Mcast	Send the multicast configuration information by Sip Notify
	signaling, and the device will configure the information to the
	system for multicast listening or cancel the multicast listening in the
	system after receiving the information
Auto Mcast Timeout	When a multicast call does not end normally, but for some reason
Delete Time	the device can no longer receive a multicast RTP packet, this
	configuration cancels the listening after a specified time
Priority	The priority defined in the current call, 1 is the highest priority and
	10 is the lowest.
Enable Page Priority	Regardless of which of the two multicast groups is called in first,
	the device will receive the higher priority multicast first.
Name	Listened multicast server name
Host:port	Listened multicast server's multicast IP address and port.

Multicast:

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

8.3 Hotspot

SIP hotspot is a simple utility. Its configuration is simple, can realize the function of group vibration, can 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 answer, 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 5 - 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.

		The second se	Section.	
IP	MAC	Alias	Line	
P Hotspo	t 😧			
Enable	e Hotspot	Enable 💌		
Mode		Client 💌		
Monito	r Type	Broadcast 💌		
Monito	or Address	224.0.2.0		
Remote	e Port	16360		
Local P	ort	16360		
Name		SIP Hotspot		

Figure 13 - 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 log into the device's web page to manage user device information and operate the device. Users must provide the correct user name and password to log in. If the password is entered incorrectly three times, it will be locked and can be entered again after 5 minutes.

The details are as follows:

- If an IP is logged in more than the specified number of times with a different user name, it will be locked
- If a user name logs in more than a specified number of times on a different IP, it is also locked

9.2 System >> Information

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

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

And summarization of network status,

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

Besides, summarization of SIP account status,

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

9.3 System >> Account

Username	admin
Web Authentication Password	*****
Confirm Password	
Privilege	Administrators 💌
	bbA
ser Accounts	
User	Privilege
admin	Administrators
	Users

Figure 14 - WEB Account

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

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

9.4 System >> Configurations

Users with administrator rights can view, export or import device configuration on this page, and can also restore the device to factory settings.

Export Configurations	
	Right click here to SAVE configurations in 'txt' format.
	Right click here to SAVE configurations in 'xml' format.
Import Configurations	
	Configuration file: Select Import
Reset to factory defaults	
	Click the [Reset] button to reset the phone to factory defaults.
	ALL USER'S DATA WILL BE LOST AFTER RESET!
	Reset

Figure 15 - System Setting

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

Export Configurations

Right click to select target save as, that is, to download the device's configuration file, suffix ".txt". (note: profile export requires administrator privileges)

Import Configurations

Import the configuration file of Settings. The device will restart automatically after successful import, and the configuration will take effect after restart

Reset Phone

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

9.5 System >> Upgrade

- farmer and -				
oftware upgrade				
	Current Software Version:	2.8.0.6948		
	System Image File		Select	Upgrade

Figure 16 - Upgrade

Upgrade the software version of the device, and upgrade to the new version through the webpage. After the upgrade, the device will automatically restart and update to the new version. Click select, select the version and then click upgrade

9.6 System >> Auto Provision

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

Upgrade the device software version and upgrade to the new version through the web page. After the upgrade is completed, the device will automatically restart and update to the new version. Click select, select the version and click upgrade.

Common Cottings		
Common Settings		
Current Configuration Version		
General Configuration Version		
CPE Serial Number	00100400FV02001000000d84a000e78 admin	
Authentication Name		
Authentication Password		
Configuration File Encryption Key		
General Configuration File Encryption Ke	y	
Download Fail Check Times	5	
Enable Get Digest From Server		
DHCP Option >>		
SIP Plug and Play (PnP) >>		
Static Provisioning Server >>		
TR069 >>		
	Apply	

Figure 17 - Auto Provision

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

PNP>DHCP>TR069> Static Provisioning

Transferring protocol: FTP、 TFTP、 HTTP、 HTTPS

Details refer to Fanvil Auto Provision

http://www.fanvil.com/Uploads/Temp/download/20180920/5ba38170d79fb.pdf

Table 6 - Auto Provision

Parameters	Description
Basic settings	
	Show the current config file's version. If the version of configuration
Current	downloaded is higher than this, the configuration will be upgraded. If
Configuration	the endpoints confirm the configuration by the Digest method, the
Version	configuration will not be upgraded unless it differs from the current configuration
	Show the common config file's version. If the configuration
General	downloaded and this configuration is the same, the auto provision will
Configuration	stop. If the endpoints confirm the configuration by the Digest method,
Version	the configuration will not be upgraded unless it differs from the current configuration.
CPE Serial Number	Serial number of the equipment
Authentication	Username for configuration server. Used for FTP/HTTP/HTTPS. If this
Name	is blank the phone will use anonymous
Authentication	
Password	Password for configuration server. Used for FTP/HTTP/HTTPS.
Configuration File	
Encryption Key	Encryption key for the configuration file
General	
Configuration File	Encryption key for common configuration file
Encryption Key	
Save Auto	Save the auto provision username and password in the phone until
Provision	the server url changes
Information	
Download Fail	The default value is 5. If the download configuration fails, it will be
Check Times	downloaded 5 times.
Enable Server	When the feature is enable, if the configuration of server is changed,
Digest	phone will download and update.
DHCP Option	
Option Value	The equipment supports configuration from Option 43, Option 66, or a
	Custom DHCP option. It may also be disabled.

Custom Ontion	
Custom Option Value	Custom option number. Must be from 128 to 254.
Enable DHCP	
-	Set the SIP server address through DHCP option 120.
Option 120	(D=D)
SIP Plug and Play	
	Whether enable PnP or not. If PnP is enable, phone will send a SIP
Enable SIP PnP	SUBSCRIBE message with broadcast method. Any server can
	support the feature will respond and send a Notify with URL to phone.
	Phone could get the configuration file with the URL.
Server Address	Broadcast address. As default, it is 224.0.0.0.
Server Port	PnP port
Transport Protocol	PnP protocol, TCP or UDP.
Update Interval	PnP message interval.
Static Provisioning	g Server
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address
	can be an IP address or Domain name with subdirectory.
	The configuration file name. If it is empty, phone will request the
Configuration File	common file and device file which is named as its MAC address.
Name	The file name could be a common name, \$mac.cfg, \$input.cfg. The
	file format supports CFG/TXT/XML.
Protocol Type	Transferring protocol type ,supports FTP、TFTP、HTTP and HTTPS
Lindata Interval	Configuration file update interval time. As default it is 1, means phone
Update Interval	will check the update every 1 hour.
	Provision Mode.
	1. Disabled.
Update Mode	2. Update after reboot.
	3. Update after interval.
TR069	
Enable TR069	Enable TR069 after selection
Enable TR069	
Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting.
ACS Server Type	There are 2 options Serve type, common and CTC.
ACS Server URL	ACS server address
ACS User	ACS server username (up to is 59 character)
ACS Password	ACS server password (up to is 59 character)
TR069 Auto Login	Enable/Disable TR069 Auto Login.
STUN	
server address	Enter the STUN address
Enable the STUN	Enable the STUN
	1

9.7 System >> FDMS

ommunity Name	
lding Number	
m Number	

Figure 18 - FDMS

Table 7 - FDMS

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

9.8 System >> Tools

This page gives the user the tools to solve the problem.

