IP Intercom User Manual



Single button



Dual button

Safety Notices

- 1. Please use the specified power adapter. If special circumstances need to use the power adapter provided by other manufacturers, please make sure the voltage and current provided 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, or forcefully twist it Stretch pull or banding, and not to be under heavy pressure or between items, Otherwise may cause the power cord damage, thus lead to fire or get an electric shock.
- 3. Before use, please confirm the temperature and environment humidity suitable for the product 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. Non-technical staff not remove or repair, improper repair or may cause electric shock, fire or malfunction, etc, Which can lead to injury accident, and also can cause your product damage.
- 5. Do not use fingers, pins, wire and other metal objects, foreign body into the vents and gaps. It may cause current through the metal or foreign body, which even cause electric shock and injury accident. If any foreign body or objection falls into the product please stop usage.
- 6. Please do not discard the packing bags or stored 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.

Directory

Α.	PRC	DDUCT INTRODUCTION	.5
1	. Арр	EARANCE OF THE PRODUCT	5
2	. Вит	TON DESCRIPTION	. 5
в.	STAR	USING	.5
1	. Con	INECTING THE POWER SUPPLY AND THE NETWORK	.5
	(1)	Connecting network	
	(2)	Interface specification	.6
	a)	Schematic diagram of peripherals	. 6
	b)	Interface specification	. 7
	c)	Port instructions	. 8
2	. Qui	ck Setting	10
C.	BASIC	OPERATION	LO
1	. Ans	WER A CALL	10
2			
3	. End	CALL	10
4	. CAL	L RECORD	10
D.	PAC	GE SETTINGS	11
1	BRO	WSER CONFIGURATION	11
2		sword Configuration	
3		IFIGURATION VIA WEB	
-	(1)	BASIC	
	`́а)	STATUS	12
	b)	WIZARD	13
	c)	CALL LOG	15
	d)	LANGUAGE	٤5
	e)	TIME&DATE	۱6
	(2)	NETWORK	۲7
	a)	WAN	۲7
	b)	LAN	20
	c)	QoS&VLAN	20
	d)	WEB FILTER	22
	e)	FIREWALL	23
	f)	VPN	24
	g)	SECURITY	
	(3)	VOIP	
	a)	SIP	
	b)	STUN	
	(4)	Intercom	30

	a)	FUNCTION KEY
	b)	AUDIO
	c)	FEATURE
	d)	MCAST
	e)	Action URL
(5)	SAFEGUARDING
(6)	MAINTENANCE
	a)	AUTO PROVISION
	b)	SYSLOG 42
	c)	CONFIG
	d)	UPDATE
	e)	ACCESS
	f)	REBOOT
(7)	LOGOUT
E. 4	APPE	NDIX46
1.	TEC	HNICAL PARAMETERS
2.	Bas	IC FUNCTIONS
3.	Sсн	EMATIC DIAGRAM
4.	Тне	RADIO TERMINAL CONFIGURATION NOTICE
5.	Тне	OTHER FUNCTION SETTINGS

A. Product introduction

This product is a fully digital network intercom equipment, its core part adopts mature VOIP solutions (Broadcom 1190), the performance is stable and reliable; the digital full duplex hands-free, voice loud and clear; the keys feel comfortable, simple installation, appearance, durable, low power consumption.

1. Appearance of the product



2. Button description

Buttom	Description	Function
	programmable keys	Can be set to a variety of functions, in order to meet the needs of different occasions

B. Start Using

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

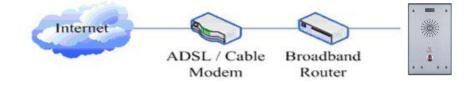
1. Connecting the power supply and the network

(1) Connecting network

In prior to this step, please check if your network can work normally and have capacity of broadband internet access.

Broadband Router

Connect one end of the network cable to the intercom WAN port, the other end is connected to your broadband router's LAN port, so that the completion of the network hardware connections. In most cases, you must configure your network settings to DHCP mode. Please refer to the detailed setting ways: **D**, **3**, **(2)**, **a) WAN**.



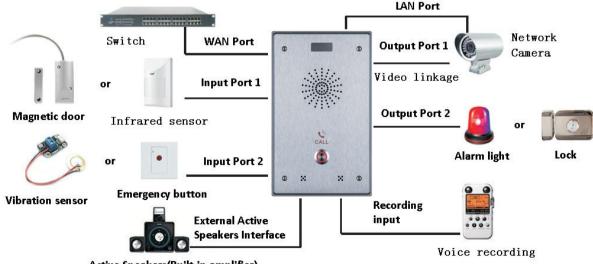
No Broadband Router

Connect one end of the network cable to the intercom WAN port, the other end is connected to the broadband modem to your LAN port, so that the completion of the network hardware connections. In most cases, if you are using the cable broadband, you must configure your network settings to DHCP mode; if you are using the ADSL, you must configure your network settings to PPPoE mode. Please refer to the detailed setting ways: **D**, **3**, **(2)**, **a) WAN**.



(2) Interface specification

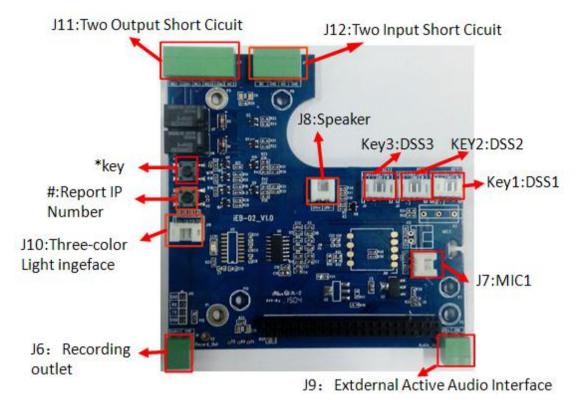
a) Schematic diagram of peripherals



Active Speakers(Built-in amplifier)

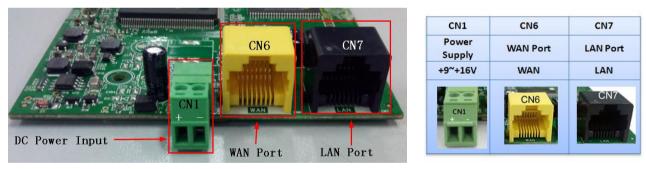
b) Interface specification

• Expansion board interface



[Notice] Press "#"key for 3 seconds, the controller will report it IP number by itself.

Motherboard interface



[Notice]LAN port Support two modes:

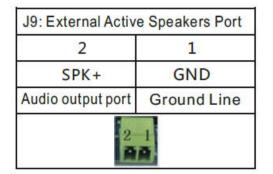
- ♦ Routing mode (It can assign IP Address to LAN port the via the DHCP for each connected device)
- ♦ Bridge Mode (LAN port and WAN port are in the same network segment)

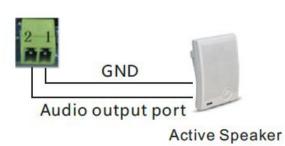
• Port description

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

c) Port instructions

• External Active Speakers





Audio Recording output port

J6: Audio Record	ding output port		
2	1	2-1	
Audio+	GND	AA	
Audio Recording output port	Ground Line	GND	- Contraction
2	1	Audio Recording output	PC Recordin

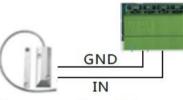
Two short circuit output port

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

1	J11: Short circuit output Port							
Outp	ut Port1(OUT2)	Output Port1(OUT1)					
6	5	4	3	2	1			
NC2	COM2	NO2	NC1	COM1	NO1			
	Common terminal		Normal close	Common terminal	Normal Open			
	close terminal Open close terminal Open							

Two short circuit input port

Input Po	rt2(IN2)	Input Port1(IN1)		
4	3	2	1	
GND	IN2	GND	IN1	
Input Port2	Input Port2	Input Port1	Input Port1	



0 0 0

NC COM

M=1

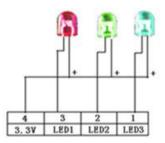
Alarm light

Door magnetic switch

12V DC Power Supply

Status lamp interface

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



2. Quick Setting

The product provides a rich and complete function and parameter setting; users may need to have a network with SIP protocol in order to understand the related knowledge on behalf of all the significance of the parameters. In order to high quality voice service and low cost advantage, allowing users to enjoy the facility brought fast, especially in the listed in this section the basic and necessary to set options users can quickly get started, no without understanding the complicated SIP protocol.

In this step, please confirm the Internet broadband access can be normal operation, and complete the connection to the network hardware. The intercom default for DHCP mode.

- A long press # key 3 seconds, automatic voice playing device's IP address, or use the "iDoorPhoneNetworkScanner.exe" software to find the IP address of the device.
- > Log on to the WEB device configuration.
- In a SIP page configuration service account, user name, parameters that are required for server address register.
- You can settings DSS key in the Webpage(functions key settings -> function key).
- > You can settings function parameters in the Webpage (Intercom-> feature).

