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.

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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:

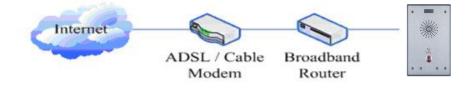
1. Connecting the power supply and the network

(1) Connecting network

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

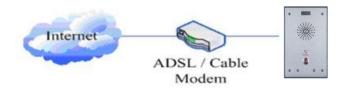
Broadband Router

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.

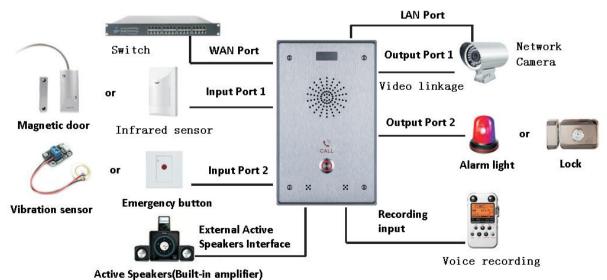


• No Broadband Router

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.

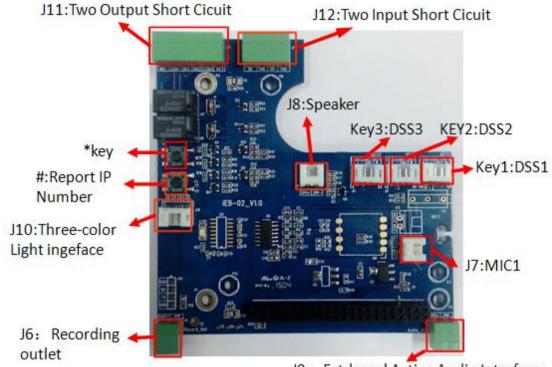


- (2) Interface specification
- a) Schematic diagram of peripherals



b) Interface specification

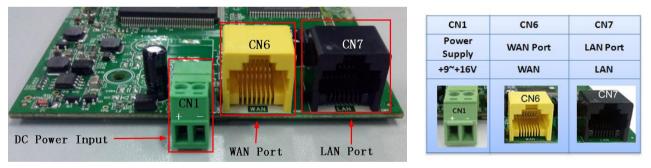
• Expansion board interface



J9: Extdernal Active Audio Interface

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

• Motherboard interface



[Notice]LAN port Support two modes:

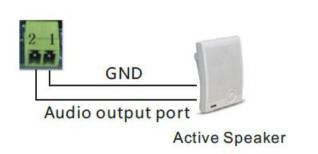
- Routing mode (It can assign IP Address to LAN port the via the DHCP for each connected device)
- ♦ Bridge Mode (LAN port and WAN port are in the same network segment)

• Port description

Port	Description	Feature	Picture
CN1	DC Power Input port	Input Range:+9~+16V DC (Notice: Plus-n-Minus connection of the Power)	
CN6	WAN port	10M/100M Adaptive Ethernet port, connected to the network	CN6
CN7	LAN Port	10M/100M Adaptive Ethernet port, connected to the computer (which can be configured to routing mode, or to bridge mode)	CN7
J ð	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)	AA
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	AAAAA
J12	Short circuit Input detection Port	Used to connect to infrared detector, magnetic switch, vibration sensor and other input devices	8888
J10	Status indicator light port	For an external status instructions (calling, ringing, network/registered)	

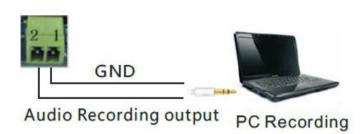
- c) Port instructions
- External Active Speakers

J9: External Activ	e Speakers Port
2	1
SPK+	GND
Audio output port	Ground Line
2	1



• Audio Recording output port

2	1
Audio+	GND
Audio Recording output port	Ground Line



OM

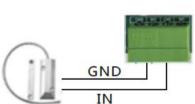
Alarm light

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

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

• Two short circuit input port

	2: Short cir ort2(IN2)		ort1(IN1)
4	3	2	1
GND	IN2	GND	IN1
Input Port2	Input Port2	Input Port1	Input Port1
Input Port2	-	2 1	Input Port1

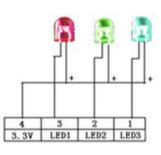


Door magnetic switch

12V DC Power Supply

• Status lamp interface

	J10: Status la	mp interrace	-
4	3	2	1
3.3V	LED1	LED2	LED3
Power supply	Network	Call	Ringing
	Network	Call	Ringir
	4	1	



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).

