DP20 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:

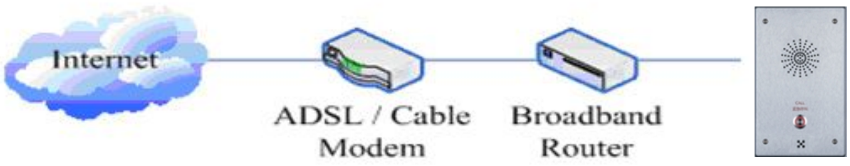
1. Connecting the power supply and the network

(1) Connecting network

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

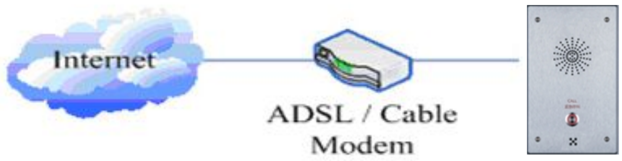
- **Broadband Router**

  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.



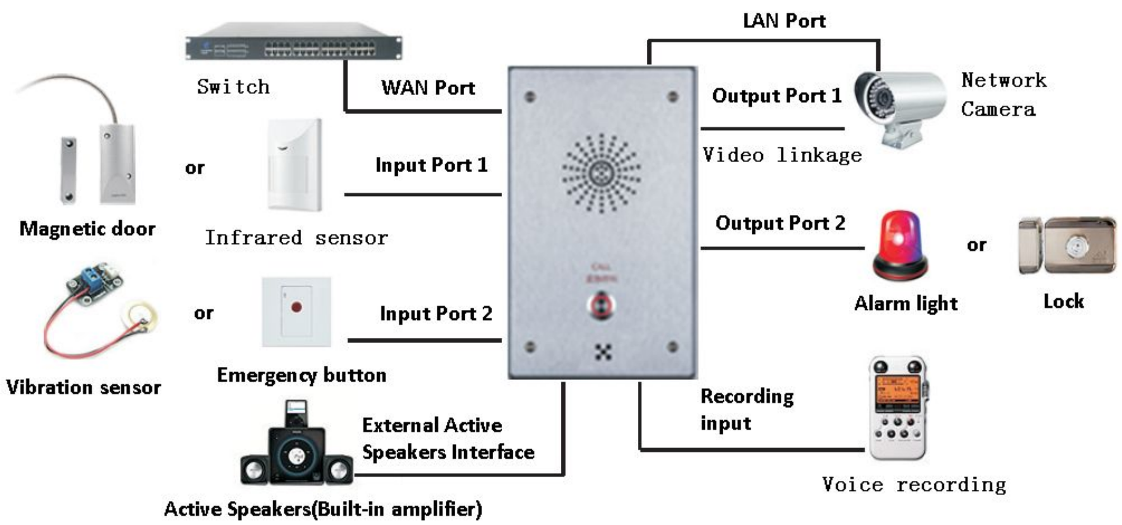
- **No Broadband Router**

  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.



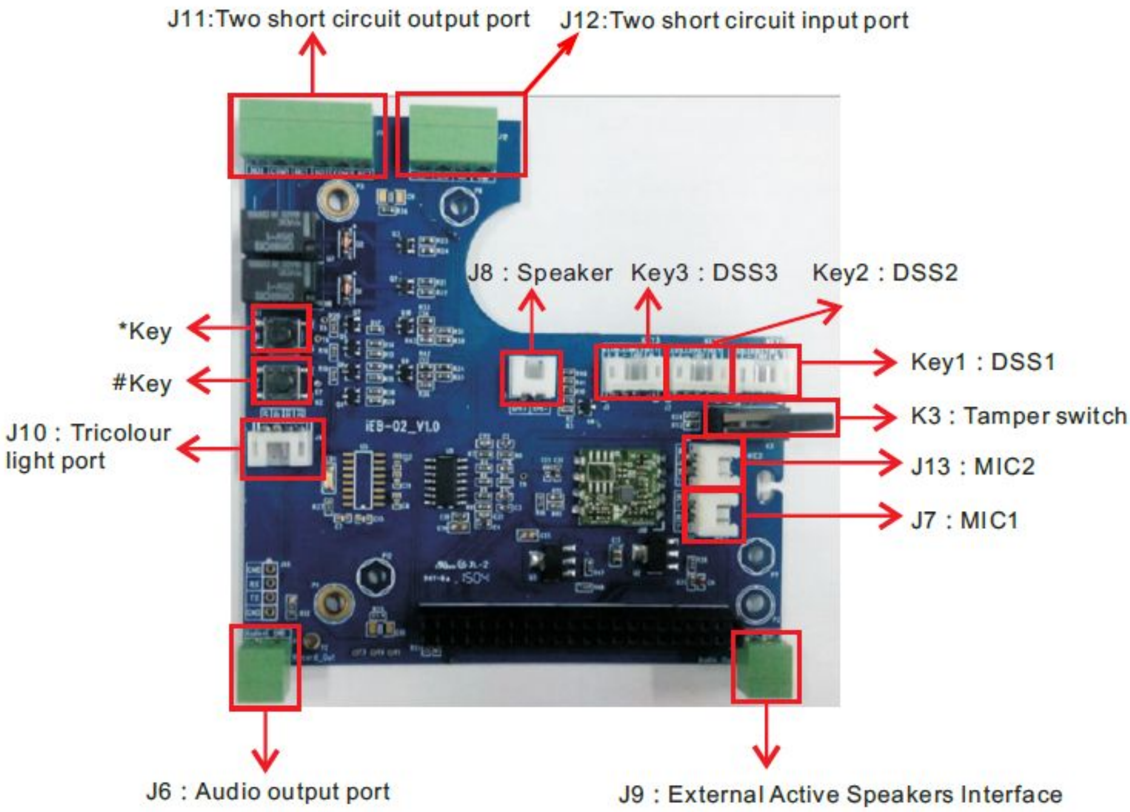
(2)Interface specification

a) Schematic diagram of peripherals



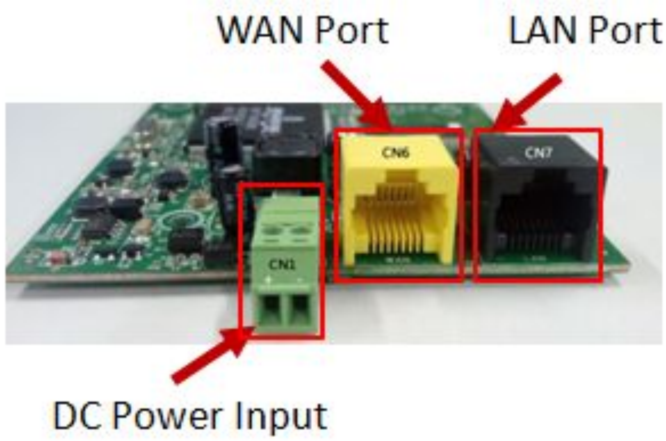
b) Interface specification

- **Expansion board interface**



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

- **motherboard interface**



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

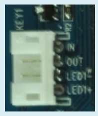
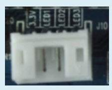
[Notice]LAN port Support two modes:

✧ **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**

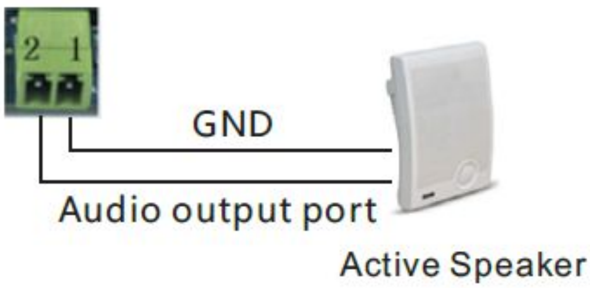
Port	Description	Feature	Picture
CN1	DC Power Input port	Input Range:+9~+16V DC (Notice: Plus-n-Minus connection of the Power)	

CN6	**WAN port**	**10M/100M Adaptive Ethernet port, connected to the network**	
CN7	**LAN Port**	**10M/100M Adaptive Ethernet port, connected to the computer(which can be configured to routing mode, or to bridge mode)**	
J9	**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)**	
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)**	
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**	
K3	**Tamper switch**	**To prevent the remove of host.** **Need to be reset by serve or web after the alarm ring.**	
J10	**Status indicator light port**	**For an external status instructions (calling, ringing, network/registered)**	

c) Port instructions

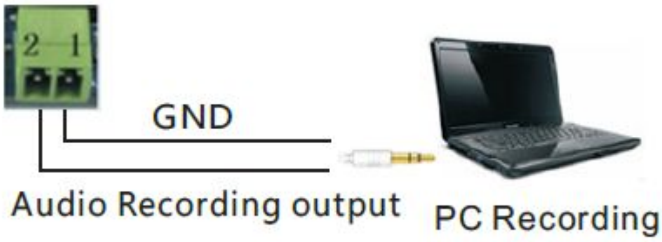
- **External Active Speakers**

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



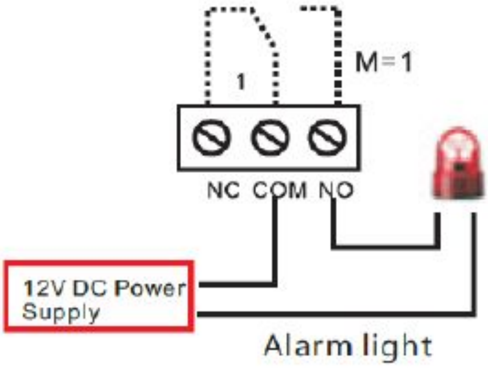
- **Audio Recording output port**

J6: Audio Recording output port	
2	1
Audio+	GND
Audio Recording output port	Ground Line
	



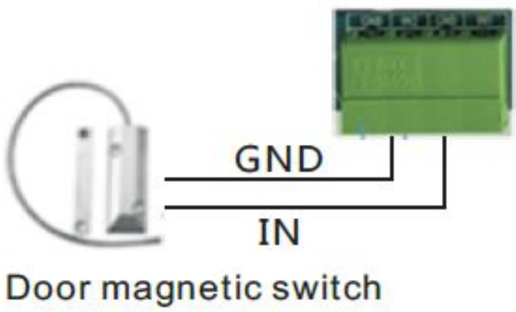
- **Two short circuit output port**
- ➢ **NO: Under the idle state is disconnected (normally open);**
- ➢ **COM: Contactor of the Relay (middle);**
- ➢ **NC: Under the idle state is connected (normally close);**

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



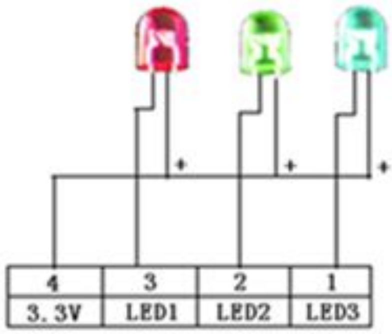
- **Two short circuit input port**

J12: Short circuit Input Port			
Input Port2(IN2)		Input Port1(IN1)	
4	3	2	1
GND	IN2	GND	IN1
Input Port2	Input Port2	Input Port1	Input Port1
			



- **Status lamp interface**

J10: Status lamp interface			
4	3	2	1
3.3V	LED1	LED2	LED3
Power supply	Network	Call	Ringing
			

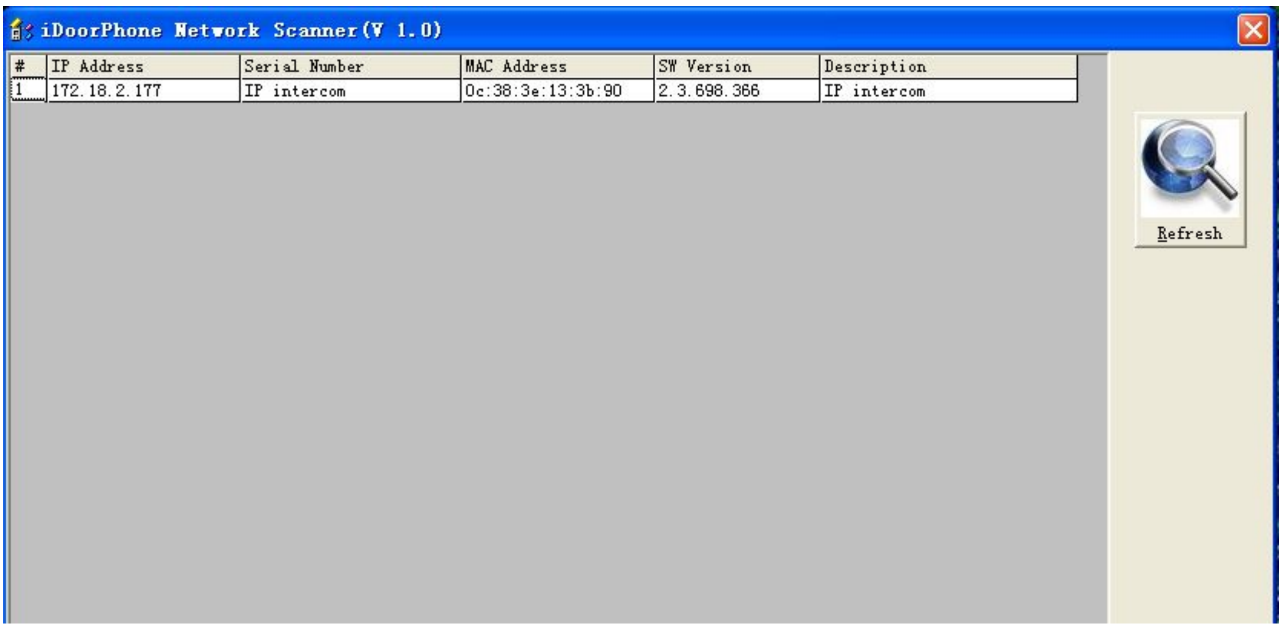


2. Quick Setting

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);



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.

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 or IAX2.

- **Default user with general level:**
  - **Username: guest**

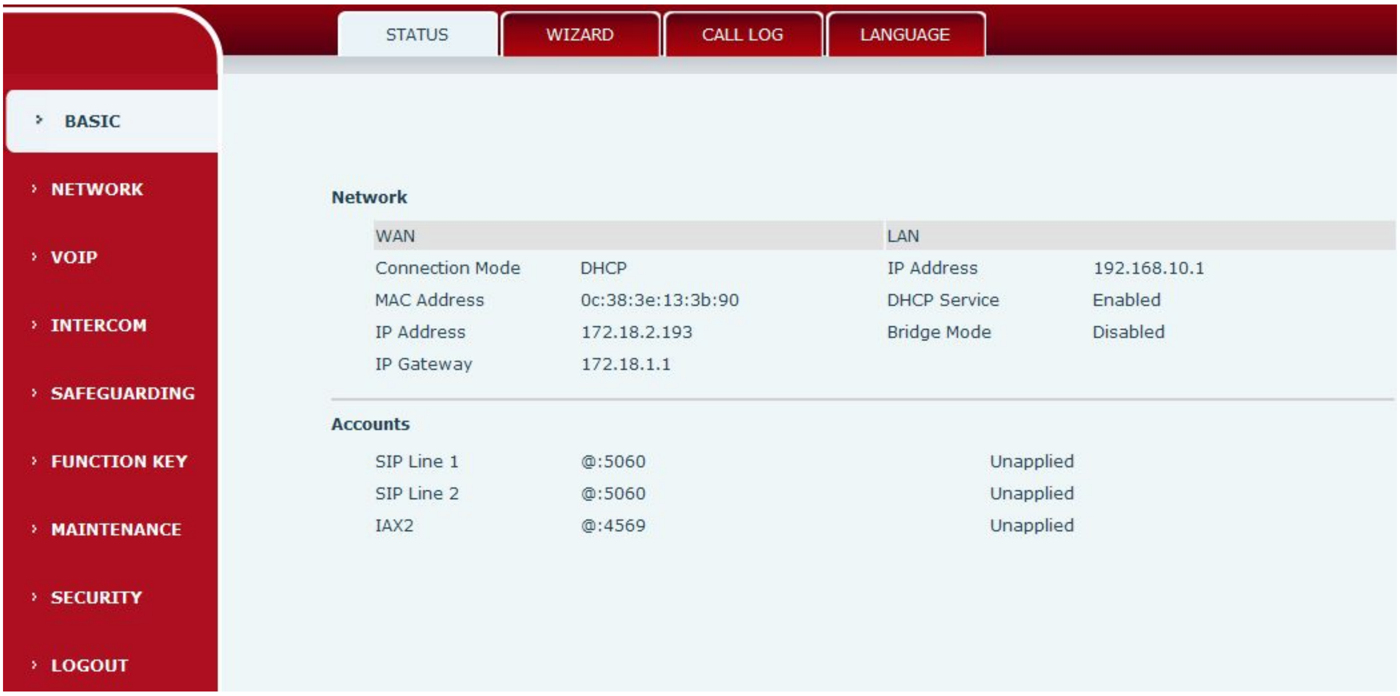
- Password: guest

- **Default user with root level:**
  - **Username: admin**
  - **Password: admin**