🖇 iDoorPhone Network Scanner(V 1.0)						
#	IP Address	Serial Number	MAC Address	SW Version	Description	
#	172.18.2.177	IP intercom	Oc:38:3e:13:3b:90	2.3.698.366	IP intercom	
						<u>R</u> efresh

C. Basic operation

1. Answer a call

When calling come, the device automatically answer, in cancel automatic answer and settings automatic answer time, will hear the bell in the set time, automatic answer after a timeout.

2. call

Configuration shortcut as hot key and setup a number, then press shortcut can call the configured number immediately.

3. End call

Enable Release key hang up to end call.

4. Call record

The device provides 300 call recording, when the storage space is exhausted, will cover the first call records. When the device is powered down or reboot, call records will be removed. You can view the three call records in the Webpage (Basic->call log)

D. Page settings

1. Browser configuration

When the device and your computer successfully connected to the network, the on browsers enter the IP address of the device. You can see the Webpage management interface the login screen. Enter the user name and password and click [logon] button to enter the settings screen.

	(
User:		
Password:		
Language:	English •	-

After configuring the equipment, remember to click SAVE under the Maintenance tab. If this is not done, the equipment will lose the modifications when it is rebooted.

2. Password Configuration

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

- Default user with general level:
 - Username: guest
 - Password: guest
- Default user with root level:
 - Username: admin
 - Password: admin

3. Configuration via WEB

(1) BASIC

a) STATUS

	STATUS	WIZARD CALL LOG	LANGUAGE TIM	IE&DATE
ASIC	Network			
TWORK	WAN		LAN	
	Connection Mode	DHCP	IP Address	192.168.10.1
VoIP	MAC Address	00:d8:4a:00:02:ba	DHCP Service	Enabled
	IP Address	172.18.2.112	Bridge Mode	Disabled
INTERCOM	IP Gateway	172.18.1.1		
	Accounts			
SAFEGUARDING	SIP Line 1	@:5060	Unappl	lied
MAINTENANCE	SIP Line 2	@:5060	Unappl	lied
∟ogout				

Status	
Field Name	Explanation
	Shows the configuration information for WAN and LAN port, including connection
Notwork	mode of WAN port (Static, DHCP, PPPoE), MAC address, IP address of WAN port and
Network	LAN port, DHCP server, status for LAN port (ENABLED or DISABLED).
	Default Static IP: 192.168.1.128
Accounts	Shows the phone numbers and registration status for the 2 SIP LINES and 1 IAX2
Accounts	server.

b) WIZARD

	STATUS	WIZARD	CALL LOG	LANGUAGE	TIME&DATE	
> BASIC	WAN Connection Ma	de				
> NETWORK	Static IP	۲				
	DHCP	۲				
> VoIP	PPPoE	0				
> INTERCOM				Next		
> SAFEGUARDING						
MAINTENANCE						
> LOGOUT						

Wizard		
Field Name	Explanation	
Select the approp	riate network mode. The equipment supports three network modes:	
Static IP mode	The parameters of a Static IP connection must be provided by your ISP.	
	In this mode, network parameter information will be obtained automatically from a	
DHCP mode:	DHCP server.	
PPPoE mode:	In this mode, you must enter your ADSL account and password.	
Static IP mode is	selected; Click Next to go to Quick SIP Settings, Click Back to return to the Wizard	
screen.		
Static IP Settings		
IP Address	192.168.1.179	
Subnet Mask	255.255.0	
IP Gateway	192.168.1.1	
DNS Domain		
Primary DNS Secondary DNS	202.96.134.133 202.96.128.68	
Secondary SNS	Back Next	
Static IP address	Please enter the Static IP address	
Subnet Mask	Please enter the Subnet Mask	
IP Gateway	Please enter the IP Gateway	
	Set the DNS domain suffix. When the user enter the domain name DNS address	
DNS Domain	cannot be resolved, the domain equipment to resolve in the domain name.	
Primary DNS	Please enter the Primary DNS server address	
Secondary DNS	Please enter the Secondary DNS server address	

Field Name	Explanation
Quick SIP Setting	S
Quick SIP Settings	
Display Name Server Address	603 172.18.1.200
Server Port	5060
Authentication User	603
Authentication Password	•••
SIP User	603
Enable Registration	
	Back
Display Name	The name shown in caller ID
Server Address	SIP server address either IP address or URI
Server Port	SIP server port (usually 5060)
User	Login name or Authentication ID.
Password	SIP password
SIP User	Phone number
Enable	Submits registration information. Normally checked
Registration	
Displays detailed	information for manual configuration.
WAN	
Connection Mode	Static IP
Static IP Address	192.168.1.179
IP Gateway	192.168.1.1
SIP	
Server Address	172.18.1.200
Account	603
Phone Number	603
Registration	Enabled
	Back
After selecting DI	HCP and clicking NEXT, the Quick SIP Settings screen will appear. Click Back to return

to the Wizard screen. Click Next to go to the Summary screen.

If PPPoE is selected, this screen will appear. Enter the information provided by the ISP. Click Next to go to Quick SIP Setting. Click Back to return to the Wizard screen.

Click Finish button to save settings and reboot. After the reboot, SIP calls can be made.

c) CALL LOG

Outgoing call logs can be seen on this page

	STATUS	WIZARD	CALL LOG	LANGUAGE	TIME&DATE	
BASIC	Call Information					
NETWORK	Start Time	Dur	ation	Peer Calls		Туре
> NETWORK	February 26 14:	01 7 s	econd(s)	172.18.2.40@1	72.18.2.40	Received
VoIP	February 26 14:	00 10	second(s)	8207@1		Received
> INTERCOM						

Call log		
Field Name	Explanation	
Start time	Start time of the outgoing call	
Duration	Duration of the outgoing call	
Dialed calls	Account, protocol, and line of the outgoing call	
Туре	The call records of type	

d) LANGUAGE

Set the current language.

	STATUS	WIZARD CALL LOO	3 LANGUAGE	TIME&DATE
> BASIC	Language			
> NETWORK	Language Selection	English 🔻]	
› VoIP			Apply	
> INTERCOM				

e) TIME&DATE

	STATUS WIZARD CALL LOG LANGUAGE TIME&DATE
	System Current Time
	2016/02/26 16:53:43
> BASIC	Simple Network Time Protocol (SNTP) Settings
	Enable SNTP 🗹
> NETWORK	Enable DHCP Time
	Primary Server 0.pool.ntp.org
› VoIP	Secondary Server time.nist.gov
	Timezone (GMT+08:00)Beijing,Chongqing,Hong Kong,Urumqi
INTERCOM	Resync Period 60 second(s)
	12-Hour Clock
> SAFEGUARDING	Apply
	Daylight Saving Time Settings
BASIC	Enable 🗐
	Offset 60 minutes(s)
> NETWORK	Month March
	Week 5 T
> VoIP	Day Sunday T
	Hour 2
> INTERCOM	Minute 0
	Apply
> SAFEGUARDING	
	Manual Time Settings
> MAINTENANCE	Year
	Month
> LOGOUT	Day
	Hour
	Minute
	Apply

TIME&DATE	
Field Name	Explanation
System Current T	ime
Display the currer	it time
SNTP Settings	
Enable SNTP	Enable or Disable SNTP
DHCP Time	If this is enabled, equipment will synchronize time with DHCP server
Primary Server	IP address of Primary SNTP Server
Secondary	IP address of Secondary SNTP Server
Server	
Time zone	Local Time Zone
Resync Period	Time between resync to SNTP server. Default is 60 seconds.
12-Hour Clock	If checked, clock is 12 hour mode. If unchecked, 24 hour mode. Default is 24 hour
	mode.
Date Format	Specify the date format. Fourteen different formats are available.

Field Name	Explanation				
Daylight Saving T	Daylight Saving Time Settings				
Enable	Enable daylight saving time				
Offset(minutes)	DST offset. Default is 60 minutes				
Month	Start and end month for DST				
Week	Start and end week for DST				
Day	Start and end day for DST				
Hour	Start and end hour for DST				
Minute	Start and end minute for DST				
Manual Time Settings					
Enter the values for the current year, month, day, hour and minute. All values are required.					
Be sure to disable SNTP service before entering manual time and date.					