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
BASIC	Network			
IETWORK	WAN		LAN	
and the second	Connection Mode	DHCP	IP Address	192,168,10,1
/oIP	MAC Address	00:d8:4a:00:02:ba	DHCP Service	Enabled
041	IP Address	172,18.2.112	Bridge Mode	Disabled
TERCOM	IP Gateway	172,18.1.1		
	Accounts			
AFEGUARDING	SIP Line 1	@:5060	Unappl	lied
IAINTENANCE	SIP Line 2	@:5060	Unappl	ied
одонт				

Status	
Field Name	Explanation
Network	Shows the configuration information for WAN and LAN port, including connection mode of WAN port (Static, DHCP, PPPoE),MAC address, IP address of WAN port and 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 server.

b) WIZARD

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

Wizard	Wizard				
Field Name	Explanation				
Select the appropriate 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				
DHCP mode:	from a 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	255.255.2					
IP Gateway	192.168.1.1					
DNS Domain						
Primary DNS	202.96.134.133					
Secondary DNS	202.96.128.68					
	Back					
Static IP						
addraaa	Please enter the Static IP address					
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					
DNS Domain	address cannot be resolved, the domain equipment to resolve in the domain					
	name.					
Primary DNS	Please enter the Primary DNS server address					
Secondary						
Please enter the Secondary DNS server address						
DNS						

Field Name	Explanation					
Quick SIP Settings						
Quick SIP Settings						
Display Name	603					
Server Address	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.						

After selecting DHCP 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	Di	uration	Peer Calls		Туре
	February 26 14:	01 7	second(s)	172.18.2.40@1	72.18.2.40	Received
> ¥oI₽	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 LOG	LANGUAGE	TIME&DATE
• BASIC	Language				
> NETWORK	Language Select	ion	English 🔻		
› VoIP				Apply	
> INTERCOM					

e) TIME&DATE

	STATUS	WIZARD	CALL LOG	LANGUAGE	TIME&DATE	
	System Current Time					
> BASIC	2016/02/26 16:5		Cattings			
> NETWORK	Simple Network Time Enable SNTP	Ø) seconds			
→ ¥oIP	Enable DHCP Tim Primary Server	ne 🔲 O.pool.r	ntp.org			
	Secondary Serve Timezone			igqing,Hong Kong,U	rumqi 🔹	
> INTERCOM	Resync Period 12-Hour Clock	60	second(s)			
> SAFEGUARDING				Apply		

	Daylight Saving Time	Settings	
BASIC	Enable Offset	60 minutes(s)	
NETWORK	Month Week	March •	October •
/oIP	Day Hour	Sunday 🔻	Sunday 🔻
INTERCOM	Minute	0 Apply	0
SAFEGUARDING	Manual Time Settings		
1AINTENANCE	Year Month		
.OGOUT	Day Hour		
	Minute	Apply	

TIME&DATE				
Field Name	Explanation			
System Current Time				
Display the curre	ent 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 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:l	pa			
<u>v</u>	MAC Timestamp	20150428				
> VoIP	WAN Settings					
	- Obtain DNS Server Automaticall	ly Enabled 🔻				
INTERCOM	Static IP O	DHCP		PPPoE 🔘		
	State IP	DHCP @		FFFUC U		
> SAFEGUARDING			Apply			
> MAINTENANCE	802.1X Settings					
	User	admin				
> LOGOUT	Password	••••				
	Enable 802.1X					
			Apply			
MAINTENANCE	Service Port Settings					
7 MAINTENANCE	Web Server Type	HTTP 🔻				
› LOGOUT	HTTP Port	80				
1 200001	HTTPS Port	443				
	Telnet Port	23				
	RTP Port Range Start	10000				
	RTP Port Quantity	200				
]	Apply			

WAN			
Field Name	Explanation		
WAN Status			
Active IP Ac	ldress	172.18.2.193	
Current Subnet Mask		255.255.0.0	
Current IP	Gateway	172.18.1.1	
MAC Addres	55	0c:38:3e:13:3b:90	
Active IP address	The current IP a	ddress of the equipment	

Current subnet mask	The current Subnet Mask				
Current IP gateway	The current Gateway IP address				
MAC address	The MAC address of the equipment				
MAC	Get the MAC address of time.				
Timestamp	Get the MAC address of time.				
WAN Settings					
Obtain DNS	Server Automatically Enabled 💌				
Static IP 🔘	DHCP PPPoE				
	Apply				
Select the appro	priate network mode. 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 parameters are provided automatically by a DHCP server.				
PPPoE	Account and Password must be input manually. These are provided by your ISP.				
If Static IP is ch	nosen, the screen below 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	Please enter the Static IP address				
address					
Subnet mask	Please enter the Subnet Mask				
Gateway	Please enter the IP Gateway				
	Set the DNS domain suffix. When the user enter the domain name DNS				
DNS Domain	address cannot be resolved, the domain equipment to resolve in the domain				
Primary DNS	Please enter the Primary DNS server address				
Secondary DNS	Please enter the Secondary DNS server address				

Field Name	Explanation
If PPPoE is cho	sen, 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 t	he new settings, click the APPLY button. The equipment will save the new				
settings and app	oly them. If a new IP address was entered for the equipment, it must be used to				
•	ne after clicking the APPLY button.				
802.1X Settings					
User	admin				
Password	•••••				
Enable 802.1					
User	802.1X user account				
Password	802.1X password				
Enable 812.1X	Open/Close 812.1X				
Service Port Se	ettings				
Web Server	Specify Web Server Type UTTD or UTTDS				
type	Specify Web Server Type – HTTP or HTTPS				
	Port for web browser access. Default value is 80. To enhance security, change				
	this from the default. Setting this port to 0 will disable HTTP access.				
HTTP port	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				
HTTPS port	must be 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	Cat the beginning value for DTD Darte. Darte are domentically allocated				
start	Set the beginning value for RTP Ports. Ports are dynamically allocated.				
RTP port	Cat the mention mention of DTD Darts. The default is 000				
quantity	Set the maximum quantity of RTP Ports. The default is 200.				
Note:					
1) Any changes made 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 reserved.					
3) If the HTTP port is set to 0, HTTP service will be disabled.					

