# **SPA2 SIP Gateway User Manual**



Document VER	Firmware VER	Explanation	Time
V1.0	2.1.1.2545	Initial issue	20170518

# **Safety Notices**

- 1. Please use the specified power adapter. If you need to use the power adapter provided by other manufacturers under special circumstances, please make sure that the voltage and current provided is in accordance with the requirements of this product, meanwhile, please use the safety certificated products, otherwise may cause fire or get an electric shock.
- 2. When using this product, please do not damage the power cord either by forcefully twist it, stretch pull, banding or put it under heavy pressure or between items, otherwise it may cause damage to the power cord, lead to fire or get an electric shock.
- 3. Before using, please confirm that the temperature and environment is humidity suitable for the product to work. (Move the product from air conditioning room to natural temperature, which may cause this product surface or internal components produce condense water vapor, please open power use it after waiting for this product is natural drying).
- 4. Please do not let non-technical staff to remove or repair. Improper repair may cause electric shock, fire, malfunction, etc. It would lead to injury accident or cause damage to your product.
- 5. Do not use fingers, pins, wire, other metal objects or foreign body into the vents and gaps. It may cause current through the metal or foreign body, which may even cause electric shock or injury accident. If any foreign body or objection falls into the product please stop using.
- 6. Please do not discard the packing bags or store in places where children could reach, if children trap his head with it, may cause nose and mouth blocked, and even lead to suffocation.
- 7. Please use this product with normal usage and operating, in bad posture for a long time to use this product may affect your health.
- 8. Please read the above safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

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	b) c) d) e) f) g) (2) a) b) (3) a) b) (4)	Account错误!未定义书签 Configurations	<ol> <li>12</li> <li>12</li> <li>13</li> <li>14</li> <li>15</li> <li>16</li> <li>16</li> <li>17</li> <li>18</li> <li>23</li> <li>24</li> </ol>
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	(6)	Function Key	错误!未定义书签。
	a)	Function Key Settings	
V. /	APPE	NDIX	
1.	TEC	HNICAL PARAMETERS	
2.	BAS	SIC FUNCTIONS	

### I. Product introduction

SPA2 is dedicated to the needs of industry users to develop a SIP audio and video intercom, voice transmission using standard IP / RTP protocol, video transmission using RTSP. It is well inherited Voptech phone stability, carrier-class sound quality advantages, and perfectly compatible with all current SIP-based mainstream IPPBX / softswitch / IMS platform, such as Asterisk, Broadsoft, 3CX, Elastix and so on. It is a variety of functional interface in one, call, radio, video, security, recording, to adapt to a variety of use of the environment, convenient and rapid deployment of equipment, is ideal for everyone.

#### **1.** Appearance of the product



#### 2. Description



Label	Description	Label	Description	Label	Description
1	Speaker interface	2	Headset interface	3	LED interface
4	Function key interface	5	Microphone interface	6	Switch input interface
7	Switch output interface	8	Power input interface	9	Camera interface
10	Ethernet interface	(11)	Registration/Network LED	(12)	Volume control key

(13)	Restore factory key	(14)	Recording output interface	(15)	Grounding screw
------	---------------------	------	----------------------------	------	-----------------

## **II. Start Using**

Before you start to use the equipment, please make the following installation.

### 1. Confirm the connection

Confirm whether the equipment of the power cord, network cable and the boot-up is normal. (Check the network state of light



Label	Explanation		
① Speaker interface	according to the device input voltage adaptive output maximum power; $4\Omega$ speaker, POE / 8W, 12V / 10W, 18V / 20W, 24V / 30W. The greater the horn impedance, the smaller the output power. Suggested wire diameter: 18AWG or larger diameter.		
2 Headset interface	speaker audio signal output, for external headset or active speakers.		
③ LED interface	5V output, can be an external LED, indicating the network status, call status, registration status.		
Function key interface connection switch, you can log on page set the call number or IP address			

	one interfacethe proposed use of electret condenser microphone, sensitivity: -38dB, bias voltage 2.2V. Microphone signal cable it is recommended to use a shielded cable and connect the shield cable to the grounding screw, improve anti-interference.upput interfaceConnect an infrared probe or emergency switch or Doorsensor and other switch components.upput interfaceCorresponding to the short-circuit input interface, login device security page settings, you can control the alarm light, electric locks and other equipment; with the adjacent (®) power port connection for external equipment power supply.put interface12V ~ 24V 2A input, according to the input voltage to determine the maximum output power amplifier.nterfaceStandard RJ45 interface, connect the original camera, the proposed use of five or five sub-network cableinterfaceWAN port, standard RJ45 interface, 10 / 100M adaptive, support POE input, it is recommended to use five or super five network cable.on/Networkindicates network status, call status, registration status. Fast flashing: network anomaly or SIP account exception. Slow flashing: during a call. Always bright: successful registration.standby to adjust the volume of the ringtone, call only adjust the call volume, broadcast only adjust the broadcast volume. Long press the volume down key to
<b>(5)</b> Microphone interface	2.2V. Microphone signal cable it is recommended to use a shielded cable and connect
(5) Microphone interfacethe proposed use of 2.2V. Microphone sign the shield cable to the Connect an infrared components.(6) Switch input interfaceConnect an infrared components.(7) Switch output interfacecorresponding to the you can control the a (®) power port conne(8) Power input interface12V ~ 24V 2A input, a power amplifier.(9) Camera interfacestandard RJ45 interfa sub-network cable(10) Ethernet interfaceWAN port, standard recommended to use indicates network star or SIP account excer registration.(11) Registration/Network LEDstandby to adjust th broadcast only adjus broadcast the IP add acquisition mode (spe press and hold for 3 settings.(12) Recording output interfacepress and hold for 3 settings.	the shield cable to the grounding screw, improve anti-interference.
	Connect an infrared probe or emergency switch or Doorsensor and other switch
<ul><li>6 Switch input interface</li><li>7 Switch output</li></ul>	components.
3 Cuitate autout	corresponding to the short-circuit input interface, login device security page settings,
() Switch output	you can control the alarm light, electric locks and other equipment; with the adjacent
Interface	8 power port connection for external equipment power supply.
	12V ~ 24V 2A input, according to the input voltage to determine the maximum output
<ul> <li>(8) Power input interface</li> <li>(9) Camera interface</li> </ul>	power amplifier.
	standard RJ45 interface, connect the original camera, the proposed use of five or five
<ul> <li>④ Camera interface</li> <li>⑩ Ethernet interface</li> </ul>	sub-network cable
Image: Sub-Network cable         Image: Su	WAN port, standard RJ45 interface, 10 / 100M adaptive, support POE input, it is
	recommended to use five or super five network cable.
1 Ethernet interface	indicates network status, call status, registration status. Fast flashing: network anomaly
(II) Registration/Network	or SIP account exception. Slow flashing: during a call. Always bright: successful
LED	registration.
	standby to adjust the volume of the ringtone, call only adjust the call volume,
<ul> <li>(2) Switch output interface</li> <li>(3) Power input interface</li> <li>(3) Power input interface</li> <li>(3) Camera interface</li> <li>(4) Camera interface</li> <li>(5) Camera interface</li> <li>(7) Camera interface</li> <li>(8) Power input interface</li> <li>(9) Camera interface</li> <li>(9) Camera interface</li> <li>(12V ~ 24V 2A input, according to the input voltage to determine the maximum ou power amplifier.</li> <li>(9) Camera interface</li> <li>(9) Ethernet interface</li> <li>(9) Ethernet interface</li> <li>(12V ~ 24V 2A input, according to the original camera, the proposed use of five or sub-network cable</li> <li>(10) Ethernet interface</li> <li>(11) WAN port, standard RJ45 interface, 10 / 100M adaptive, support POE input, recommended to use five or super five network cable.</li> <li>(11) Indicates network status, call status, registration status. Fast flashing: network anor or SIP account exception. Slow flashing: during a call. Always bright: succe registration.</li> <li>(12) Volume control key</li> <li>(13) Restore factory key</li> <li>(13) Restore factory key</li> <li>(13) Recording output interface</li> <li>(14) Recording output interface</li> </ul>	
(12) Volume control key	broadcast the IP address. Long press the volume plus key to switch the IP address
	acquisition mode (specific operation see below search door phone).
	press and hold for 3 seconds to flash the device to restart and restore the factory
(13) Restore factory key	settings.
(1) Recording output	Wicrophone interface       2.2V. Microphone signal cable it is recommended to use a shielded cable and connect the shield cable to the grounding screw, improve anti-interference.         Switch input interface       Connect an infrared probe or emergency switch or Doorsensor and other switch components.         Switch output interface       Corresponding to the short-circuit input interface, login device security page settings, you can control the alarm light, electric locks and other equipment; with the adjacent (®) power port connection for external equipment power supply.         Power input interface       12V ~ 24V 2A input, according to the input voltage to determine the maximum output power amplifier.         Camera interface       standard RJ45 interface, connect the original camera, the proposed use of five or five sub-network cable         Ethernet interface       WAN port, standard RJ45 interface, 10 / 100M adaptive, support POE input, it is recommended to use five or super five network cable.         ED       indicates network status, call status, registration status. Fast flashing: network anomaly or SIP account exception. Slow flashing: during a call. Always bright: successful registration.         Folume control key       standby to adjust the volume of the ringtone, call only adjust the call volume, broadcast only adjust the broadcast volume. Long press the volume down key to broadcast the IP address. Long press the volume plus key to switch the IP address acquisition mode (specific operation see below search door phone).         estore factory key       press and hold for 3 seconds to flash the device to restart and restore the factory settings.         e
interface	
	external parts metal housing needs to be connected to this interface to prevent static
(15) Grounding screw	electricity and other interference caused the equipment to work abnormally.