	Click [Reboot] button to restart the phone!	
Reboot Phone		
	Start	
Network Packets Capture		
	Apply	
SIP Log Level	None	
APP Log Level	None	
Server Port	514	
Server Address	0.0.0	
Enable Syslog		

Figure 19 - Tools

Syslog: When enabled, set the syslog software address, and log information of the device will be recorded in the syslog software during operation. If there is any problem, log information can be

analyzed by Fanvil technical support.

9.9 Network >> Basic

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

IP:	172.16.7.132	
Subnet mask:	255.255.255.0	
Default gateway:	172.16.7.1	
MAC:	00:d8:4a:00:0e:78	
MAC Timestamp:	2018/10/10 16:27:44	
tting		
Static IP 🔘	DHCP (PPPoE 🔘
DNS Server Configured by	DHCP	
Primary DNS Server		
Secondary DNS Server		
	Apply	
rvice Port Settings 😡		
Web Server Type	HTTP 💌	
HTTP Port	80	
HTTPS Port	443	
	Apply	

Figure 20 - Network Basic Setting

Table 8 - Network Basic Setting

Field Name	Explanation
Network Status	
IP	The current IP address of the equipment
Subnet mask	The current Subnet Mask
Default gateway	The current Gateway IP address
MAC	The MAC address of the equipment
MAC Time stamp	Get the MAC address of time.
Settings	
Select the appropri	ate network mode. The equipment supports three network modes:
Static IP	Network parameters must be entered manually and will not change. All parameters are provided by the ISP.
DHCP	Network parameters are provided automatically by a DHCP server.

PPPoE	Account and Password must be input manually. These are			
	provided by your ISP.			
If Static IP is chose	en, the screen below will appear. Enter values provided by the ISP.			
DNS Server	Coloct the Configured mode of the DNC Conver			
Configured by	Select the Configured mode of the DNS Server.			
Primary DNS	Enter the conver address of the Drimony DNS			
Server	Enter the server address of the Primary DNS.			
Secondary DNS	Enter the conver address of the Secondary DNS			
Server	Enter the server address of the Secondary DNS.			

attention:

1) After setting the parameters, click [submit] to take effect.

2) If you change the IP operation, the web page will no longer respond, at this time should be entered in the address bar new IP to connect to the device.

3) f the system USES DHCP to obtain IP at start up, and the network address of the DHCP Server is the same as the network address of the system LAN, then after the system obtains the DHCP IP, it will add 1 to the last bit of the network address of LAN and modify the IP address segment of the DHCP Server of LAN. If the DHCP access is reconnected to the WAN after the system is started, and the network address assigned by the DHCP server is the same as that of the LAN, then the WAN will not be able to obtain IP access to the network

Service Port	Settings
Web Server	
Туре	Specify Web Server Type – HTTP or HTTPS
	Port for web browser access. Default value is 80. To enhance security,
	change this from the default. Setting this port to 0 will disable HTTP
HTTP Port	access.
	Example: The IP address is 192.168.1.70 and the port value is 8090,
	the accessing address is http://192.168.1.70:8090.
	Port for HTTPS access. Before using https, an https authentication
	certification must be downloaded into the equipment.
HTTPS Port	Default value is 443. To enhance security, change this from the
	default.

9.10 Network >> Advanced

Link Layer Discovery Protocol (LLDP) Settings			
Enable LLDP 😡		Packet Interval(1~3600)	60	Second(s)
Enable Learning Function				
ARP Cache Life				
ARP Cache Life	2 Minute			
VLAN Settings				
Enable VLAN		VLAN ID	256	(0~4095)
802.1p Signal Priority	0 (0~7)	802.1p Media Priority	0	(0~7)
LAN Port VLAN Settings				
Mode	Disable 💌	LAN Port VLAN ID	254	(0~4095)
		802.1p Priority	0 (0/	~7)
DHCP VLAN Settings				
Option Value	Disabled 💌	DHCP Option Vlan(128-254)	0	
Quality of Service (QoS) Settin	gs			
Enable DSCP QoS		Signal QoS Priority	46	(0~63)
Media QoS Priority	46 (0~63)			

Figure 21 - Basic network settings

Advanced network settings are usually configured by IT administrators to improve the quality of equipment services

Table 9	- Basic network parameters
---------	----------------------------

Parameters	Description
LLDP setting	
Report	Enable LLDP
Interval	LLDP requests interval time
Learning	apply the learned VLAN ID to the phone configuration
QoS	
QoS Mode	Voice quality assurance (default closed)
DHCP VLAN Settings	
Parameter value	128-254, get the VLAN value through DHCP
WAN VLAN	
WAN VLAN	WAN port VLAN configuration
LAN VLAN	
LAN VLAN	LAN port VLAN configuration

9.11 Network >> VPN

Virtual Private Network (VPM	N) Status			
	VPN IP Addre	ess:	0.0.0	
VPN Mode				
	Enable VPN			
	L2TP		OpenVPN 💿	
Layer 2 Tunneling Protocol (L2TP)			
	L2TP Server	Address		
	Authenticatio	on Name	admin	
	Authenticatio	on Password	••••	
			Apply	
OpenVPN Files				
OpenVPN Configuration fi	ile: client.ovpn	N/A	Upload	Delete
CA Root Certification:	ca.crt	N/A	Upload	Delete
Client Certification:	client.crt	N/A	Upload	Delete
Client Key:	client.key	N/A	Upload	Delete

Figure 22 - VPN

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

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

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

■ L2TP

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

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

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

OpenVPN

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

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

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

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

9.12 Network >> Web Filter

A user can set up a configuration management device that allows only machines with a certain network segment IP to access the configuration management device

Start IP Address	End IP Address	Option
Web Filter Table Settings		
Start IP Address	End IP Address	Add
Web Filter Setting		
Enable Web Filter 🔳	Apply	
网页过滤表		
开始IP	结束IP	选项
192.168.1.1	192.168.254.254	更改

Figure 23 - WEB Filter Table

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

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

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

9.13 Line >> SIP

Configure the service configuration of the line on this page.

c Settings >> Line Status	Registered	SIP Proxy Server Address	172.16.1.2
Phone number	36	SIP Proxy Server Port	5060
Display name		Backup Proxy Server Address	
Authentication Name		Backup Proxy Server Port	5060
Authentication Password		Outbound proxy address	
Activate		Outbound proxy port	
		Realm	

Figure 24 - SIP

Table 10 - SIP

SIP	
Field Name	Explanation
Basic Settings (Choose the SIP line to configured)	
Line Status	Display the current line status at page loading. To get the up to date
	line status, user has to refresh the page manually.
Username	Enter the username of the service account.
Display name	Enter the display name to be sent in a call request.
Authentication	Enter the authentication name of the service account
Name	
Authentication	Enter the authentication password of the service account
Password	
Activate	Whether the service of the line should be activated
SIP Proxy Server	Enter the IP or FQDN address of the SIP proxy server
Address	