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

LAN				
Field Name	Explanation			
IP address	LAN static IP			
Subnet mask	LAN Subnet Mask			
	If Bridge Mode is activated, the equipment will not provide an IP address for the			
Enable bridge	LAN port. Instead, the LAN and WAN will be part of the same network. If this is			
mode	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.

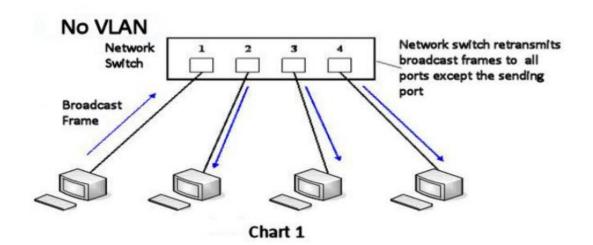
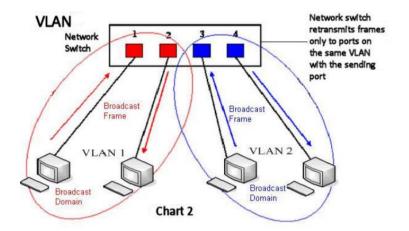


Chart 2 shows an example with two VLANs indicated by red and blue. In this example, frames broadcast from Port 1 will only go to Port 2 since Ports 3 and 4 are in a different VLAN. VLANs can be used to divide a network by restricting the transmission of broadcast frames.



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

	WAN LAN	QoS&VLAN	WEB FILTER FIREWALL	VPN SECURITY
> BASIC	Link Layer Discovery Protocol ((LLDP) Settings		
	Enable LLDP 9		Packet Interval	60 (1~3600)second(s)
NETWORK	Enable Learning Function			
> VoIP	Quality of Service (QoS) Settin	igs		
	Enable DSCP		SIP DSCP	46 (0~63)
> INTERCOM	Audio RTP DSCP	46 (0~63)		
> SAFEGUARDING	WAN Port VLAN Settings			
	Enable WAN Port VLAN		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			
an an an Arabitan Cara	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
	Enables the telephone to synchronize its VLAN data with the Network
Enable Learning	Switch. The telephone will automatically synchronize DSCP, 802.1p, and
Function	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)

Specify the value of the SIP DSCP in decimal			
Specify the value of the Audio DSCP in decimal			
Explanation			
lings			
Enchle er Dischle WAN Dert VI AN			
Enable or Disable WAN Port VLAN			
Specify the value of the WAN Port VLAN ID. Range is 0-4095			
Specify the value of the signal 8021.p priority. Range is 0-7			
Creatify the value of the value 200 in rejerity. Dense is 0.7			
Specify the value of the voice 802.1p priority. Range is 0-7			
ngs			
Follow WAN: LAN Port ID is same as WAN ID.			
Disable: Disable Port VALN			
Enable: Specify a VLAN ID for the LAN port which is different from WAN			
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 Addres	55	End I	P Address		Option	
› VoIP	Web Filter Table So Start IP Addres		End 1	P Address		Add	
> INTERCOM	Web Filter Setting						
> SAFEGUARDING	Enable Web Fil	lter 🔲		pply			
> MAINTENANCE							
> LOGOUT							

Web filter					
The Web filter is us	The Web filter is used to limit access to the equipment. When the web filter is enabled, only the				
IP addresses betwe	een the start IP and end IP can access the equipment.				
Field Name	Explanation				
Web Filter Table					
Webpage access a	Illows 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					