3. Configuration via WEB

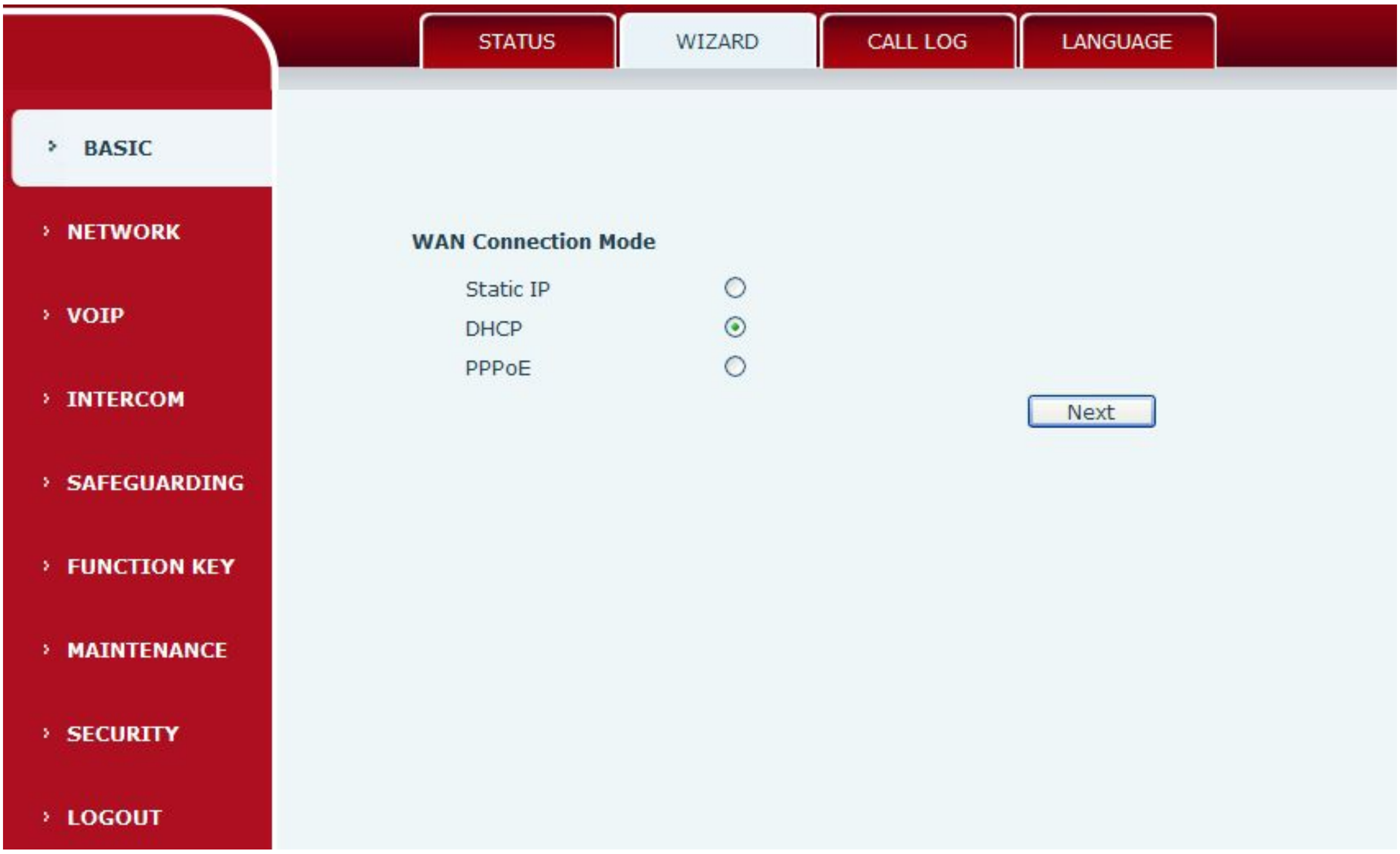
(1)BASIC

a) STATUS



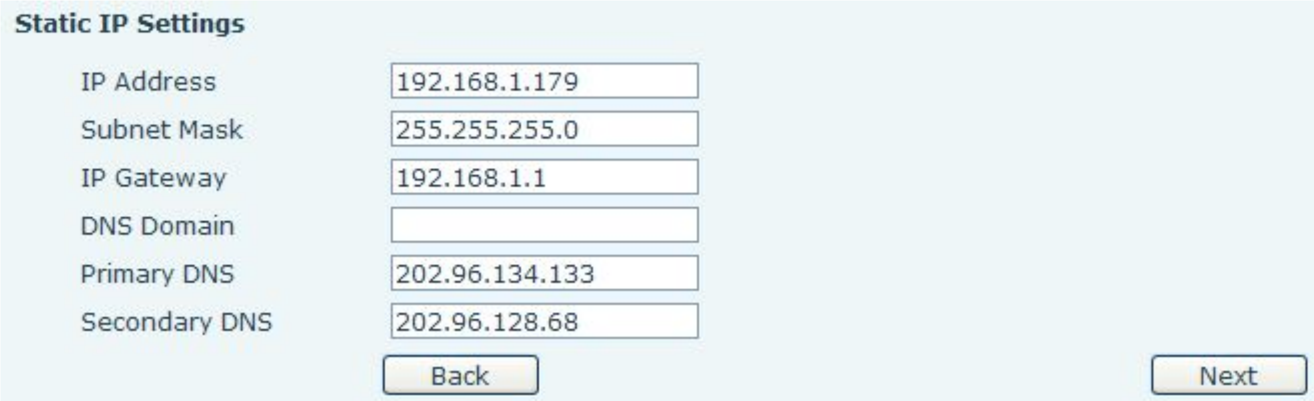
Status	
Field Name	**Explanation**
Network	**Shows the configuration information for WAN and LAN port, including connection mode of WAN port (Static, DHCP, PPPoE),MAC address, IP address of WAN port and LAN port, DHCP server, status for LAN port (ENABLED or DISABLED).**
Accounts	**Shows the phone numbers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

b) 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.**
DHCP mode：	**In this mode, network parameter information will be obtained automatically 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.	

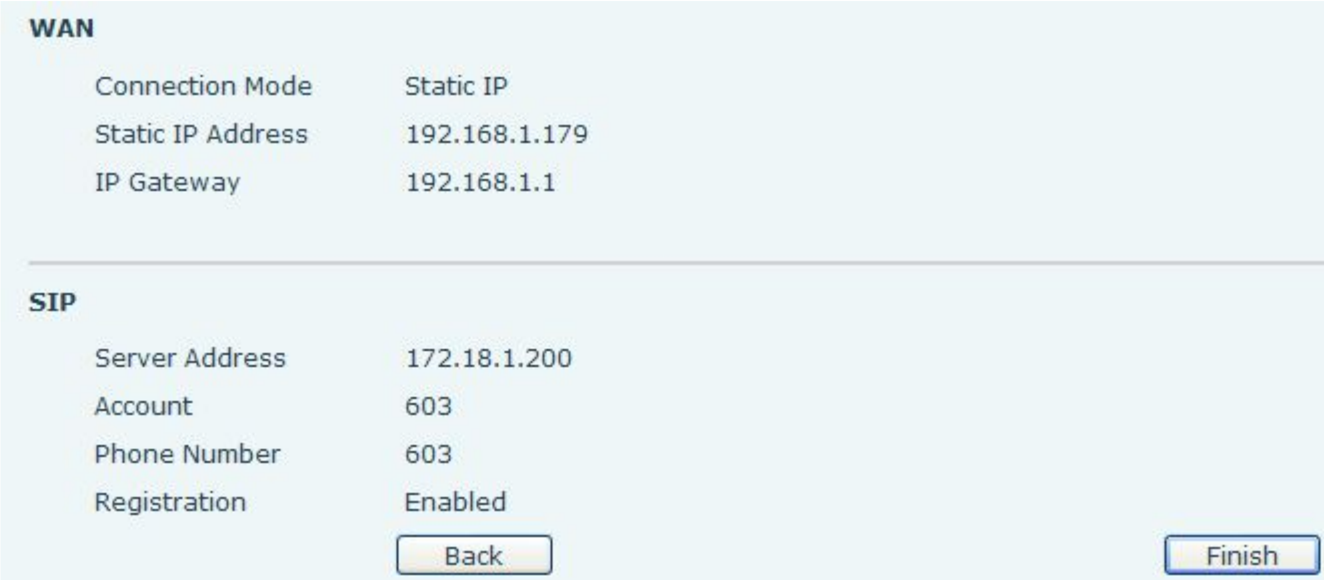
Field Name	Explanation



Static IP address	**Please enter the Static IP address**
Subnet Mask	**Please enter the Subnet Mask**
IP Gateway	**Please enter the IP Gateway**
DNS Domain	**Set the DNS domain 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**
Quick SIP Settings	

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 Registration	**Submits registration information. Normally checked**

Field Name	**Explanation**
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.

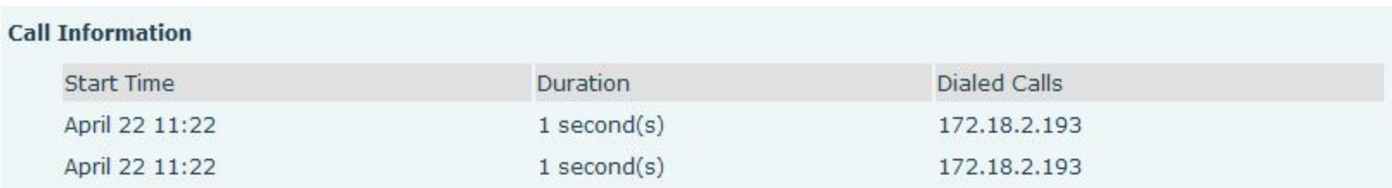
If PPPoE is selected, this screen will appear. Enter the information provided by the ISP. Click Next to go to Quick SIP Setting. Click Back to return to the Wizard screen.

Service Name	**PPPoE Service name, Usually the default value.**
User	**ADSL user account**
Password	**ADSL password**

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

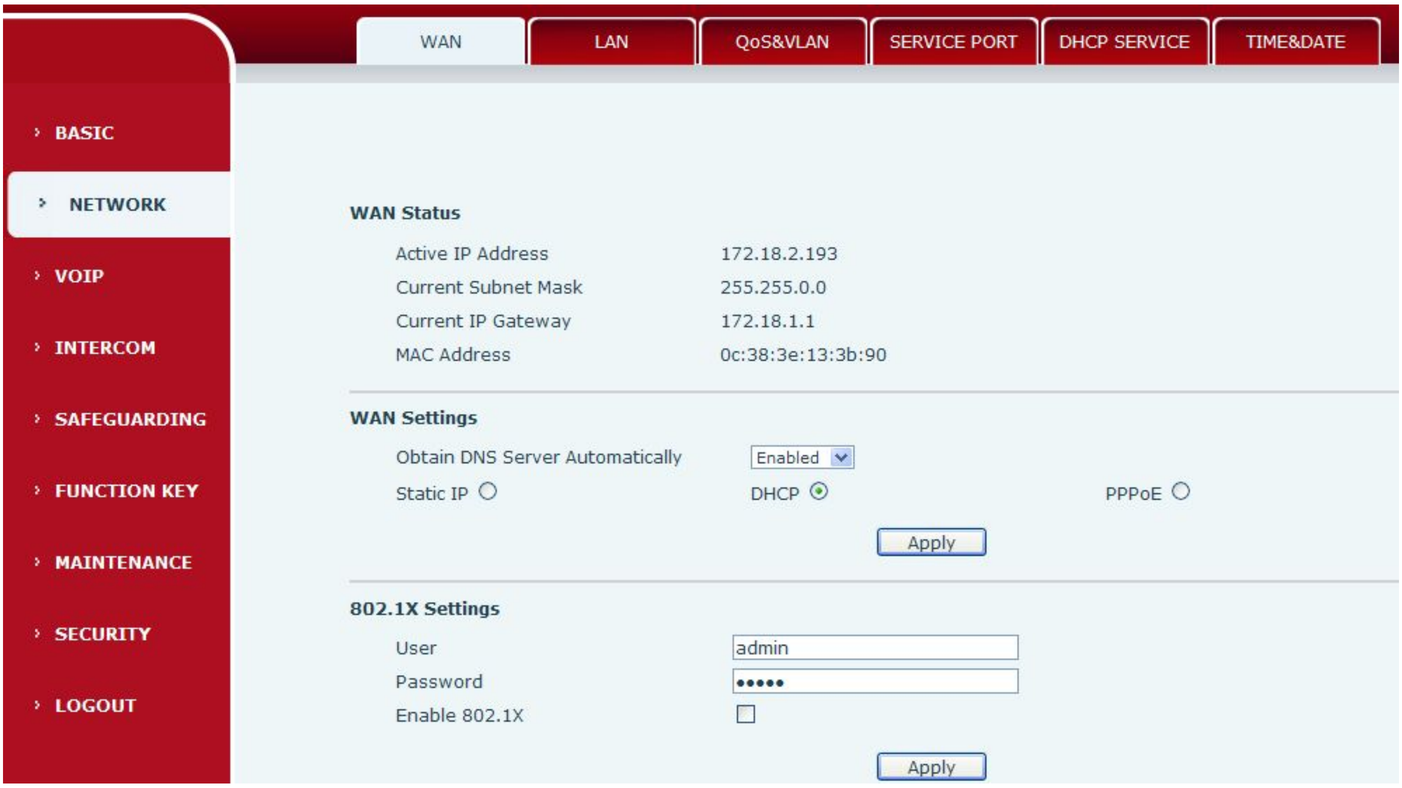


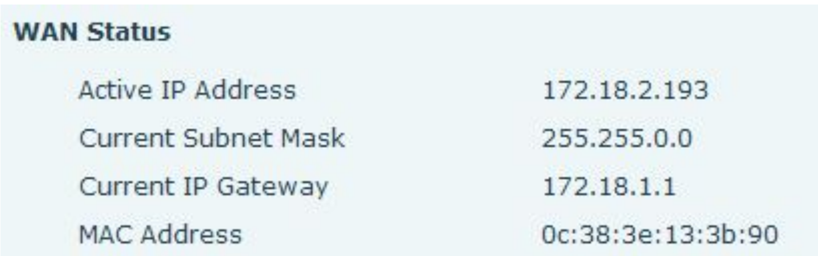
Call log

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**

(2)NETWORK

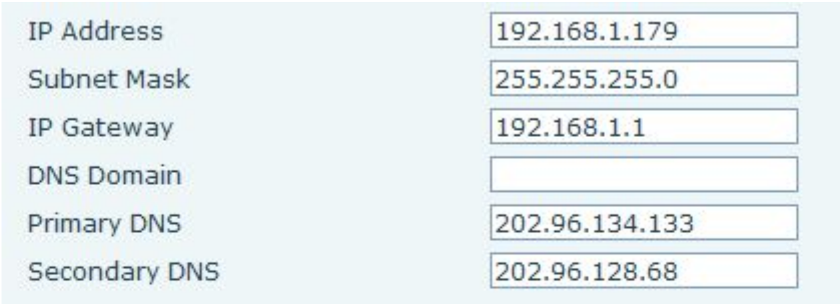
a) WAN



WAN	
Field Name	**Explanation**
	

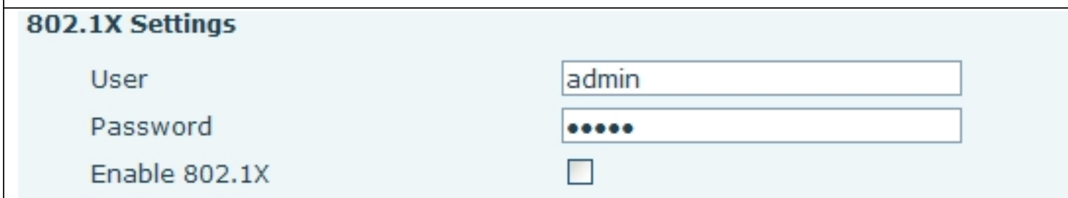
Field Name	Explanation
Active IP address	The current IP address of the equipment
Current subnet mask	The current Subnet Mask
Current IP gateway	The current Gateway IP address
MAC address	The MAC address of the equipment

Select the appropriate 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 chosen, the screen below will appear. Enter values provided by the ISP.	



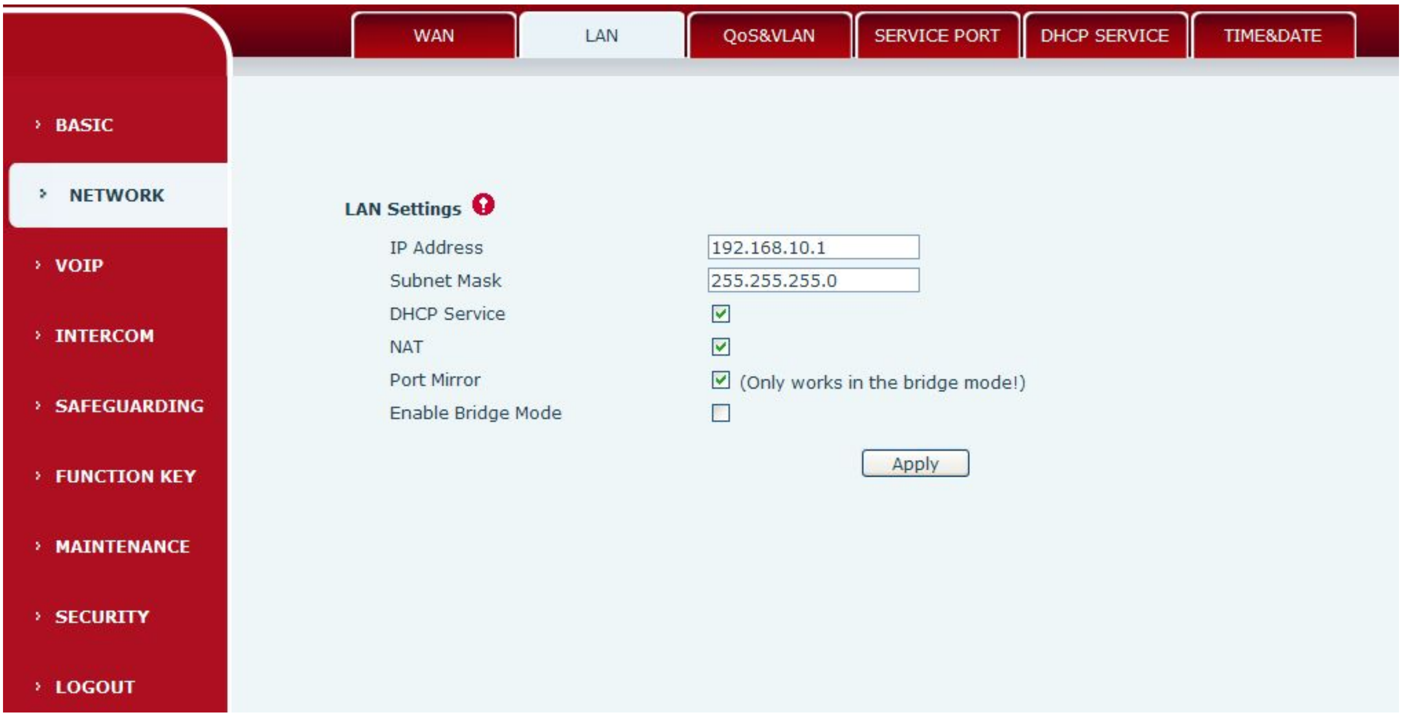
Static IP address	Please enter the Static IP address
Subnet mask	Please enter the Subnet Mask
Gateway	Please enter the IP Gateway
DNS Domain	Set the DNS domain suffix. When the user enter the domain name DNS address cannot be resolved, the domain equipment to resolve in the domain name.
Primary DNS	Please enter the Primary DNS server address
Secondary DNS	Please enter the Secondary DNS server address

Field Name	Explanation
802.1X Settings	



User	802.1X user account
Password	802.1X password
Enable 812.1X	Open/Close 812.1X
After entering the new settings, click the APPLY button. The equipment will save the new settings and apply them. If a new IP address was entered for the equipment, it must be used to login to the phone after clicking the APPLY button.	

b) LAN



LAN	
Field Name	**Explanation**
LAN IP address	LAN static IP
Subnet mask	LAN Subnet Mask
DHCP Service	Activate DHCP server for LAN port. The equipment must be rebooting for the DHCP server setting to take effect.
NAT	Enable NAT operation
Field Name	**Explanation**
Port Mirror	Port Mirror can only be activated in bridge mode. If activated, the data stream from the WAN port is copied to the LAN port of the equipment.
Enable bridge mode	If Bridge Mode is activated, the equipment will not provide an IP address for the LAN 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.