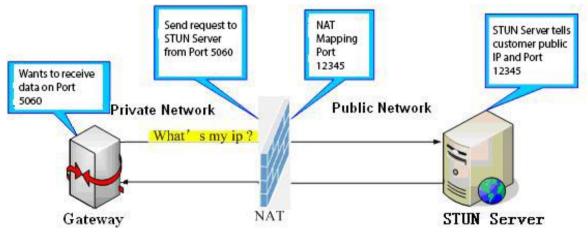
SIP Proxy Server Port	Enter the SIP proxy server port, default is 5060		
Outbound proxy	Enter the IP or FQDN address of outbound proxy server provided by		
address	the service provider		
Outbound proxy port	Enter the outbound proxy port, default is 5060		
Realm	Enter the SIP domain if requested by the service provider		
Codecs Settings			
	vailability of the codecs by adding or remove them from the list.		
Advanced Settings			
g~	Enable the device to subscribe a voice message waiting notification,		
Subscribe For Voice	if enabled, the device will receive notification from the server if there		
Message	is voice message waiting on the server		
Voice Message			
Number	Set the number for retrieving voice message		
Voice Message			
Subscribe Period	Set the interval of voice message notification subscription		
	Enable Do-not-disturb, any incoming call to this line will be rejected		
Enable DND	automatically		
Diaglian	automatically		
Blocking	Reject any incoming call without presenting caller ID		
Anonymous Call			
Use 182 Response	Set the device to use 182 response code at call waiting response		
for Call waiting			
Anonymous Call	Set the standard to be used for anonymous		
Standard			
Dial Without	Set call out by proxy without registration		
Registered			
Click To Talk	Set Click To Talk		
User Agent	Set the user agent, the default is Model with Software Version.		
Response Single	If setting enabled, the device will use single codec in response to an		
Codec	incoming call request		
Ring Type	Set the ring tone type for the line		
	Set the type of call conference, Local=set up call conference by the		
Conference Type	device itself, maximum supports two remote parties, Server=set up		
	call conference by dialing to a conference room on the server		
Server Conference	Set the conference room number when conference type is set to be		
Number	Server		
Transfer Timeout	Set the timeout of call transfer process		
Enable Long	Allow more peremeters in contact field per PEC 2010		
Contact	Allow more parameters in contact field per RFC 3840		
Enable the Inactive	Active capture package SDP is inactive, while the hold is sendrecv.		
Hold	Active capture package has no response of 400, etc. Hold the hair		
	, ,		

	·	
	inactive	
	After closing the grab packet, you can see that the DSP is sendonly	
Use Quote in	and the hold is sendrecv	
	Whether to add quote in display name	
Display Name	Set the line to collaborate with apositic convertives	
Specific Server Type Registration	Set the line to collaborate with specific server type	
Expiration	Set the SIP expiration interval	
Use VPN	Set the line to use VPN restrict route	
Use STUN	Set the line to use STUN for NAT traversal	
Convert URI	Convert not digit and alphabet characters to %hh hex code	
	Set the DTMF sending mode, there are four types: In-band	
	RFC2833	
DTMF Type	SIP INFO	
	AUTO	
	Different service providers may offer different models	
	When the device's DTMF type is set to SIP_INFO	
DTMF SIP INFO	The DTMF_SIP_INFO type is configured to send */#, and when the device presses the */# key, the actual value sent is */#;	
Mode	Configured to send 10/11, when the device presses the */# key, the	
	actual value sent is 10/11.	
Transportation		
Protocol	Set the line to use TCP or UDP for SIP transmission	
Local Port	Set the Local Port	
SIP Version	Set the SIP version	
Caller ID Header	Set the Caller ID Header	
	Enables the use of strict routing. When the phone receives packets	
Enable Strict Proxy	from the server, it will use the source IP address, not the address in	
	via field.	
Enable user=phone	one Sets user=phone in SIP messages.	
Enable SCA	Enable/Disable SCA (Shared Call Appearance)	
Enable DNS SRV	Set the line to use DNS SRV which will resolve the FQDN in proxy	
	server into a service list	
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT	
	pinhole opened	
Keep Alive Interval	Set the keep alive packet transmitting interval	
Enable Session	Set the line to enable call ending by session timer refreshment. The	
Timer	call session will be ended if there is not new session timer event	
	update received after the timeout period	
Session Timeout	Set the session timer timeout period	

Enable Rport	Set the line to add rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
Enable DNS SRV	Set the line to use DNS SRV which will resolve the FQDN in proxy
	server into a service list
Auto Change Port	Enable/Disable Auto Change Port
Keep Authentication	Keep the authentication parameters from previous authentication
	Using TCP protocol to guarantee usability of transport for SIP
Auto TCP	messages above 1500 bytes
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
RTP Encryption	Set the pass phrase for RTP encryption
With Mac field	When enabled, all SIP messages strip Mac fields
Register with the	When eachied register the measure rikken Mac field
Mac field	When enabled, register the message ribbon Mac field

9.14 Line >> Basic Settings

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



Local SIP Port	5060	
Registration Failure Retry Interval	32	Second(s)
Transaction TimerT1(0.5~10s)	500	millisecond
Transaction TimerT2(2~40s)	4000	millisecond
Transaction TimerT4(2.5~60s)	5000	millisecond
Enable Strict UA Match		
Strict Branch		
	Apply	
STUN Settings		
STUN NAT Traversal	FALSE	
Server Address		
Server Port	3478	
Binding Period	50	Second(s)
SIP Waiting Time	800	millisecond

Figure 25 - Line Basic Setting

Field Name	Explanation	
SIP Settings		
Local SIP Port	Set the local SIP port used to send/receive SIP messages.	
Registration Failure	Set the retry interval of SIP REGISTRATION when registration	
Retry Interval	failed.	
Enable Strict UA Match	Enable or disable Strict UA Match	
Field Name	Explanation	
STUN Settings		
Server Address	STUN Server IP address	
Server Port	STUN Server Port – Default is 3478.	
Pinding Dariad	STUN blinding period – STUN packets are sent at this interval to	
Binding Period	keep the NAT mapping active.	
SIP Waiting Time	Waiting time for SIP. This will vary depending on the network.	

9.15 Line >> SIP Hotspot

SIP hotspot is a simple and practical function. It is simple to configure, can realize the function of group vibration, and can expand the number of SIP accounts.

See $\underline{8.3 \text{ Hotspot}}$ for details.

Device Tab	le		
IP	MAC	Alias	Line
SIP Hotspo	ot 😧		
Enable	e Hotspot	Disabled 💌	
Mode		Client	
Monito	or Type	Broadcast 💌	
Monitor Address		224.0.2.0	
Remot	e Port	16360	
Local F	Port	16360	
Name		SIP Hotspot	
Line Setting	gs		
SIP 1		Enable	×
SIP 2		Enable	
		Apply	

Figure 26 - SIP Hotspot

9.16 Line >> Blacklist

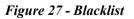
The function of restricting incoming calls is added to the webpage, and incoming calls can be restricted by setting a number or prefix. The rules are as follows:

Add x, the type is number, then x cannot call. Add x and type as prefix, then the number starting with x cannot be called.

x can be a number or ip. To add a whitelist rule, you need to add "-" in front of the number/IP, and then add a ".",

After adding, only the numbers in the whitelist are allowed to call, and the numbers outside the whitelist are all rejected.

		Add	Delete	Delete A
	Caller ID	Block on Line		Туре
cted Outgoing C	alls			
cted Outgoing C	alls	Add	Delete	Delete A



9.17 Intercom Settings >> Function Settings

Configure the intercom function settings.