e) FIREWALL

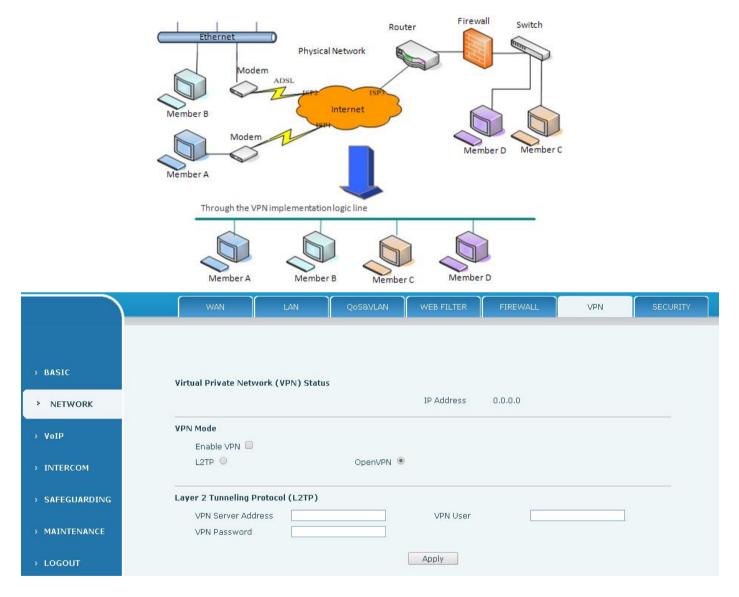
)	WAN	LAN	QoS&VLAN	WEB FILTER	FIREWALL	VPN	SECURITY
	Firewall Type						
> BASIC		Enable Input F	tules 🗖	Apply	Enable Outpu	it Rules 🔲	
• NETWORK	Firewall Input	t Rule Table					
	Index D	eny/Permit Protocol Src 4	Address Src Mas	sk Dest Addi	ress Dest Mask	Range	Port
> VoIP							
> INTERCOM	Firewall Outp		Address Src Mag	sk Dest Add	ress Dest Mask	Bassa	D - ±
	Index D	eny/Permit Protocol Src 4	Address Src Mas	sk Dest Addi	ress Dest Mask	Range	Port
> SAFEGUARDING	Firewall Setti	ngs					
	Input/Ou			Src Address			
> MAINTENANCE	Deny/Per			Dest Address			Add
› LOGOUT	Protocol Port Rani	ge more that	an 🔻 📃	Src Mask Dest Mask			
	Rule Delete O	ption					
	Input/Ou	tput Input	T	Index To Be Delete	d	[Delete

Firewall				
Firewall rules can be used to prevent unauthorized Internet users from accessing private				
networks conne	ected to this phone (input rule), or prevent unauthorized devices connected to this			
phone from acc	cessing the Internet (output rule). Each rule type supports a maximum of 10 items.			
Field Name	Explanation			
Firewall Rules	Settings			
Enable Input Rules	Enable rules limiting access from the Internet.			
Enable Output Rules	Enable rules limiting access to the Internet.			
Firewall Settin	gs			
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	Set source address. It can be a single IP address or use * as a wild card. For			
Address	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.			

Source Mask	Set the source address mask. For example: 255.255.255.255 points to one host
Source Mask	while 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
Mask	host 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		Selec	t Security File:		Browse	odate	
Vate	Delete Security File						
› VoIP		Se	lect Security File:	nttps.pem	▼ Delete		
> INTERCOM	SIP TLS Files						
> SAFEGUARDING	HTTPS Files						
• MAINTENANCE			https.pem		(4555 Bytes)		
	OpenVPN Files						
› LOGOUT							

Field Name	Explanation				
Update Security	Select the accurity file to be undeted. Click the Undete button to undete				
File	Select the security file to be updated. Click the Update button to update				
Delete Security	Oslastika sa wita fila ta ka dalata di Olislakka Dalata kaitan ta Dalata				
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 Serv	er 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)
	YE		

DTMF SIP INFO Mode	Send */# 🔻	L	ocal Port	5060	
Enable Rport		K	eep Authentication		
Enable PRACK		А	ns. With a Single Co	dec 📃	
Enable Strict Proxy		A	uto TCP		
Enable DNS SRV		U	se VPN		
Transport Protocol	UDP 🔻				
SIP Global Settings >>					
				-	
Strict Branch			Enable Group		
Registration Failure Retry Time	32	second(s)	DND Return Code	480(Temporarily Not Available)	¥
Reject Return Code	603(Decline)	•	Busy Return Code	486(Busy Here)	•

RFC Protocol Edition

RFC3261 •

DTMF Type

AUTO

۲

SIP	
Field Name	Explanation
Basic Settings (C	choose the SIP line to configured)
Status	Shows registration status. If the registration is successful will display has
Status	been 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						
User	SIP account name (Login ID).					
Authentication	SID registration personal					
password	SIP registration password.					
	Phone number assigned by VoIP service provider. Equipment will not register					
SIP user	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 Se	ettings
Proxy server	SIP proxy server IP address or URI, (This is normally the same as the SIP
address	Registrar 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 address	server is unavailable)
Backup Proxy	Paalkup SID Sonvor Part
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 Timer	If enabled, this will refresh the SIP session timer per RFC4028.
Registration	SIP re-registration time. Default is 60 seconds. If the server requests a
Expires	different time, the phone will change to that value.
Session Timeout	Refresh interval if Session Timer is enabled.
Keep Alive Type	Specifies the NAT keep alive type. If SIP Option is selected, the equipment will send SIP Option sip messages to the server every NAT Keep Alive Period. The server will 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	Set the NAT Keep Alive interval. Default is 60 seconds