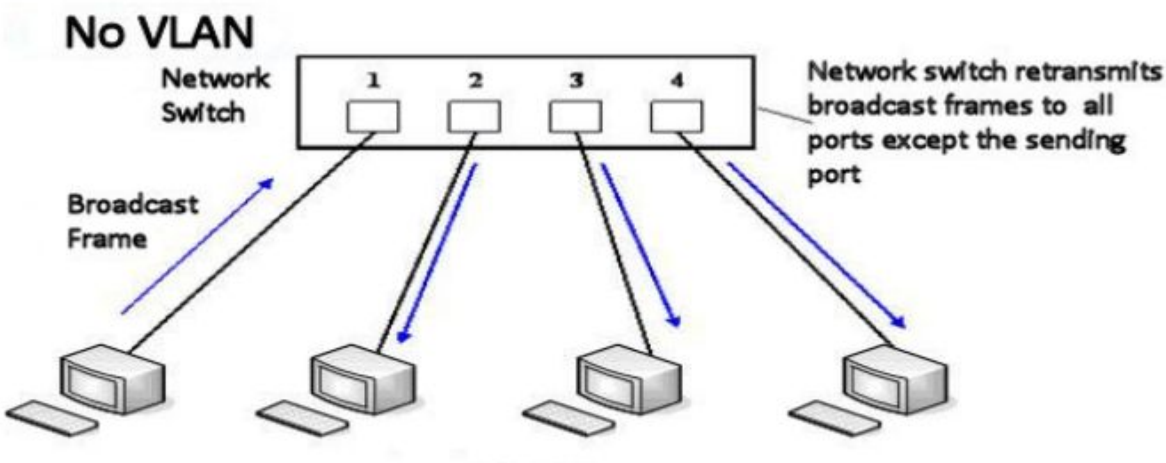


Chart 1

➢ **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.**

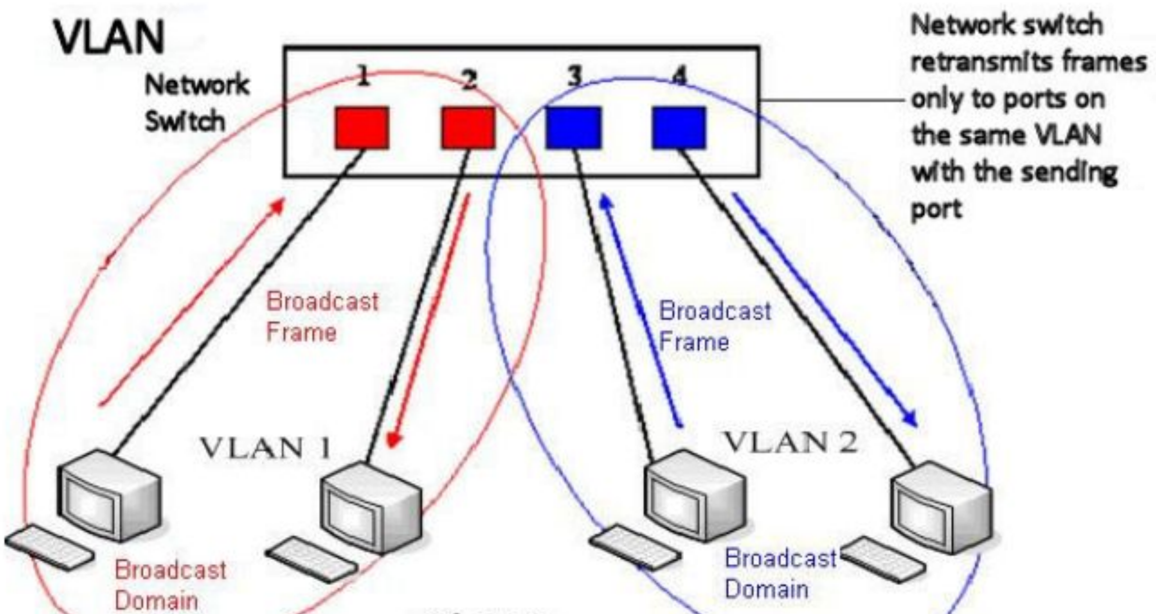
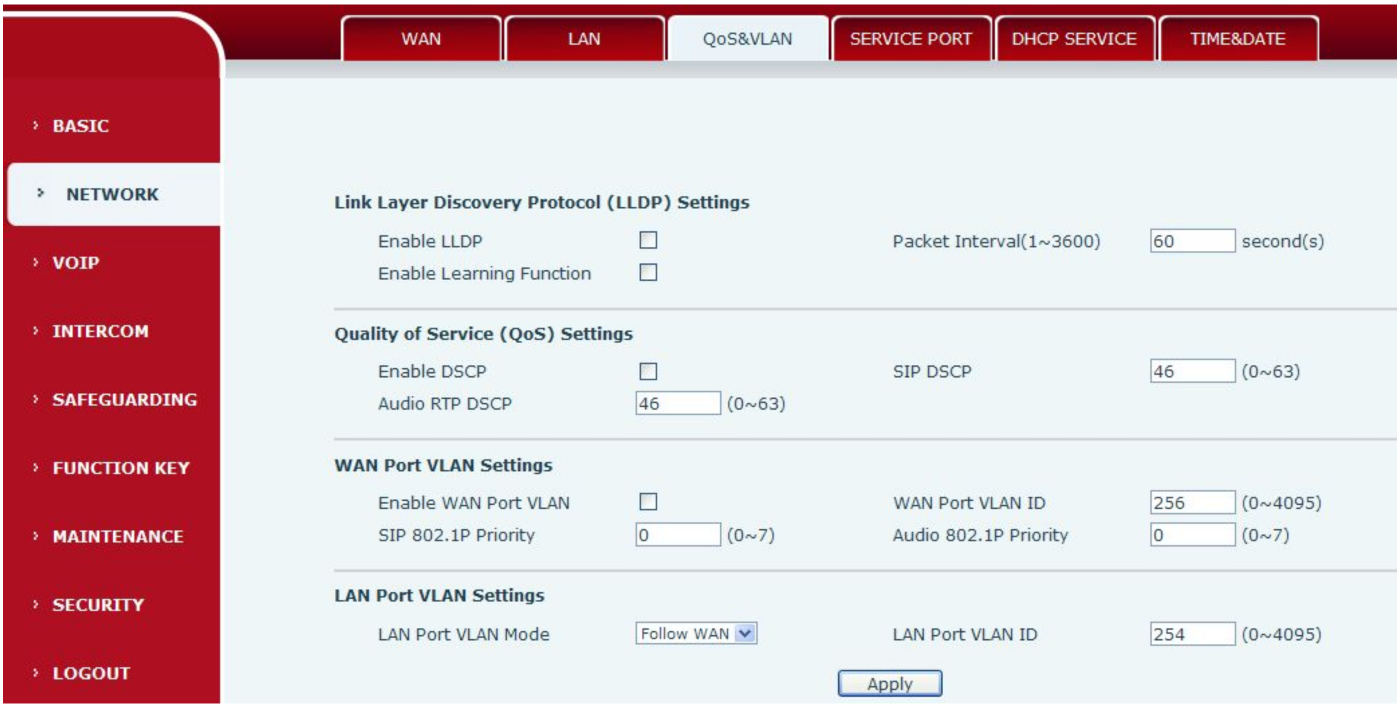


Chart 2

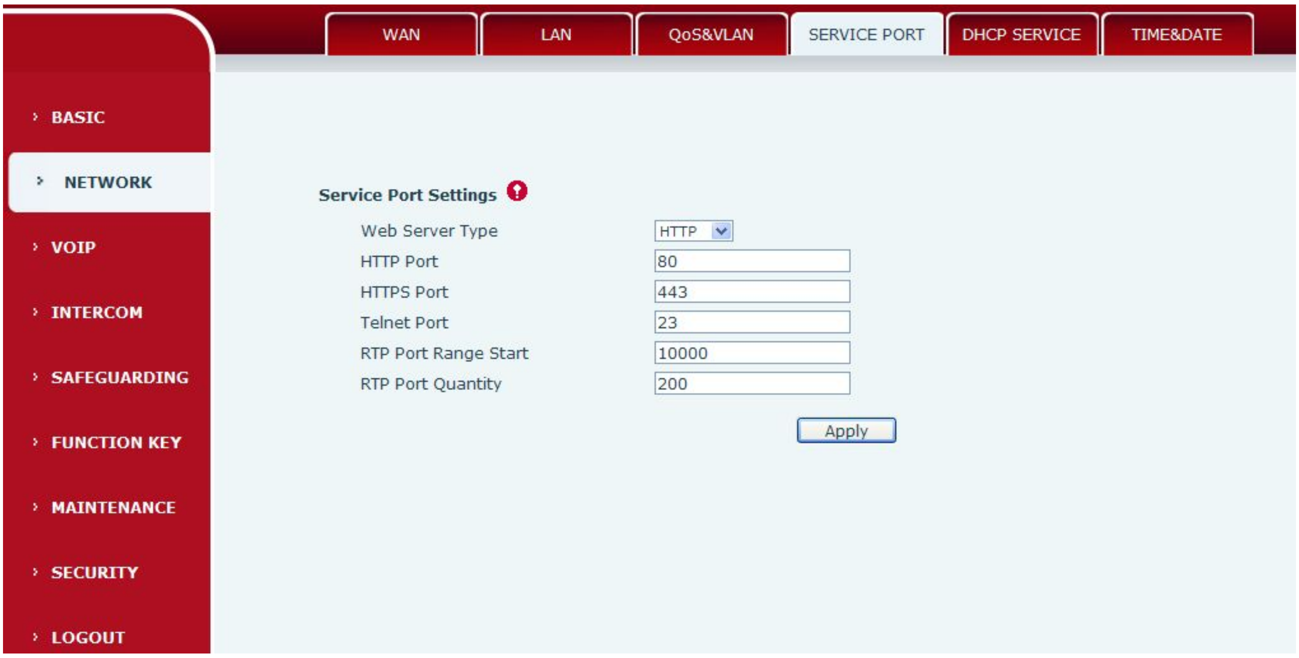
Note: In practice, VLANs are distinguished by the use of VLAN IDs.



QoS&VLAN	
Field Name	**Explanation**
LLDP Settings	
Enable LLDP	Enable or Disable Link Layer Discovery Protocol (LLDP)
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.
Packet Interval	The time interval for sending LLDP Packets
QOS Settings	
Enable DSCP	Enable or Disable Differentiated Services Code Point (DSCP)
Audio RTP DSCP	Specify the value of the Audio DSCP in decimal
SIP DSCP	Specify the value of the SIP DSCP in decimal
WAN Port VLAN Settings	
Enable WAN Port VLAN	Enable or Disable WAN Port 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	
LAN Port VLAN	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
LAN Port VLAN ID	Used when the VLAN ID is different from WAN ID. Range is 0-4095

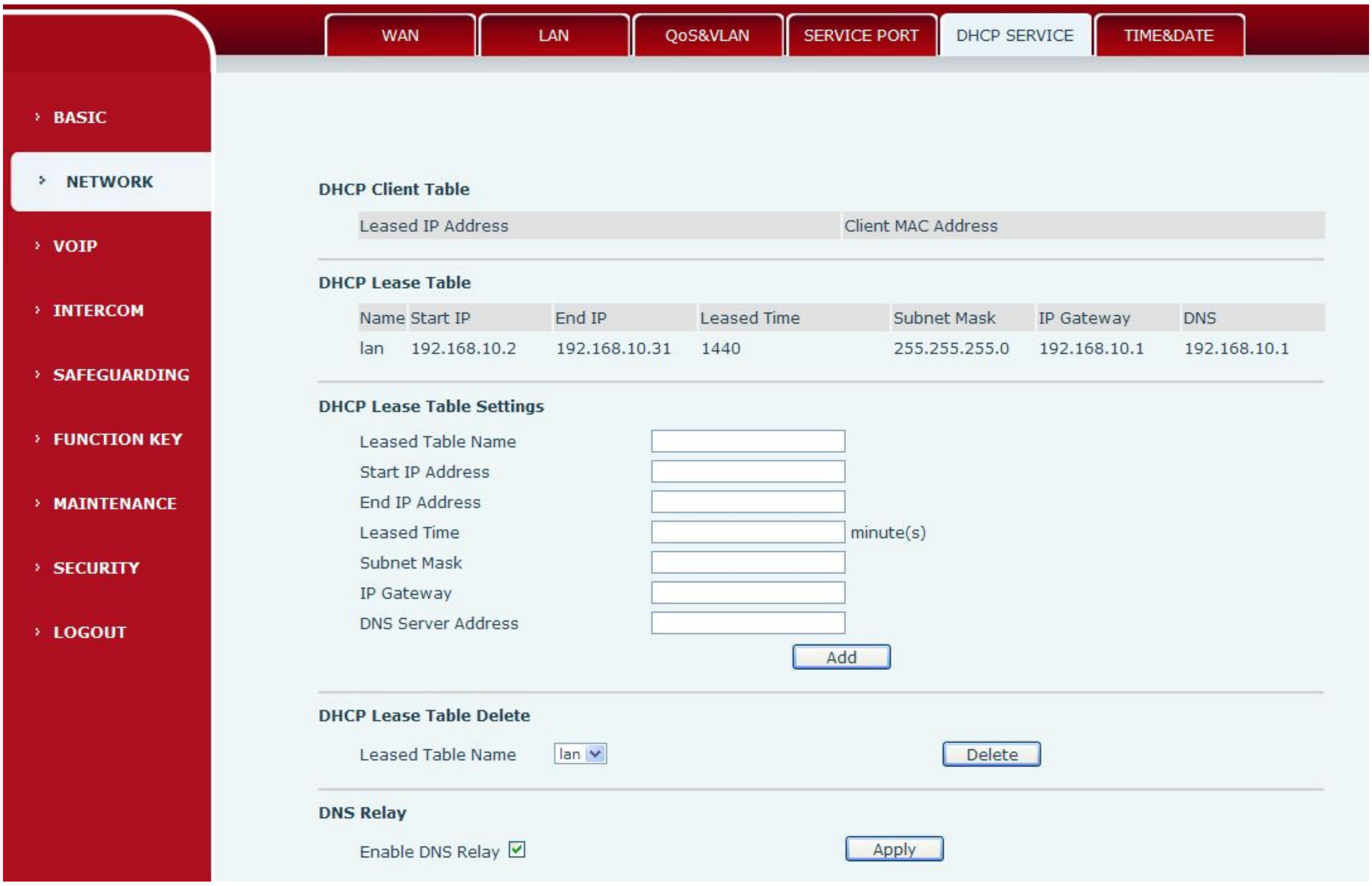
d) SERVICE PORT

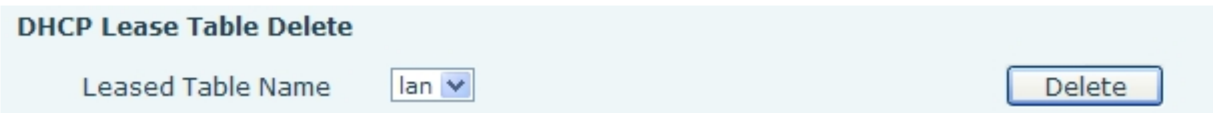
Set the port values for Telnet/HTTP/RTP on this page.