(2) NETWORK

a) WAN

	WAN LAN	QoS&VLAN	WEB FILTER	FIREWALL	VPN	SECURITY
	WAN Status					
	Active IP Address	172.18.2.112				
BASIC	Current Subnet Mask	255.255.0.0				
	Current IP Gateway	172.18.1.1				
• NETWORK	MAC Address	00:d8:4a:00:02:b	a			
	MAC Timestamp	20150428				
VoIP	WAN Settings					
INTERCOM	Obtain DNS Server Automatically	Enabled 🔻				
INTERCOM	Static IP 🥥	рнср 🖲		PPPoE 🔘		
SAFEGUARDING		[Apply			
MAINTENANCE	802.1X Settings					
	User	admin				
	Password	• • • • •				
	Enable 802.1X					
			Apply			
	Service Port Settings 9					
MAINTENANCE						
	Web Server Type	HTTP V				
LOGOUT	HTTP Port	80				
	HTTPS Port	443				
	Telnet Port	23				
	RTP Port Range Start	10000				
	RTP Port Quantity	200				
		[Apply			

WAN					
Field Name	Explanation				
WAN Status					
Active IP Address		172.18.2.193			
Current Sub	ana an	255.255.0.0			
Current IP G		172.18.1.1			
MAC Addres	100	0c:38:3e:13:3b:90			
	1	000000000000000000000000000000000000000			
Active IP	The current IP ad	dress of the equipment			
address					
Current subnet	The current Subn	et Mask			
mask					
Current IP	The current Gate	way IP address			
gateway					
MAC address	The MAC address	s of the equipment			
MAC Timestamp	Get the MAC add	ress of time.			
WAN Settings					
Obtain DNS	Server Automatical	ly Enabled 💌			
Static IP 🔘		DHCP PPPoE			
		Apply			
Select the approp		le. The equipment supports three network modes:			
Static	Network parameters must be entered manually and will not change. All parameters				
		are provided by the ISP.			
DHCP	Network parame	ters 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 chos	en, the screen belo	ow will appear. Enter values provided by the ISP.			
IP Address		192.168.1.179			
Subnet Mask		255.255.255.0			
IP Gateway		192.168.1.1			
DNS Domain					
Primary DNS		202.96.134.133			
Secondary DNS		202.96.128.68			
Static IP address	Please enter the				
Subnet mask	Please enter the				
Gateway	Please enter the				
DNS Domain		ain suffix. When the user enter the domain name DNS address			
cannot be resolved, the domain equipment to resolve in the domain name.					
Primary DNS	Please enter the Primary DNS server address				
Secondary DNS	Please enter the	Secondary DNS server address			

Field Name	Explanation				
If PPPoE is chose	n, the screen below will appear. Enter values provided by the ISP.				
Service Name	admin				
User	user123				
Password					
Service Name	PPPoE Service name, Usually the default value.				
User	ADSL user account				
Password	ADSL password				
After entering the	e new settings, click the APPLY button. The equipment will save the new settings and				
apply them. If a i	new IP address was entered for the equipment, it must be used to login to the phone				
after clicking the	APPLY button.				
802.1X Settings					
User	admin				
Password	•••••				
Enable 802.1	LX 🔲				
User	802.1X user account				
Password	802.1X password				
Enable 812.1X	Open/Close 812.1X				
Service Port Setti	ngs				
Web Server type	Specify Web Server Type – HTTP or HTTPS				
	Port for web browser access. Default value is 80. To enhance security, change this from				
HTTP port	the default. Setting this port to 0 will disable HTTP 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				
HTTPS port	downloaded into the equipment.				
	Default value is 443. To enhance security, change this from the default.				
Telnet port	Port for Telnet access. The default is 23.				
RTP port range	Set the beginning value for RTP Ports. Ports are dynamically allocated.				
start	Set the beginning value for KTF Forts. Forts are dynamically anotated.				
RTP port	Set the maximum quantity of RTP Ports. The default is 200.				
quantity	Set the maximum quantity of KTF Folts. The default is 200.				
Note:					
1) Any changes m	ade on this page require a reboot to become active.				
2) It is suggested that changes to HTTP Port and Telnet ports be values greater than 1024. Values less					
than 1024 are res	erved.				
3) If the HTTP por	t is set to 0, HTTP service will be disabled.				

3) If the HTTP port is set to 0, HTTP service will be disabled.

b) LAN

	WAN	LAN	QoS&VLAN	WEB FILTER	FIREWALL	VPN	SECURITY
> BASIC	LAN Settings 이						
• NETWORK	IP Address		192.168.10.1				
	Subnet Mask		255.255.255.0				
> VoIP	Enable Bridge M	Mode					
• INTERCOM				Apply			

LAN		
Field Name	Explanation	
IP address	LAN static IP	
Subnet mask	LAN Subnet Mask	
Enable bridge	If Bridge Mode is activated, the equipment will not provide an IP address for the LAN	
mode	port. Instead, the LAN and WAN will be part of the same network. If this is activated,	
	clicking Apply, will cause the equipment will reboot.	
Note: If bridge mode is chosen, static LAN configuration will be disabled automatically.		

c) QoS&VLAN

The equipment supports 802.1Q/P protocol and DiffServ configuration. Use of a Virtual LAN (VLAN) allows voice and data traffic to be separated.

Chart 1 shows a network switch with no VLAN. Any broadcast frames will be transmitted to all other ports. For example, and frames broadcast from Port 1 will be sent to Ports 2, 3, and 4.

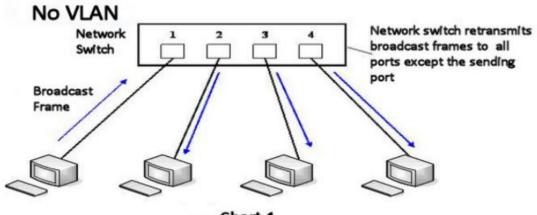
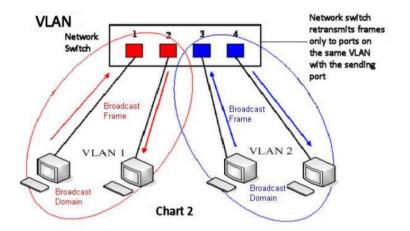


Chart 1

Chart 2 shows an example with two VLANs indicated by red and blue. In this example, frames broadcast from Port 1 will only go to Port 2 since Ports 3 and 4 are in a different VLAN. VLANs can be used to divide a network by restricting the transmission of broadcast frames.



Note: In practice, VLANs are distinguished by the use of VLAN IDs.

	WAN	LAN QoS&VLAN	WEB FILTER FIREW	ALL VPN SECURITY
> BASIC	Link Layer Discovery Pr	otocol (LLDP) Settings		
	Enable LLDP 🚺		Packet Interval	60 (1~3600)second(s)
NETWORK	Enable Learning Fur	nction 🔲		
> VoIP	Quality of Service (QoS			
	Enable DSCP		SIP DSCP	46 (0~63)
> INTERCOM	Audio RTP DSCP	46 (0~63)		
> SAFEGUARDING	WAN Port VLAN Settings			
	Enable WAN Port VL	AN 🔲	WAN Port VLAN ID	256 (0~4095)
> MAINTENANCE	SIP 802.1P Priority	0 (0~7)	Audio 802.1P Priority	0 (0~7)
> LOGOUT	LAN Port VLAN Settings			
	LAN Port VLAN Mode	Follow WAN V	LAN Port VLAN ID	254 (0~4095)
			Apply	

QoS&VLAN	
Field Name	Explanation
LLDP Settings	
Enable LLDP	Enable or Disable Link Layer Discovery Protocol (LLDP)
Packet Interval	The time interval for sending LLDP Packets
Enable Learning Function	Enables the telephone to synchronize its VLAN data with the Network Switch. The telephone will automatically synchronize DSCP, 802.1p, and VLAN ID values even if these values differ from those provided by the LLDP server.
QOS Settings	
Enable DSCP	Enable or Disable Differentiated Services Code Point (DSCP)
SIP DSCP	Specify the value of the SIP DSCP in decimal

Audio RTP DSCP	lio RTP DSCP Specify the value of the Audio DSCP in decimal			
Field Name	Explanation			
WAN Port VLAN Setting	gs			
Enable WAN Port	Enable or Disable WAN Port VLAN			
VLAN				
WAN Port VLAN ID	Specify the value of the WAN Port VLAN ID. Range is 0-4095			
SIP 802.1P Priority	Specify the value of the signal 8021.p priority. Range is 0-7			
Audio 802.1P Priority	Specify the value of the voice 802.1p priority. Range is 0-7			
LAN Port VLAN Settings	5			
	Follow WAN: LAN Port ID is same as WAN ID.			
LAN Port VLAN	Disable: Disable Port VALN			
	Enable: Specify a VLAN ID for the LAN port which is different from WAN ID			
LAN Port VLAN ID	Used when the VLAN ID is different from WAN ID. Range is 0-4095			

d) WEB FILTER

	WAN	LAN	QoS&VLAN	WEB FILTER	FIREWALL	VPN	SECURITY
> BASIC	Web Filter Table						
• NETWORK	Start IP Address		End IP	Address		Option	
> VoIP	Web Filter Table Set		End IP	Address		Add	
> INTERCOM	Web Filter Setting						
> SAFEGUARDING	Enable Web Filt	er 🔲	Apt	oly			
> MAINTENANCE							
> LOGOUT							

Web filter			
The Web filter is used to limit access to the equipment. When the web filter is enabled, only the IP			
addresses between t	he start IP and end IP can access the equipment.		
Field Name	Explanation		
Web Filter Table	Web Filter Table		
Webpage access allo	Webpage access allows display the IP network list;		
Web Filter Table Settings			
Beginning and Ending IP Address for MMI Filter, Click add this filter range to the Web Filter Table			
Web Filter Setting			
Select to enable MMI Filter. Click [apply] Make filter settings effective.			