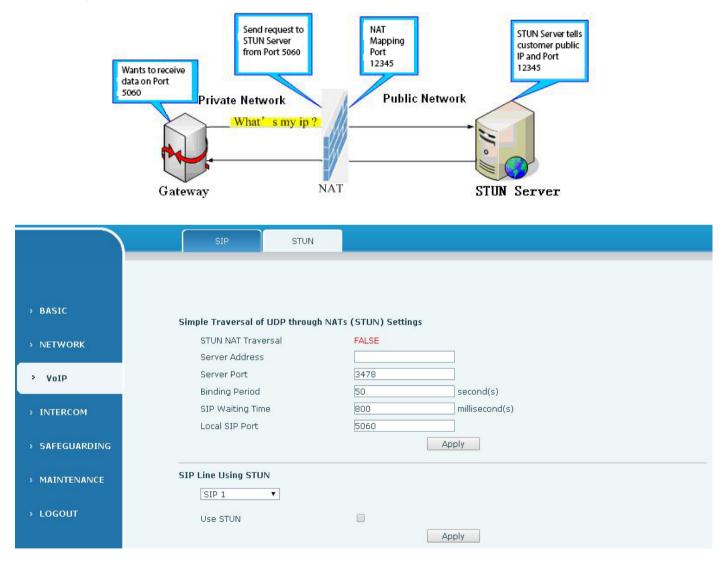
Interval	
User Agent	Set SIP User Agent value.
Server Type	Configures phone for unique requirements of selected server.
	DTMF sending mode. There are four modes:
	● In-band
	• RFC2833
DTMF Type	• SIP_INFO
	• AUTO
	Different VoIP Service providers may require different modes.
RFC Protocol	Select SIP protocol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

Field Name	Explanation				
Кеер	Enable /disable registration with authentication. It will use the last				
Authentication	authentication field which passed authentication by server. This will decrease				
Authentication	the load on the server if enabled				
Enable PRACK	Enable or disable SIP PRACK function. Default is OFF. It is suggested this be				
	used.				
Ans. With a	If enabled phone will respond to incoming calls with only one codec.				
Single Codec	If enabled phone will respond to incoming calls with only one codec.				
Enable Strict	Enables the use of strict routing. When the phone receives packets from the				
Proxy	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				
Auto TOP	messages above 1500 bytes				
Enable DNS	Enables use of DNS SRV records				
SRV					
Use VPN	Enable SIP use VPN for every line individually, not all of them				
Transport	Configuration using the transport protocol, TCP, TLS or UDP, the default is				
Protocol	UDP.				
SIP Global Settin	ngs				
	Enable Strict Branch - The value of the branch must be after"z9hG4bK" in				
Strict Branch	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				

Failure Retry	to register again after registration failure retry time. This will affect all lines					
Time						
DND Return	Specify SIP Code returned for DND. Default is 480 - Temporarily Not					
Code	Available.					
Reject Return	Specify SID Code returned for Dejected cell, Defeult is 602 Decline					
Code	Specify SIP Code returned for Rejected call. Default is 603 – Decline.					
Busy Return	Specify SID Code returned for Duoy, Default is 496 Duoy Llare					
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	Shows whether or not STUN NAT Traversal was successful.

Traversal					
Server Address	STUN Server IP address				
Server Port	STUN Server Port – Default is 3478.				
Rinding Poriod	STUN blinding period – STUN packets are sent at this interval to keep the				
Binding Period	NAT mapping active.				
SIP Waiting Time	Waiting time for SIP. This will vary depending on the network.				
Local SIP Port	Port configure the local SIP signaling				
SIP Line Using STU	N (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	t configuration of the STUN server IP and port (usually the default is 3478),				
and select the Use S	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 •	Speed Dial
/oIP	DSS Key 2	None 🔹			SIP1 T	Speed Dial
	DSS Key 3	None 🔹			SIP1 T	Speed Dial
						112

> 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 💉	Release
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
	Handfree	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	4
DSS Key 4	Multicast			SIP1 V	Speed Dial	×

DSS key type	Number	Line	Subtype	Usage
Hot Key	Fill the called party's SIP	The SIP account corresponding	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
	account or address	lines	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.

G.711A

G.711U G.722

G.723.1 G.726-32

729AB

mougi	IT THE DOO NEY CO	ingulation multicas	st calling WLD is a	13 10110103.	
Key	Type	Number 1	Number 2	Line	Subtype
DSS Key 1	Multicast 🗸		-	SIP1 V	5.711A

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

DSS key type	Number	Subtype	Usage	
Multicast	Set the host IP address and port number, the middle separated by a colon	G.711A	Narrowband speech coding (4Khz) Wideband speech coding (7Khz)	
		G./ 110		
		G.722		
		G.723.1		
		G.726-32	Narrowband speech coding (4Khz)	
		G.729AB		

\diamond operation mechanism

None

Line

Hot Key

Key Event

DSS Key 2

DSS Key 3

DSS Key 4

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.