### 2. Quick Setting

The product provides a complete function and parameter setting. Users may need to have the network and SIP protocol knowledge to understand the meaning all parameters represent. In order to let equipment users, enjoy the high quality of voice service and low cost advantage brought by the device immediately, here we list some basic but necessary setting options in this section to let users know how to operate device without understanding such complex SIP protocols.

In prior to this step, please make sure your broadband Internet can be normally operated, and you must complete the connection of the network hardware. The product factory default network mode is static IP address 192.168.1.128.

Press and hold volume down key for 3 seconds; the door phone would report the IP address by voice. Or you can also use the "iDoorPhoneNetworkScanner.exe" software to find the IP address of the device. (download address <u>www.Voptech.com</u>)

- Long press the volume plus key for 10 seconds, the speaker issued a rapid beep, and then quickly press the three volumes plus the key, beep stopped. Wait 10 seconds, successfully switch to dynamic IP after the system automatically voice broadcast IP address. Switching again will become a fixed IP address.
- > Note: when the device is powered on, 30s waiting is needed for device running.
- Log on to the WEB device configuration.
- In a line configuration page, service account, user name, server address and other parameters are required for server address registration.

8	DoorPhone Network	Scanner(V 1.0)				×
#	IP Address	Serial Number	MAC Address	SW Version	Description	
1	192.168.1.128	SPA2	00:a8:34:68:23:a3	2.1.1.2834	SPA2	
						<u>R</u> efresh

### **III. Basic operation**

#### 1. Answer a call

When a call comes in, the device would answer automatically. If you cancel auto answer feature and set auto answer time, you would hear the ring at the set time and the device would auto answer after configured timer.

#### 2. Call

Configure Function key as hot key and then set up a number; after that you might press the Function key for making call to the configured extension(s).

#### 3. End call

Enable Release (You can set Function key to Release) key for hanging up feature to end call.

#### 4. Security linkage

- switch input interface received a door or emergency button and other sensor signals, the output port connected to the alarm lights or electric locks and other equipment will automatically respond to the server and send alarm information.
- The output port defaults to the call automatically triggers the response and supports call triggering with DTMF number triggering and short message triggering.

#### 5. Video linkage

Use another manufacturers camera please connect to the switch, the device 

 interface can only connect the original camera.

 Landing page configuration camera user name, password, port number and other information. For more information, please refer to the Video settings

## **IV.Page settings**

### 1. Browser configuration

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

Enter the user **admin** and password **admin** and click the Logon button to enter the settings screen.

User:	
Password:	
Language:	English 🗸
	Logon

### 2. Password Configuration

There are two levels of access: root level and general level. A user with root level can browse and set all configuration parameters, while a user with general level can set all configuration parameters except server parameters for SIP.

- General level: It is not be set by default, you can add the feature when you need
- User uses root level by default:
  - User name: admin
  - Password: admin
- 3. Configuration via WEB
- (1) System
- a) Information

	Information	Account	Configurations	Upgrade	Auto Provision	FDMS	Tools
> System							
› Network	System Informat	ion					
› Line	Hardware: Software:		SPA2 2.1 2.1.1.2843				
› Intercom settings	Uptime: Last uptime:		28:06:3 41:52:03	3			
> Security settings	MEMInfo:		ROM: 0.8/8	(M) RAM: 2.1/16	5(M)		
<ul> <li>Intercom settings</li> <li>Security settings</li> <li>Function Key</li> </ul>	Network mod	e:	DHCP				
	MAC: IP:		00:a8:23:0	70			
	Subnet mask: Default gatew	: /ay:	255.255.0 172.18.1.1	.0			
	SIP Accounts						
	Line 1 Line 2	8308 801	In Re	active gistered			

Information	
Field Name	Explanation
System	Display equipment model, hardware version, software version, uptime, last uptime and
Information	MEMInfo.
Notwork	Shows the configuration information of WAN port, including connection mode of WAN
Network	port (Static, DHCP, PPPoE), MAC address, IP address of WAN port.
SIP Accounts	Shows the phone numbers and registration status of the 2 SIP LINES.

### b) Account

Through this page, administrator can add or remove user accounts depend on their needs or modify existed user accounts permission.

	Information Account	Configurations	Upgrade	Auto Provision	FDMS	Tools	ן
> System							
<ul> <li>System</li> <li>Network</li> <li>Line</li> <li>Intercom settings</li> <li>Security settings</li> <li>Function Key</li> </ul>	Change Web Authentication Pa	assword					
	Old Password:						
	New Password:						
<ul> <li>System</li> <li>Network</li> <li>Line</li> <li>Intercom settings</li> <li>Security settings</li> <li>Function Key</li> </ul>	Confirm Password:						
<ul> <li>Intercom settings</li> <li>Security settings</li> </ul>			Apply				
	Add New User						
<ul> <li>System</li> <li>Network</li> <li>Line</li> <li>Intercom settings</li> <li>Security settings</li> <li>Function Key</li> </ul>	Username						
	Web Authentication Passwor	rd 📃					
<ul> <li>System</li> <li>Network</li> <li>Line</li> <li>Intercom settings</li> <li>Security settings</li> <li>Function Key</li> </ul>	Confirm Password						
	Privilege	Ad	Iministrators 🔻				
			Add				
	User Accounts						
	User	Privilege	3				
	admin	Administra	tors		Delete		

Account				
Field Name	Explanation			
Change Web Au	thentication Password			
You can modify the login password of the account				
Add New User				
You can add new	v user			
User Accounts				
Show the existed	user accounts' information			

## c) Configurations

	Information Account	Configurations Upgrade Auto Provision FDMS Tools
> System		
> Network	Export Configurations	Right click here to SAVE configurations in 'txt' format.
> Line	Import Configurations	Right click here to SAVE configurations in 'xml' format.
> Intercom settings		Configuration file: Select Import
<ul> <li>Security settings</li> </ul>	Reset to factory defaults	
› Function Key		ALL USER'S DATA WILL BE LOST AFTER RESET! Reset

Configurations	
Field Name	Explanation
Export	Save the equipment configuration to a txt or xml file. Please right click on the
Configurations	choice and then choose "Save Link As."
Import	Find the config file, and pross <b>Undate</b> to load it to the equipment
Configurations	Find the coming me, and press <b>Opdate</b> to load it to the equipment.
Reset to factory	SPA2 would restore to factory default configuration and remove all configuration
defaults	information.