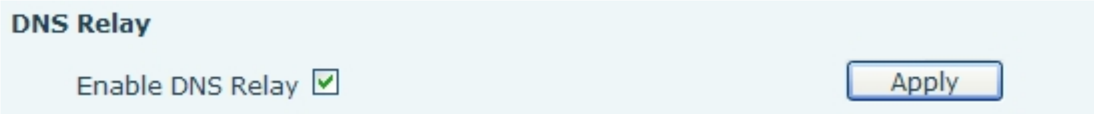


Service port	
Field Name	**Explanation**
Web Server type	Specify Web Server Type – HTTP or HTTPS
HTTP port	Port for web browser access. Default value is 80. To enhance security, change this from the default. Setting this port to 0 will disable HTTP access. Example: The IP address is 192.168.1.70 and the port value is 8090, the accessing address is http://192.168.1.70:8090.
HTTPS port	Port for HTTPS access. Before using https, an https authentication certification 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 start	Set the beginning value for RTP Ports. Ports are dynamically allocated.
RTP port 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.	

e) DHCP SERVICE

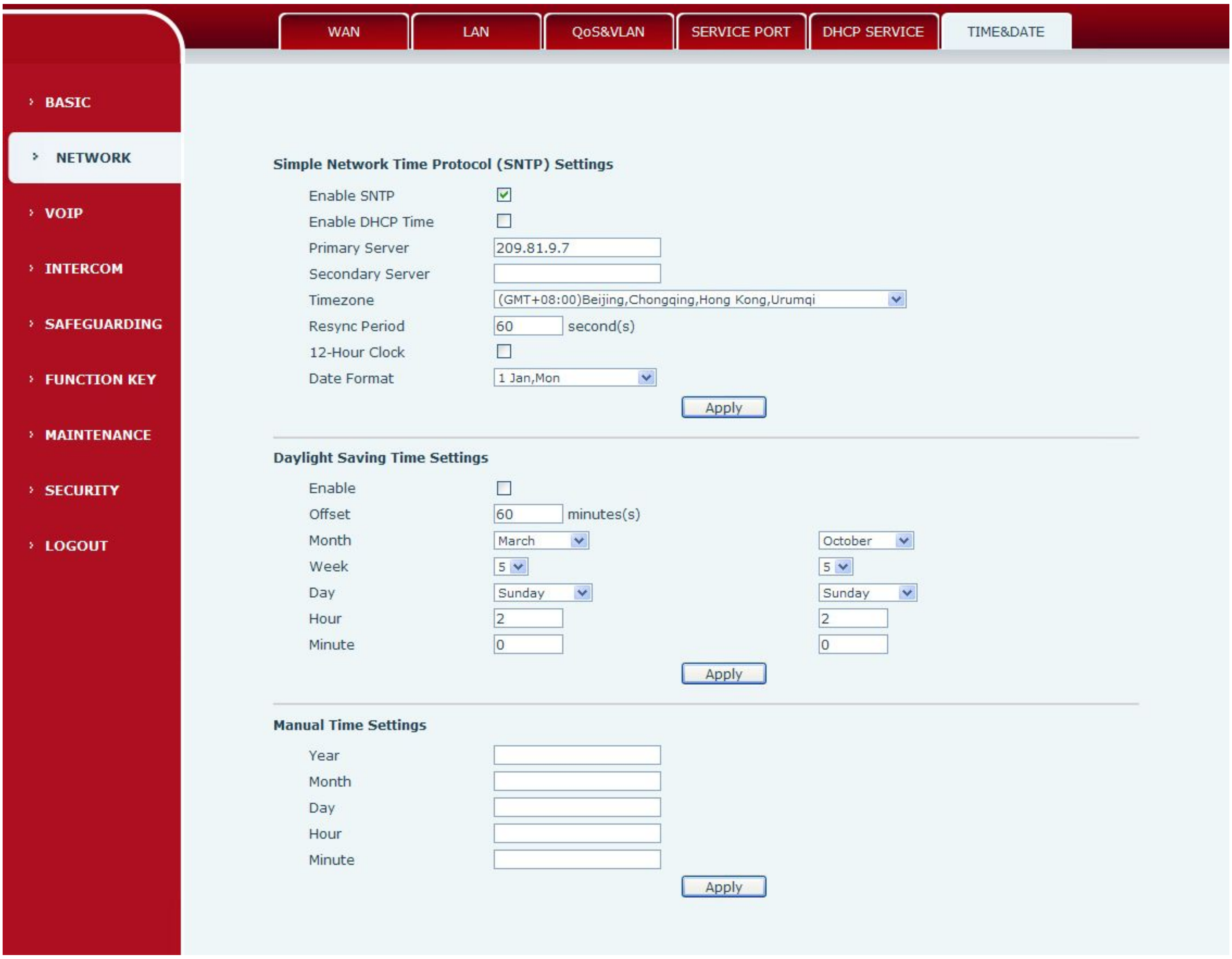


DHCP Server	
Field Name	**Explanation**
DHCP Lease Table	IP-MAC mapping table. If the LAN port of the device connects to a device, this table will show its IP and MAC address.
DHCP Lease Table Settings	
Leased table name	Name of the lease table.
Start IP address	Beginning IP address of the lease table.
End IP address	Ending IP address of the lease table. A device connected to the LAN port will get an IP address between Start IP and End IP.
Leased time	Time IP address assignments will persist. Unit is minutes.
Subnet mask	Subnet Mask of the lease table.
Gateway	Network Gateway of the lease table
DNS server address	IP address of DNS server.
Field Name	**Explanation**
DHCP Lease Table Delete	
	

Enter the table name and click the Delete button to remove a DHCP lease table.	
DNS Relay	
	
Enable DNS Relay	Activates DNS Relay in the equipment. Default is enabled.
Note: 1) The size of lease table cannot be larger than the quantity of C network IP address. It is recommended to use the default lease table without modification 2) If the DHCP lease table is modified, the equipment must be rebooted.	

f) TIME&DATE

Set the time zone and SNTP (Simple Network Time Protocol) server on this page. Daylight savings time configuration and manual time and date entry are also done on this page.



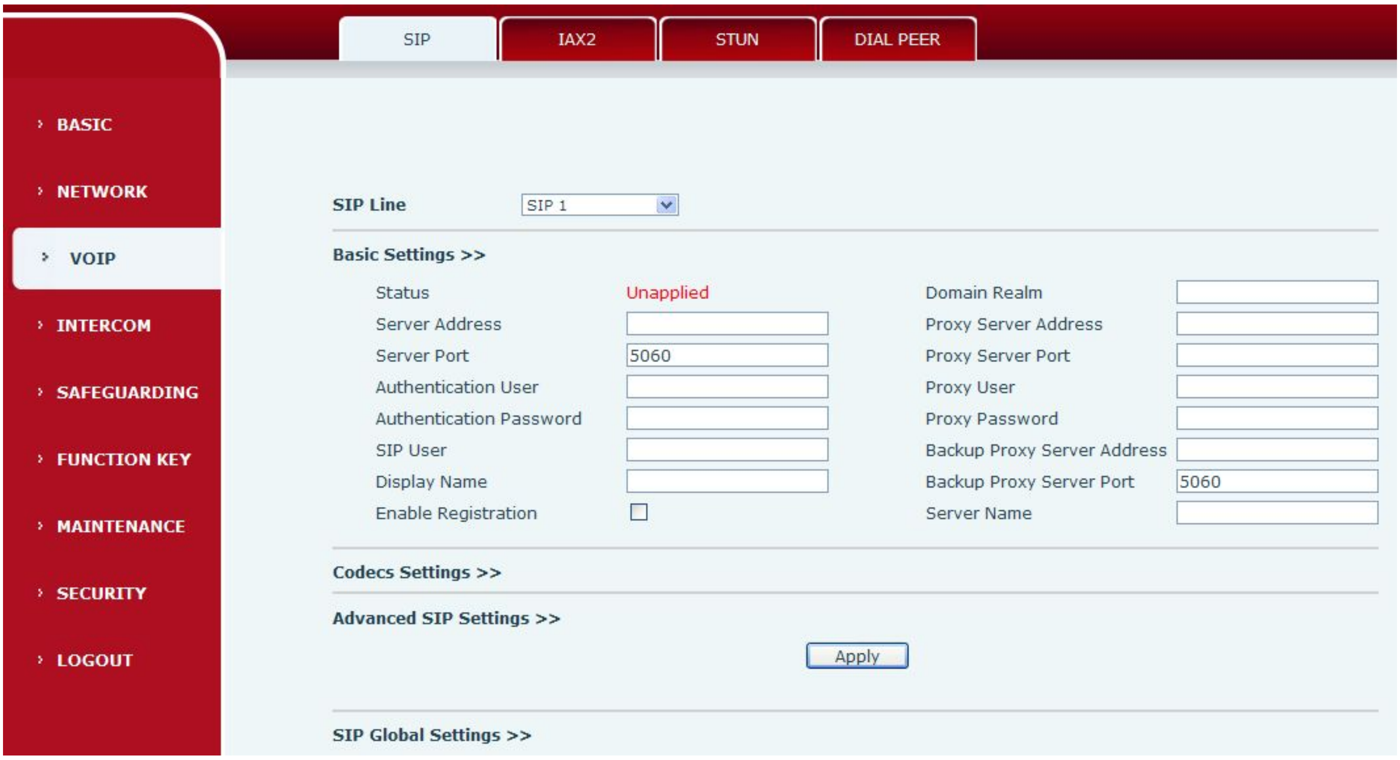
TIME&DATE	
Field Name	Explanation

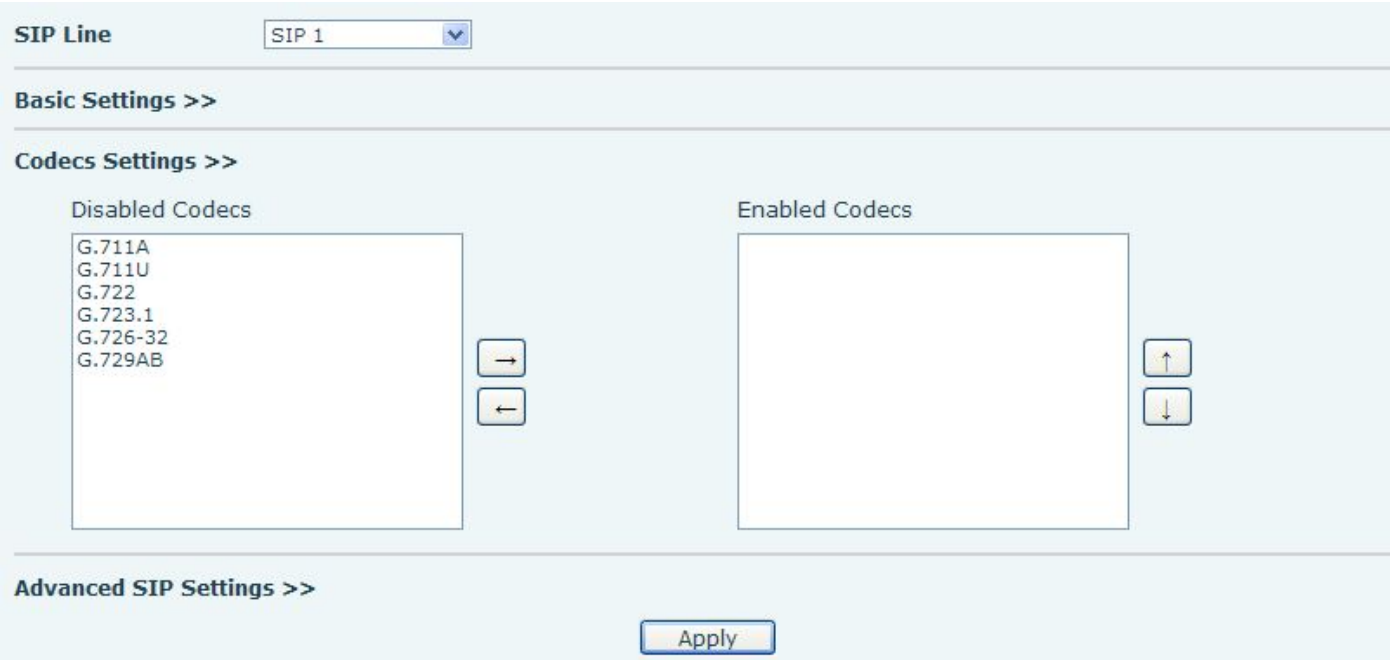
SNTP Settings	
Enable SNTP	Enable or Disable SNTP
DHCP Time	If this is enabled, equipment will synchronize time with DHCP server
Primary Server	IP address of Primary SNTP Server
Secondary Server	IP address of Secondary SNTP Server
Time zone	Local Time Zone
Resync Period	Time between resync to SNTP server. Default is 60 seconds.
Field Name	Explanation
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.
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.	

(3)VOIP

a) SIP

Configure a SIP server on this page


SIP
IAX2
STUN
DIAL PEER
BASIC
NETWORK
VOIP
INTERCOM
SAFEGUARDING
FUNCTION KEY
MAINTENANCE
SECURITY
LOGOUT
SIP Line
SIP 1
Basic Settings >>
Status
Unapplied
Server Address
Server Port
5060
Authentication User
Authentication Password
SIP User
Display Name
Enable Registration
Domain Realm
Proxy Server Address
Proxy Server Port
Proxy User
Proxy Password
Backup Proxy Server Address
Backup Proxy Server Port
5060
Server Name
Codecs Settings >>
Advanced SIP Settings >>
Apply
SIP Global Settings >>


SIP Line
SIP 1
Basic Settings >>
Codecs Settings >>
Disabled Codecs
G.711A
G.711U
G.722
G.723.1
G.726-32
G.729AB
Enabled Codecs
Advanced SIP Settings >>
Apply

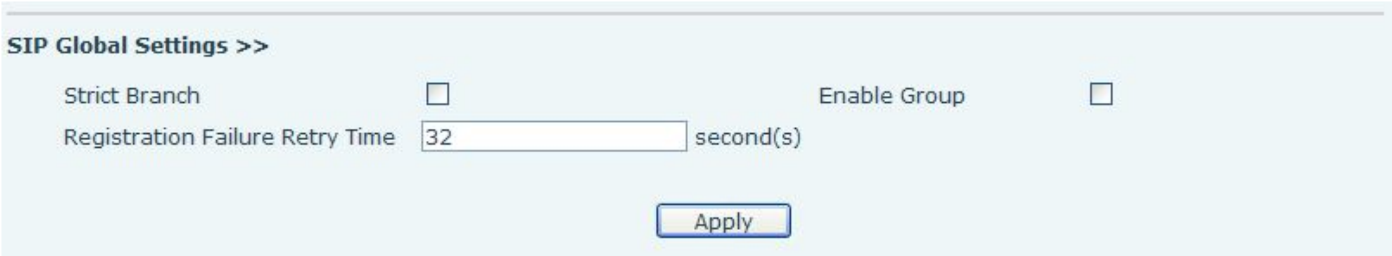
Codecs Settings >>

Advanced SIP Settings >>

Always Forward	☐	Enable Hotline	☐
Always Fwd Number		Hotline Number	
Busy Forward	☐	Warm Line Wait Time	0 (0~9)second(s)
Busy Fwd Number		Keep Alive Type	SIP Option
No Answer Forward	☐	Keep Alive Interval	60 second(s)
NoAnswer Fwd Number		BLF Server	
No Ans. Fwd Wait Time	60 (0~120)second(s)	Transfer Timeout	0 second(s)
SIP Encryption	☐	Enable Auto Answer	☐
SIP Encryption Key		Auto Answer Timeout	60 second(s)
RTP Encryption	☐	Enable Session Timer	☐
RTP Encryption Key		Session Timeout	0 second(s)
		Session Refresher	UAS
Subscribe For MWI	☐	Conference Type	Local
MWI Number		Conference Number	
Subscribe Period	3600 second(s)	Registration Expires	3600 second(s)

Enable Service Code	☐		
DND On Code		DND Off Code	
Always CFwd On Code		Always CFwd Off Code	
Busy CFwd On Code		Busy CFwd Off Code	
No Ans. CFwd On Code		No Ans. CFwd Off Code	
Ban Anonymous On Code		Ban Anonymous Off Code	
User Agent		Server Type	COMMON
DTMF Type	AUTO	RFC Protocol Edition	RFC3261
DTMF SIP INFO Mode	Send 10/11	Local Port	5060
Ring Type	Default	Anonymous Call Edition	None
Enable Rport	☐	Keep Authentication	☐
Enable PRACK	☐	Ans. With a Single Codec	☐
Enable Long Contact	☐	Auto TCP	☐
Convert URI	☑	Enable Strict Proxy	☐
Dial Without Registered	☐	Enable GRUU	☐
Ban Anonymous Call	☐	Enable Displayname Quote	☐
DNS Mode	A	Enable user=phone	☑
Enable Missed Call Log	☑	Click To Talk	☐
BLF List Number		Transport Protocol	UDP
Enable BLF List	☐	Use VPN	☑

Apply



SIP	
Field Name	**Explanation**
Choose the sip line to configured (SIP 1 – SIP2). Click the dropdown arrow to select the line.	
Basic Settings	
Status	Shows registration status. If the registration is successful will display has 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.
User	SIP account name (Login ID).
password	SIP registration password.
Field Name	**Explanation**
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.
Enable Registration	Check to submit registration information.
Domain Realm	SIP Domain if different than the SIP Registrar Server.
Proxy server address	SIP proxy server IP address or URI, (This is normally the same as the SIP 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 server address	Backup SIP Server Address or URI (This server will be used if the primary server is unavailable)
Backup Proxy server port	Backup SIP Server Port
Server name	Name of SIP Backup server
Codecs Settings	
Disable Codecs /Enable Codecs	Click on the desired codec to select it. Then click the Left/right arrow to move to the Enabled or Disabled List. Use the Up/Down arrow to change the priority of enabled codecs.
Advanced SIP Settings	
Always Forward	All incoming calls will be forwarded to the specified number.

Always Fwd Number	Always to which calls are to be forwarded the number.
Busy Forward	If the line is busy, incoming calls will be forwarded to the specified number.
Busy Fwd Number	When the line busy to which calls are to be forwarded the number.
No Answer Forward	If there in after a specified time no answer, incoming calls will be forwarded to the specified number.
No Answer Fwd Number	When the no answer to which calls are to be forwarded the number.
No Ans. Fwd Wait Time	Used in conjunction with Call Forward No Answer. Wait time in seconds before call is forwarded.
Enable Hotline	Activate Hot Line feature. Automatically call a number by going off hook.
Hotline Number	Number to be called in Hot Line Mode.