Limit Talk Duration	Disable 💌	Talk Duration	120	(20~600) Second(s)
DND Mode	Phone 💌	Ban Outgoing		
Enable Call Waiting		Enable Call Waiting Tone		
Enable Intercom		Enable Intercom Barge		
Enable Intercom Mute				
Enable Auto Dial Out		Auto Dial Out Time	5	(3~30)Second(s)
Enable Auto Answer	Lines and IP Call 🕞	Auto Answer Timeout	0	(0~60)Second(s)
Dial Fixed Length to Send		Send length	4	
Voice Read IP	Enable 💌	System Language	English	
Description	i12 IP Intercom Phone	Enable DND		
HangUp Delay	3 Second(s)(1~60)	Call Timeout	90	(1~3600)Second(s)
Dial Number Voice Play	Disable 💌	Ring Timeout	120	(1~3600)Second(s)
Hotline Number		Hotline Delay	0	(0~9)Second(s)

Figure 28 - Function setting

Function setting	
Field Name	Description
General settings	
Limit call duration	After enabling, hang up the call after timeout
Call time	Hang up after timeout
DND (Do Not	DND might be disabled phone for all SIP lines, or line for SIP individually.
Disturb)	But the outgoing calls will not be affected
Ban Outgoing	If enabled, no outgoing calls can be made.
Enable Call	The default value is enabled. Allow users to answer the second call while
Waiting	maintaining the call.
Enable Call	The default value is enabled. When enabled, the call waiting tone can be
_	heard while waiting for a call. If this function is turned off, when waiting for a
Waiting Tone	call, the beep will not be heard.
	When the intercom system is enabled, the device will accept the SIP header
Turn on intercom	Call-Info of the incoming call request
	Instruction to answer the phone automatically
The second	Automatically answer the call in intercom mode during the call, if the current
intercom	call is in intercom mode
answering	Mode, refuse to answer the new intercom mode
Mute the	Configure intercom mode to turn on the mute function during a call
intercom	
Turn on intercom	When the intercom mode is configured, the incoming call will hear a ringing
ringing	tone.
Turn on timeout	
dialing	The system will automatically dial after timeout
Timeout dial time	

Turn on auto answer	Configure to turn on the auto answer function
Auto answer time	Configure auto answer time
Hang up	
automatically if	Configure to enable automatic hangup when no answer
no answer	
Auto hang up	Configure to hang up automatically when there is no answer within a set
timeout	time
Fixed length dial	When enabled, the number entered by the user reaches a fixed length and
Length of	When enabled, the number entered by the user reaches a fixed length and
receiving number	automatically dials out
Report IP	Turn on or off the device's voice broadcast IP address
System language	Configure the language of the voice prompt
Description	Descriptive information displayed on the IP scanning tool software or FDMS.
Description	The default is "i12"
Auto hang up	Configure the automatic hang-up time, if it is in hands-free mode, the device
time	will automatically return to standby after the auto handdown time is
	exceeded

9.18 Intercom Settings >> Voice Settings

Change voice settings

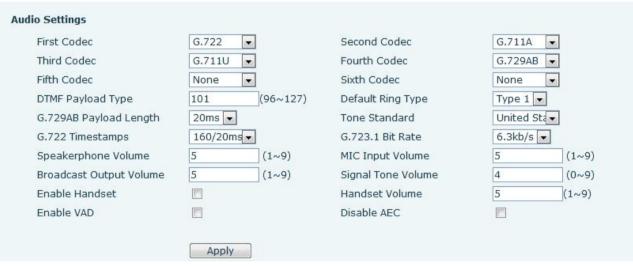




Table 13 - Audi	o Setting
-----------------	-----------

Voice settings		
Field Name	Description	
Cadaa	Select DSP priority speech coding algorithm, including: G.711A/u, G.722,	
Codec	G.723, G.729, G.726-32.	