### d) Upgrade

	Information A	ccount Configurations	Upgrade	Auto Provision	FDMS	Tools
> System						
> Network	Software upgrade	Current Software Version:	2.1.1.2843			
> Line		System Image File		Sele	ct Upgra	de

Upgrade	
Field Name	Explanation
Software upgrad	de

### e) Auto Provision

	Information	Account	Configurations	Upgrade	Auto Provision	FDMS	Tools	
> System								
	Common Settin	gs						
> Network	Information         Account         Configurations         Upgrade         Auto Provision         FDMS         Tools           Common Settings         Current Configuration Version         Ostoreal Configuration File Encryption         Static Provisioning Server >>         TR069 >>         Static Provisioning Server >>         TR069 >>         TR069 >>         TR069 >>         TR069 >>         Apply         Static Provision Information         Show the current config file's version. If the config file to be downloaded is higher than current version, the configuration would be upgraded. If the endpoints confirm the configuration by the Digest method, the configuration would not be upgraded unless it differs from the current configuration to be downloaded and this configuration is the same, the auto provision would stop. If the endpoints confirm the configuration by the Digest method, the configuration would not be upgraded unless it differs from the current configuration.         Serial number of the equ							
> Line	General Con CPE Serial N	figuration Version umber	00100400FV02	0010000000a823	6a6d76			
> Intercom settings	Authenticati	on Name						
	Configuratio	on Password n File Encryption Ke	у					
<ul> <li>Security settings</li> </ul>	General Con Key	figuration File Encry	ption					
> Function Key	Save Auto P	rovision Information						
	STP Plug and Pl	, av (DnD) >>						
	Static Provision	ing Server >>						
	TR069 >>							
			Apply					
Auto Provision								
Field Name	Explanatio	n						
Common Settings								
	Show the c	urrent conf	ig file's versi	on. If the c	onfig file to	be downlo	aded is higher	
Current	than current version, the configuration would be upgraded. If the endpoints							
Configuration	confirm the configuration by the Digest method, the configuration would not be							
Version	ungraded unloss it differs from the current configuration							
	Show the common config file's version. If the configuration to be downloaded and							
General	this config		o como tho		ion would	stop If the	andnointe	
Configuration	this configuration is the same, the auto provision would stop. If the endpoints							
Version	contirm the configuration by the Digest method, the configuration would not be							
	upgraded u		ers from the	current co	ntiguration	•		
CPE Serial Number	Serial num	$\frac{\text{per of the e}}{c}$	quipment		c /	- /		
Authentication	Username	for configur	ration server.	It is used	for FTP/HTT	P/HTTPS. It	this is blank,	
Name	the phone	would use a	anonymous a	access				
Authentication	Password f	or configura	ation server.	It is used fo	or FTP/HTTF	P/HTTPS.		
Password					,	,		
Configuration File	Encryption	key for the	configuratio	n file				
Encryption Key			comparatio	e				
General								
Configuration File	Encryption	key for con	nmon config	uration file				
Encryption Key								
Save Auto Provision	Save the au	ito provisio	n username	and passw	ord in the p	hone until	the server url	
Information	changed							
DHCP Option								
	The equipn	nent suppo	rts configura	tion from (	Option 43, C	Option 66, c	or a Custom	
Option Value	DHCP optic	on. It may a	lso be disabl	ed.				

Custom Option	Custom option number. It must be from 128 to 254.
Value	
SIP Plug and Play (Pnl	P)
	If it is enabled, the equipment would send SIP SUBSCRIBE messages to the server
Enable SID DeD	address when it boots up. Any SIP server compatible with that message would
	reply with a SIP NOTIFY message containing the Auto Provisioning Server URL
	where the phones can request their configuration.
Server Address	PnP Server Address
Server Port	PnP Server Port
Transportation	
Protocol	PhP transfer protocol – ODP of TCP
Update Interval	Interval time for querying PnP server. Default is 1 hour.
Static Provisioning Se	rver
Conver Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be an IP
Server Address	address or domain name with subdirectory.
Configuration File	Specify configuration file name. The equipment would use its MAC ID as the
Name	config file name if this is blank.
Protocol Type	Specify the Protocol type FTP, TFTP or HTTP.
Update Interval	Specify the update interval time. Default is 1 hour.
	1. Disable – not to update
Update Mode	2. Update after reboot – update only after reboot.
	3. Update at time period – update at periodic update period
TR069	
Enable TR069	Enable/Disable TR069 configuration
ACS Server Type	Select Common or CTC ACS Server Type.
ACS Server URL	ACS Server URL.
ACS User	User name of ACS.
ACS Password	ACS Password.
TR069 Auto Login	Enable/Disable TR069 Auto Login.
INFORM Sending	Time between transmissions of "Inform", the unit is second
Period	

f) FDMS

	Information Account	Configurations	Upgrade	Auto Provision	FDMS	Tools	]
> System							
> Network	FDMS Settings Enable FDMS						
› Line	FDMS Interval	3600					
> Intercom settings	Doorphone Info Settings Community Name						
> Security settings	Building Number Room Number						
> Function Key		Apply					

FDMS Settings	
Enable FDMS	Enable/Disable FDMS configuration
EDMS Interval	The time to send sip Subscribe information to the FDMS server on a regular basis.
FDIVIS Interval	Unit seconds
Doorphone Info Setti	ngs
Community Name	The name of the community where the device is installed
Building Number	The name of the building where the equipment is installed
Room Number	The name of the room where the equipment is installed

#### g) Tools

	Information Account	Configurations	Upgrade	Auto Provision	FDMS	Tools
> System						
> Network	<b>Syslog</b> Enable Syslog					
> Line	Server Address Server Port	0.0.0.0				
› Intercom settings	APP Log Level SIP Log Level	None None	<b>v</b>			
Security settings	Network Packets Capture	енни				
Function Key	Robert Dhone	Start				
	Rebot Fibile	Click [Reboot] b	outton to restart th	e phone!		

Syslog is a protocol used to record log messages using a client/server mechanism. The Syslog server receives the messages from clients and classifies them based on priority and type. Then these messages would be written into a log by rules which the administrator has configured.

There are 8 levels of debug information.

Level 0: emergency; System is unusable. This is the highest debug info level.

Level 1: alert; Action must be taken immediately.

- Level 2: critical; System is probably working incorrectly.
- Level 3: error; System may not work correctly.
- Level 4: warning; System may work correctly but needs attention.

Level 5: notice; It is normal but significant condition.

Level 6: Informational; It is normal daily messages.

Level 7: debug; Debug messages normally used by system designer. This level can only be displayed via telnet.

Tools	
Field Name	Explanation
Syslog	
Enable Syslog	Enable or disable system log.
Server Address	System log server IP address.
Server Port	System log server port.
APP Log Level	Set the level of APP log.
SIP Log Level	Set the level of SIP log.
Network Packet	s Capture
Capture a packe	t stream from the equipment. This is normally used to troubleshoot problems.
Reboot Phone	
Some configurat	ion modifications require a reboot to become effective. Clicking the Reboot button
would lead to re	boot immediately.
Note: Be sure to	save the configuration before rebooting.