\diamond 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 AUDI	O FEATURE	MCAST Action U	RL
	Audio Settings			
> BASIC	First Codec Third Codec	G.711A • G.722 •	Second Codec Fourth Codec	G.711U V G.729AB V
> NETWORK	DTMF Payload Type G.729AB Payload Length	101 (96~127) 20ms T	Default Ring Type Tone Standard	Type 1 V China
→ VoIP	G.722 Timestamps Enable VAD	160/20ms 🔻	G.723.1 Bit Rate	6.3kb/s 🔻
• 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 <u>5</u> (1~11)	Handsfree Hardware Speaker;	phone 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 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 Length	G.729AB Payload Length – Adjusts from 10 – 60 mSec.		
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.		
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 Set	Talk Volume Settings		
SPK Output	Set the speaker calls the volume level.		
Volume			
MIC Input	Set the MIC cells the values level		
Volume	Set the MIC calls the volume level.		
Media Volume Settings			
Broadcast	Set the broadcast the output volume level.		
Output Volume			
Signal Tone	Set the audio signal the output volume level.		
Volume			
Codec Gain Sett	Codec Gain Settings		
Hands-free Hardware		Settings Hands-free Hardware MIC Gain	
MIC Gain			
Hands-free Hardware		Settings hands-free Hardware Speakerphone Gain	
Speakerphone Gain			

c) FEATURE

	FUNCTION KEY AU	DIO FEATURE		ction URL
> BASIC	Feature Settings DND (Do Not Disturb)		Ban Outgoing	
> NETWORK	Enable Intercom Mute Enable Auto Answer	✓ Lines and IP Call ▼	Enable Intercom ⁻ Auto Answer Time	
> VoIP	No Answer Handdown Dial Fixed Length to Sen		No Ans. Handdow Send length	
• INTERCOM	Enable Speed Dial Hand Use Function Key to Ans	22	Dial Number Voice Status Led Reuse	
> SAFEGUARDING	Hot Key Dial Mode Selec Day Start Time	t Main-Secondary • 06:00 (00:00~23:59)	Call Switched Tim Day End Time	e 16 (5~50s) 18:00 (00:00~23:59)
> MAINTENANCE	Description	IP Intercom	Apply	
> LOGOUT	Block Out Settings			
		Add	Block Out	Delete

Field Name	Explanation		
Feature Settings			
DND (Do Not	DND might be disabled phone for all SIP lines, or line for SIP		
Disturb)	individually.But the outgoing calls will not be affected		
Ban Outgoing	If enabled, no outgoing calls can be made.		
Enable Intercom	If enabled, mutes incoming calls during an intercom call.		

Mute		
Field Name	Explanation	
Enable Intercom	If enabled, plays intercom ring tone to alert to an intercom call.	
Tone		
Enable Auto	Enable Auto Answer function	
Answer		
Auto Answer		
Timeout	Set Auto Answer Timeout	
No Answer		
Handdown	Enable automatically hang up when no answer	
No Answer		
Handdown Time	Configuration in a set time, automatically hang up when no answer	
Dial Fixed Length	Enable or disable dial fixed length to cond	
to Send	Enable or disable dial fixed length to send.	
Send length	The number will be sent to the server after the specified numbers of digits	
	are dialed.	
Enable Speed Dial	Enable Speed Dial Hand Up function	
Handdown		
Dial Number Voice	Configuration Open / Close Dial Number Voice Play	
Play	Configuration Open / Close Dial Number Voice Flay	
Use Function Key	Configure whether to enable the function keys, is disabled by default.	
to Answer	Configure whether to enable the function keys, is disabled by default.	
Status Led Reuse	Enable the function, the registered status indicator will reuse the call	
Mode	instructions function, which means the LED will flashes in the call state.	
	<primary secondary="">mode allow system to call primary extension first, if</primary>	
	there were no answer, it would cancel the call and then call secondary	
Hot Key Dialed	extension automatically.	
Mode Selection	<day night="">mode allow system to check the calling time is belong to Day</day>	
Mode Selection	or 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 entered, the phone will not dial any numbers beginning with 001.		
X and x are wildcards which match single digits. For example, if 4xxx or 4XXX is entered, the		
phone will not dial any 4 digit numbers beginning with 4. It will dial numbers beginning with 4		
which are longer or shorter than 4 digits.		

d) MCAST

	FUNCTION KEY AUDIO	D FEATURE	MCAST	Action URL
	MCAST Settings			
> BASIC	Priority	1 *		
	Enable Page Priority			
> NETWORK	Index/Priority	Name	Host:port	
	1][
> VoIP	2]	
	3			
INTERCOM	4			
	5][
SAFEGUARDING	6			
	7			
> MAINTENANCE	8		1	
	9			
> LOGOUT	10		1	
			debi	
			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, 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:
 - \diamond 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
 - \diamond 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:	
---------------	--

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, 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)

"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		-172		
> BASIC	Registration Su	ccess	-			
> NETWORK	Registration Dis	abled				
7 NETWORK	Registration Fa	iled				
> VoIP	Off Hook					
	On Hook					
INTERCOM	Incoming Call					
	Outgoing Call					
> SAFEGUARDING	Call Established	1				
	Call Terminated	E.				
> 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