Field Name	Explanation
Warm Line Wait Time	Used in Hot Line Mode. Time the waits after off hook before dialing the hot line number.
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 Interval	Set the NAT Keep Alive interval. Default is 60 seconds
BLF Server	BLF server address
Transfer Timeout	Time interval between sending “bye” message and hanging up after the equipment transfers a call.
SIP Encryption	Enable/Disable SIP Encryption.
SIP Encryption Key	SIP Encryption key.
RTP Encryption	Enable/Disable RTP Encryption.
RTP Encryption Key	Enable/Disable RTP Encryption key.
Enable Auto Answer	Activate Auto Answer mode.
Auto Answer Timeout	Used in conjunction with auto answer. The equipment will answer an incoming call after the Auto Answer Timeout
Enable Session Timer	If enabled, this will refresh the SIP session timer per RFC4028.
Session Timeout	Refresh interval if Session Timer is enabled.
Session Refresher	Refresh mode configuration
Subscribe For MWI	If enabled, the phone will send Message Waiting Indication(MWI) Subscribe message to the SIP Server
MWI Number	Specify the number to call to retrieve Voice Messages.

Subscribe Period	Time interval between MWI Subscribe Messages.
Conference Type	Choose Conference Type, either local or network
Conference Number	Number to dial to access network conference server. Not needed if Local conference mode is chosen
Registration Expires	SIP re-registration time. Default is 3600 seconds. If the server requests a different time, the phone will change to that value.

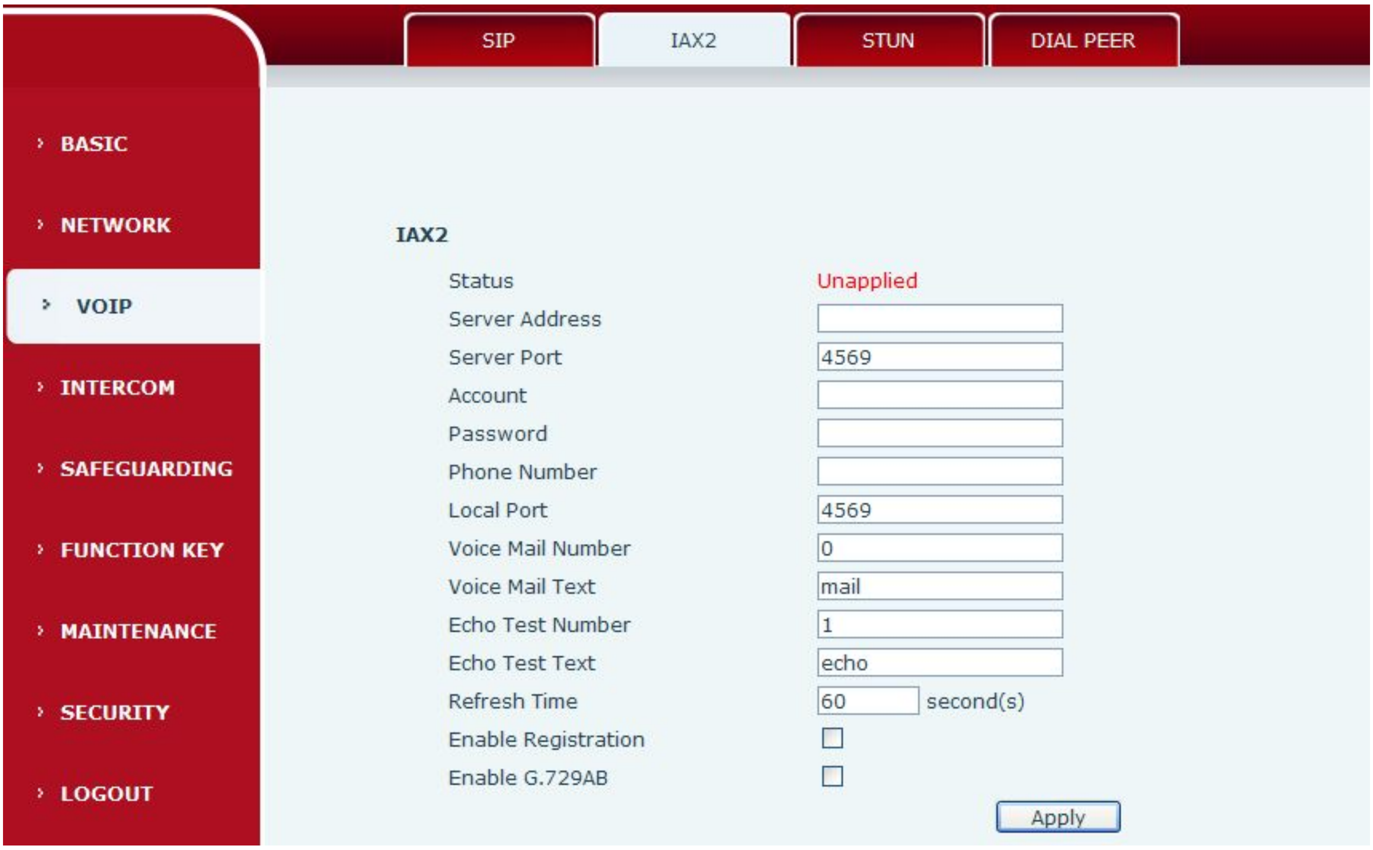
Field Name	Explanation
Enable Service Code	Enables or disables the services described below. These codes will be sent to the SIP server to activate or deactivate the service.
DND On Code	Do Not Disturb (DND) – When this hot key is pressed, all calls to the extension to be rejected by the server. The incoming call record will not be displayed in the Call History.
Always CFwd On Code	Always Call Forward On – When this function is enabled, the server will forward all calls to a designated number. The incoming call record will not be displayed in the Call History
Busy CFwd On Code	Busy Call Forward On - When this function is enabled, the server will forward all calls to a designated number if the telephone is busy. The call record will not be displayed in Call History.
No Answer CFwd On Code	No Answer Call Forward On - When this function is enabled, the server will forward all calls to a designated number if there is no answer within a designated time. The incoming call record will not be displayed in the Call History.
Ban Anonymous On Code	Allow Anonymous Calling function described above. In other words “Anonymous” will be transmitted for Caller ID.
DND Off Code	Disable Server DND as described above.
Always CFwd Off Code	e Disable Server Always CFwd as described above.
Busy CFwd Off Code	Disable Server Busy CFwd as described above.
No Answer. CFwd Off Code	Disable Server No Ans. CFwd as described above.
Ban Anonymous Off Code	Allow Anonymous Calling function described above. In other words “Anonymous” will be transmitted for Caller ID.
User Agent	Set SIP User Agent value.
DTMF Type	DTMF sending mode. There are four modes: ● In-band ● RFC2833 ● SIP_INFO ● AUTO Different VoIP Service providers may require different modes.
DTMF SIP INFO Mode	You can chose Send 10/11 or Send */#

Ring Type	**Set ring tone. There are 9 standard options and 3 user options.**
Enable Rport	**Enable/Disable support for NAT traversal via RFC3581 (Rport).**
Enable PRACK	**Enable or disable SIP PRACK function. Default is OFF. It is suggested this be used.**
Field Name	**Explanation**
Enable Long Contact	**Allow more parameters in contact field per RFC 3840**
Convert URI	**Converts # to %23 when sending URI information.**
Dial Without Registered	**Allow outgoing calls without registration.**
Ban Anonymous Call	**Refuse Anonymous Calls**
DNS Mode	**DNS mode configuration, Select A, SRV, NAPTR three models, the default is A.**
Enable Missed Call Log	**If enabled, the phone will save missed calls into the call history record.**
BLF List Number	**BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.**
Enable BLF List	**Enable/Disable BLF List**
Server Type	**Configures phone for unique requirements of selected server.**
RFC Protocol Edition	**Select SIP protocol version RFC3261 or RFC2543. Default is RFC3261. Used for servers which only support RFC2543.**
Local Port	**SIP port. Default is 5060.**
Anonymous Call Edition	**Set privacy support RFC3323, RFC3325 or none**
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 server if enabled**
Answer With a Single Codec	**If enabled phone will respond to incoming calls with only one codec.**
Auto TCP	**Force the use of TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes**
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.**
Enable GRUU	**Support for Globally Routable User-Agent URI (GRUU)**
Enable Display name Quote	**Puts quotation marks around the display-name in SIP messages. For servers that require this.**
Enable user=phone	**Sets user=phone in SIP messages. For compatibility with servers that require this.**
Click To Talk	**Set click to Talk (needs support from server).**

Field Name	**Explanation**
Transport	**Configuration using the transport protocol, TCP, TLS or UDP, the default is UDP.**

Protocol	
Use VPN	**Enable SIP use VPN for every line individually, not all of them**
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 Failure Retry Time	**Registration failures retry time – If registrations fails, the phone will attempt to register again after registration failure retry time. This will affect all lines**

b) IAX2

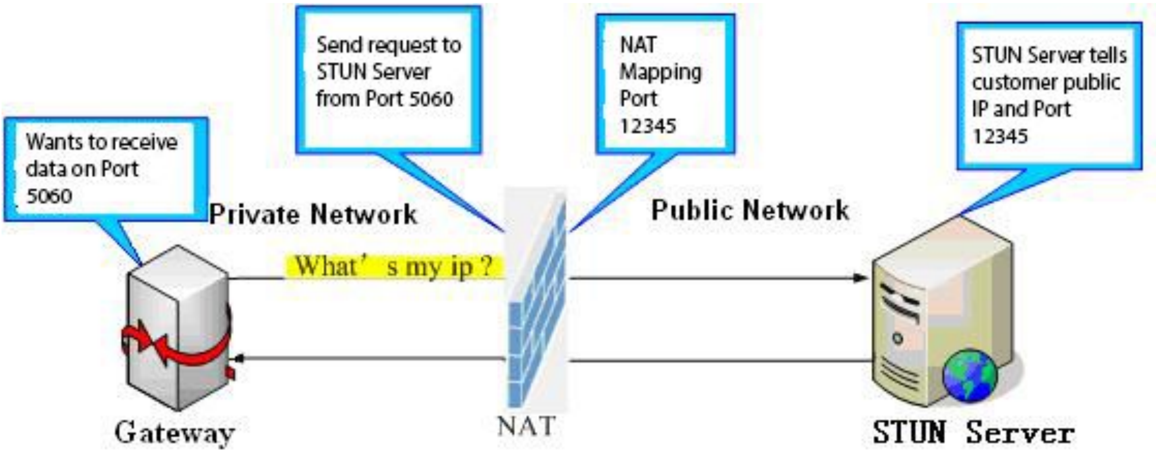


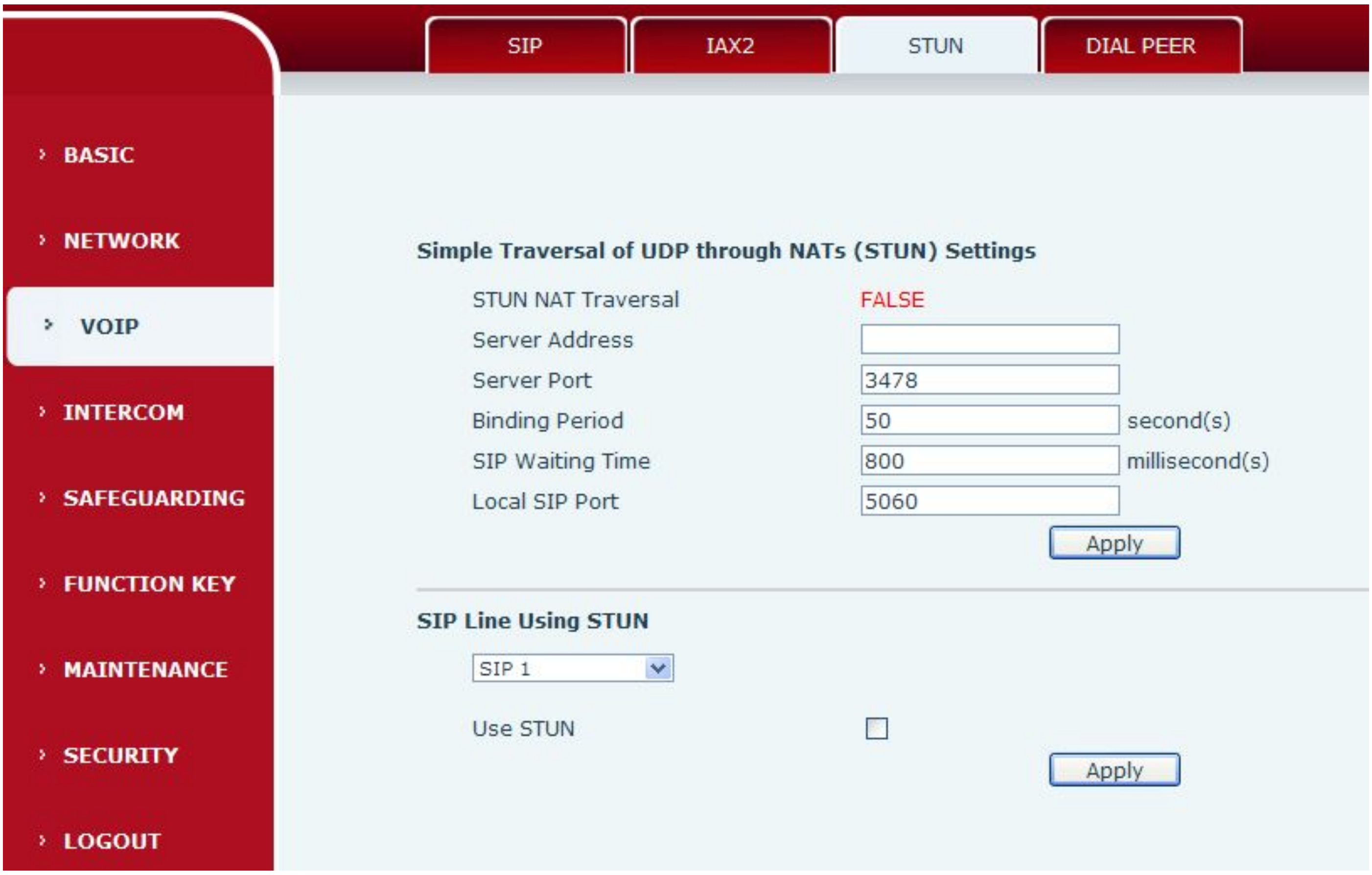
IAX2	
Field Name	**Explanation**
Status	**Shows registration status. Will show "Registered" if registered or "Unapplied" if not registered.**
Server Address	**IAX2 server address.**
Server Port	**IAX2 server port. Default is 4569.**
Account	**IAX2 account name for registration**
Password	**IAX2 registration password.**
Phone Number	**IAX2 phone number (usually the same as IAX2 account name).**
Local Port	**IAX2 local port. Default is 4569.**
Voice Mail Number	**Voice mail number.**

Voice Mail Text	Voice mail name.
Echo Test Number	If the IAX2 server supports echo test and the echo test number is non- numeric, this number can be used to replace the echo test text. This allows dialing a number to perform an echo voice test. This function is provided to test whether communication through the server.
Echo Test Text	Echo test text
Refresh Time	Expiration time of IAX2 server registration. Allowed values are between 60 and 3600 seconds.
Enable Registration	Enable/Disable IAX2 registration.
Enable G.729AB	Enable/Disable G.729 codec.

c) 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 Transversal was successful.
Server Address	STUN Server IP address
Server Port	STUN Server Port – Default is 3478.
Binding Period	STUN blinding period – STUN packets are sent at this interval to keep the 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
Select the SIP account configuration the first few lines, two lines are available. The selection switch to the line account configuration.	
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.	

d) DIAL PEER

This feature allows the user to create rules to make dialing easier. There are several different options for dial rules. The examples below will show how this can be used.

- Substitution – Assume that it is desired to place a direct IP call to IP address 192.168.119.Using this

feature, 156 can be substituted for 192.168.1.119.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
156	192.168.1.119	5060	SIP	no alias	no suffix	0

- **Substitution – To dial a long distance call to Beijing requires dialing area code 010 before the local phone number. Using this feature 1 can be substituted for 010. For example, to call 62213123 would only require dialing 162213123 instead of 01062213123.**

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
1T	0.0.0.0	5060	SIP	no alias	no suffix	0

- **Addition – Two examples are shown. In the first case, it is assumed that 0 must be dialed before any 11 digit number beginning with 13. In the second case, it is assumed that 0 must be dialed before any 11 digit number beginning with 135, 136, 137, 138, or 139. Two different special characters are used.**

x – Matches any single digit that is dialed.

[] – Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
13xxxxxxxxx	0.0.0.0	5060	SIP	no alias	no suffix	0
13[5-9]xxxxxxxx	0.0.0.0	5060	SIP	no alias	no suffix	0

We can also realize the equipment at the same time, using a different account, without switching fast call, will make the following specific configuration.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
13xxxxxxxxx	0.0.0.0	5060	SIP	no alias	no suffix	0
13[5-9]xxxxxxxx	0.0.0.0	5060	SIP	no alias	no suffix	0
156	192.168.1.119	5060	SIP	no alias	no suffix	0
1T	0.0.0.0	5060	SIP	no alias	no suffix	0

Add Dial Peer

Phone Number

Destination(Optional)

Port(Optional)

Alias(Optional)

Call Mode SIP

Suffix(Optional)

Deleted Length(Optional)

Apply

Dial Peer Option

13xxxxxxxxx Delete Modify

DIAL PEER

Field Name	Explanation
Phone Number	There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules. In prefix matching, only part of the number is entered followed by T. The mapping with then take place whenever these digits are dialed. Prefix mode supports a maximum of 30 digits.
Destination(Optional)	Set Destination address. This is optional. For a peer to peer call, enter the destination IP address or domain name. To use a dial rule on the SIP2 line, enter 0.0.0.2. For SIP3 enter 0.0.0.3
Port(Optional)	Set the Signaling port, the default is 5060.
Alias(Optional)	Set the Alias. This is the text to be added, replaced, or deleted. It is optional.
Note: There are four types of aliases. 1) Add: xxx – xxx will be dialed before any phone number. 2) All: xxx – xxx will replace the phone number. 3) Del: The characters will be deleted from the phone number. 4) Rep: xxx – xxx will be substituted for the specified characters.	