### (2) Network

### a) Basic

	Basic VPN		
→ Svstem	Network Status		
	IP:	172.18.2.170	
> Network	Subnet mask:	255.255.0.0	
- Hetwork	Default gateway:	172.18.1.1	
> Line	MAC:	00:a8:23:6a:6d:76	
	Settings		
> Intercom settings	Static IP 🔘	DHCP   PPPoE	
	DNS Server Configured by	DHCP	
> Security settings	Primary DNS Server		
	Secondary DNS Server		
Function Key		Apply	
	Service Port Settings 😯		
	Web Server Type	НТТР 🔻	
	HTTP Port	80	
	HTTPS Port	443	
		Apply	
	HTTPS Certification File: http	ps.pem N/A Upload Delete	

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 Timestamp	Get the MAC address's time.		
Settings			
Select the appropriate netwo	ork mode. The equipment supports three network modes:		
Static ID	Network parameters must be entered manually and would not change. All		
Static IP	parameters are provided by the ISP.		
DHCP	Network parameters are provided automatically by a DHCP server.		
	Account and Password must be input manually. These are provided by		
PPPOE	your ISP.		
If Static IP is chosen, the screen below would appear. Enter values provided by the ISP.			
DNS Server Configured by	Select the Configured mode of the DNS Server.		
Primary DNS Server	Enter the server address of the Primary DNS.		
Secondary DNS Server	Enter the server address of the Secondary DNS.		
After entering the new settings, click the <b>Apply</b> button. The equipment would save the new settings and			
apply them. If a new IP address was entered for the equipment, it must be used to login to the phone			
after clicking the Apply b	after clicking the <b>Apply</b> button.		

#### b) VPN

The device supports remote connection via VPN. It supports both Layer 2 Tunneling Protocol (L2TP) and OpenVPN protocol. This allows users at remote locations on the public network to make secure connections to local networks.



	Basic VPN			
tem V	'irtual Private Network (VPN	I) Status		
		VPN IP Add	ress:	0.0.0
letwork V	/PN Mode			
ne		Enable VPN		
		L2TP 🔘		OpenVPN 🖲
tercom settings	aver 2 Tunneling Protocol (	L2TP)		
	-,	L2TP Serve	er Address	
curity settings		Authentica	tion Name	
nction Key		Authentica	tion Password	
			[	Apply
-	penVPN Files			
	OpenVPN Configuration file	: 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

Field Name	Explanation		
VPN IP Address	Shows the current VPN IP address.		
VPN Mode			
Enable VPN	Enable/Disable VPN.		
L2TP	Select Layer 2 Tunneling Protocol		
	Select OpenVPN Protocol. (Only one protocol may be activated. After the selection		
Орентири	is made, the configuration should be saved, and the phone be rebooted.)		
Layer 2 Tunneling Protocol (L2TP)			
L2TP Server	Set VPN L2TP Server IP address.		
Address			
Authentication	Set User Name access to VPN L2TP Server.		
Name			
Authentication	Set Dessword access to VDN LOTD Server		
Password			
Open VPN Files			
Upload or delete Open VPN Certification Files			

# (3) Line

### a) SIP

You can configure a SIP server on this page.

	SIP	Basic Settings					
) System							
, system							
> Network	Line Basic Sel	tings >>					
> Line	Line	Status	Registered		SIP Proxy Server Address	172.18.1.88	
	Pnor	ne number	5521		SIP Proxy Server Port	5060	
<ul> <li>Intercom settings</li> </ul>	Disp	iay name ientication Name	5521		Backup Proxy Server Address Backup Proxy Server Port	5060	
> Security settings	Auth	entication Password	•••••		Outbound proxy address		
	Activ	vate			Outbound proxy port Realm		
> Function Key	Codecs S	ettings >>					
	Advance	d Settinas >>					
		-	Apply				
Codecs Settings >>							
Disabled Codecs			Er	abled Cod	lecs		
	^	$\rightarrow$	G	.722	<b>^</b> ↑		
		<u> </u>	G	.7110 .711A			
	L		G	.729AB	× •		
Advanced Settings >>	>						
Call Forward Uno	onditional			Enable A	uto Answering		
Call Forward Num	ber for				uco Answering		Casand(a)
Unconditional				AULO ANS	wenng Delay	D	Second(s)
Call Forward on E	Busy			Subscribe	e For Voice Message		
Call Forward Num	ider for Busy			Voice Me	ssage Number		
Call Forward on N	No Answer			Period		3600	Second(s)
Call Forward Num Answer	iber for No						
Call Forward Dela	ay for No	5		Enable H	otline		
Hotline Delay		0 (0~120)Second(s)	9)Second(s)	Hotline N	umber		
,			-,(-,				
Enable DND				Ring Type	2	Default 🗸	
Blocking Anonym	ous Call			Conferen	ce Type	Local 🗸	
Use 182 Respons waiting	e for Call			Server Co	onference Number		
Anonymous Call 9	Standard	None ~		Transfer	Timeout	0	Second(s)
Dial Without Regi	stered			Enable L	ong Contact		
Click To Talk				Enable U	se Inactive Hold		
User Agent				Enable M	issed Call Log	$\checkmark$	
Use Quote in Dis	play Name			Response	e Single Codec		

Use Feature Code			
Enable DND		DND Disabled	
Enable Call Forward Unconditional		Disable Call Forward Unconditional	
Enable Call Forward on Busy		Disable Call Forward on Busy	
Enable Call Forward on No Answer		Disable Call Forward on No Answer	
Enable Blocking Anonymous Call		Disable Blocking Anonymous Call	
Specific Server Type	COMMON ~	Enable DNS SRV	
Registration Expiration	60 Second(s)	Keep Alive Type	UDP 🗸
Use VPN	$\checkmark$	Keep Alive Interval	30 Second(s)
Use STUN		Sync Clock Time	
Convert URI	$\checkmark$	Enable Session Timer	
DTMF Type	AUTO 🗸	Session Timeout	0 Second(s)
DTMF SIP INFO Mode	Send */# ~	Enable Rport	$\checkmark$
Transportation Protocol	UDP 🗸	Enable PRACK	$\checkmark$
SIP Version	RFC3261 V	Keep Authentication	
Caller ID Header	FROM 🗸	Auto TCP	
Enable Strict Proxy		Enable Feature Sync	
Enable user=phone	$\checkmark$	Enable GRUU	
Enable SCA		BLF Server	
Enable BLF List		BLF List Number	
SIP Encryption		RTP Encryption	
SIP Encryption Key		RTP Encryption Key	
	Apply		

SIP				
Field Name	Explanation			
Basic Settings (Choose th	ne SIP line to configured)			
Line Chature	Display the current line status after page loading. To get the up to date line			
	status, user must 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 Name	Enter the authentication name of the service account			
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 ID or FODN address of the SID providence			
Address	Enter the P or FQDN address of the SIP proxy server			
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 the			
address	service provider			
Outbound proxy port	Enter the outbound proxy port, default is 5060			
Realm	Enter the SIP domain if it is needed by the service provider			

Codecs Settings				
Set the priority and availability of the codecs by adding or removing them from the list.				
Advanced Settings				
Call Forward	Enable unconditional call forwarding, all incoming calls would be forwarded to			
Unconditional	the number specified in the next field			
Call Forward Number for Unconditional	Set the number of unconditional call forwarding			
Call Forward on Busy	Enable call forward on busy, when the phone is busy, any incoming call would be forwarded to the number specified in the next field			
Call Forward Number for Busy	Set the number of call forwarding when the SPA2 is busy			
Call Forward on No Answer	Enable call forward on no answer, when an incoming call is not answered within the configured delay time, the call would be forwarded to the number specified in the next field			
Call Forward Number for No Answer	Set the number of call forward on no answer			
Call Forward Delay for No Answer	Set the delay time of not answered call before being forwarded			
Hotline Delay	Set the delay for hotline before the system automatically dial it			
Enable Auto Answering	Enable auto-answering, the incoming calls would be answered automatically after the delay time			
Auto Answering Delay	Set the delay for incoming call before the system automatically answered answer it			
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if you enable it , the device would receive notification from the server if there is voice message waiting on the server			
Voice Message Number	Set the number for retrieving voice message			
Voice Message Subscribe Period	Set the period of voice message notification subscription			
Enable Hotline	Enable hotline configuration, the device would dial to the specific number immediately at audio channel opened by off-hook or turning on hands-free speaker or headphone			
Hotline Number	Set the hotline dialing number			
Enable DND	Enable Do-not-disturb, any incoming call on this line would be rejected automatically			
Blocking Anonymous Call	Reject any incoming call without presenting caller ID			
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response			
Anonymous Call Standard	Set the standard to be used for anonymous call			
Dial Without Registered	Set call out by proxy without registration			