	Input Settings			
	🔲 Input 1:		🔲 Input 2 :	
> BASIC	Trigger Mode Low Leve	el Trigger(Close Trigger) 🛛 🔻	Trigger Mode	ow Level Trigger(Close Trigger)
	Response Mode 📝 Remot	te Response	Response Mode 🗹	Remote Response
> NETWORK	Output Settings			
> VoIP	Output 1:			
	Output Level	High Level(NO:closed) 🔻	Output Duration	5 (1~600)s
INTERCOM	Output Trigger Mode	🖉 Input 1 Trigger	🔲 Input 2 Trigger	
		🗷 Remote DTMF Trigger	1234	Output Last By Duration 🔻
SAFEGUARDING		🗷 Remote SMS Trigger	ALERT=OUT1_SOS	
		Call State Trigger	Talking	T
MAINTENANCE		Emergency Key Trigger		
	Output 2:			
LOGOUT	Output Level	High Level(NO:closed) 🔻	Output Duration	5 (1~600)s
	Output Trigger Mode	🔲 Input 1 Trigger	🗹 Input 2 Trigger	
		🗹 Remote DTMF Trigger	5678	Output Last By Duration 🔻
		🗹 Remote SMS Trigger	ALERT=OUT2_SOS	
		Call State Trigger	Talking	•
		Emergency Key Trigger		
SAFEGUARDING				
· JAPEGUARDING	Server & Trigger Ring Type Setti	ings		
MAINTENANCE	Server Address			
THATLEMANCE	Input 1 Trigger Ring		Input 2 Trigger Ring	default 🔻
LOGOUT	Remote DTMF Trigger Ring		Remote SMS Trigger Rin	ng default 🔻
	Alarm Ring Duration	5 (1~600)s		
		Apt	bly	

Security Settin	gs
Field Name	Explanation
Input settings	
Input 1	Open /Close Input port1
	When choosing the low level trigger (closed trigger), detect the input port 1
Trioren Marda	(low level) closed trigger.
Trigger Mode	When choosing the high level trigger (disconnected trigger), detect the input
	port 1 (high level) disconnected trigger.
Response	Onen (Class Innut north the Demote Despenses
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 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	1		
Output Setting	S			
Output 1/2	Open/close, Output 1/Output 2			
	When choos	sing the low level trigger (NO: normally open), when meet the		
	trigger cond	lition, trigger the NO port disconnected.		
Output Level	When choos	sing the high level trigger (NO: normally close), when meet the		
	trigger cond	lition, trigger the NO port close.		
Output Duration	Changes in	port, the duration of. The default is 5 seconds.		
Output Trigger I	Mode: There	are many kinds of trigger modes, multiple choices.		
Input port1	When the in	put port1 meet to trigger condition, the output port1 will trigger(The		
trigger	Port level tir	ne change, By < Output Duration > control)		
Input port2	When the input port2 meet to trigger condition, the output port2 will trigger(The			
trigger	Port level tir	ne change, By < Output Duration > control)		
	By duration	Received the terminal equipment to send the DTMF password, if correct, which triggers the corresponding output port (The Port level time change, By < Output Duration > control)		
Remote DTMF trigger	By Calling State	During the call, receive the terminal equipment to send the DTMF password, if correct, which triggers the corresponding output port (The 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	e device or server to send instructions to ALERT=[instructions], if		
trigger	correct, whi	ch triggers the corresponding output port		
Call state trigger	The port output continuous time synchronization and trigger state changes, including the trigger conditions: 1, call; 2, call and singing; 3, singing; three models. (for example: the call trigger output port, will be in conversation state continued to output the corresponding level)			
Emergency key trigger	When the emergency call button to trigger the equipment shell, which triggers the corresponding output port(after the end of the call, port to return the			
Server & Trigger Ring Type Settings				
Server & Trigge				
Server Address		gure remote response server address(including remote response		
Input 1 trigger ri	Wher	server address and tamper alarm server address) When the input port 1 triggering condition is satisfied, the		
		sponding ring tone or alarm		

Input 2 trigger ring	When the input port 2 triggering condition is satisfied, the
	corresponding ring tone or alarm
Remote DTMF trigger	When received the remote DTMF command, whether to output the
ring	ringtone
Remote SMS trigger	When receiving the remote SMS instructions, whether to output the
ring	ringtone
Alarm Ring Duration	duration of alarm ring(not including tamper alarm)

(6) MAINTENANCE

a) AUTO PROVISION

	AUTO PROVISION SYSLOG	CONFIG UPDATE	ACCESS	REBOOT	
> BASIC	Auto Provision Settings Current Config Version				
• NETWORK	Common Config Version CPE Serial Number	00000000000000000000000000000000000000	02ba		
> VoIP	User Password				
> INTERCOM	Config Encryption Key Common Config Encryption Key				
> SAFEGUARDING	Save Auto Provision Information	0			
• MAINTENANCE	DHCP Option Settings >> Plug and Play (PnP) Settings >>				
> LOGOUT	Phone Flash Settings >>				
	TR069 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	Auto Provision Settings			
	Show the current config file's version. If the version of configuration			
Current Config	downloaded is higher than this, the configuration will be upgraded. If the			
Version	endpoints confirm the 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			
Common	this configuration is the same, the auto provision will stop. If the endpoints			
Config Version	confirm the configuration by the Digest method, the configuration will not be			
	upgraded unless it differs from the current configuration.			
CPE Serial	Serial number of the equipment			