Field Name	Explanation
Alias(Optional)	Protocol configuration option, the default is SIP
Suffix(Optional)	Characters to be added at the end of the phone number. This is optional.
Deleted Length(Optional)	Sets the number of characters to be deleted. For example, if this is set to 3, the phone will delete the first 3 digits of the phone number. This is optional.
Here's how to realize multiple accounts at the same time using the configuration number IP configuration:	

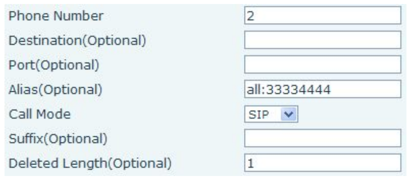
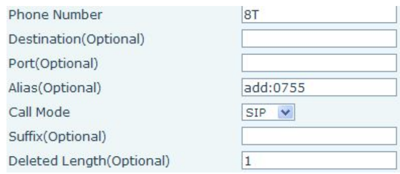
Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
9T	0.0.0.0	5060	SIP	del	no suffix	1
8T	0.0.0.0	5060	SIP	del	no suffix	1

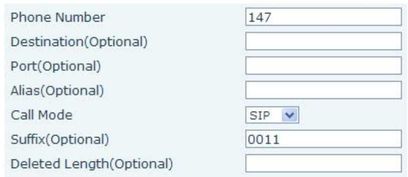
9T mapping shows that when the user to configure the SIP1 server, and the user registration, all through the SIP1 call number to dial 9;

8T mapping shows that when the user to configure the SIP2 server, and the user registration, all through the SIP2 call number to dial 8;

The following for each alias types for example:

Web Interface	Explanation	Example
Phone Number: 9T Destination(Optional): 255.255.255.255 Port(Optional): Alias(Optional): del Call Mode: SIP Suffix(Optional): Deleted Length(Optional): 1	Set phone number, Destination, Alias and Delete Length. Phone number is XXXT; Destination is 255.255.255.255 (0.0.0.2) and Alias is del. Any phone number that begins with XXX will be sent via SIP2 after the first several digits are deleted depending on the delete length.	Dial “93333” The SIP2 server will receive “3333”

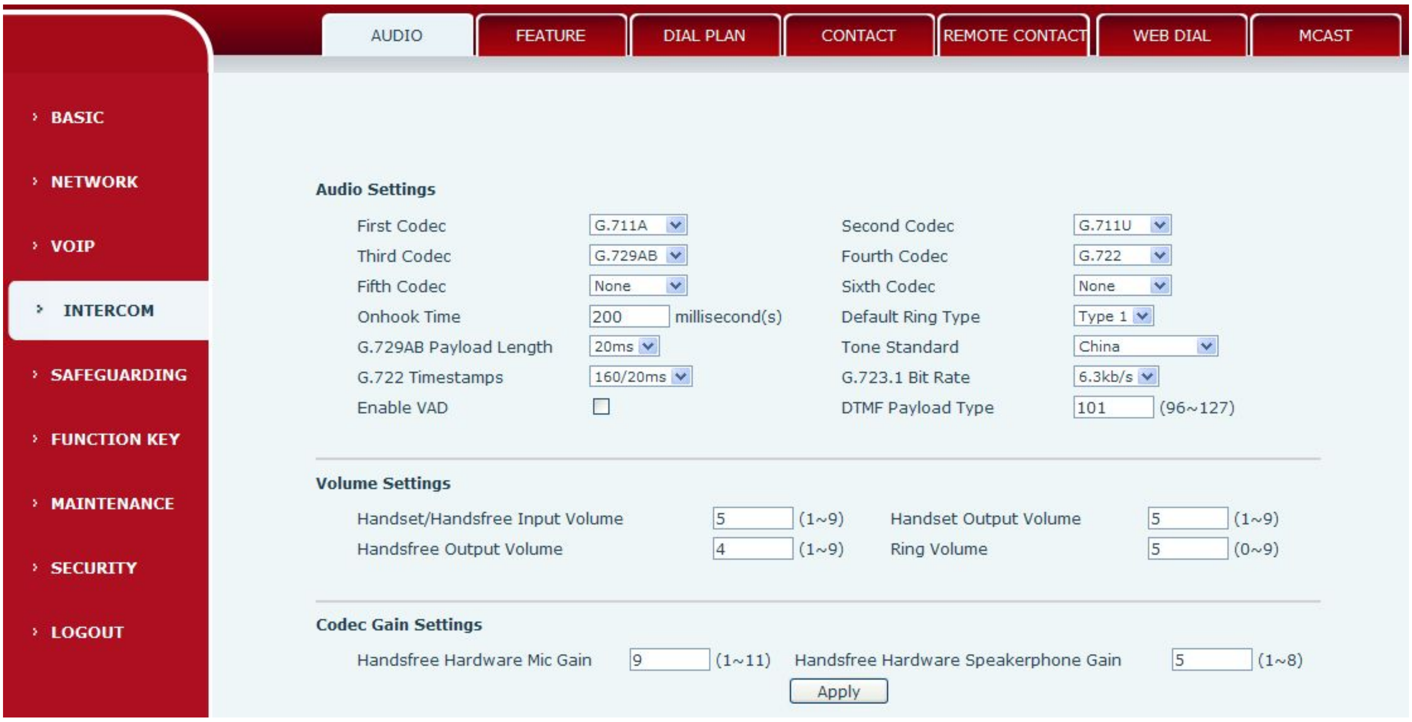
	This creates a speed dial function. Dialing "2", will cause the entire alias number to be sent out.	Dial "2" The SIP1 server will receive 33334444
	The equipment will add the alias to the end of the dialed number if the dialed number matches the template in the Phone Number box.	Dial "8309" The SIP1 server will receive "07558309"
	Set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is rep: xxx If the dialed phone number starts with the digits in the Phone Number box, the matching digits will be replaced by the alias number.	Dial "0106228" The SIP1 server will receive "86106228"

Web Interface	Explanation	Example
	If the dialed phone number starts with the digits in the Phone Number box, the phone will send out the dialed phone number and add the suffix number.	Dial "147" The SIP1 server will receive "1470011"

(4)INTERCOM

a) AUDIO

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

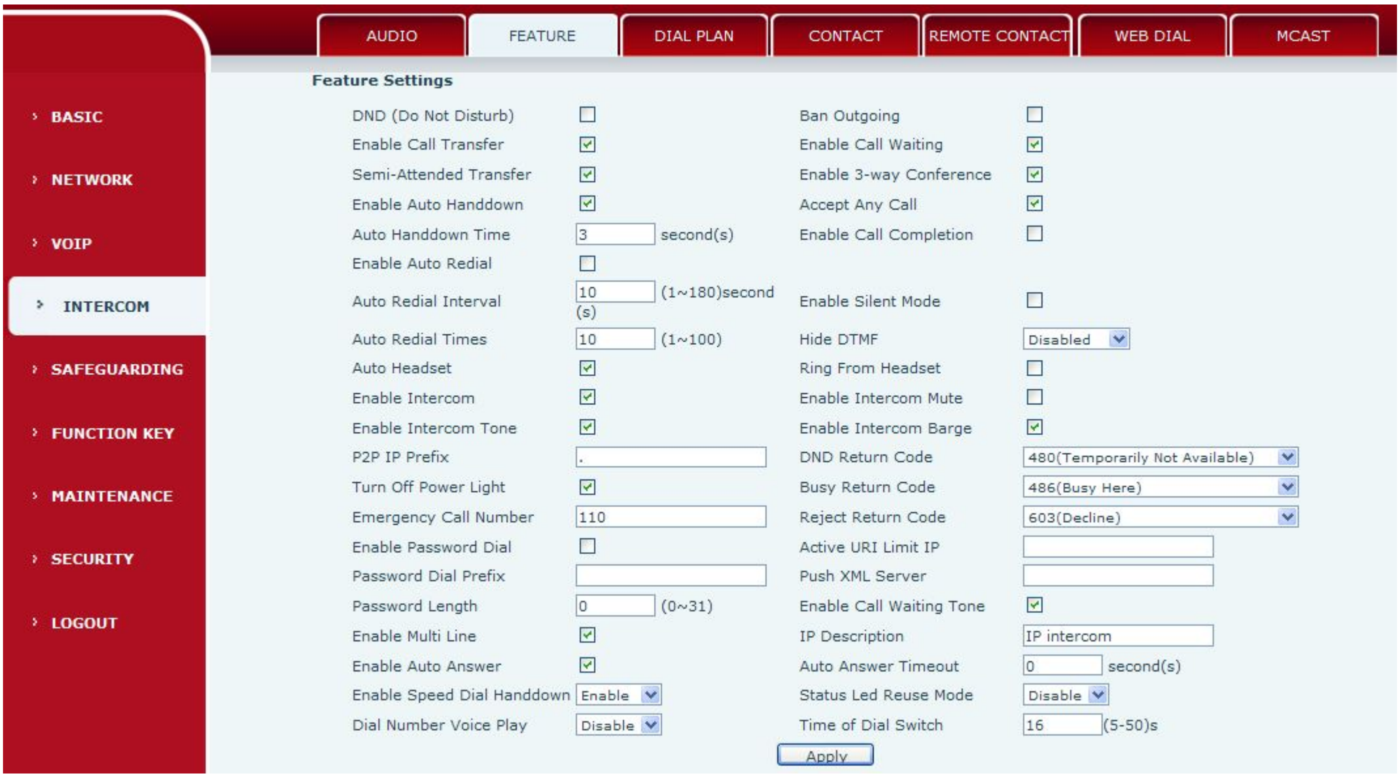


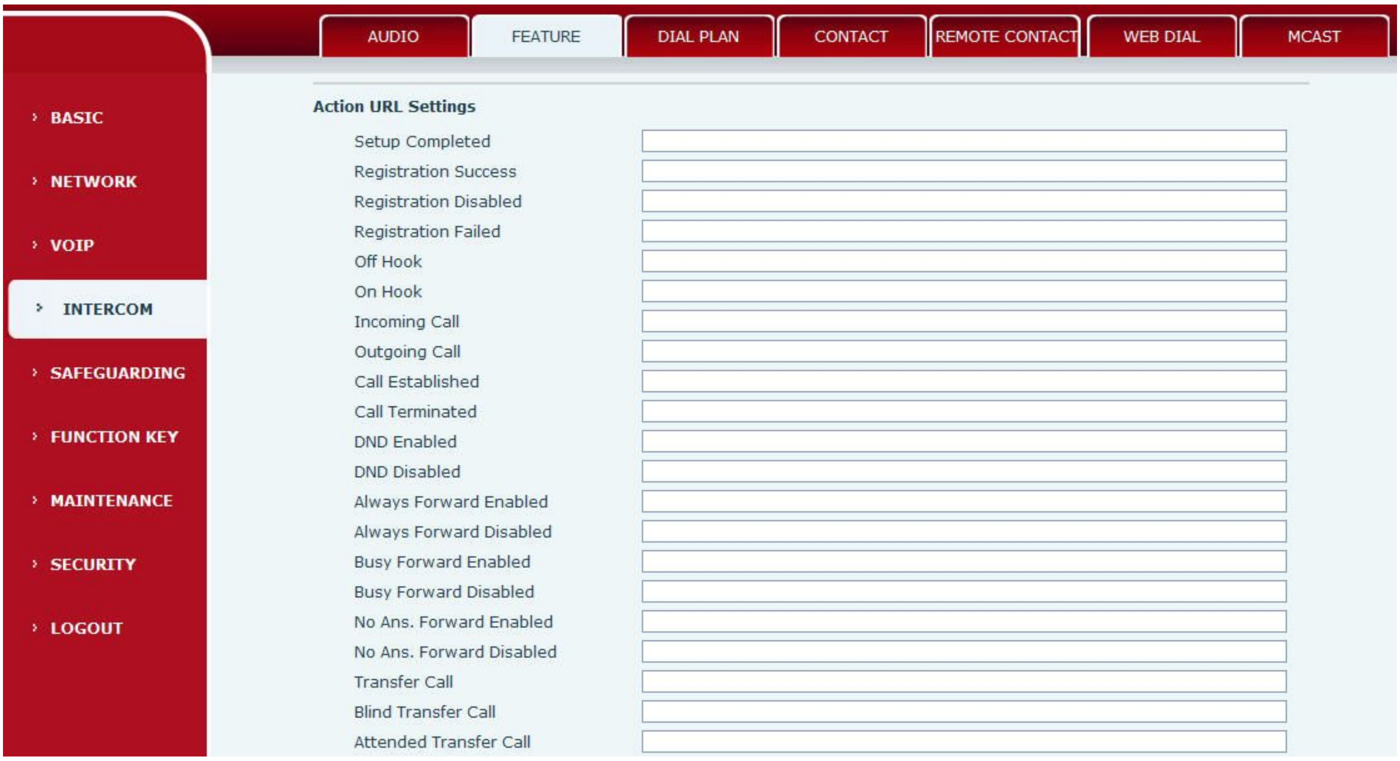
Audio settings	
Field Name	**Explanation**
First Codec	The first codec choice: G.711A/U, G.722, G.723, G.729, G.726
Second Codec	The second codec choice: G.711A/U, G.722, G.723, G.729, G.726, None
Third Codec	The third codec choice: G.711A/U, G.722, G.723, G.729, G.726, None
Fourth Codec	The forth codec choice: G.711A/U, G.722, G.723, G.729, G.726, None
Fifth Codec	The fifth codec choice G.711A/U, G.722, G.723, G.729, G.726, None
Sixth Codec	The sixth codec choice G.711A/U, G.722, G.723, G.729, G.726, None
On hook Time	Time the handset must be on hook to disconnect a call. Default is 200ms.
Default Ring Type	Ring Sound – There are 9 standard types and 3 User types

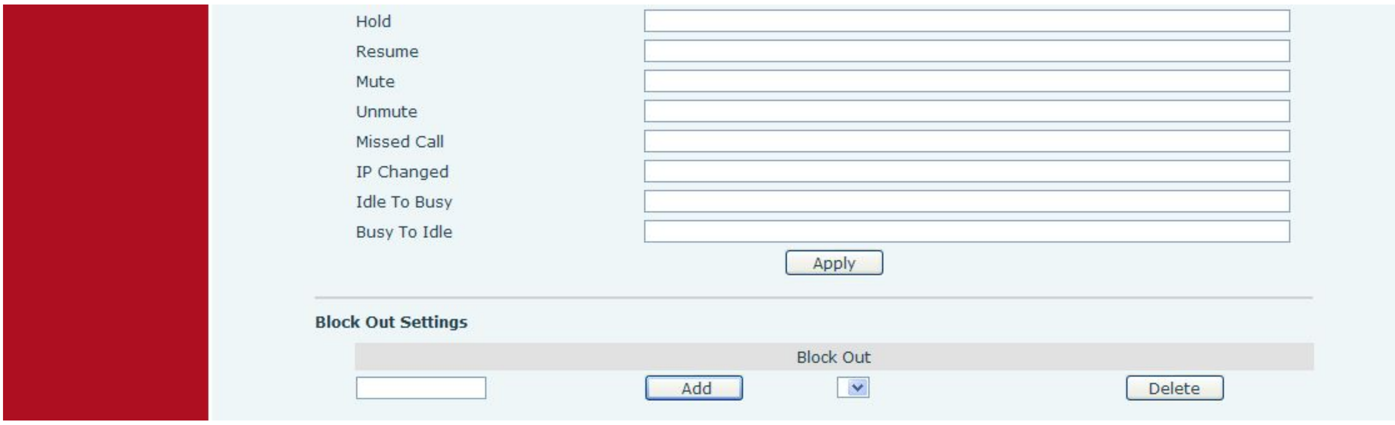
Field Name	Explanation
G.729AB Payload Length	G.729 Payload Length – Adjusts from 10 – 60 mSec
Tone Standard	Select tone plan for the country of operation
G.722 Timestamps	Choices are 160/20ms or 320/20ms
G.723.1 Bit Rate	Choices are 5.3kb/s or 6.3kb/s
Enable VAD	Enable or disable Voice Activity Detection (VAD). If VAD is enabled, G729 Payload length cannot be set greater than 20 mSec.
DTMF Payload Type	The RTP Payload type that indicates DTMF. Default is 101
Volume Settings	
Handset/Hands-free Input Volume	Handset/Hands-free Input Volume levels
Handset Output Volume	Handset Output Volume levels
Hands-free Output Volume	Hands-free Output Volume levels
Ring Volume	Speaker Ring Volume levels
Codec Gain Settings	
Hands-free Hardware MIC Gain	Settings Hands-free Hardware MIC Gain
Hands-free Hardware Speakerphone Gain	Settings hands-free Hardware Speakerphone Gain

b) FEATURE

This page configures various features such as Hotline, Call Transfer, Call Waiting and Block Out.







Field Name	Explanation
Feature Settings	
DND (Do Not Disturb)	DND might be disabled, phone for all SIP lines, or line for SIP individually.
Ban Outgoing	If enabled, no outgoing calls can be made.
Enable Call Transfer	If enabled, Call Transfer is allowed.
Enable Call Waiting	If enabled, notifies user of a second call during a call. Caller ID of the new caller will be displayed. Press HOLD button to place existing call on hold and answer new call. Press HOLD again to return to first call.
Semi-Attended Transfer	If enabled, Semi-Attended Transfer is allowed.
Enable 3-way Conference	If enabled, allows 3-way conference.
Enable Auto Hand-down	If enabled in speakerphone mode, the equipment will automatically hang up and return to idle when the distant party terminates the call. In handset mode, it will play dial tone instead of returning to idle.
Accept Any Call	If enabled, the equipment will accept a call even if the called number does not belong to the phone.
Auto Hand-down Time	Wait time before the equipment performs the Auto Hand-down behavior described above.
Enable Call Completion	If this feature is enabled, digits dialed on-hook will be transmitted when the phone goes off-hook.
Enable Auto Redial	If enabled, the equipment will automatically redial a call if a busy tone is received.
Auto Redial Interval	Wait time between auto redial attempts in seconds.
Enable Silent Mode	If enabled, the equipment will not ring to indicate a new call. Instead, the light below the key pad will blink to indicate a new call.