Click To Talk	Set Click To Talk		
User Agent	Set the user agent, the default is Model with Software Version.		
Use Quote in Display Name	Whether to add quote in display name		
Ring Type	Set the ring tone type for the line		
	Set the type of call conference, Local=set up call conference by the device		
Conference Type	itself; SPA2 maximally 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 he Server		
Number	Set the contenence room number when contenence type is set be server		
Transfer Timeout	Set the timeout of call transfer process		
Enable Long Contact	Allow more parameters in contact field per RFC 3840		
Enable Missed Call Log	If it is enabled, the phone would save missed calls into the call history record.		
Response Single Codec	If it is enabled, the device would use single codec in response to an incoming		
	call request		
	When this setting is enabled, the features in this section would not be handled		
Liso Easturo Codo	by the device itself but by the server instead. In order to control the		
Use realure code	authorization of the features, the device would send feature code to the server		
	by dialing the number specified in each feature code field.		
Specific Server Type	Set the line to collaborate with specific server type		
Registration Expiration	Set the SIP expiration period		
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		
DTMF Type	Set the DTMF type to be used for the line		
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'		
Transportation Protocol	Set the line to use TCP or UDP for SIP transmission		
SIP Version	Set the SIP version		
Caller ID Header	Set the Caller ID Header		
Fachla Chilat Dae	Enables the use of strict routing. When the phone receives packets from the		
Enable Strict Proxy	server, it would use the source IP address, not the address in via field.		
Enable user=phone	Sets user=phone in SIP messages.		
Enable SCA	Enable/Disable SCA (Shared Call Appearance )		
Enable BLF List	Enable/Disable BLF List		
Enable DNS SRV	Set the line to use DNS SRV which would 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 Timer	Set the line to enable call ending by session timer refreshment. The call session		
	would be ended if there is not new session timer event updating 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
Keep Authentication	Keep the authentication parameters of previous authentication
	Using TCP protocol to guarantee usability of transport when SIP messages
Auto TCP	have more than 1500 bytes
Enable Feature Sync	Feature Sync with server
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
	The registered server would receive the subscription package from ordinary
DIE Comion	application of BLF phone.
BLF Server	Please enter the BLF server, if the sever does not support subscription
	package, the registered server and subscription server would be separated.
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists
	are supported.
SIP Encryption	Enable SIP encryption such that SIP transmission would be encrypted
SIP Encryption Key	Set the pass phrase for SIP encryption
RTP Encryption	Enable RTP encryption such that RTP transmission would be encrypted
RTP Encryption Key	Set the pass phrase for RTP encryption

#### b) 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.



	SIP Basic Settings	
s System		
- System		
> Network	SIP Settings	
	Local SIP Port	5060
	Registration Failure Retry Interval	32 Second(s)
> Line	Enable Strict UA Match	
	Enable DHCP Option 120	
> Intercom settings	[	Apply
Security settings	STUN Settings	
	STUN NAT Traversal	FALSE
> Function Key	Server Address	
- Tuncuon ney	Server Port	3478
	Binding Period	50 Second(s)
	SIP Waiting Time	800 millisecond
		Apply
	TLS Certification File: sips.pem	N/A Upload Delete

Basic Settings			
Field Name	Explanation		
SIP Settings			
Local SIP Port	Set the local SIP port used to send/receive SIP messages.		
<b>Registration Failure</b>	Sat the rate interval of SID registration when registration failed		
Retry Interval	Set the retry interval of SP registration when registration failed.		
STUN Settings			
Server Address	STUN Server IP address		
Server Port	STUN Server Port – Default is 3478.		
Binding Period	STUN blinding period – STUN packets are sent once every this period to keep the		
	NAT mapping active.		
SIP Waiting Time	Waiting time for SIP. This would vary depending on the network.		
SIP Line Using STUN(SIP1 or SIP2)			
Use STUN	Enable/Disable STUN on the selected line.		
TLS Certification File			
Upload or delete the TLS certification file used for encrypting SIP transmission.			
Note: the SIP STUN is used to achieve the penetration of SIP NAT; it is a realization of service, when the			
equipment is configured the STUN server IP and port (usually the default is 3478), and selected "Use Stun			
SIP server", you can m	SIP server", you can make common SIP equipment achieve penetration.		

# (4) Intercom settings

a) Features

	Features	Audio	Video	MCAST	Action URL	Time/Date	
<ul> <li>System</li> <li>Network</li> <li>Line</li> <li>Intercom settings</li> <li>Security settings</li> <li>Function Key</li> </ul>	m ork Enable DND Enable Intercom Mute Enable Auto Answer No Answer Auto Hangup Voice Read IP Enable Delay Start Description rity settings tion Key		Lines and IP Call  Lines and IP Call  SPA2	Ban Outr Enable In Auto Ans Auto Har Voice Pla Delay St	going htercom Ringing swer Timeout ngup Timeout y Language art Time	0 (0~60 30 (1~60 English • 1 (1~18	))Second(s) ))Second(s) 30)Second(s)
Features							
Field Name	Explana	tion					
Basic Settings	1						
Enable DND	DND feature can refuse all incoming calls for all SIP lines, or for individual SIP						
	line. But the outgoing calls would not be affected						
Ban Outgoing	If it is enabled, no outgoing calls can be made.						
Enable Intercom Mute	If it is enabled, device would mute incoming calls during an intercom call.				com call.		
Enable Intercom	If it is er	habled, de	evice would pla	ay intercon	n ring tone	to alert that	t there is a
Ringing	new inc	oming ca	ll during an int	ercom call.			
Enable Auto Answer	Enable A	Auto Ansv	wer function				
Auto Answer Timeout	Set Auto	o Answer	Timeout				
No Answer Auto	Enable	utomati	cally bang up fo	aturo who	n thoro is n	o answor	
Hangup							
Auto Hangun Timeout	Configuration in a set time, the device would automatically hang up when						
	there is no answer						
Voice Read IP	Enable or disable voice broadcast IP address						
Voice Play Language	Set lang	uage of t	he voice prom	pt			
Enable Delay Start	Enable o	or disable	the start delay	y			
Delay Start Time	Set start delay time						
Description	Device description displayed on IP scanning tool software or FDMS.						

### b) Audio

This page configures audio parameters such as voice codec, speak volume, mic volume and ringer volume.