Number	
Lloor	Username for configuration server. Used for FTP/HTTP/HTTPS. If this is blank
User	the phone will use anonymous
Password	Password for configuration server. Used for FTP/HTTP/HTTPS.
Config	Encyration kew far the configuration file
Encryption Key	Encryption key for the configuration file

Field Name	Explanation			
Common				
Config	Encryption key for common configuration file			
Encryption Key				
Save Auto	Save the auto provision uppersone and peopulard in the phone until the conver			
Provision	Save the auto provision username and password in the phone until the server			
Information	url changes			
DHCP Option S	ettings			
DHCP Option	The equipment supports configuration from Option 43, Option 66, or a Custom			
Setting	DHCP option. It may also be disabled.			
Custom DHCP	Custom option number. Must be from 128 to 254.			
Option				
Plug and Play(F	PnP)Settings			
	If this is enabled, the equipment will send SIP SUBSCRIBE messages to a			
Enable PnP	multicast 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 Se	ettings			
Server	Set FTP/TFTP/HTTP server IP address for auto update. The address can be an			
Address	IP address or Domain name with subdirectory.			
Config File	Specify configuration file name. The equipment will 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 – 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		Enable or disable TR069 Warning Tone	
Warning Tor	ne		
ACS Ser	ver		
Туре		Select Common or CTC ACS Server Type.	
ACS Ser	ver	ACS Server URL.	
URL			
ACS User		User name for ACS.	
ACS Passwo	ord	ACS Password.	
TR069 Auto Login Enable/Disable TR069 Auto Login.			
		Enable/Disable TRuos 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	None 🔻			
	SIP Log Level	None 🔻			
> INTERCOM	Enable Syslog				
			Apply		
> SAFEGUARDING	Web Capture				
• MAINTENANCE	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 problems.	
Stop	Stop capturing the packet stream	

c) CONFIG

	AUTO PROVISION	SYSLOG	CONFIG	UPDATE	ACCESS	REBOOT	
> BASIC	Save Configuration						
> NETWORK			Click "Save" butto	n to save the config Save	juration files!		
› VoIP	Backup Configuratio	n					
> INTERCOM			Save all ne	twork and VoIP set	tings!		
				e to Save as Config			
> SAFEGUARDING			Right Click here	e to Save as Config	File(.xml)		
MAINTENANCE	Clear Configuration		Click the "Clear" but	ton to clear the con	figuration files!		
› LOGOUT			С	lear ETC File 🗖 Clear			

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
Configuration	on the choice and then choose "Save Link As."
	Logged in as Admin, this will restore factory default and remove all
Clear	configuration 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 COM	IFIG UPDATE	ACCESS	REBOOT
> BASIC	Web Update				
• NETWORK		lect File:	Browse (*.z,*.t	xt,*.xml,*.au,*.wav) [Update
› VoIP					

Field Name	Explanation
	Browse to the config file, and press Update to load it to the equipment.
Web Update	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.

		SYSLOG	CONFIG	UPDATE	ACCESS	REBOOT	
> BASIC	User Settings						
> NETWORK	oser settings	User			User Level		
		admin			Root		
> VoIP		guest			General		
> INTERCOM	Add User User						
> SAFEGUARDING	Password Confirm					Apply	
• MAINTENANCE	User Level		Root 🔻				
› LOGOUT	User Management		D	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
User level	can only 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			
User level	only read the configuration.			
User Management				
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

	1	
> 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

Communicati	on 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)
	MIC	One (XH2.54 port)
	Speaker	One (XH2.54 port)
Port	An external active speaker	One (3.5mm port)
FOIL	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
		10/100BASE-TX s Auto-MDIX, RJ-45
power supply	^v mode	9V~16V/1A DC or POE
Cables		CAT5 or better
working temp	perature	-40°C to 70°C
working hum	idity	10% - 95%
storage temp	erature	-40°C to 70°C
overall dimen	sion	195x120x39mm
Package dim	ensions	260x165x62mm

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	
R. Construction and the second	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	

♦ 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 U	IRL
> BASIC	Audio Settings			
> NETWORK	First Codec	G.711A 🔻	Second Codec	G.711U 🔻
	Third Codec	G.722 🔻	Fourth Codec	G.729AB 🔻
> VoIP	DTMF Payload Type	101 (96~127)	Default Ring Type	Type 1 🔻
	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	
BASIC	Feature Settings		Rap Outpoing	
	DND (Do Not Disturb) Enable Intercom Mute		Ban Outgoing Enable Intercom Tone	
NETWORK				
NET HONK	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	

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.