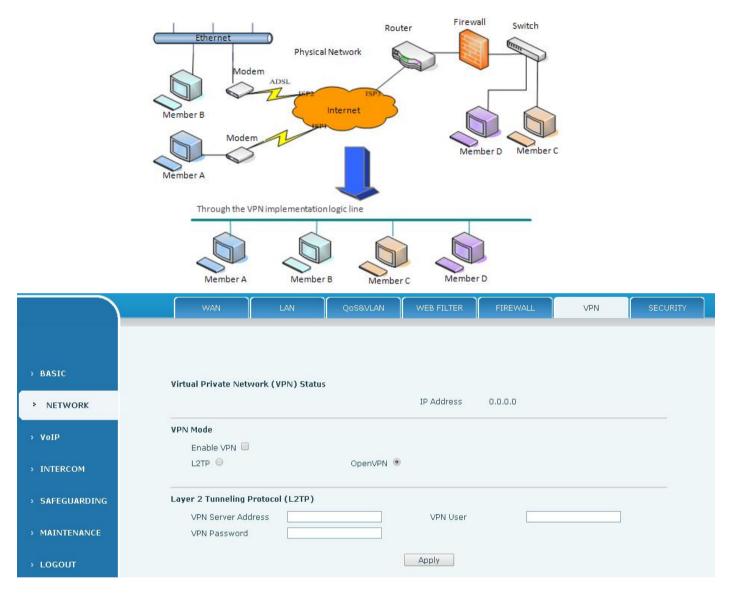
e) FIREWALL

	WAN	LAN	QoS&VLAN	WEB FILTER	FIREWALL	VPN	SECURITY
	Firewall Type						
> BASIC		Enable Input Rule	95 🗖	Apply	Enable Outpu	ut Rules 🔲	
• NETWORK	Firewall Input Rule	Table					
	Index Deny/P	ermit Protocol Src Add	iress Src Mas	k Dest Addre	ss Dest Mask	. Range	Port
> VoIP							
> INTERCOM	Firewall Output Rul				-		
A INTERCOM	Index Deny/P	ermit Protocol Src Add	Iress Src Mas	k Dest Addre	ss Dest Mask	k Range	Port
> SAFEGUARDING	Firewall Settings						
	Input/Output	Input 🔻		Src Address			
> MAINTENANCE	Deny/Permit	Deny 💌		Dest Address			Add
	Protocol	UDP 🔻		Src Mask			
> LOGOUT	Port Range	more than	•	Dest Mask			
	Rule Delete Option						
	Input/Output	Input 🔻		Index To Be Deleted			Delete

Firewall			
Firewall rules can be used to prevent unauthorized Internet users from accessing private networks			
connected to this phone (input rule), or prevent unauthorized devices connected to this phone from			
accessing the Int	ernet (output rule). Each rule type supports a maximum of 10 items.		
Field Name	Explanation		
Firewall Rules Se	ettings		
Enable Input Rules	Enable rules limiting access from the Internet.		
Enable Output Rules	Enable rules limiting access to the Internet.		
Firewall Settings			
Input / Output	Specify if the current rule is input or output.		
Deny/Permit	Specify if the current rule is Deny or Permit.		
Protocol type	Filter protocol type (TCP/ UDP/ ICMP/ IP)		
Port Range	Set the filter Port range		
Source Address	Set source address. It can be a single IP address or use * as a wild card. For example: 192.168.1.14 or *.*.*.14.		
Destination	Set destination address. It can be a single IP address or use * as a wild card. For		
Address	example: 192.168.1.14 or *.*.*.14.		
	Set the source address mask. For example: 255.255.255.255 points to one host while		
Source Mask	255.255.255.0 points to a C type network.		
Destination	Set the destination address mask. For example: 255.255.255.255 points to one host		
Mask	while 255.255.255.0 points to a C type network.		

f) 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.



Field Name	Explanation	
IP Address	Shows the current VPN IP address.	
VPN Mode		
Enable VPN	Enable/Disable VPN.	
L2TP	Select Layer 2 Tunneling Protocol	
OpenVPN	Select OpenVPN Protocol. (Only one protocol may be activated. After the selection is	
	made, the configuration should be saved and the phone rebooted.)	
L2TP		
VPN Server		
address	Set VPN L2TP Server IP address.	
VPN user	Set User Name access to VPN L2TP Server.	

VPN password	Set Password access to VPN L2TP Server.
--------------	---

g) SECURITY

	WAN LAN QOS&VLAN WEB FILTER FIREWALL VPN SECURITY	
> BASIC	Update Security File	
• NETWORK	Select Security File: Browse Update	
> VoIP	Delete Security File Select Security File: https.pem Delete	
> INTERCOM	SIP TLS Files	
> SAFEGUARDING	HTTPS Files	
> MAINTENANCE	https.pem (4555 Bytes)	
› LOGOUT	OpenVPN Files	
Field Name	Explanation	
Update Security File	Select the security file to be updated. Click the Update button to update.	
Delete Security File Select the security file to be deleted. Click the Delete button to Delete.		
SIP TLS Files	Show SIP TLS authentication certificate.	
HTTPS Files	Show HTTPS authentication certificate.	

OpenVPN Files	Show OpenVPN File authentication certificate file.

(3) VOIP

a) SIP

Configure a SIP server on this page.

	SIP			
BASIC	SIP Line SIP 1			
	Basic Settings >>			
NETWORK	Status	Registered		
	Server Address	172.18.1.88		
VoIP	Server Port	5060		
INTERCOM	Authentication User	5104		
INTERCOM	Authentication Password	*****		
SAFEGUARDING	SIP User	5104		
	Display Name	5104		
MAINTENANCE	Enable Registration			
	Advanced SIP Settings >>			
> LOGOUT		Apply		
	SIP Global Settings >>			
Advanced SIP S	ettings >>			
Proxy Serve		Proxy Server Port		
Proxy User		Proxy Password		
Backup Server Address		Backup Server Port	5060	
Domain Real		Server Name		
RTP Encrypti		Enable Session Timer		
Registration		Session Timeout	0 second(s)	
Keep Alive T		Keep Alive Interval	60 second(s)	
the prime of the second s	YE			

Enable PRACK		A	ns. With a Single Co	dec 🖌 🔲	
Enable Strict Proxy		Δ	uto TCP		
Enable DNS SRV		ι	Ise VPN		
Transport Protocol	UDP 🔻				
		Apply			
SIP Global Settings >>					
Strict Branch			Enable Group		
Registration Failure Retry Time	32	second(s)	DND Return Code	480(Temporarily Not Available)	T
Reject Return Code	603(Decline)	•	Busy Return Code	486(Busy Here)	•

RFC Protocol Edition

Keep Authentication

Local Port

RFC3261 🔻

5060

DTMF Type

Enable Rport

DTMF SIP INFO Mode

AUTO

Send */# 🔻

Ŧ

SIP			
Field Name Explanation			
Basic Settings (Choose the SIP line to configured)			
Status	Shows registration status. If the registration is successful will display has been		
Status	registered, not successful display not registered, the wrong password is displayed		

	403 errors, account number failure display timeout.	
Server address	SIP server IP address or URI.	
Server port	SIP server port. Default is 5060.	
Authentication	SID account name (Login ID)	
User	SIP account name (Login ID).	
Authentication	SID registration password	
password	SIP registration password.	
SIP user	Phone number assigned by VoIP service provider. Equipment will not register if there	
	is no phone number configured.	
Display name	Set the display name. This name is shown on Caller ID.	