	Features Audio	Video	MCAST Action URL	Time/Date
› System				
	Audio Settings			
> Network	First Codec	G.722 ▼	Second Codec	G.711A 🔻
	Third Codec	G.711U 🔻	Fourth Codec	G.729AB 🔻
> Line	Fifth Codec	None 🔻	Sixth Codec	None 🔻
	DTMF Payload Type	101 (96~127)	Default Ring Type	Type 1 🔻
Intercom settings	G.729AB Payload Length	20ms 🔻	Tone Standard	United St: 🔻
	G.722 Timestamps	160/20m 🔻	G.723.1 Bit Rate	6.3kb/s 🔻
Security settings	Speakerphone Volume	5 (1~9)	MIC Input Volume	5 (1~
	Broadcast Output Volume	5 (1~9)	Signal Tone Volume	4 (0~
> Function Key	Enable VAD			

Audio Setting								
Field Name	Explanation							
First Codec	The first codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB							
Second Codec	The second codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None							
Third Codec	The third codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None							
Fourth Codec	The forth codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, None							
DTMF Payload Type	The RTP Payload type that indicates DTMF. Default is 101							
Default Ring Type	Ring sound – there are 9 standard types and 3 user types.							
G.729AB Payload	G.729AB Payload length – adjust from 10 – 60 msec.							
Length								
Tone Standard	Configure tone standard area.							
G.722 Timestamps	Choices are 160/20ms or 320/20ms.							
G.723.1 Bit Rate	Choices are 5.3kb/s or 6.3kb/s.							
Speakerphone	Set the speaker call volume lovel							
Volume								
MIC Input Volume	Set the MIC call volume level.							
Broadcast Output	Set the breadcast output volume level							
Volume								
Signal Tone Volume	Set the audio signal output volume level.							
Enable VAD	Enable or disable Voice Activity Detection (VAD). If VAD is enabled, G729 Payload							
	length cannot be set greater than 20 msec.							

### c) Video

This page allows you to set the video capture and video encode.

Local Mode

	Features Audio	Video	ICAST Action URL	Time/Date
> System				
> Network	Ip Camera Connect Settings Connect Mode	Local <b>T</b>		
> Line	Video Conturo	Apply		
> Intercom settings	IRCUT Mode	Automatic	Day/Night Mode	Automatic V
> Security settings	Anti Flicker IR Swap	Disable	Vertical Flip DNC Threshold	Enable  (10~50)
› Function Key	Backlight Compensation	Disable <b>v</b>	AutoFill Sensitivity	5 <b>(</b> 1~10)
		Default	Apply	
> Network	Video Encode>>			
› Line	Encode Format	Main Stream	Sub Stream	
> Intercom settings	Resolution Frame Rate	720P V	CIF <b>T</b>	
<ul> <li>Security settings</li> </ul>	Bitrate Control Quality	VBR ▼ General ▼	VBR ▼ General ▼	
Function Key	Bitrate I Frame Interval Activate	1700 ▼ 2 (2~12)S €	318 ▼ 2 (2~12)S	
		Default	Apply	
	RTSP Information Main Stream Url : _rtsp://17	2.18.2.170/user=admin&nassw	ord=tlJwpbo6&channel=1&stre	am=0.sdo?real_stream_Preview
	Sub Stream Url : rtsp://17	2.18.2.170/user=admin&passw	ord=tlJwpbo6&channel=1&strea	am=1.sdp?real_stream Preview

Camera Connect Settings							
Field Name	Explanation						
Connect Mode	Local: Connect the original camera						
Connect Mode	External: Connect to another manufacturers camera						
Video Capture (Loca	l Mode)						
	Auto: IRCUT switches according to the actual ambient light level of the camera						
IRCUT Mode	Synchronization: The switching of the IRCUT is determined by the actual brightness						
	of the IR lamp.						
	Automatic: automatically switches according to the DNC Threshold and the						
	brightness of the actual environment where the camera is located						
Day/Night Mada	Day Mode: The camera's video screen is always colored, if there is IR-cut will be						
Day/Night Wode	synchronized to switch.						
	Night Mode: the camera's video screen is always black and white, if there is IR-cut						
	will be synchronized switch.						
	Automatic: Automatically adjusts according to the actual environment in which the						
M/hite Deleves	camera is located.						
white Balance	Outdoor: installed in the outdoor preferred.						
	Indoor: installed in the room preferred.						
Horizon Flip	The video is flipped horizontally						
Anti Flicker	Enable the option. In a fluorescent environment can eliminate the video horizontal						

	scroll			
Vertical Flip	The video is flipped horizontally			
IR Swap IR-cut filter switch				
DNCThreehold	In the Day / Night mode Auto option, the color switching black and white threshold			
DINC Inreshold	is set			
Backlight	In front of a very strong background light can see people or objects clearly			
Compensation				
	In the environment changes in light and shade, the higher the sensitivity the faster			
Autoriii Sensitivity	the video changes			

Video Encode(Local Mode)						
Field Name	Explanation					
Encode Format	Only H.264 encoding format is supported					
Decolution	Main stream: support 720P					
Resolution	Sub-stream: you can select CIF (352 * 288), D1 (720 * 576)					
Frama Rata	The larger the value is, the more coherent the video would be got; not recommend					
Frame Rate	adjusted.					
	CBR: If the code rate (bandwidth) is insufficient, it is preferred.					
Bitrate Control	VBR: Image quality is preferred, not recommended.					
Quality	Video quality adjustment, the better the quality needs to transfer faster					
Bit rate	It is proportional to video file size, not recommend adjusted.					
I Frame Interval	The greater the value is, the worse the video quality would be, otherwise the better					
I Frame interval	video quality would be; not recommend adjusted.					
Activate When you selected it, the stream is enabled, otherwise disabled						
Droviow	copy and paste the main stream or sub-stream Url into the VLC player, or click					
PIEVIEW	[Preview] to display the current camera video.					

#### **External Mode**

	Features	Audio	Video	MCAST	Action URL	Time/Date	
> System							
> Network	Ip Camera Conn Connect Moo	aect Settings de	External  Apply				
› Line	Ip Camera Setti	ngs					
> Intercom settings	Position User		ipCameraName admin		(40 Characters)		
> Security settings	Password Ip Camera B	rand	XM V				
› Function Key	IP Port		172.18.3.64 554 Apply				
	RTSP Information	on Url: rtsp://172	.18.3.64:554/user=ad	min&password=&cl	nannel=1&stream=0	).sdp?real_stream	Preview

Connection	Select external, click [Apply], restart the device
mode	
IP Camera Settin	<b>gs</b> (External Mode)
Field Name	Explanation
User name	External camera login required account
Password	External camera login password required
Camera type	Select the camera manufacturers
ID address	IP address of the camera, please use the camera matching scan tool to obtain the IP
IP autress	address
Port	Camera port number
RTSP	Click [Apply], the connection automatically shows the camera does not show the
information	reverse
Droviow	Copy and paste the main stream or sub-stream Url into the VLC player, or click
Preview	[Preview] to display the current camera video

#### d) MCAST

	Features	Audio	Video	MCAST	Action URL	Time/Date	
> System							
	MCAST Settings						
> Network	Priority		1	▼			
> Line	Enable Page P	Enable Page Priority					
	Index/Pr	iority	Name			Host:port	
> Intercom settings	1						
	3						
> Security settings	4						
	5						
> Function Key	6						
	7						
	8						
	9						
	10		Apply				

It is easy and convenient to use multicast function to send notice to each member of the multicast via setting the multicast key on the device and sending multicast RTP stream to pre-configured multicast address. By configuring monitoring multicast address on the device, the device monitors and plays the RTP stream which sent by the multicast address.

#### **MCAST Settings**

Equipment can be set up to monitor up to 10 different multicast addresses, used to receive the multicast RTP stream sent by the multicast address.

Here are the ways to change equipment receiving multicast RTP stream processing mode in the web interface: set the ordinary priority and enable page priority.

• Priority:

In the drop-down box to choose priority of ordinary calls, if the priority of the incoming streams of multicast RTP, lower precedence than the current common calls, device would automatically ignore the

group RTP streams. If the priority of the incoming stream of multicast RTP is higher than the current common calls priority, device would automatically receive the group RTP streams, and keep the current common calls in maintained status. You can also choose to disable the function in the receiving threshold drop-down box, the device would automatically ignore all local network multicast RTP streams.