Field Name	Explanation
Auto Redial Times	Maximum numbers of auto redial attempts.
Hide DTMF	This feature sets how DTMF digits are displayed after a call is in progress.
Auto Headset	Automatically answers call on headset.
Ring From Headset	If this is enabled and a headset is connected, ring tone will be played in the

	headset.
Enable Intercom	If enabled, allows intercom calls.
Enable Intercom Mute	If enabled, mutes incoming calls during an intercom call
Enable Intercom Tone	If enabled, plays intercom ring tone to alert to an intercom call.
Enable Intercom Barge	If enabled, the equipment wills auto-answer an intercom call during an outside call. If an intercom call is established, a second intercom call will be rejected.
P2P IP Prefix	Set Prefix for peer to peer IP call. For example: You wish to dial 192.168.1.119. If the P2P IP Prefix is defined as 192.168.1., it is only necessary to dial #119. The default is ".". If this box is left blank, IP dialing is disabled.
DND Return Code	Specify SIP Code returned for DND. Default is 480 - Temporarily Not Available.
Turn Off Power Light	Disables Power Light if selected.
Busy Return Code	Specify SIP Code returned for Busy. Default is 486 – Busy Here.
Emergency Call Number	And multi numbers can be added by ",", such as 911,999
Reject Return Code	Specify SIP Code returned for Rejected call. Default is 603 – Decline.
Enable Password Dial	When a number is entered beginning with the password prefix, the following N numbers after the password prefix will be displayed as *. N is the value entered in the Password Length field. For example: If the password prefix is 3 and the Password Length is 2, then dialing the number 34567 will display 3**67 on the equipment.
Active URI Limit IP	IP address of the server for the Action URL messages described below.
Password Dial Prefix	Prefix for password dialing as described above.
Push XML Server	IP address for XML server which can send display content to the equipment.
Password Length	Length for password dialing as described above.
Enable Call Waiting Tone	Enables audible notification of call waiting.
Enable Multi Line	Enable phone to make calls for 10 lines max, or disable for 2 lines max.
IP Description	device IP description
Enable Auto Answer	Enable Auto Answer function
Auto Answer Timeout	Set Auto Answer Timeout

Field Name	Explanation
Enable Speed Dial Hand-down	Enable Speed Dial Hand-down function
Status Led Reuse Mode	Configuration Open / Close state light multiplexing mode.
Dial Number Voice Play	Configuration Open / Close Dial Number Voice Play
Time of Dial Switch	Set time of Dial Switch
Action URL Settings	
URL for various actions performed by the phone. These actions are recorded and sent as xml files to the	

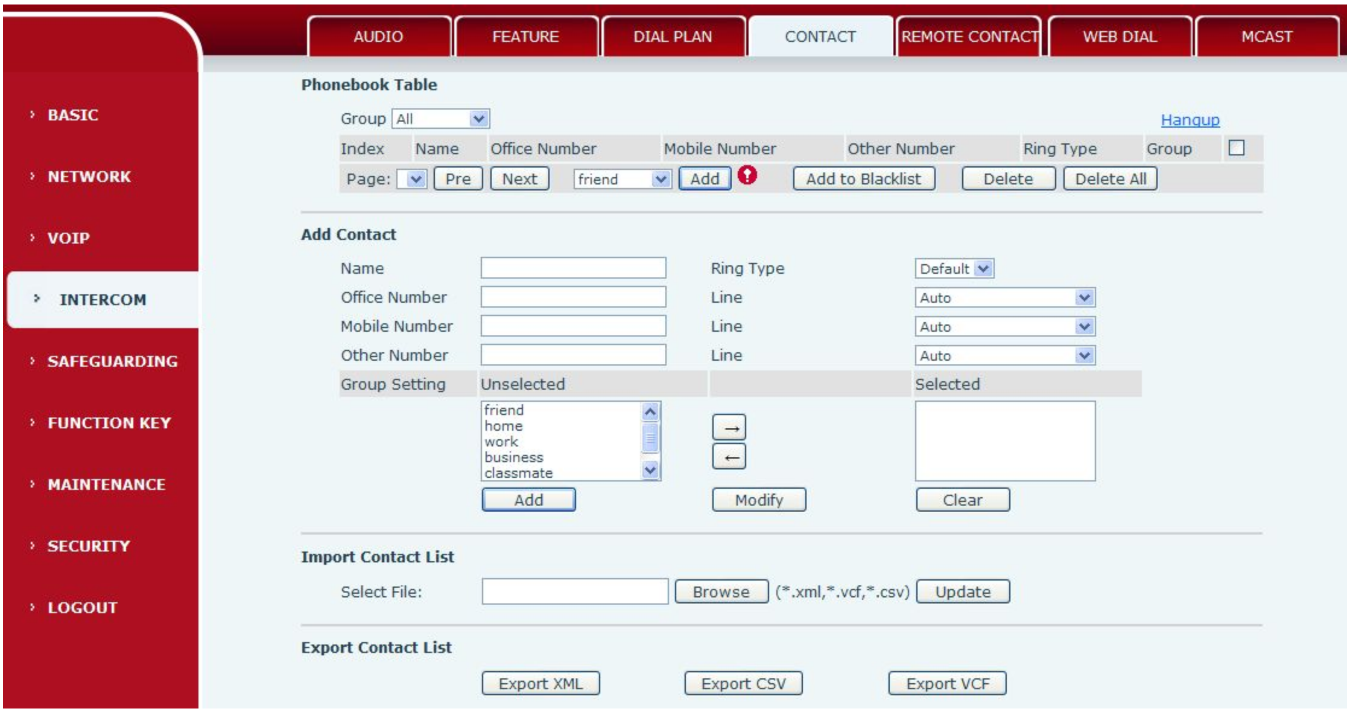
server. Sample format is http://InternalServer /FileName.xml
Block Out Settings
Add or Delete Blocked numbers – Enter the prefix of numbers which should not be dialed by the phone. For example, if 001 is entered, the phone will not dial any numbers beginning with 001. X and x are wildcards which match 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.

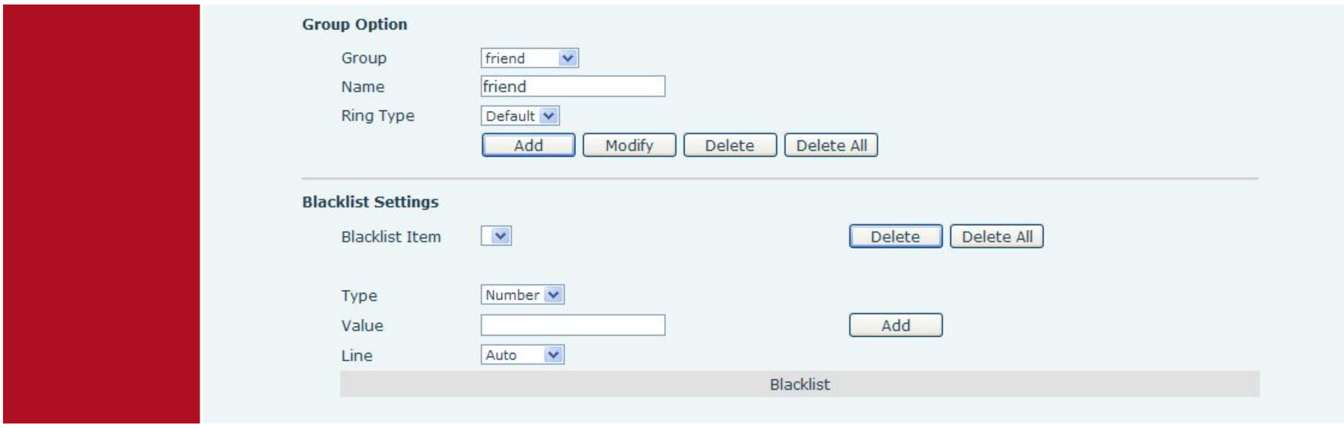
c) DIAL PLAN

Dial plan	
Field Name	**Explanation**
Basic Settings	
End with “#"	Dial the desired number, and press # to send it to the server.
Fixed Length	The number will be sent to the server after the specified numbers of digits are dialed.
Time out	Number will be sent to the server after the specified time.
Press # to Do Blind Transfer	Press # after entering the target number for the transfer. The equipment will transfer the current call to the third party.
Blind Transfer on hook	Hang up after entering the target number for the transfer. The equipment will transfer the current call to the third party.
Attended Transfer on hook	Hang up after the third party answers. The equipment will transfer the current call to the third party.

d) CONTACT

Enter the name, phone number and ring type for each contact here.

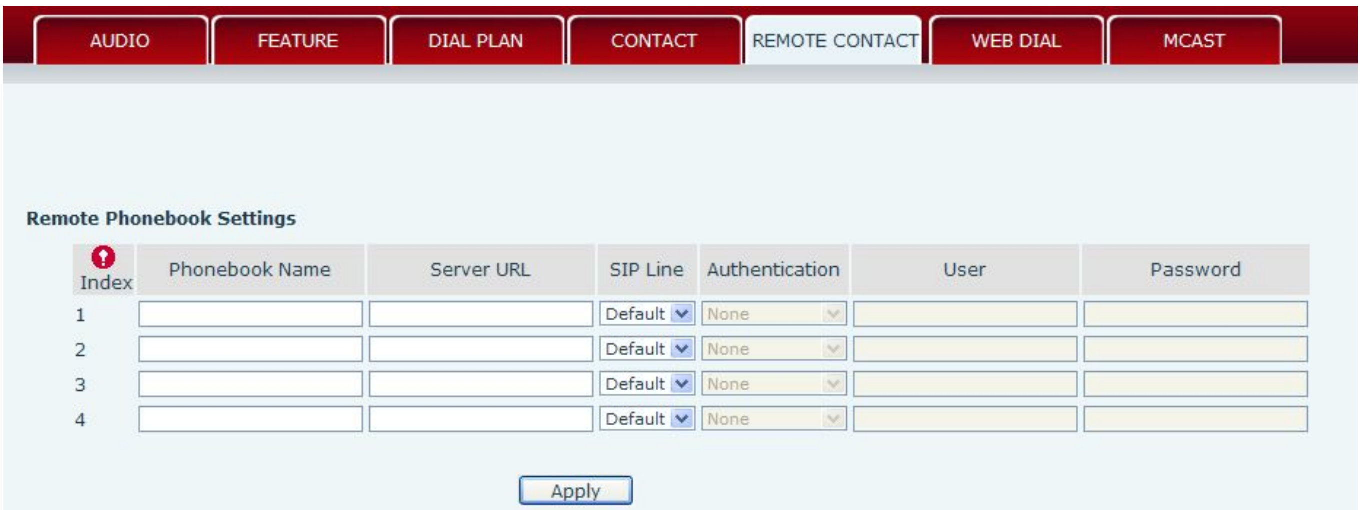




Phonebook	
Field Name	**Explanation**
Phonebook Table	
Name	Contact name
Number	Contact phone numbers
Ring Type	Ring type for this contact
Group	Dropdown box to select group
Note: the capacity specified phone book is up to 500 records. You can add one or more add a contact to a group or a black list, click Delete to delete multiple contacts, click delete all delete all contacts have been added.	
Add Contact	
Name	Contact name
Office Number	Contact phone numbers

Mobile Number	
Other Number	
Ring Type	**Ring type for this contact**
Line	**Select line for associated contact number**
Group Setting	**Choose the group or groups for this contact and move them to the Selected list on the right.**
Note: click on the Add button to add a new contact, click the Edit button can modify add contact information, click the delete button can fill the empty has contact information.	
Import Contact List	
Click the browse button to select the phonebook file to import. Then click the update button and the selected file will be added to the phone. File must be xml, vcf or csv format.	
Export Contact List	
Export contacts to xml file, csv file, vcf file.	
Field Name	**Explanation**
Group Option	
Group	**Lists existing groups**
Name	**Enter name for new group**
Ring Type	**Ring type for group**
Blacklist Settings	
Note: The maximum capability of the phonebook is 500 contacts. Note: “x” and “.” are special characters in the black list. “x” matches any single digit and “.” matches any number of digits. For example, “4xxx” matches any 4 digit number beginning with 4. “6.” Matches any digit string beginning with 6. Note: There is also an allowed number list feature if the user only wants to allow a limited access to the phone. To use this, precede the number with “-“. For example, -123456, or -1234xx. Allowed number lists must end with an entry which is only a “.”.	

e) REMOTE CONTACT

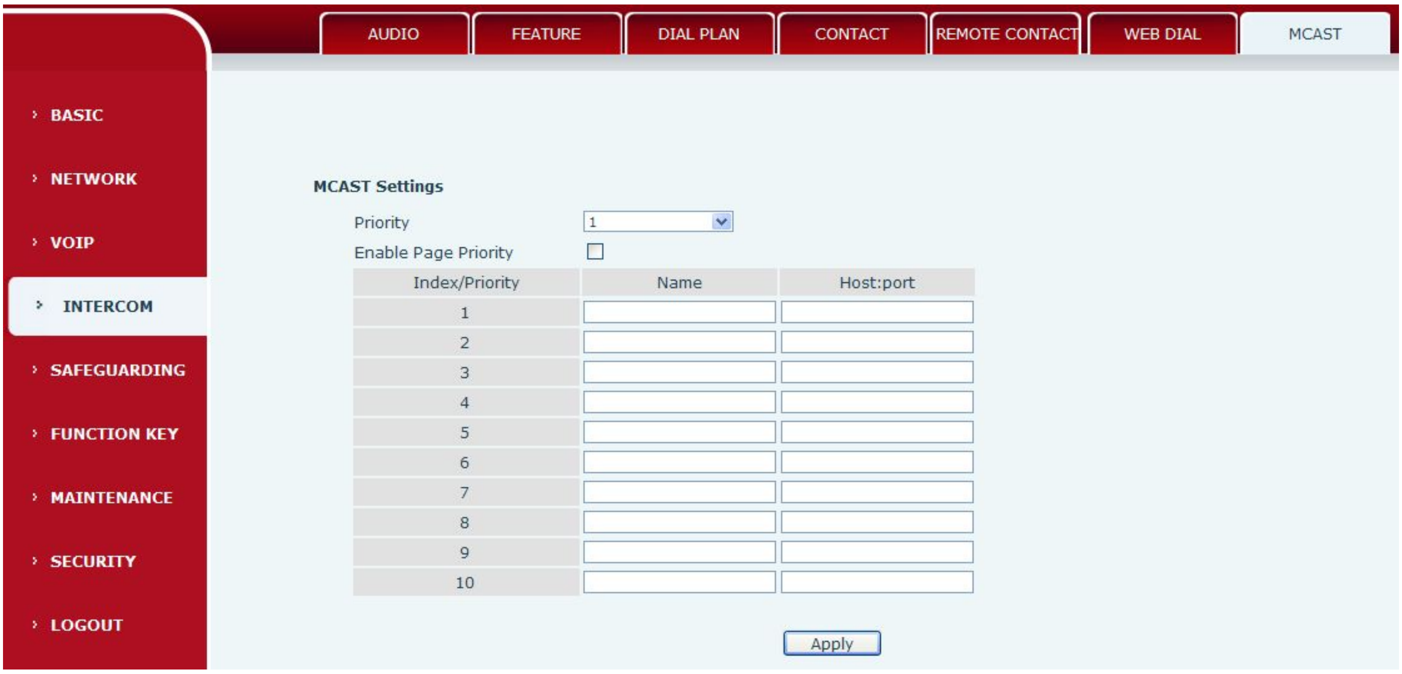


Remote Phonebook Settings	
Field Name	**Explanation**
Phonebook name	Phonebook name displayed on the phone.
Server URL	Server URL of the remote phonebook.
SIP line	SIP line for the remote phonebook.
Authentication	Authentication mode for remote phonebook.
User	Authentication username.
Password	Authentication password.

f) WEB DIAL

This feature allows a call to be initiated by a computer. To place a call, enter the number in the Dial Number box, select the line in the Line Selection box and press the Dial button. To end the call, press the Hang-up button.

g) MCAST



Using multicast functionality can be simple and convenient to send notice to each member of the

multicast, through setting the multicast key on the device, sending multicast RTP stream to pre-configured multicast address. By on the device configuration monitoring multicast address, listen to and play the group multicast address send RTP stream.

MCAST Settings

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

In the Web interface setting change equipment receiving multicast RTP stream processing mode are: set the ordinary priority and enable page priority.

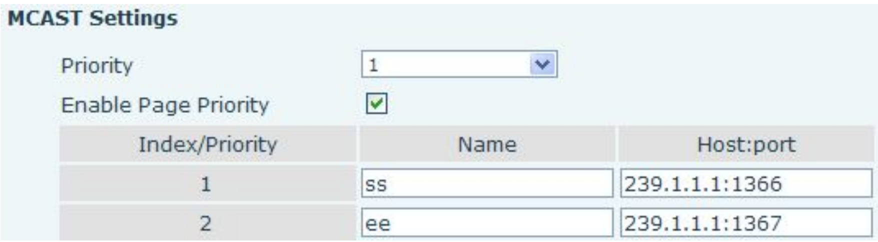
- **Priority:**

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 flow. 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: The definition of common call priority, 1 is the most advanced, most low 10**
  - ✧ **Disable: ignore all incoming stream multicast RTP**
  - ✧ **enable the page priority:**

  Page determines the priority equipment current in multicast session, how to deal with the new receiving multicast RTP stream, enabling the Page switch priority, the device will automatically ignore the low priority of multicast RTP stream, receive priority multicast RTP stream, and keep the current multicast session in state; If is not enabled, the device will automatically ignores all receive multicast RTP stream.