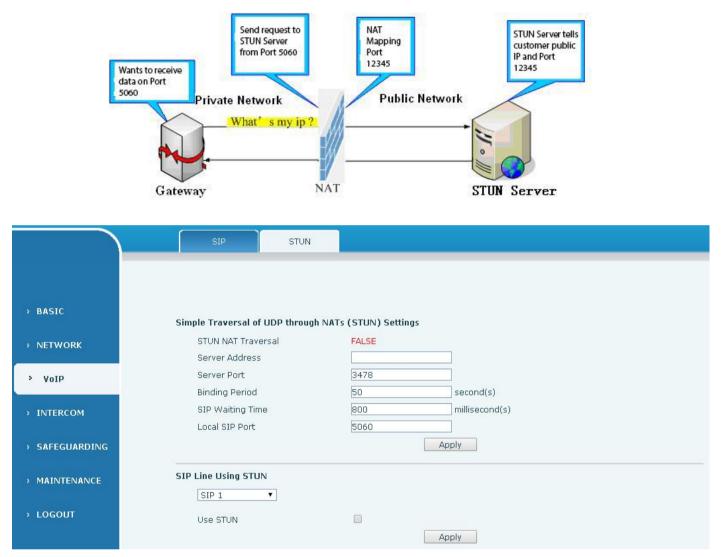
Field Name	Explanation			
Advanced SIP Settings				
Proxy server	SIP proxy server IP address or URI, (This is normally the same as the SIP Registrar			
address	Server)			
Proxy server port	SIP Proxy server port. Normally 5060.			
Proxy user	SIP Proxy server account.			
Proxy password	SIP Proxy server password.			
Backup Proxy	Backup SIP Server Address or URI (This server will be used if the primary server is			
server address	unavailable)			
Backup Proxy	Packup SID Server Port			
server port	Backup SIP Server Port			
Domain Realm	SIP Domain if different than the SIP Register Server.			
Server name	Name of SIP Backup server			
RTP Encryption	Enable/Disable RTP Encryption.			
Enable Session	If enabled, this will refresh the SIP session timer per RFC4028.			
Timer				
Registration	SIP re-registration time. Default is 60 seconds. If the server requests a different time,			
Expires	the phone will change to that value.			
Session Timeout	Refresh interval if Session Timer is enabled.			
	Specifies the NAT keep alive type. If SIP Option is selected, the equipment will send			
Keep Alive Type	SIP Option sip messages to the server every NAT Keep Alive Period. The server will			
Keep Alive Type	then respond with 200 OK. If UDP is selected, the equipment will send a UDP			
	message to the server every NAT Keep Alive Period.			
Keep Alive Interval	Set the NAT Keep Alive interval. Default is 60 seconds			
User Agent	Set SIP User Agent value.			

Server Type	Configures phone for unique requirements of selected server.		
	DTMF sending mode. There are four modes:		
	In-band		
	• RFC2833		
DTMF Type	• SIP_INFO		
	• AUTO		
	Different VoIP Service providers may require different modes.		
RFC Protocol	Select SIP protocol version RFC3261 or RFC2543. Default is RFC3261. Used for		
Edition	servers which only support RFC2543.		
DTMF SIP INFO	Vou can share Sand 10/11 or Sand */#		
Mode	You can chose Send 10/11 or Send */#		
Local Port	SIP port. Default is 5060.		
Enable Rport	Enable/Disable support for NAT traversal via RFC3581 (Rport).		

Field Name	Explanation		
Keep Authentication	Enable /disable registration with authentication. It will use the last authentication field which passed authentication by server. This will decrease the load on the servit enabled		
Enable PRACK	Enable or disable SIP PRACK function. Default is OFF. It is suggested this be used.		
Ans. With a Single Codec	If enabled phone will respond to incoming calls with only one codec.		
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from the server it will use the source IP address, not the address in via field.		
Auto TCP	Force the use of TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes		
Enable DNS SRV	Enables use of DNS SRV records		
Use VPN	Enable SIP use VPN for every line individually, not all of them		
Transport Protocol	Configuration using the transport protocol, TCP, TLS or UDP, the default is UDP.		
SIP Global Settings			
Strict Branch	Enable Strict Branch - The value of the branch must be after"z9hG4bK" in the VIA field of the INVITE message received, or the phone will not respond to the INVITE. Note: This will affect all lines		
Enable Group	Enable SIP Group Backup. This will affect all lines		
Registration	Registration failures retry time – If registrations fails, the phone will attempt to		
Failure Retry Time	register again after registration failure retry time. This will affect all lines		
DND Return Code	Specify SIP Code returned for DND. Default is 480 - Temporarily Not Available.		
Reject Return Code	Specify SIP Code returned for Rejected call. Default is 603 – Decline.		
Busy Return Code	Specify SIP Code returned for Busy. Default is 486 – Busy Here.		

b) STUN

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.



STUN				
Field Name	Explanation			
STUN NAT Traversal	Shows whether or not STUN NAT Traversal was successful.			
Server Address	STUN Server IP address			
Server Port	Port STUN Server Port – Default is 3478.			
	STUN blinding period – STUN packets are sent at this interval to keep the NAT			
Binding Period	mapping active.			
SIP Waiting Time	SIP Waiting Time Waiting time for SIP. This will vary depending on the network.			
Local SIP Port	Local SIP Port Port configure the local SIP signaling			
SIP Line Using STUN (SIP1 or SIP2)				
Use STUN	Enable/Disable STUN on the selected line.			

Note: the SIP STUN is used to achieve the SIP penetration of NAT, is the realization of a service, when the equipment configuration of the STUN server IP and port (usually the default is 3478), and select the Use Stun SIP server, the use of NAT equipment to achieve penetration.

(4) Intercom

a) FUNCTION KEY

	FUNCTION KEY	AUDIO	FEATURE	MCAST Actio	n URL	
> BASIC						
	Function Key Set	tings				
> NETWORK	Key	Туре	Number 1	Number 2	Line	Subtype
	DSS Key 1	None 🔻			SIP1 V	Speed Dial 🔹
> VoIP	DSS Key 2	None 🔹			SIP1 T	Speed Dial 🔹
	DSS Key 3	None 🔹			SIP1 V	Speed Dial 🔹
INTERCOM	DSS Key 4	None 🔹			SIP1 V	Speed Dial 🔹
> SAFEGUARDING				Apply		

Key Event Settings

Set the key type to the Key Event.

Key	Туре	Number 1	Number 2	Line	Subtype
DSS Key 1	Key Event 💽			SIP1 🗸	ОК
DSS Key 2	None Hot Key			SIP1 🕑	None Dial
DSS Key 3	Line Key Event			SIP1 V	Release OK
DSS Key 4	Multicast			SIP1 💉	Handfree Speed Dial

DSS key type	Subtype	Usage
	None	Not responding
	Dial	Dial function
Key Event	Release	End calls
	ОК	Identify key

Ha	landfree	The hand-free key(with hook dial, hang up)
----	----------	--

Hot key Settings

Enter the phone number in the input box, when you press the shortcut key, equipment will dial set telephone number. This button can also be used to set the IP address, press the shortcut key IP direct dial call.

Key	Туре	Number 1	Number 2	Line	Subtype	
DSS Key 1	Hot Key			SIP1 ¥	Speed Dial	~
DSS Key 2	None Hot Key			SIP1 V	Speed Dial Intercom	
DSS Key 3	Line Key Event			SIP1 V	Speed Dial	×
DSS Key 4	Multicast			SIP1 M	Speed Dial	×

DSS key type	Number	Line	Subtype	Usage
Hot Key	Fill the called party's SIP account or address	The SIP account corresponding lines	Speed Dial	In Speed dial mode, with Enable Speed Dial Enable Can define whether this call is allowed to be hang up by re-press the speed dial
			Intercom	In Intercom mode, if the caller's IP phone support intercom feature, can realize auto answer

Multicast Settings

Multicast function is launched will voice messages sent to set the multicast address, all equipment to monitor the group multicast address can receive sponsors speech information, etc. Using multicast functionality can be simple and convenient to send notice to each member in the multicast.

Through the DSS Key configuration multicast calling WEB is as follows:

Key	Туре	Number 1	Number 2	Line	Subtyp	e
DSS Key 1	Multicast 💌			SIP1 M	G.711A	¥
DSS Key 2	None Hot Key			SIP1 V	G.711A G.711U	
DSS Key 3	Line Key Event			SIP1 💉	G.722 G.723.1	
DSS Key 4	Multicast			SIP1 🗸	G.726-32 G.729AB	

DSS key type	Number	Subtype	Usage	
Multicast	Cat the heat ID address	G.711A	Narrowband speech coding (4Khz)	
	Set the host IP address and port number, the middle separated by a	G.711U		
		G.722	Wideband speech coding (7Khz)	
		G.723.1	Narrowband speech coding (4Khz)	

\diamond operation mechanism

Device through the DSS Key configuration of multicast address and port and started coding; set by WEB to monitor the multicast address and port; device sends a multicast, listens to the address of the device can receive the multicast content.

♦ calling configuration

The call is already exists, and three party or initiated multicast communication, so it will not be able to launch a new multicast call.

b) AUDIO

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

	FUNCTION KEY AUDIO FEATURE MCAST Action URL
	Audio Settings
	First Codec G.711A Second Codec G.711U
> BASIC	Third Codec G.722 Fourth Codec G.729AB DTMF Payload Type 101 (96~127) Default Ring Type Type 1
> NETWORK	G.729AB Payload Length 20ms ▼ Tone Standard China ▼
> VoIP	G.722 Timestamps 160/20ms ▼ G.723.1 Bit Rate 6.3kb/s ▼ Enable VAD
• INTERCOM	Talk Volume Settings
> SAFEGUARDING	SPK Output Volume 5 (1~9) MIC Input Volume 5 (1~9)
> MAINTENANCE	Media Volume Settings
> LOGOUT	Broadcast Output Volume 5 (1~9) Signal Tone Volume 5 (0~9)
	Codec Gain Settings
	Handsfree Hardware MIC Gain 5 (1~11) Handsfree Hardware Speakerphone Gain 3 (1~8)
	Apply

Field Name	Explanation
Audio Settings	
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	The RTP Payload type that indicates DTMF. Default is 101

Туре	
Default Ring Type	Ring Sound – There are 9 standard types and 3 User types.
G.729AB Payload	C 720AB Dayload Longth Adjusts from 10 C0 mSec
Length	G.729AB Payload Length – Adjusts from 10 – 60 mSec.
Tone Standard	Configure tone standard area.
G.722	Chairea are 100/20ms or 220/20ms
Timestamps	Choices are 160/20ms or 320/20ms.
G.723.1 Bit Rate	Choices are 5.3kb/s or 6.3kb/s.
Enable VAD	Enable or disable Voice Activity Detection (VAD). If VAD is enabled, G729 Payload
	length cannot be set greater than 20 mSec.