DTMF payload typeHend DTMF payload type, ranging from 96 to 127, and the default is 101.Presel ringtone typeConfigure the default ringtoneG.729AB payload lengthConfigure the length of the G.729AB voice coding payload lengthSignal tome standardConfigure the length of the G.729AB voice coding payload lengthG.729XSelect time stamp for G.722 encoding, 160/20ms and 320/20ms can be timestampG.723.1 timestampConfigure the signal tone standard area electedG.723.1 timestampFor G723 rate selection, you can choose 5.3kb/s and 6.3kb/sHands-free volumeConfigure the call volume levelNicrophome input volumeConfigure the call volume levelNicrophome volumeConfigure the output volume level of the microphone input volumeSignal tome volumeConfigure the output volume level of the signal toneSignal tome volumeConfigure the output volume level of the signal toneSignal tome volumeConfigure the output volume level of the signal toneSignal tome volumeConfigure the output volume level of the signal toneSignal tome volumeConfigure the output volume level of the signal toneSignal tome volumeConfigure the output volume level of the signal toneSignal tome volumeConfigure the output volume level of the signal toneSignal tome volumeConfigure the signal tone standard areaSignal tome volumeConfigure the signal tone standard areaSignal tome volumeConfigure tope standard areaSignal tome volume <t< th=""><th></th><th>-</th></t<>		-	
payload type Configure the default ringtone Preset ringtone type Configure the default ringtone G.729AB Configure the length of the G.729AB voice coding payload length Signal tone standard Configure the signal tone standard area Signal tone standard Select time stamp for G.722 encoding, 160/20ms and 320/20ms can be selected Select time stamp for G.722 encoding, 160/20ms and 320/20ms can be G.723.1 bit rate For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s Hands-free Volume Volume Configure the call volume level setting Configure the call volume level of the microphone invut volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone signal tone Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone signal tone Configure the output volume level of the signal tone volume Configure the output vol	DTMF	Set the DTMF payload type, ranging from 96 to 127, and the default is 101.	
ringtone type Configure the default ringtone G.729AB Configure the length of the G.729AB voice coding payload length Configure the signal tone standard area Signal tone Select time stamp for G.722 encoding, 160/20ms and 320/20ms can be standard Selected G.723.1 bit For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s rate For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s rate Configure the call volume level setting Configure the call volume level wolume Configure the output volume level of the microphone input volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume leve	payload type		
ringtone type The up of the G.729AB G.729AB Configure the length of the G.729AB voice coding payload length Signal tome Signal tome Configure the signal tone standard area standard Select time stamp for G.722 encoding, 160/20ms and 320/20ms can be imestamp selected G.723.1 bit For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s rate For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s Hands-free volume volume Configure the call volume level setting Configure the call volume level Microphone Configure the output volume level of the microphone input volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone <	Preset	Configure the default ringtone	
payload length Configure the length of the G.729AB voice coding payload Signal tone standard Configure the signal tone standard area Signal tone standard Configure the signal tone standard area G.722 Select time stamp for G.722 encoding, 160/20ms and 320/20ms can be selected G.723.1 bit rate For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s Hands-free volume Configure hands-free call volume level setting Configure the call volume level Microphone input volume Configure the call volume level of the microphone Signal tone volume Configure the output volume level of the signal tone Signal tone volume Configure the output volume level of the signal tone Signal tone volume Configure the output volume level of the signal tone Ringtone upgrade Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Ringtone upgrade Optional ringtone upgrade with .wav suffix Ringtone upgrade The upgraded ringtones are displayed in the delete list and can be deleted selectively Value of notification message 1 to 10 Set to point the value of the specified ringtone type	ringtone type		
length Image: Configure the signal tone standard area Signal tone standard Configure the signal tone standard area G.722 Select time stamp for G.722 encoding, 160/20ms and 320/20ms can be selected G.723.1 bit rate For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s Hands-free configure hands-free call volume level volume Configure hands-free call volume level setting Configure the call volume level of the microphone introphone configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone selectivity Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Ringtone Optional ringtone upgrade with .wav suffix Ringtone Optional ringtone upgrade with .wav suffix Value of	G.729AB		
Signal tone standardConfigure the signal tone standard areaG.722Select time stamp for G.722 encoding, 160/20ms and 320/20ms can be selectedG.723.1 bit rateFor G723 rate selection, you can choose 5.3kb/s and 6.3kb/sHands-free volumeConfigure hands-free call volume levelSettingConfigure the call volume levelMicrophone input volumeConfigure the call volume level of the microphoneBroadcast outputConfigure the output volume level when broadcastingVolumeConfigure the output volume level of the signal toneSignal tone volumeConfigure the output volume level of the signal toneSignal tone volumeConfigure the output volume level of the signal toneSignal tone volumeConfigure the output volume level of the signal toneSignal tone volumeConfigure the output volume level of the signal toneSignal tone volumeConfigure the output volume level of the signal toneSignal tone volumeConfigure the output volume level of the signal toneSignal tone volumeConfigure the output volume level of the signal toneSignal tone volumeConfigure the output volume level of the signal toneSignate to upgradeOptional ringtone upgrade with .wav suffixRingtone upgradeThe upgraded ringtones are displayed in the delete list and can be deleted selectivelyIncoming call designated ring type setting (alert-info)Set to point the value of the specified ringtone type notification message 1 to 10	payload	Configure the length of the G.729AB voice coding payload	
StandardConfigure the signal tone standard areaG.722Select time stamp for G.722 encoding, 160/20ms and 320/20ms can be selectedG.723.1 bit rateFor G723 rate selection, you can choose 5.3kb/s and 6.3kb/sHands-free volumeConfigure hands-free call volume levelSettingConfigure the call volume levelMicrophone 	length		
standard Image: Construction of Configure thands-free call volume level Setting For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s Hands-free Configure hands-free call volume level setting Configure hands-free call volume level Microphone Configure the call volume level of the microphone input volume Configure the output volume level of the microphone Broadcast Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone setection Thu a 20ms Ringtone Optional ringtone upgrade with .wav suffix Ringtone Coptional ringtone	Signal tone	Configure the signal tone standard area	
timestamp selected G.723.1 bit rate For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s Hands-free volume Configure hands-free call volume level setting Microphone input volume Configure the call volume level of the microphone input volume Broadcast output Configure the output volume level of the signal tone volume Signal tone volume Configure the output volume level of the signal tone volume Enable voice activity detection Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Ringtone upgrade Optional ringtone upgrade with .wav suffix Ringtone upgrade The upgraded ringtones are displayed in the delete list and can be deleted selectively Incoming call - signated ring type setting (alert-info) Value of notification message 1 to 10 Set to point the value of the specified ringtone type	standard		
G.723.1 bit rate For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s Hands-free volume Configure hands-free call volume level Setting Configure hands-free call volume level Microphone input volume Configure the call volume level of the microphone Broadcast output Configure the output volume level when broadcasting Volume Configure the output volume level of the signal tone Signal tone volume Configure the output volume level of the signal tone Signal tone volume Configure the output volume level of the signal tone Ringtone upgrade/delete Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Ringtone upgrade/delete Optional ringtone upgrade with .wav suffix Ringtone upgraded ringtone sare displayed in the delete list and can be deleted selectively Incoming call tesignated ring type setting (alert-info) Value of notification message 1 to 10 Set to point the value of the specified ringtone type	G.722	Select time stamp for G.722 encoding, 160/20ms and 320/20ms can be	
rate For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s Hands-free Configure hands-free call volume level setting Configure hands-free call volume level Microphone Configure the call volume level of the microphone input volume Configure the call volume level of the microphone Broadcast Configure the output volume level when broadcasting volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Mute detection; if VAD is enabled, G.729 payload length cannot be set greater fingtone upgrade/delete Ringtone Uptional ringtone upgrade with .wav suffix Ringtone Optional ringtone upgrade with .wav suffix Ringtone Set copint the value of the specified ringtone type Value of Set to point the value of the specified ringtone type Value of Set to point the value of the specified ringtone type 10 Intervention	timestamp	selected	
volume setting Configure hands-free call volume level Microphone input volume Configure the call volume level of the microphone Broadcast Configure the call volume level of the microphone output Configure the output volume level when broadcasting volume Configure the output volume level of the signal tone Signal tone volume Configure the output volume level of the signal tone Signal tone volume Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Ringtone upgrade/delete Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Ringtone upgrade/delete Optional ringtone upgrade with .way suffix Ringtone upgrade ringtone upgrade with .way suffix Detectively Ringtone The upgraded ringtones are displayed in the delete list and can be deleted selectively Value of notification Set to point the value of the specified ringtone type notification message 1 to 10 Intervent the value of the specified ringtone type		For G723 rate selection, you can choose 5.3kb/s and 6.3kb/s	
setting Image: Configure the call volume level of the microphone Microphone Configure the call volume level of the microphone Broadcast Configure the output volume level when broadcasting volume Configure the output volume level of the signal tone Volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Mute detection; if VAD is enabled, G.729 payload length cannot be set greater activity Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Optional ringtone upgrade with .wav suffix upgrade Optional ringtone upgrade with .wav suffix Ringtone The upgraded ringtones are displayed in the delete list and can be deleted delete selectively Incoming call Set to point the value of the specified ringtone type volification Set to point the value of the specified ringtone type notification Image: Im	Hands-free		
Microphone input volume Configure the call volume level of the microphone Broadcast output Configure the output volume level when broadcasting volume Configure the output volume level when broadcasting Signal tone volume Configure the output volume level of the signal tone Signal tone volume Configure the output volume level of the signal tone Enable voice activity detection Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Ringtone upgrade/delete Optional ringtone upgrade with .wav suffix Ringtone Optional ringtone upgrade with .wav suffix Incoming call the upgraded ring type setting (alert-info) Value of notification message 1 to 10 Set to point the value of the specified ringtone type	volume	Configure hands-free call volume level	
input volume Configure the call volume level of the microphone Broadcast Configure the output volume level when broadcasting output Configure the output volume level when broadcasting volume Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone volume Mute detection; if VAD is enabled, G.729 payload length cannot be set greater activity Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Mute detection; if VAD is enabled, G.729 payload length cannot be set greater Ringtone Uptional ringtone upgrade with .wav suffix Ringtone Optional ringtone upgrade with .wav suffix Ringtone The upgraded ringtones are displayed in the delete list and can be deleted delete selectively Incoming call Set to point the value of the specified ringtone type notification Set to point the value of the specified ringtone type notification Set to point the value of the specified ringtone type notification Intervention message 1 to Intervention 10 Intervention	setting		
input volume Configure the output volume level when broadcasting output Configure the output volume level when broadcasting Signal tone Configure the output volume level of the signal tone volume Configure the output volume level of the signal tone Enable voice Mute detection; if VAD is enabled, G.729 payload length cannot be set greater activity Mute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20ms Optional ringtone upgrade with .wav suffix Ringtone Optional ringtone upgrade with .wav suffix Ringtone The upgraded ringtones are displayed in the delete list and can be deleted delete selectively Incoming call Set to point the value of the specified ringtone type notification Set to point the value of the specified ringtone type notification Set to point the value of the specified ringtone type notification Let upgrade ringtone type notification Set to point the value of the specified ringtone type notification Let upgrade ringtone type	Microphone		
output volumeConfigure the output volume level when broadcastingSignal tone volumeConfigure the output volume level of the signal toneEnable voice activity detectionMute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20msRingtone upgrade/deleteOptional ringtone upgrade with .wav suffixRingtone upgradeOptional ringtone upgrade with .wav suffixRingtone deleteSet civelyIncoming calSet to point the value of the specified ringtone typeValue of notification message 1 to 10Set to point the value of the specified ringtone type	input volume	Configure the call volume level of the microphone	
volumeConfigure the output volume level of the signal toneSignal tone volumeConfigure the output volume level of the signal toneEnable voice activity detectionMute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20msRingtone up=rade/de1eteRingtone up=rade/de1eteRingtone upgradeOptional ringtone upgrade with .wav suffixRingtone deleteSelectivelyIncoming call designated ring type setting (alert-info)Value of notification message 1 to 10Set to point the value of the specified ringtone type	Broadcast		
Signal tone volumeConfigure the output volume level of the signal toneEnable voice activity detectionMute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20msRingtone upgrade•Optional ringtone upgrade with .wav suffixRingtone delete•Set to point the value of the specified ringtone typeValue of notification message 1 to 10Set to point the value of the specified ringtone type	output	Configure the output volume level when broadcasting	
VolumeConfigure the output volume level of the signal toneEnable voice activity detectionMute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20msRingtone upgrade/deleteRingtone upgrade/deleteRingtone upgradeOptional ringtone upgrade with .wav suffixRingtone upgradeThe upgraded ringtones are displayed in the delete list and can be deleted selectivelyIncoming call designated ring type setting (alert-info)Value of notification message 1 to 10Set to point the value of the specified ringtone type	volume		
volumeAn analysisEnable voice activity detectionMute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20msRingtone upgradeOptional ringtone upgrade with .wav suffixRingtone upgradeOptional ringtone upgrade with .wav suffixRingtone upgradeThe upgraded ringtones are displayed in the delete list and can be deleted selectivelyIncoming callSet to point the value of the specified ringtone typeValue of notification message 1 to 10Set to point the value of the specified ringtone type	Signal tone	Configure the output volume level of the signal tang	
activity detectionMute detection; if VAD is enabled, G.729 payload length cannot be set greater than 20msRingtone upgradeImage: Comparison of the set greaterRingtone upgradeOptional ringtone upgrade with .wav suffixRingtone upgradeImage: Comparison of the set greaterRingtone deleteSet comparison of the specified ringtone typeNature of 	volume		
activity than 20ms Ringtone upgrade/delete Ringtone Optional ringtone upgrade with .wav suffix upgrade The upgraded ringtones are displayed in the delete list and can be deleted delete selectively Incoming call designated ring type setting (alert-info) Value of Set to point the value of the specified ringtone type notification Message 1 to 10 Incoming call designated ring type	Enable voice	Mute detection: if VAD is enabled, C 720 perford length connet be set greater.	
detection Ringtone upgrade/delete Ringtone upgrade Optional ringtone upgrade with .wav suffix upgrade The upgraded ringtones are displayed in the delete list and can be deleted delete selectively Incoming call tesignated ring type setting (alert-info) Value of Set to point the value of the specified ringtone type notification Set to point the value of the specified ringtone type 10 Here is a set is a se	activity		
Ringtone upgradeOptional ringtone upgrade with .wav suffixRingtone deleteThe upgraded ringtones are displayed in the delete list and can be deleted selectivelyIncoming call designated ring type setting (alert-info)Value of notification message 1 to 10Set to point the value of the specified ringtone type	detection		
Optional ringtone upgrade with .wav suffix upgrade The upgraded ringtones are displayed in the delete list and can be deleted delete selectively Incoming call designated ring type setting (alert-info) Value of Set to point the value of the specified ringtone type notification message 1 to 10 Incoming call designated ring type setting (alert-info)	Ringtone up	grade/delete	
upgrade The upgraded ringtones are displayed in the delete list and can be deleted Ringtone The upgraded ringtones are displayed in the delete list and can be deleted delete selectively Incoming call designated ring type setting (alert-info) Value of Set to point the value of the specified ringtone type notification message 1 to 10 Incoming call designated ring type	Ringtone	Ontional ringtone ungrade with way suffix	
delete selectively Incoming call designated ring type setting (alert-info) Value of Set to point the value of the specified ringtone type notification Set to point the value of the specified ringtone type 10 Incomine type	upgrade		
Incoming call designated ring type setting (alert-info) Value of notification Set to point the value of the specified ringtone type notification 10	Ringtone	The upgraded ringtones are displayed in the delete list and can be deleted	
Value of notification Set to point the value of the specified ringtone type 10 10	delete	selectively	
notification message 1 to 10	Incoming call designated ring type setting (alert-info)		
message 1 to 10	Value of	Set to point the value of the specified ringtone type	
10	notification		
	message 1 to		
Ring type Type1-Type9	10		
	Ring type	Туре1-Туре9	