- The options are as follows:
  - ✤ 1-10: To definite the priority of the common calls, 1 is the top level while 10 is the lowest
  - ♦ Disable: ignore all incoming multicast RTP streams
  - $\diamond$  Enable the page priority:

Page priority determines the device how to deal with the new receiving multicast RTP streams when it is in multicast session currently. When Page priority switch is enabled, the device would automatically ignore the low priority multicast RTP streams but receive top-level priority multicast RTP streams and keep the current multicast session in maintained status; If it is not enabled, the device would automatically ignore all receiving multicast RTP streams.

#### Web Settings:

Μ

CA	ST Settings		
	Priority	1 💙	
	Enable Page Priority		
	Index/Priority	Name	Host:port
	1	SS	239.1.1.1:1366
	2	ee	239.1.1.1:1367

The multicast ss priority is higher than that of ee; ss has the highest priority.

Note: when you press the multicast key for multicast session, both multicast sender and receiver would beep.

#### Listener configuration

CAST Settings		
Priority	3	
Enable Page Priority		
Index/Priority	Name	Host:port
1	group 1	224.0.0.2:2366
2	group 2	224.0.0.2:1366
3	group 3	224.0.0.6:3366
4		
5		
6		
7		
8		
9		
10		

• Blue part (name)

"Group 1", "Group 2" and "Group 3" are your setting monitoring multicast name. The group name would be displayed on the screen when you answer the multicast. If you have not set, the screen would display the IP: port directly.

#### • Purple part (host: port)

It is a set of addresses and ports to listen, separated by a colon.

#### • Pink part (index / priority)

Multicast is a sign of listening, but also the monitoring multicast priority. The smaller number refers to higher priority.

#### • Red part (priority)

It is the general call, non-multicast call priority. The smaller number refers to higher priority. The followings would explain how to use this option:

- ♦ The purpose of setting monitoring multicast "Group 1" or "Group 2" or "Group 3" is to launch a multicast call.
- ♦ All equipment has one or more common non multicast communication.
- ♦ When you set the priority as disabled, any level of multicast would not be answered , multicast call is rejected.
- when you set the priority as some value, only the multicast higher than the priority can come in. If you set the priority as 3, group 2 and group 3 would be rejected, for its priority level is equal to 3 and less than 3; multicast 1 priority is set up with 2, higher than ordinary call priority, device can answer the multicast message, at the same time, holding the other call.
- Green part (Enable Page priority)

Set whether to open multicast comparison function, multicast priority is pink part number. Following explains how to use:

- The purpose of setting monitoring multicast "group 1" or "group 3" is listening "group of 1" or "group 3" multicast call of multicast address.
- ♦ The device has a path or multi-path multicast calls, such as listening to "multicast information group 2".
- ♦ If multicast is a new "group 1", and because the priority of group 1" is 2, higher than the current call priority 3 of "group 2", so multicast call would come in.
- ♦ If multicast is a new "group 3", and because the priority of group 3" is 4, lower than the current call priority 3 of "group 2", the device would listen to the "group 1" and maintain the "group 2".

#### **Multicast service**

- Send: when you configure the item, pressing the corresponding key on the equipment shell, equipment would directly enter the Talking interface; the premise is to ensure no current multicast call and three-way conference, so the multicast can be established.
- **Monitor:** IP port and priority are configured to monitor the device, when the call is initiated by multicast and the call is successful; the device would directly enter the Talking interface.

### e) Action URL

	Features	Audio	Video	MCAST	Action URL	Time/Date	
→ System	Action URL Even	nt Settings					
	Active URI L Setup Comp	imit IP leted					
> Network	Registration Succeeded						
> Line	Registration Registration	Disabled Failed					
> Intercom settings	Off Hooked On Hooked						
	Incoming Ca	II					
<ul> <li>Security settings</li> </ul>	Outgoing cal Call Establish	lls ned			_		
> Function Key	Call Termina	ted					
	DND Enabled DND Disabled						
	Mute						
	Unmute Missed calls						
	IP Changed						
	Busy To Idle						
			Apply				

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

URL for various actions performed by the phone. These actions are recorded and sent as xml files to the server. Sample format is http://InternalServer /FileName.xml

### f) Time/Date

	Features A	udio Video	MCAST	Action URL	Time/Date			
› System	Network Time Server Se	ttings						
	Time Synchronized vi	a SNTP 🕑						
> Network	Time Synchronized vi	a DHCP						
	Primary Time Server	time.nist.gov						
> Line	Secondary Time Serve	pool.ntp.org						
	Time zone	(UTC+8) Chir	a,Singapore,Australi	i 🔻				
> Intercom settings	Resync Period	60	(1~500	0)Second(s)				
Intercom Seconds	Data Format							
· Cocurity cottings	Date Format							
v Security setungs	Date Format	1 JAN MO	N T					
> Function Key		Apply						

> Line	Daylight Saving Time Settings		
	Location	China(Beijing) 🔹	
> Intercom settings	DST Set Type	Automatic 🔹	
	Fixed Type	Disabled 🔹	
> Security settings	Offset	0 Minu	ite
		Start	End
> Function Key	Month	January 🔻	January 🔻
	Week	1 •	1
	Weekday	Sunday 🔻	Sunday 🔻
	Hour	0 •	0
		Apply	
	Manual Time Settings		
	2017.05.19	10 V Apply	
	2017-05-18	Appiy	

Time/Date						
Field Name	Explanation					
Network Time Server S	ettings					
Time Synchronized via SNTP	Enable time-sync through SNTP protocol					
Time Synchronized via DHCP	Enable time-sync through DHCP protocol					
Primary Time Server	Set primary time server address					
Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device would 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					
Date Format						
12-hour clock	Set the time display in 12-hour mode					
Date Format	Select the time/date display format					
Daylight Saving Time S	ettings					
Location	Select the user's time zone according to specific area					
DST Set Type	Select automatic DST according to the preset rules of DST, or you can 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						
The time might be set ma	The time might be set manually, needed user to disable SNTP service first.					

# (5) Security settings

rstem	Input Settings			
Network	Trigger Mode Output Settings	Low Level Trigger(Close Trigger)	▼	nd to server
Line	✓ Output Response Output Level	High Level(NO:closed) •	Output Duration	5 (1~600) s
Intercom settings	Alert Trigger Setting			
Security settings	Alarm Ring Duration <ul> <li>Input Trigger</li> </ul>	5 (1~600) s Enable Ring ▼	DTMF Output Last	By Duration V
Function Key	<ul> <li></li></ul>	Enable Ring   Enable Ring Talking	DTMF Trigger Code Trigger Message Format	1234 ALERT=OUT1_SOS
			Apply	
	Server Settings			
	Server Address		Send message to the server	when the alarm is triggered
	Message:Alarm_Info:Descrip	otion=PA2;SIP User=;Mac=00:a8	:23:6a:6d:76;IP=172.18.2.17	0;port=Input1