Field Name	Explanation			
Talk Volume Settin	Talk Volume Settings			
SPK Output				
Volume	Set the s	peaker calls the volume level.		
MIC Input	Sot the N	AIC calls the volume level.		
Volume	Set the w			
Media Volume Set	Media Volume Settings			
Broadcast Output	Set the broadcast the output volume level.			
Volume				
Signal Tone	Sot tho a	udio signal the output volume level.		
Volume	Set the a			
Codec Gain Setting	Codec Gain Settings			
Hands-free Hardware MIC		Settings Llands free Llandware MIC Cain		
Gain		Settings Hands-free Hardware MIC Gain		
Hands-free Hardwa	ire	Settings hands-free Hardware Speakerphone Gain		
Speakerphone Gain				

c) FEATURE

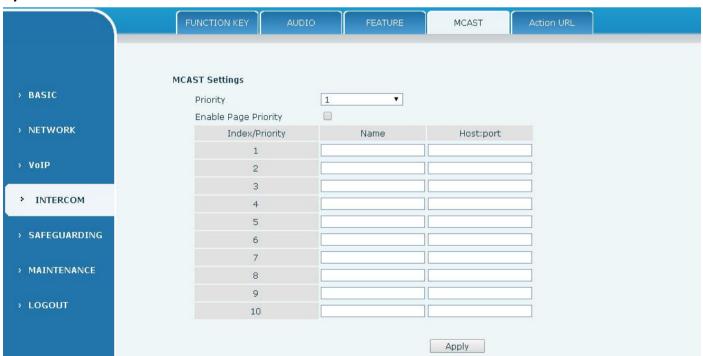
	FUNCTION KEY AUDIO	FEATURE	MCAST Action URL	
	Feature Settings			
BASIC	DND (Do Not Disturb)		Ban Outgoing	
	Enable Intercom Mute	2	Enable Intercom Tone	
NETWORK	Enable Auto Answer	Lines and IP Call 🔻	Auto Answer Timeout	0 (0~60s)
11-70	No Answer Handdown		No Ans. Handdown Time	30 (1~60s)
VoIP	Dial Fixed Length to Send		Send length	11
	Enable Speed Dial Handdowr	Enable 🔻	Dial Number Voice Play	Disable 🔻
INTERCOM	Use Function Key to Answer	Disable 🔻	Status Led Reuse Mode	Disable 🔻
	Hot Key Dial Mode Select	Main-Secondary 🔻	Call Switched Time	16 (5~50s)
SAFEGUARDING	Day Start Time	06:00 (00:00~23:59)	Day End Time	18:00 (00:00~23:59)
	Description	IP Intercom		
MAINTENANCE	Construction (INC) account (C)		Apply	
LOGOUT	Block Out Settings			
	-	DI	ock Out	
		Add		Delete

Field Name	Explanation		
Feature Settings			
DND (Do Not Disturb)	DND might be disabled phone for all SIP lines, or line for SIP individually.But the		
	outgoing calls will not be affected		
Ban Outgoing	If enabled, no outgoing calls can be made.		
Enable Intercom	If anabled, mutos incoming calls during an intersom call		
Mute	If enabled, mutes incoming calls during an intercom call.		
Field Name	Explanation		
Enable Intercom	If anothed plays intersom ring tong to playt to an intersom call		
Tone	If enabled, plays intercom ring tone to alert to an intercom call.		
Enable Auto Answer	Enable Auto Answer function		
Auto Answer	Set Auto Answer Timeout		
Timeout	Set Auto Answer Timeout		
No Answer	Enable automatically hang up when no answer		
Handdown	Enable automatically hang up when no answer		
No Answer	Configuration in a set time, automatically hang up when no answer		
Handdown Time	Configuration in a set time, automatically hang up when no answer		
Dial Fixed Length to	Enable or disable dial fixed length to send		
Send	Enable or disable dial fixed length to send.		
Sandlangth	The number will be sent to the server after the specified numbers of digits are		
Send length	dialed.		
Enable Speed Dial	Enable Speed Dial Hand Up function		
Handdown			
Dial Number Voice	Configuration Open / Close Dial Number Voice Play		
Play			

Use Function Key to	Configure whether to enable the function keys, is disabled by default.	
Answer	compare whether to enable the function keys, is disabled by default.	
Status Led Reuse	Enable the function, the registered status indicator will reuse the call instructions	
Mode	function, which means the LED will flashes in the call state.	
	<primary secondary="">mode allow system to call primary extension first, if there</primary>	
	were no answer, it would cancel the call and then call secondary extension	
Hot Key Dialed Mode	automatically.	
Selection	<day night="">mode allow system to check the calling time is belong to Day or</day>	
	Night time, and then decide to call the number 1 or number 2 automatically.	
	Users just press speed dial key once.	
Call Switched Time	The period between hot key dialing to the first and second number.	
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>	
Description	Device description displayed on IP scanning tool software.	
Block Out Settings		
Add or Delete Blocked	numbers – Enter the prefix of numbers which should not be dialled by the phone.	
For example, if 001 is e	entered, the phone will not dial any numbers beginning with 001.	
X and x are wildcards v	which match single digits. For example, if 4xxx or 4XXX is entered, the phone will	
not dial any 4 digit nun	nbers beginning with 4. It will dial numbers beginning with 4 which are longer or	
i		

shorter than 4 digits.

d) MCAST



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, monitor and play the RTP stream which sent by the multicast address.

MCAST Settings

Equipment can be set up to monitor up to 10 different multicast address, 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 the priority, if the priority of the incoming flows of multicast RTP, lower precedence than the current common calls, device will automatically ignore the group RTP stream. If the priority of the incoming flow of multicast RTP is higher than the current common calls priority, device will automatically receive the group RTP stream, and keep the current common calls in state. You can also choose to disable in the receiving threshold drop-down box, the device will automatically ignore all local network multicast RTP stream.

- 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 stream
 - ♦ Enable the page priority:

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

Web Settings:

Priority	1	*
Enable Page Priority		and the second design of the s
Index/Priority	Name	e 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, which is the highest priority.

Note: when pressing the multicast key for multicast session, both multicast sender and receiver will beep.

Listener configuration

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)

M

"Group 1", "Group 2" and "Group 3" are your setting monitoring multicast name. The group name will be displayed on the screen when you answer the multicast. If you have not set, the screen will 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 high priority. The followings will explain how to use this option:

- ☆ The purpose of setting monitoring multicast "Group 1" or "Group 2" or "Group 3" launched a multicast call.
- ♦ All equipment has one or more common non multicast communication.
- ↔ When you set the Priority for the disable, multicast any level will not answer, multicast call is rejected.
- when you set the Priority to a value, only higher than the priority of multicast can come in, if you set the Priority is 3, group 2 and group 3 for priority level equal to 3 and less than 3 were rejected, 1 priority is 2 higher than ordinary call priority device can answer the multicast message at the same time, keep the hold the other call.

• Green part (Enable Page priority)

Set whether to open more priority is the priority of multicast, multicast is pink part number. Explain how to use:

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

Multicast service

• Send: when configured ok, our key press shell on the corresponding equipment, equipment directly into the Talking interface, the premise is to ensure no current multicast call and 3-way of the case, the multicast can be established.

Lmonitor: IP port and priority configuration monitoring device, when the call is initiated and incoming multicast, directly into the Talking interface equipment.