9.19 Intercom Settings >> Video Settings

Camera Status	Inactive			
Max Access Num 😧	N/A			
Max M Num	N/A	Use		0
Max S Num	N/A	Use		0
p Camera Settings>>				
Position	ipCameraName		(40 Characters)	
User	admin			
Password	••••			
Ip Camera Brand	XM			
IP				
Port	554			
Main Stream Url				
Sub Stream Url				
User Agent				
H.264 Stream No SPS&PPS				
	Apply			

Figure 30 - Video Setting

Table 14 - Video Setting

Connection mode	Sel	lect external, click submit, restart the device		
Camera settings (external mode)				
Field Name		Description		
Name		Camera name		
Username		External camera login name		
password		External camera login password		
Camera type		Choose a camera manufacturer		
		The IP address of the camera, please use the scanning tool matching the		
IP address		camera to obtain the IP address		
port		Camera port number		
Main stream l	Irl	Click Submit, the camera Url information will be automatically displayed if the		
	JII	connection is successful, and it will not be displayed if it fails		
Substream Url		Click Submit, the camera Url information will be automatically displayed if the		
		connection is successful, and it will not be displayed if it fails		
H.264 stream				
without		Compatible with cameras without SPS&PPS, can display video normally		
SPS&PPS				
Advanced Se	Advanced Settings			
Video Directio	'n	Sendonly: establish video call, and the SDP packet in the invite packet is		
		Sendonly;		

	Sendrecv: to create a call, the SDP package in the invite package is		
	Sendrecv		
RTSP Over TCP	The RTSP goes over the TCP protocol		
H.264 Payload	Set the h. 264 Payload type. The range is between 96 and 127. The default is		
Туре	117		
Default Call	Ontional main stream and substream		
Stream	Optional main stream and substream		
RTSP Information	RTSP Information		
Main Stream Url	Access the main address of RTSP		
Sub Stream Url	Access the child address of RTSP		

9.20 Intercom Settings >> Multicast

The multicast function can be used to send announcements to each member of the multicast simply and conveniently. By setting the multicast key on the device, the multicast RTP stream can be sent to the pre-configured multicast address. By configuring the monitoring multicast address on the device, the RTP stream sent by the multicast address is monitored and played.