Security Settings	Security Settings					
Field Name	Field Name					
Input settings						
Input Detect	Enable or disable Input Detect					
<b>T</b> (	When choosing the low level trigger (Closed Trigger), detect the input port (low level)					
	closed trigger.					
ingger wode	When choosing the high level trigger (Disconnected Trigger), detect the input port					
	(high level) disconnected trigger.					
Alert message	Enable or disable input port cond message to server					
send to server	Enable of disable input port send message to server					
Output Settings						
Output	Enable or dicable Output Perpanse					
Response						
	When choosing the low level (NO: open), when meet the trigger condition, trigger the					
Output Loval	NO port disconnected.					
Output Level	When choosing the high level (NO: closed), when meet the trigger condition, trigger					
	the NO port close.					
Output Duration	The port changes the duration. The default is 5 seconds.					
Alert Trigger Setti	ng					
Alarm Ring	duration of alarm ring					
Duration						
Input Trigger	When the input port meet to trigger condition, the output port will trigger (The Port					

	level time change, I	By < Output Duration > control).				
	You can choose to e	enable or disable the ringtone				
	By duration	The Port level time change, By < Output Duration > control				
	Dy Calling State	By call state control, after the end of the call, port to return the				
Lasi	By Calling State	default state				
Romoto DTME	Received the term	inal equipment to send the DTMF password, if correct, which				
	triggers the corresp	onding output port.				
Ingger	You can choose to e	enable or disable the ringtone				
DTMF Trigger	During the call, rece	eive the terminal equipment to send the DTMF password, if correct,				
Code	which triggers the corresponding output port. The default is 1234.					
Remote SMS	Enable or disable Remote SMS Trigger.					
Trigger	You can choose to enable or disable the ringtone					
Trigger Message	In the remote device or server to send instructions to ALERT=[instructions], if correct,					
Format	which triggers the c	which triggers the corresponding output port				
	The port output continuous time synchronization and trigger state changes, including					
	the trigger conditions. Four models, such as: call trigger output port, will be in a call					
	state to continue to respond)					
Call State Trigger	1, Taking;					
	2, Taking and ringing;					
	3, ringing;					
	4, Call.					
Server Settings						
	Configure remote re	esponse server address (including remote response server address				
Sonver Address	and tamper alarm s	erver address). When the input port is triggered will send a short				
Server Address	message to the serv	ver, the message format is as follows: Alarm Info: Description=SPA2;				
	SIP User=;Mac=00:a	a8:34:68:23:d1;IP=172.18.2.243;port=Input1				

# (6) Function Key

### a) Function Key Settings

› System						
Notwork	Function Key Setting	js				
/ NELWOIK	🔲 Input port Mu	ltiplexing as DSS Ke	ey2			
	Key	Туре	Number 1	Number 2	Line	Subtype
> Line	DSS Key 1	Key Event 🔻	DSS Key2          Number 1       Number 2       Line       Subtype         V       SIP1 V       OK       V         V       SIP1 V       Speed Dial       V         Enable V       Enable Speed Dial Hangup       Enable V       Enable V         Main-Secondary V       Sipay Start Time 06:00 (00:00~23:59) Day End Time 18:00 (00:00~23:59)       Sipay End Time 18:00 (00:00~23:59)			
	DSS Key 2	None 🔻			SIP1 T	Speed Dial
Security settings	Advanced Settings Use Function Ke	y to Answer En	able 🔻	Enable Speed Dial Hangur	e Enab	ole ▼
> Function Key	Hot Key Dial Mod	le Select Ma	in-Secondary 🔻			
	Call Switched T	īme <mark>16</mark> (5~	50)S Day Start Time 06:0	00 (00:00~23:59) Day	End Time 1	8:00 (00:00~23:59)
			Ар	pply		

#### Key Event

You might set up the key type with the Key Event.

Key	Туре	Number 1	Number 2	Line	Subtype			
DSS Key 1	Key Event 🔻			SIP1 T	OK 🔻			
	Apply				None Dial Release			
		OK Handfroe						
					Hallullee			
Туре	Subtype	Usage						
	None	No res	oonding					
	Dial	Dialing	function					
Key Event	Release	Delete	Delete password input, cancel dialing input and end call					
	ОК	identifi	cation key					

#### ≻ Hot Key

Handfree

You might enter the phone number in the input box. When you press the shortcut key, equipment would dial preset telephone number. This button can also be used to set the IP address: you can press the shortcut key to directly make a IP call.

The hand-free key (with hooking dial, hanging up functions)

Key	Туре	Number 1	Number 2	Line	Subtype	
DSS Key 1	Hot Key 🔻			SIP1 V	Speed Dial	•
	Speed Dial					
		Ar	vlac		Intercom	

Туре	Number	Line	Subtype	Usage
Fill the called Hot Key party's SIP account or IP address The SIP account correspon lines	The SIP account corresponding lines	Speed Dial	Using Speed Dial mode together with Enable Speed Dial Hangup Enable, can define whether this call is allowed to be hung up by re-pressing the speed dial key.	
	IP address		Intercom	In Intercom mode, if the caller's IP phone supports Intercom feature, the device can automatically answer the Intercom calls

### > Multicast

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

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

Key	Туре	Number 1	Number 2	Line	Subtype	
DSS Key 1	Multicast 🔹			SIP1 V	G.722	•
		A	pply		G.711A G.711U G.722 G.723.1 G.726-32 G.729AB	

Туре	Number	Subtype	Usage
Multicast	Set the host IP address and port number; they must be separated by a colon	G.711A	Narrowband speech coding (4Khz)
		G.711U	
		G.722	Wideband speech coding (7Khz)
		G.723.1	
		G.726-32	Narrowband speech coding (4Khz)
		G.729AB	

#### operation mechanism

You can define the Function Key configuration with multicast address, port and used codec. The device can configure via WEB to monitor the multicast address and port. When the device make a multicast, all devices monitoring the address can receive the multicast data.

#### ♦ calling configuration

If the device is in calls, or it is three-way conference, or initiated multicast communication, the device would not be able to launch a new multicast call.

#### b) Advanced Settings

Advanced Settings					
Use Function Key to Answer	Enable 🔻	Enable Speed Dial Hangup	Enable 🔻		
Hot Key Dial Mode Select	Main-Secondary </th <th></th> <th></th>				
Call Switched Time 16 (5~50)S Day Start Time 06:00 (00:00~23:59) Day End Time 18:00 (00:		Time 18:00 (00:00~23:59)			
		Apply			

Advanced Settings				
Field Name	Field Name			
Input Port Multiplexing 2	Enable or disable input port reuse for DSS key 2.			
Use Function Key to Answer	Enable or disable DSS Key answer			
Enable Speed Dial key Hang up	Enable or disable the DSS Key to hang up			
Hot Key Dial Mode Select	Number 1 Transfer Call Number 2 Mode Select. <primary secondary="">mode allow system to call primary extension first, if there is no answer, system would cancel the call and then call secondary extension automatically. <day night="">mode allow system to check the calling time is belong to day time or night time, and then system decides to call the number 1 or number</day></primary>			

	2 automatically.	
	Users just press speed dial key once.	
Call Switched Time	Set number 1 to transfer call number 2 time, default 16 seconds	
Day Start Time	The start time of the day when you select <day night="">mode.</day>	
Day End Time	The end time of the day when you select <day night="">mode.</day>	

# V. Appendix

# 1. Technical parameters

Communication protocol		SIP 2.0(RFC-3261)	
Main chipset		Broadcom	
Speech	Protocols	RTP	
flow	Decoding	G.729、G.723、G.711、G.722、G.726	
WAN		10/100BASE-TX s Auto-MDIX, RJ-45	
MIC(Optional)		electret condenser microphone	
		Sensitivity: -38dB, bias voltage 2.2V	
Speaker(Optional)		$4\Omega$ / 3W, internal magnetic horn, diameter: 57mm	
Camera(Optional)		1/4 "color CMOS, 1 megapixel, wide angle	
Power supply mode		$12^{\sim}24V$ 2A DC or PoE	
РоЕ		PoE 802.3af (Class 3 - 6.49~12.95W)	
Cables		CAT5 or better	
Shell Material		Aluminum alloy	
Working temperature		-20°C to 60°C	
Storage temperature		-40°C to 70°C	
Installation way		Embedded or desktop	
External size		113x83x28mm	
Package size		138x108x77mm	
Equipment weight		250g	

### 2. Basic Functions

- 2 SIP lines
- PoE Enabled
- External power supply
- Supports two lines RTSP

- Voice broadcast IP address
- button or remote volume adjustment
- switch signal input and output
- Talkback recording output
- Default auto answer
- Dynamic multicast function
- Support for Function key interface
- Support monophonic active speakers