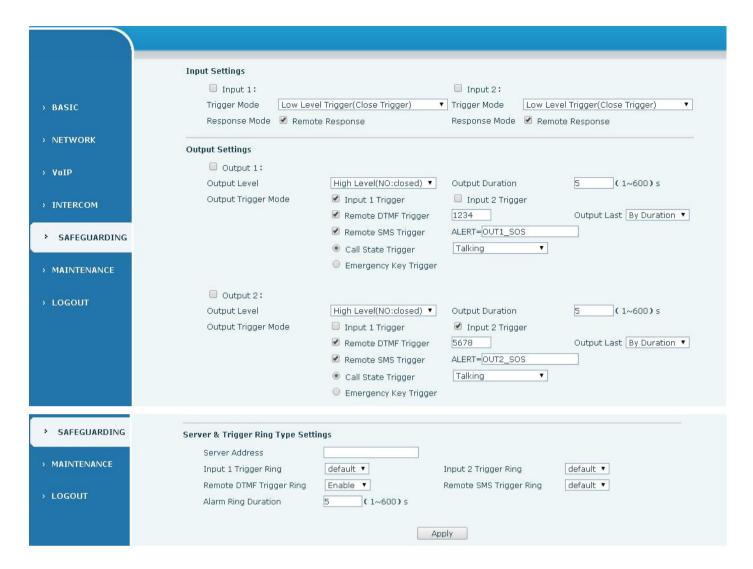
e) Action URL

	FUNCTION KEY	AUDIO	FEATURE	MCAST	Action URL	
	Action URL Settings					
	Active URI Limit	IP				
> BASIC	Setup Complete	ed				
C DHOIC	Registration Suc	ccess				
> NETWORK	Registration Dis	abled				
	Registration Fai	led				
> VoIP	Off Hook					
State and the	On Hook					
INTERCOM	Incoming Call					
	Outgoing Call					
SAFEGUARDING	Call Established	l				
	Call Terminated					
> MAINTENANCE	DND Enabled					
	DND Disabled					
› LOGOUT	Mute					
	Unmute					
	Missed Call					
	IP Changed					
	Idle To Busy					
	Busy To Idle					
				Apply		

Action URL 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

(5) SAFEGUARDING



Security Settings	Security Settings		
Field Name	Explanation		
Input settings			
Input 1	Open /Close Input port1		
	When choosing the low level trigger (closed trigger), detect the input port 1 (low		
Triggor Modo	level) closed trigger.		
Trigger Mode	When choosing the high level trigger (disconnected trigger), detect the input port 1		
	(high level) disconnected trigger.		
Response Mode	Open /Close Input port1 the Remote Response		
Input 2	Open /Close Input port2		
	When choosing the low level trigger (closed trigger), detect the input port 2 (low		
Trigger Mede	level) closed trigger.		
Trigger Mode	When choosing the high level trigger (disconnected trigger), detect the input port 2		
	(high level) disconnected trigger.		
Response Mode	Open /Close Input port2 the Remote Response		
Field Name	Explanation		
Output Settings			
Output 1/2	Open/close, Output 1/Output 2		

	When choos	ing the low level trigger (NO: normally open), when meet the trigger			
		ition, trigger the NO port disconnected.			
Output Level		n choosing the high level trigger (NO: normally close), when meet the trigger			
	condition, trigger the NO port close.				
Output					
Duration	Changes in port, the duration of. The default is 5 seconds.				
Output Trigger M	ode: There are	e many kinds of trigger modes, multiple choices.			
Input port1	When the in	When the input port1 meet to trigger condition, the output port1 will trigger(The Port			
trigger	level time ch	ange, By < Output Duration > control)			
Input port2	When the in	out port2 meet to trigger condition, the output port2 will trigger(The Port			
trigger	level time ch	ange, By < Output Duration > control)			
		Received the terminal equipment to send the DTMF password, if			
	By duration	correct, which triggers the corresponding output port (The Port level			
		time change, By < Output Duration > control)			
Remote DTMF		During the call, receive the terminal equipment to send the DTMF			
trigger	By Calling	password, if correct, which triggers the corresponding output port (The			
	State	Port level time change, (By call state control, after the end of the call,			
		port to return the default state)			
Remote SMS	In the remot	In the remote device or server to send instructions to ALERT=[instructions], if correct,			
trigger	which triggers the corresponding output port				
	The port output continuous time synchronization and trigger state changes, including				
	the trigger co	gger conditions: 1, call; 2, call and singing; 3, singing; three models. (for			
Call state trigger	example: the	ample: the call trigger output port, will be in conversation state continued to output			
	the correspo	nding level)			
Emergency key	When the er	nergency call button to trigger the equipment shell, which triggers the			
trigger	correspond	ing output port(after the end of the call, port to return the default state)			
Server & Trigger					
		gure remote response server address(including remote response server			
Server Address		dress and tamper alarm server address)			
	Wher	the input port 1 triggering condition is satisfied, the corresponding ring			
Input 1 trigger rin	tone				
	Wher	When the input port 2 triggering condition is satisfied, the corresponding ring			
Input 2 trigger rin	Ig	tone or alarm			
Remote DTMF tri	gger				
ring	When	n received the remote DTMF command, whether to output the ringtone			
Remote SMS trigg	ger				
ring	When	n receiving the remote SMS instructions, whether to output the ringtone			
Alarm Ring Durat	ion durat	ion of alarm ring(not including tamper alarm)			
		0(0 p)			

(6) MAINTENANCE

a) AUTO PROVISION

	AUTO PROVISION SYSLOG	CONFIG UPDATE	ACCESS	REBOOT	
	A GAL ORD				
> BASIC	Auto Provision Settings Current Config Version				
> NETWORK	Common Config Version CPE Serial Number OC	00000000000000000000000000000000000000)2ba		
> VoIP	User Password				
> INTERCOM	Config Encryption Key Common Config Encryption Key				
> SAFEGUARDING	Save Auto Provision Information]			
• MAINTENANCE	DHCP Option Settings >> Plug and Play (PnP) Settings >>				
› LOGOUT	Phone Flash Settings >>				
	TRO69 Settings >>	Apply			

The equipment supports PnP, DHCP, and Phone Flash to obtain configuration parameters. They will be queried in the following order when the equipment boots.

DHCP option \rightarrow PnP server \rightarrow Phone Flash

Field Name	Explanation	
Auto Provision Se	ettings	
	Show the current config file's version. If the version of configuration downloaded is	
Current Config	higher than this, the configuration will be upgraded. If the endpoints confirm the	
Version	configuration by the Digest method, the configuration will not be upgraded unless it	
	differs from the current configuration	
	Show the common config file's version. If the configuration downloaded and this	
Common Config	configuration is the same, the auto provision will stop. If the endpoints confirm the	
Version	configuration by the Digest method, the configuration will not be upgraded unless it	
	differs from the current configuration.	
CPE Serial	Carial sumber of the acuisment	
Number	Serial number of the equipment	
llcor	Username for configuration server. Used for FTP/HTTP/HTTPS. If this is blank the	
User	phone will use anonymous	
Password	Password for configuration server. Used for FTP/HTTP/HTTPS.	
Config	Encryption low for the configuration file	
Encryption Key	Encryption key for the configuration file	

Field Name	Explanation
Common Config	Encryption key for common configuration file
Encryption Key	Encryption key for common configuration file

Save Auto		
Provision	Save the auto provision username and password in the phone until the server url	
Information	changes	
DHCP Option Set	tings	
DHCP Option	The equipment supports configuration from Option 43, Option 66, or a Custom DHC	
Setting	option. It may also be disabled.	
Custom DHCP	Custom ontion number Must be from 128 to 254	
Option	Custom option number. Must be from 128 to 254.	
Plug and Play(Pn	P)Settings	
	If this is enabled, the equipment will send SIP SUBSCRIBE messages to a multicast	
Enable PnP	address when it boots up. Any SIP server understanding that message will reply with a	
	SIP NOTIFY message containing the Auto Provisioning Server URL where the phones	
	can request their configuration.	
PnP server	PnP Server Address	
PnP port	PnP Server Port	
PnP Transport	PnP Transfer protocol – UDP or TCP	
PnP Interval	Interval time for querying PnP server. Default is 1 hour.	
Phone Flash Setti	ngs	
	Set FTP/TFTP/HTTP server IP address for auto update. The address can be an IP	
Server Address	address or Domain name with subdirectory.	
Config File	Specify configuration file name. The equipment will use its MAC ID as the config file	
Name	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 – no update	
Update Mode	2. Update after reboot – update only after reboot.	
	3. Update at time interval – update at periodic update interval	
TR069 Settings		
Enable TR069	Enable/Disable TR069 configuration	
Enable TR069		
Warning Tone	Enable or disable TR069 Warning Tone	
ACS Server Type	Select Common or CTC ACS Server Type.	
ACS Server URL	ACS Server URL.	
ACS User	User name for ACS.	
ACS Password	ACS Password.	
TR069 Auto	Enable (Disable TROCO Auto Logia	
Login	Enable/Disable TR069 Auto Login.	

b) SYSLOG

	AUTO PROVISION	SYSLOG	CONFIG	UPDATE	ACCESS	REBOOT	
> BASIC	Syslog Settings						
> NETWORK	Server Address		0.0.0				
	Server Port		514				
› VoIP	MGR Log Level SIP Log Level		None 🔻				
> INTERCOM	Enable Syslog			Apply			
> SAFEGUARDING							
* MAINTENANCE	Web Capture Start		Stop				
> LOGOUT							

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 will 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 the normal but significant condition.

Level 6: Informational; It is the normal daily messages.