Enable Auto Mcast Sip Priority Enable Page Priority	Image: Auto Mcast Timeout I 0 Intercom Priority Image: Auto Mcast Timeout I	Delete Time 10 (5~10s)
Index/Priority	Name	Host:port
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		

Table 1 - MCAST parameters

 Table 2 - MCAST parameters

Parameters	Description
Enable auto Mcast	Send multicast configuration information through Sip Notify
	signaling. After receiving the information, the device configures it
	in the system for multicast monitoring or cancels multicast
	monitoring in the system
Automatic multicast timeout	When the multicast call does not end normally, but for some

reasons, the device can no longer receive the multicast rtp packet,
through this configuration, the monitoring will be cancelled after
the specified time
The priority defined in the current call, 1 is the highest priority, and
10 is the lowest.
Compared with multicast and SIP priority, high priority can be
inserted, low priority is rejected
Regardless of who calls in the two multicasts first, the device will
give priority to the multicast with the higher priority $_\circ$
Listened multicast server name
Listened multicast server's multicast IP address and port.

9.21 Intercom Settings >> Action URL

Action URL Event Settings Active URI Limit IP Setup Completed Registration Succeeded **Registration Disabled Registration Failed** Incoming Call Outgoing Call Call Established Call Terminated DND Enabled DND Disabled Mute Unmute Missed calls IP Changed Idle To Busy Busy To Idle Input1 Reset Input1

Figure 31 - Action URL

Table 15 - Action URL

Action URL Settings
Configure the URL for reporting actions to the server, for example, fill in the UR
http://InternalServer /FileName.xml?
(Internal Server is the IP address of the server, and File Name is the xml file name of the storage
device reporting action)

Note! 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

9.22 Intercom Setting >> Time/Date

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

Time Synchronized via SNTP			
Time Synchronized via DHCP			
Primary Time Server	time.nist.gov		
Secondary Time Server	pool.ntp.org		
Time zone	(UTC+8) China,Singapo	re,Austral 👻	
Resync Period	60	(1~5000)Second(s)	
	Ap	ply	
ylight Saving Time Settings			
Location	China(Beijing)		
Location DST Set Type	China(Beijing) Automatic		
		×	
DST Set Type	Automatic 🔹	Minute	
DST Set Type Fixed Type	Automatic Disabled		
DST Set Type Fixed Type	Automatic Disabled 0	Minute	v
DST Set Type Fixed Type Offset	Automatic Disabled O Start	Minute	v v
DST Set Type Fixed Type Offset Month	Automatic Disabled Start January	Minute End January	

Figure 32 - Time/Date

Table 16 - Time/Date

Field Name	Explanation		
Network Time Server Settings			
Time Synchronized via SNTP	Enable time-sync through SNTP protocol		
Time Synchronized via	Enable time-sync through DHCP protocol		
Primary Time Server	Set primary time server address		
Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device will try to connect to secondary time server to get time synchronization.		
Time zone	Select the time zone		
Resync Period	Time of re-synchronization with time server		
Daylight Saving Time Settings			

Location	Select the user's time zone specific area			
	Select automatic DST according to the preset rules of DST, or			
DST Set Type	the manually input rules			
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	The time set by hand, need to disable SNTP service first			

9.23 Intercom Settings >> Certificate Management

Set whether to enable the license certificate and conventional name verification, and select the certificate module.

Can upload and delete uploaded certificates

Update Trusted Certific	ates File			
	Load Trusted Certific	ates File	Select	Upgrade
Delete Trusted Certifica	ites File			
	Select Trusted Certifi	cates File	Delete	
Trusted Certificates File	3			
File Name	Issued To	Issued By	Expiration	File Size
Trusted Certificates Set	tings			
CA Certificates	Disable	d 💌		
	Apply	/		

Figure 33 - Certificate settings

9.24 Intercom Settings >> Equipment Certificate

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

Device Certificates				
Device Certificates	Custom Certificates Apply	×		
Import Certificates				
Load Device Certificates	File	Sele	ect Upload	
Certification File				
Index File Name	Issued To	Issued By	Expiration	File Size
				Delete

Figure 34 - Device certificate settings

9.25 Security Settings

Input Settings					
Input1					
🗹 Input Detect		Key None			
Trigger Mode	Low Level Trigger(Close T	rigger) 💽 Detection	Duration	0	(0~3600)s
Alert message send to s	server	🖾 Reset A	Alert message send to	server	
Input2					
Input Detect		Key None			
Trigger Mode	Low Level Trigger(Close T	rigger) 💽 Detection	Duration	0	(0~3600)s
Alert message send to s	server	🖾 Reset A	Alert message send to	server	
Output Settings Output1 I Output Response Output Level Output2 I Output Response Output Level	High Level(NC:closed) 💌	Output Duration Output Duration		600)s 600)s	
Alert Trigger Setting Output 1 >> Output 2 >> Ring >>					

Figure 35 - Security Settings

Table 17 - Security Settings

Security Settings	
Field Name	Explanation
Input settings	
Field Name	Explanation
Input Detect	Enable or disable Input Detect
	When choosing the low level trigger (closed trigger), detect the input port (low
Trigger Mode	level) closed trigger.
	When choosing the high level trigger (disconnected trigger), detect the input

	port (high level) disconnected trigger.					
Alert message						
sent to the server	Enable or disable the input port to send messages to the server					
Send reset						
message to server	Enable or disable sending reset messages to the server					
Output Settings						
Output port						
response	Enable or disable output response					
	When low level (NO: open) is selected, when the trigger condition is met, the					
Outrast laure l	trigger NO port is disconnected.					
Output level	When the high level (NC: Close) is selected, the trigger NO port is closed					
	when the trigger conditions are met.					
Output duration	The output port change duration, the default value is 5 seconds.					
Alarm trigger so	etting					
Output Response	Enable or disable Output Response					
	When choosing the low level trigger (NO: normally open), when meet the					
	trigger condition, trigger the NO port disconnected.					
Output Level	When choosing the high level trigger (NO: normally close), when meet the					
	trigger condition, trigger the NO port close.					
Remote DTMF	Receive the DTMF code sent by the remote device, and if it is correct, trigger					
trigger	the corresponding output port.					
	You can choose to enable or disable the ringtone					
DTMF trigger code	During the call, the receiving terminal device sends the DTMF code, and if it is					
	correct, the corresponding output port is triggered.					
Reset code	After receiving the corresponding instruction, the test equipment will reset the					
	state and stop playing the corresponding ringtone					
	Receive the active uri sent by the remote device, and if it is correct, trigger the					
Active Uri trigger	corresponding output port.					
	You can choose to enable or disable the ringtone					
Trigger message	After the test equipment receives the corresponding instruction, if it is correct,					
mgger message	it will trigger the corresponding output port					
Reset message	After receiving the corresponding instruction, the test equipment will reset the					
Reset message	state and stop playing the corresponding ringtone					
Remote SMS	Enable or disable remote SMS triggering. You can choose to enable or					
trigger	disable the ringtone					
Trigger message	Send the command to ALERT = [command] on the remote device or server, if					
	it is correct, trigger the corresponding output port					
Reset message	After receiving the corresponding instruction, the test equipment will reset the					
Root mossayo	state and stop playing the corresponding ringtone					

	1					
	The port outputs a continuous time trigger type, including the trigger					
	condition. For example, the call triggers the output port, and the output port					
	will be in the call state and continue to respond)					
	1 Talking					
	2 Talking and Ringing					
	3 Ringing					
Call status trigger	4 Calling					
	5. Call and talk					
	6. Call and talk (caller)					
	7. Calling and ringing					
	8. Talking and ringing (called)					
	9. Talking and ringing					
	10. Call, ring and talk					
Tamper Alarm S	ettings					
Town on data stice	If the terminal is violently demolished, the terminal is triggered to always play					
Tamper detection	the set alarm ringtone					
Warning	After the alarm is triggered, the command set by the Alarm command is					
instruction	to the server at the same time					
	If you need to stop the alarm ringtone, the remote end can send a short					
Alarm rocovoru	message to the terminal. The content of the short message is the value set in					
Alarm recovery	the Reset command. At this time, the test terminal will stop the alarm ringtone					
	playback					
Alarm status	Benet to stop the playback of the ringtons					
recovery	Reset to stop the playback of the ringtone					
Ring Type	Ringtone can be set to none / preset					
Server Settings						
	Configure the remote response server address (including the remote					
	response server address and trigger alarm server address, IP:PORT, SIP					
	number). When the input port is triggered, a short message will be sent to the					
Server Address	server. The message format is as follows: Alarm Info: Description=i12;SIP					
	User=;Mac=00:a8:34:68:23:d1;IP=172.18.2.243;port =Input1 (support					
	variables and strings)					
-						

9.26 Function Key >> Function Key

> Key Event

You can set the function type of these keys to Key Event, and there are multiple options for sub-types.