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

Field Name	Explanation
Syslog settings	
Server Address	System log server IP address.
Server port	System log server port.
MGR log level	Set the level of MGR log.
SIP log level	Set the level of SIP log.
Enable syslog	Enable or disable system log.
Web Capture	
Start	Capture a packet stream from the equipment. This is normally used to troubleshoot
Start	problems.
Stop	Stop capturing the packet stream

c) CONFIG

	AUTO PROVISION SYSLOG CONFIG UPDATE ACCESS REBOOT	
> BASIC	Save Configuration	
> NETWORK	Click "Save" button to save the configuration files!	
> VoIP	Backup Configuration	-
> INTERCOM	Save all network and VoIP settings!	
	Right Click here to Save as Config File(.txt)	
> SAFEGUARDING	Right Click here to Save as Config File(.xml)	
• MAINTENANCE	Clear Configuration Click the "Clear" button to clear the configuration files!	
› LOGOUT	Clear ETC File	

Field Name	Explanation
Save	Save the current equipment configuration. Clicking this saves all configuration
Configuration	changes and makes them effective immediately.
Backup	Save the equipment configuration to a txt or xml file. Please note to Right click on the
Configuration	choice and then choose "Save Link As."
	Logged in as Admin, this will restore factory default and remove all configuration
Clear	information.
Configuration	Logged in as Guest, this will reset all configuration information except for VoIP
	accounts (SIP1-6 and IAX2) and version number.

d) UPDATE

This page allows uploading configuration files to the equipment.

		SYSLOG	CONFIG	UPDATE	ACCESS	REBOOT
> BASIC	Web Update					
> NETWORK		Select File:		Browse (*.z,*.tx	t,*.xml,*.au,*.wav)	Update
> VoIP						

Field Name	Explanation
Web Update	Browse to the config file, and press Update to load it to the equipment. Various types of files can be loaded here including firmware, ring tones, local phonebook and config files in either text or xml format.

e) ACCESS

Through this page, user can add or remove users depends on their needs and can modify existing user permission.

	AUTO PROVISION	SYSLOG		UPDATE	ACCESS	REBOOT	
> BASIC							
	User Settings						
> NETWORK		User			User Level		
		admin			Root		
> VoIP		guest			General		
	Add User						
> INTERCOM	User						
	Password						
> SAFEGUARDING	Confirm		5 			Apply	
MAINTENANCE	User Level		Root 🔻				
	User Management						
> LOGOUT					-		
	admin 🔻		De	elete Modify			

Field Name	Explanation
User Settings	
User	shows the current user name
User level	Show the user level; admin user can modify the configuration. General user can only
User level	read the configuration.
Add User	
User	Set User Account name
Password	Set the password
Confirm	Confirm the password
	There are two levels. Root user can modify the configuration. General user can only
User level	read the configuration.
User Managem	ent

Select the account and click Modify to modify the selected account. Click Delete to delete the selected account. A General user can only add another General user.

f) **REBOOT**

Some configuration modifications require a reboot to become effective. Clicking the Reboot button will lead to reboot immediately.

Note: Be sure to save the configuration before rebooting.

(7) LOGOUT

	(
> BASIC	Logout	
> NETWORK		Click "Logout" button to logout the system!
> ₋ VoIP		
> INTERCOM		
> SAFEGUARDING		
> MAINTENANCE		
> LOGOUT		

Click <Logout> from the web to exit. Users need to enter their user name and password again when visit next time.

E. Appendix

1. Technical parameters

Communication protocol		SIP 2.0(RFC-3261)		
Main chipset		Broadcom		
	Protocols	RTP/SRTP		
	Decoding	G.729、G.723、G.711、G.722、G.726		
Speech flow	Audio amplifier	2.5W		
	Volume control	Adjustable		
	Full duplex speakerphone	Support (AEC)		
	DSS key	One or Two (PH2.0 port)		
	Indicating lamp	Three (PH2.0 port)		
	МІС	One (XH2.54 port)		
	Speaker	One (XH2.54 port)		
Port	An external active speaker	One (3.5mm port)		
PUIL	recording output	One (3.5mm port)		
	Short circuit input	Two (3.5mm port)		
	Short circuit output	Two (3.5mm port)		
	WAN port	10/100BASE-TX s Auto-MDIX, RJ-45		
	LAN port	10/100BASE-TX s Auto-MDIX, RJ-45		
power supply	v mode	9V~16V/1A DC or POE		
Cables		CAT5 or better		
working tem	perature	-40°C to 70°C		
working humidity		10% - 95%		
storage temperature		-40°C to 70°C		
overall dimension		195x120x39mm		
Package dime	ensions	260x165x62mm		
Package weig	ht	0.85KG		

2. Basic functions

• 2 SIP line

- Full-duplex speakerphone
- Intelligent DSS Keys(Speed dial)
- Wall-mount installation
- 2 embedded short circuit input interfaces
- 2 embedded short circuit output interfaces. Support 4 controlled events: remote DTMF; remote server's commands; interaction with short circuit input; talking status
- Output interface for active speaker
- Audio record output interface
- External Power Supply
- Multicast
- All in ONE: Radio and intercom, intelligent security function
- Industrial standard certifications: IP65, IK10, CE/FCC

3. Schematic diagram



4. The radio terminal configuration notice

♦ How to avoid an incoherency sound when the broadcast playing?

When the terminal use as broadcast, the speaker is loud, if not set mute for microphone, the AEC (echo cancellation) of equipment will be activated, which leads the sound incoherence. In order to avoid such circumstance, when the equipment turn to use as radio should be set as intercom mode, and activate the intercom mute, so as to ensure the broadcast quality.

	FUNCTION KEY AUDIO FEATURE	MCAST Action URL	
	Feature Settings		
> BASIC	DND (Do Not Disturb)	Ban Outgoing	
	Enable Intercom Mute	Enable Intercom Tone	
> NETWORK	Enable Auto Answer Lines and IP Call 🔻	Auto Answer Timeout	0 (0~60s)
	No Answer Handdown 📃	No Ans. Handdown Time	30 (1~60s)
> VoIP	Dial Fixed Length to Send 🛛 🗐	Send length	11
	Enable Speed Dial Handdown Enable 💌	Dial Number Voice Play	Disable 💌
INTERCOM	Use Function Key to Answer Disable 🔻	Status Led Reuse Mode	Disable 🔻
	Hot Key Dial Mode Select Main-Secondary 🔻	Call Switched Time	16 (5~50s)
> SAFEGUARDING	Day Start Time 06:00 (00:00~23:59) Day End Time	18:00 (00:00~23:59
	Description IP Intercom		
MAINTENANCE		Apply	

 \diamond How to improve broadcasting tone quality?

In order to obtain better broadcast quality, recommend the use of the HD (G.722) mode for broadcast.

Voice bandwidth will be by the narrow width (G.722) of 4 KHz, is extended to broadband (G.722)7 KHz, when combined with the active speaker, the effect will be better.

	FUNCTION KEY AUDI	0 FEATURE	MCAST Action	JRL
> BASIC	Audio Settings			
> NETWORK	First Codec	G.711A 🔻	Second Codec	G.711U 🔻
	Third Codec	G.722 T	Fourth Codec	G.729AB 🔻
> VoIP	DTMF Payload Type	101 (96~127)	Default Ring Type	Туре 1 🔻
and a second second	G.729AB Payload Length	20ms 🔻	Tone Standard	China 🔹
> INTERCOM	G.722 Timestamps	160/20ms 🔻	G.723.1 Bit Rate	6.3kb/s 🔻
	Enable VAD			
> SAFEGUARDING				

5. The other function settings

	FUNCTION KEY AUDIO	FEATURE	MCAST Action URL	
	Feature Settings			
BASIC	DND (Do Not Disturb)		Ban Outgoing	
	Enable Intercom Mute		Enable Intercom Tone	I
NETWORK	Enable Auto Answer	Lines and IP Call 🔻	Auto Answer Timeout	0 (0~60s)
	No Answer Handdown		No Ans. Handdown Time	30 (1~60s)
VoIP	Dial Fixed Length to Send		Send length	11
	Enable Speed Dial Handdown	Enable 🔻	Dial Number Voice Play	Disable 🔻
INTERCOM	Use Function Key to Answer	Disable 🔻	Status Led Reuse Mode	Disable 🔻
	Hot Key Dial Mode Select	Main-Secondary 🔻	Call Switched Time	16 (5~50s)
SAFEGUARDING	Day Start Time	<mark>06:00</mark> (00:00~23:59)	Day End Time	18:00 (00:00~23:59
	Description	IP Intercom		
MAINTENANCE			Apply	

1) Status Led reuse mode

Enable the function, the registered status indicator will reuse the call instructions function, which means the LED will flashes in the call state.

2) Dialing tone prompt

Enable the function; operating digital keyboard will have corresponding key tone of voice.

3) Call switching time

This function is used to define the speed dial key to call, call switching from number 1 to number 2 time interval.