Figure 36 - Function keys

Туре	Subtype	Usage
	None	No responding
	Redial	User can redial the last number dialed
	Call Back	Call the nearest missed number
	Release	Delete password input, cancel dialing input and
Function		end call
keys	ОК	Identification key
	Handfree	Use as hands-free button
	VOL UP	volume adjustment
	VOL DOWN	volume adjustment

> Hot Key

Enter the phone number in the input box. When you press the shortcut key, the device will call out the set phone number. This button can also be used to set the IP address, press the shortcut key to make an IP direct call.

Function Key Settings

Key	Туре		Number 1	Number 2	Line	e	Subtype	6
Dss Key 1	Hot Key				SIP1	-	Speed Dial	
Dss Key 2	None	-			SIP1	-	Speed Dial	
Dss Key 3	None	-			SIP1	+	Intercom	
Dss Key 4	None	-			SIP1	-	Speed Dial	
Dss Key 5	None	-			SIP1	-	Speed Dial	~
Dss Key 6	None				SIP1	-	Speed Dial	-

Figure 37 - Hot Key Settings

Table 19 - Hot Key Settings

Туре	Num ber	Line	Subtype	Usage
------	------------	------	---------	-------

Hot Key	Fill			Using Speed Dial mode together with Enable Speed Dial Hangup Enable v can
	called party'	The SIP	Speed Dial	define whether this call is allowed to be hung up by re-pressing the speed dial
	s SIP accou nt or	account correspondi ng lines	Intercom	key. In Intercom mode, if the caller's IP phone supports Intercom feature, the
	IP addre ss			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	Number 1	Number 2	Line	1	Subtype
Dss Key 1	Multicast 📃			SIP1	w	G.711A
Dss Key 2	None			SIP1	-	G.711A
Dss Key 3	None			SIP1	-	G.711U
Dss Key 4	None			SIP1	+	G.722
Dss Key 5	None 💌			SIP1	-	G.723.1
Dss Key 6	None			SIP1	-	G.726-32 G.729AB

Figure 38 - Multicast Settings

Table 20 - Multicast Settings

Туре	Number	Subtype	Usage		
	Set the host IP address and	G.711A	Newsylend encode and inc. (41/hz)		
	port number, they must be	G.711U	Narrowband speech coding (4Khz)		
	separated by a colon (The	G.722	Wideband speech coding (7Khz)		
	IP address range is	G.723.1			
Multicast	224.0.0.0 to	G.726-32			
	239.255.255.255, and the		Nerrowhand apaceh apding (4Khz)		
	port number is preferably	0 700 4 D	Narrowband speech coding (4Khz)		
	set between 1024 and	G.729AB			
	65535)				

> PTT

Keep pressing the set shortcut key to make a call, release it and hang up

Key	Type	Number 1	Number 2	Line	Э	Subtype	
Dss Key 1	PTT	•		SIP1		Speed Dial	
Dss Key 2	None			SIP1	*	Speed Dial	0.43
Dss Key 3	None			SIP1	-	Intercom	
Dss Key 4	None			SIP1	-	Multicast	-
Dss Key 5	None	•		SIP1	-	Speed Dial	-
Dss Key 6	None			SIP1	+	Speed Dial	

> Advanced Settings

Jse Function Key to Answer	Enable 💌		Enable Speed Dial Hangup	Enable 💌		
Hot Key Dial Mode Select	Main-Se	condary 💌				
Call Switched Time	16	(5~50)Second(s)				
Day Start Time	06:00	(00:00~23:59)	Day End Time	18:00	(00:00~23:59)	
Speed Dial Time	1	0				

Figure 39 - Advanced Settings

Table 21 - Advanced Settings

Advanced Settings				
Field Name	Explanation			
Input port is				
multiplexed as function	Enable or disable the input port to be multiplexed as speed dial button 2			
key 2				
Use Function Key to	Enable or disable shortcuts to answer calls			
Answer				
Enable Speed Dial	Enable or disable shortcuts to hang up calls			
Hang up				
	Number 1 call number 2 mode selection.			
	<main secondary="">: If the first number is not answered within the set</main>			
Hot Key Dial Mode	time, the second number will be automatically switched.			
Select	<day night="">: The system time is automatically detected during the call.</day>			
	If it is daytime, the first number is called, otherwise the second number			
	is called.			
Call Switched Time	Set number 1 to call number 2 time, default 16 seconds			
Day Start Time	The start time of the day when the <day night=""> mode is defined. Default</day>			
Day Start Time	"06:00"			
Day End Time	The end time of the day when the <day night=""> mode is defined. Default</day>			
Day End Time	"18:00			

10 Trouble Shooting

When the device is not working properly, users 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

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

Device information (model, software and hardware version) and Internet Information etc.

10.2 Reboot Device

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

10.3 Device Factory Reset

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

To restore the factory settings, you need to log in to the webpage [**System**] >> [**Configuration**], and click [**Reset**] button, the device will return to the factory default state.

10.4 Network Packets Capture

In order to obtain the data packet of the device, the user needs to log in to the webpage of the device, open the webpage [**System**] >> [**Tools**], and click the [**Start**] option in the "Network Packets Capture". A message will pop up asking the user to save the captured file. At this time, the user can perform related operations, such as starting/deactivating the line or making a call, and clicking the [**Stop**] button on the webpage after completion. Network packets during the device are saved in a file. Users can analyze the packet or send it to the Fanvil Technical Support mailbox.

10.5 Common Trouble Cases

Trouble Case	Solution
Device could not boot up	1. If the device enters "POST mode" (the SIP/NET and function
	button indicators are always on), the device system is damaged.

Table 22 - Common Trouble Cases

Please contact your location technical support to help you restore
your equipment system.
2. If the device enters "POST mode" (the SIP/NET and function
button indicators are always on), the device system is damaged.
Please contact your location technical support to help you restore
your equipment system.
1. Please check if the device is connected to the network. The
network cable must be connected to the 📲 [Network] interface
instead of the 🗕 [Computer] interface.
2. Please check if the device has an IP address. Check the system
information. If the IP address is Negotiating, the device has not
obtained an IP address. Please check if the network configuration is
correct.
3. If the network connection is good, please check your line
configuration again. If all configurations are correct, contact your
service provider for support, or follow the instructions in "10.4 Network
Data Capture" to obtain a registered network packet and send it to the
Fanvil Support Email to help analyze the issue.