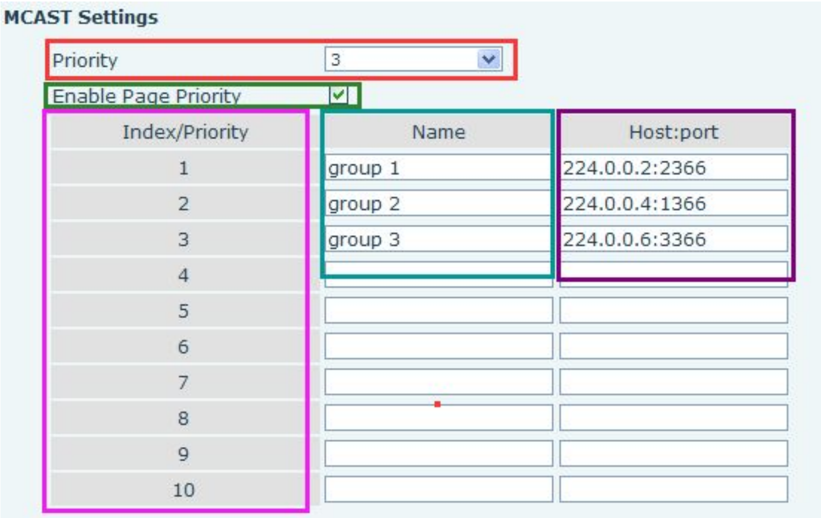
- **Web Settings:**



The multicast SS priority is higher than that of EE, the highest priority;

Note: when a multicast session key by multicast, multicast sender and receiver will beep.

Listener configuration



- **Blue part (name)**

The "group of 1" and "2" and "3" are you setting monitoring multicast name, answer time is displayed on the screen, if you do not set the screen will display the IP: port directly

- **Purple part (host: port)**

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 the number of higher priority

- **Red part (priority)**

Is the general call, non multicast call priority, the smaller the number of high priority, the following will explain how to use this option:

✧ The purpose of setting monitoring multicast "group 1" or "2" or "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".

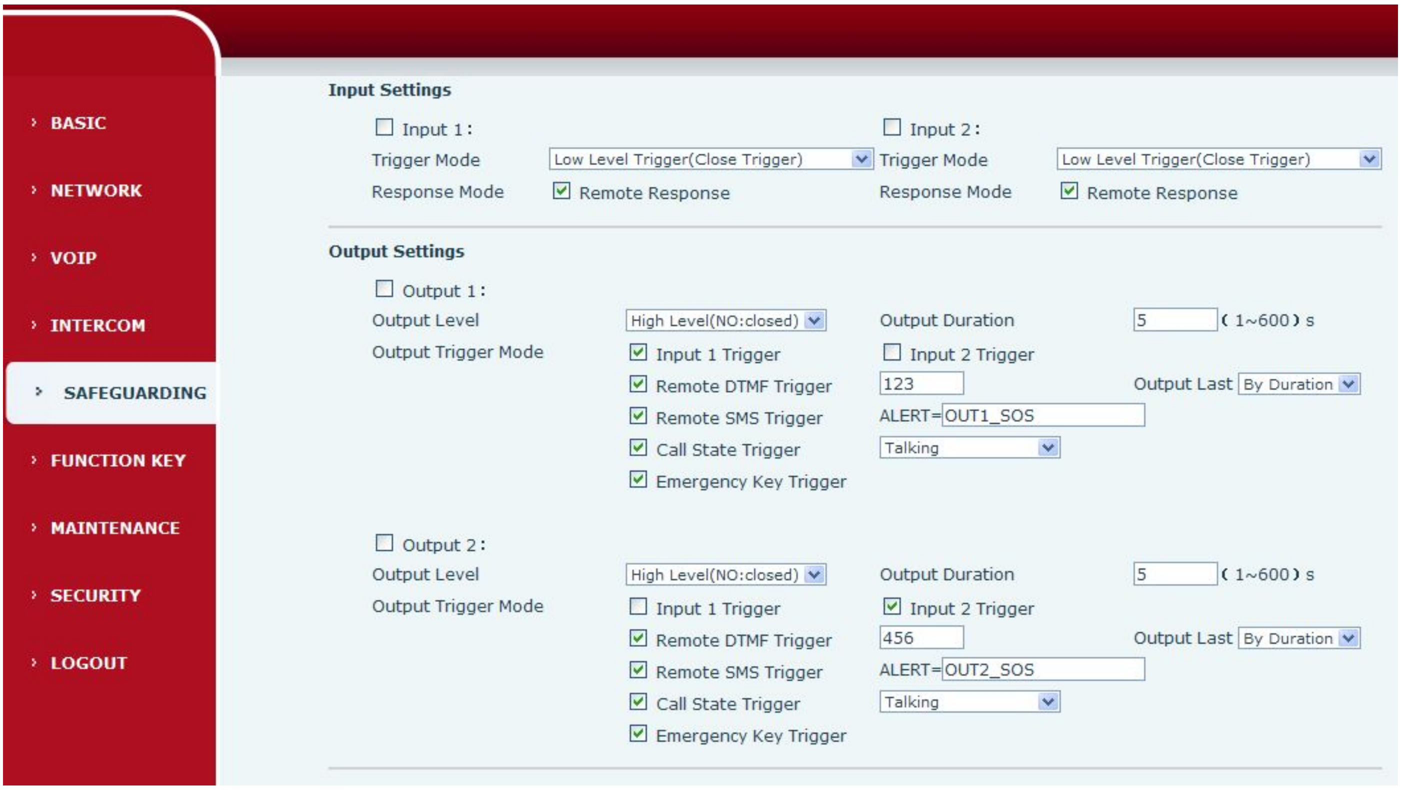
✧ If multicast is a new "group of 1", because "the priority group 1" is 2, higher than the current call "priority group 2" 3, so multicast call will can come in.

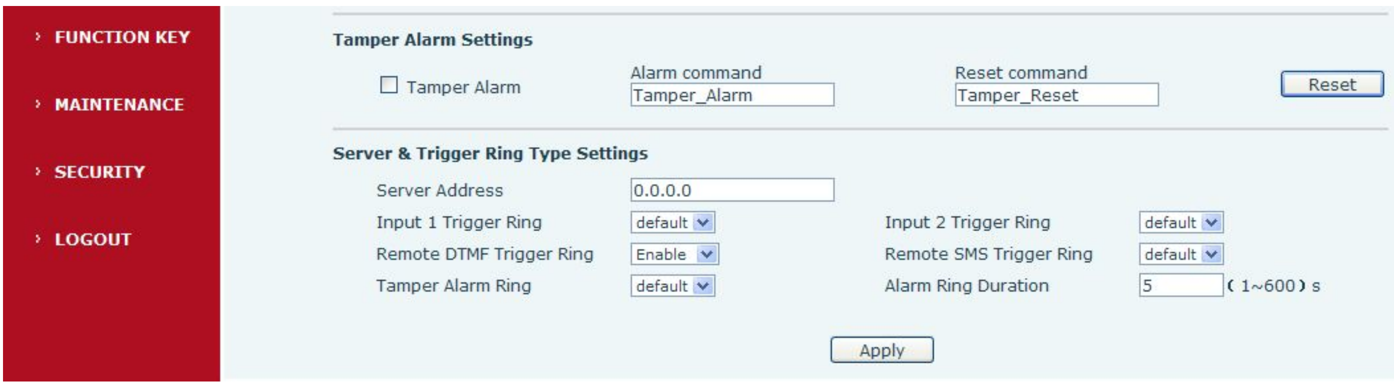
✧ If multicast is a new "group of 3", because "the priority group 3" is 4, lower than the current call "priority group 2" 3, "1" will listen to the equipment and maintain the "group of 2".

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

(5)SAFEGUARDING





Security Settings	
Field Name	**Explanation**
Input settings	
Input 1	**Open /Close Input port1**

Trigger Mode	**When choosing the low level trigger (closed trigger), detect the input port 1 (low level) closed trigger.**
	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**
Trigger Mode	**When choosing the low level trigger (closed trigger), detect the input port 2 (low level) closed trigger.**
	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**
Output Settings	
Output 1/2	**Open/close, Output 1/Output 2**
Output Level	**When choosing the low level trigger (NO: normally open), when meet the trigger condition, trigger the NO port disconnected.**
	When 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.**

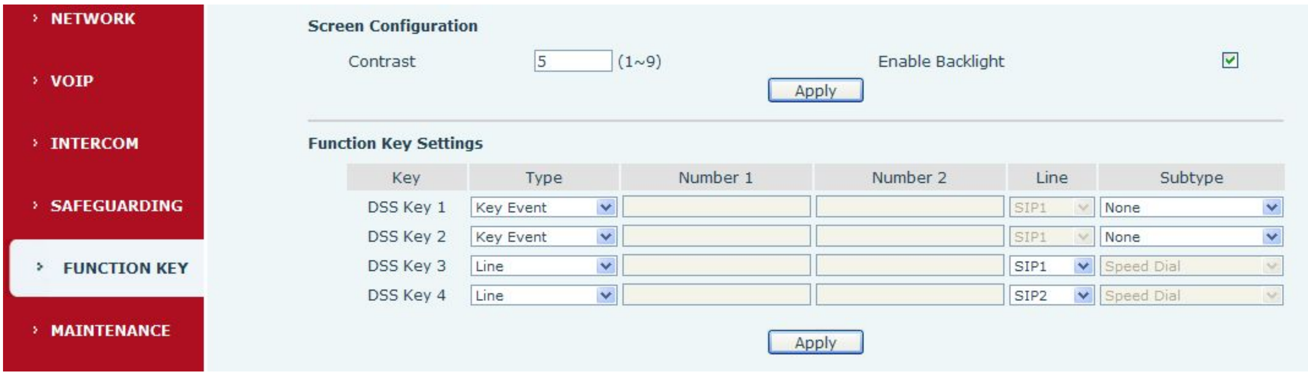
Field Name	**Explanation**	
Output Trigger Mode: There are many kinds of trigger modes, multiple choices.		
Input port1 trigger	**When the input port1 meet to trigger condition, the output port1 will trigger(The Port level time change, By < Output Duration > control)**	
Input port2 trigger	**When the input port2 meet to trigger condition, the output port2 will trigger(The Port level time change, By < Output Duration > control)**	
Remote DTMF trigger	**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)**
	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 trigger	**In the remote device or server to send instructions to ALERT=[instructions], if correct, which 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 default state)
Tamper Alarm Settings	
Tamper Alarm	When the selection is enabled, the tamper detection enabled
Alarm command	When detected someone tampering the equipment, will be sent alarm to the corresponding server
Reset command	When the equipment receives the command of reset from server, the equipment will stop alarm
Reset	Directly stop the alarm from equipment in the Webpage
Server & Trigger Ring Type Settings	
Server Address	Configure remote response server address(including remote response server address and tamper alarm server address)
Input 1 trigger ring	When the input port 1 triggering condition is satisfied, the corresponding ring tone or alarm
Input 2 trigger ring	When the input port 2 triggering condition is satisfied, the corresponding ring tone or alarm

Field Name	Explanation
Remote DTMF trigger ring	When received the remote DTMF command, whether to output the ringtone
Remote SMS trigger ring	When receiving the remote SMS instructions, whether to output the ringtone
Tamper alarm ring	When the detected someone tampering the equipment, plays the corresponding ringtone or alarm
Alarm ring duration	duration of alarm ring(not including tamper alarm)

(6)FUNCTION KEY

The equipment has four programmable keys (depending on the hardware configuration), you can set different for each key function respectively, the list below you can set up some of the functions and the related introduction, every button by default is N/A, namely the default doesn't set any function.



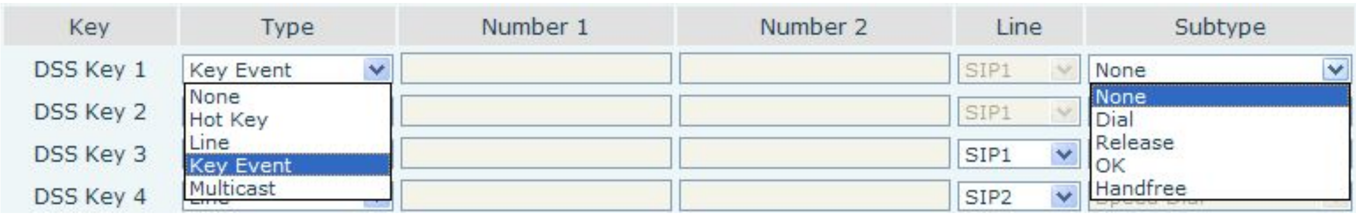
a) Screen settings

Field Name	Explanation
Contrast	Set screen contrast
Enable Backlight	Enable/disable LCD backlight.

b) Function key settings

- **Key Event Settings**

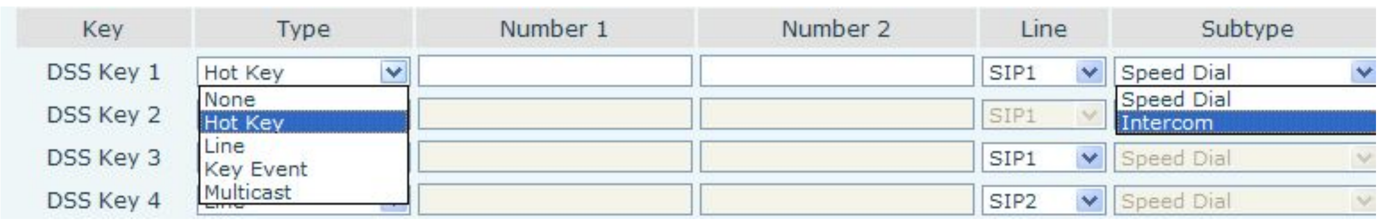
The Subtype configuration of Hot key.



DSS key type	Subtype	Usage
Key Event	None	Not responding
	Dial	Dial function
	Release	End calls
	OK	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.



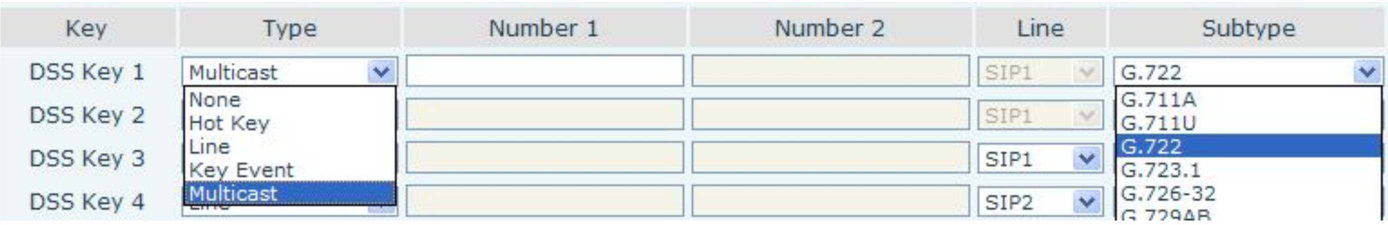
DSS key type	Number	Line	Subtype	Usage
Hot key	Fill the	The SIP	Speed	In Speed dial mode,

	called party's SIP account or address	account corresponding lines	Dial	with Enable Speed Dial Handdown 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:



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)
		G.711U	
		G.722	Wideband speech coding (7Khz)
		G.723.1	Narrowband speech coding (4Khz)
		G.726-32	
		G.729AB	

✧ **operation mechanism**

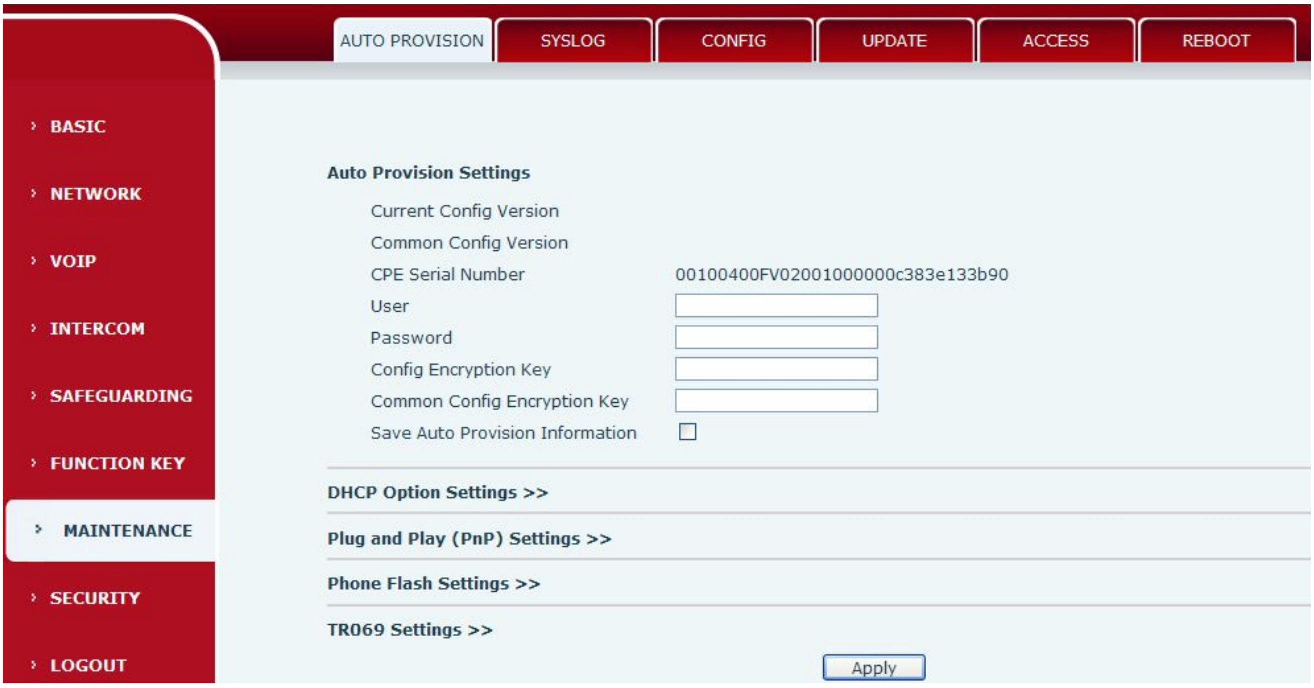
Device through the DSS Key configuration of multicast address and port and started coding; set by WEB to monitor the multicast address and port; device sends a multicast, listens to the address of the device can receive the multicast content.

✧ **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.

(7)MAINTENCE

a) AUTO PROVISION



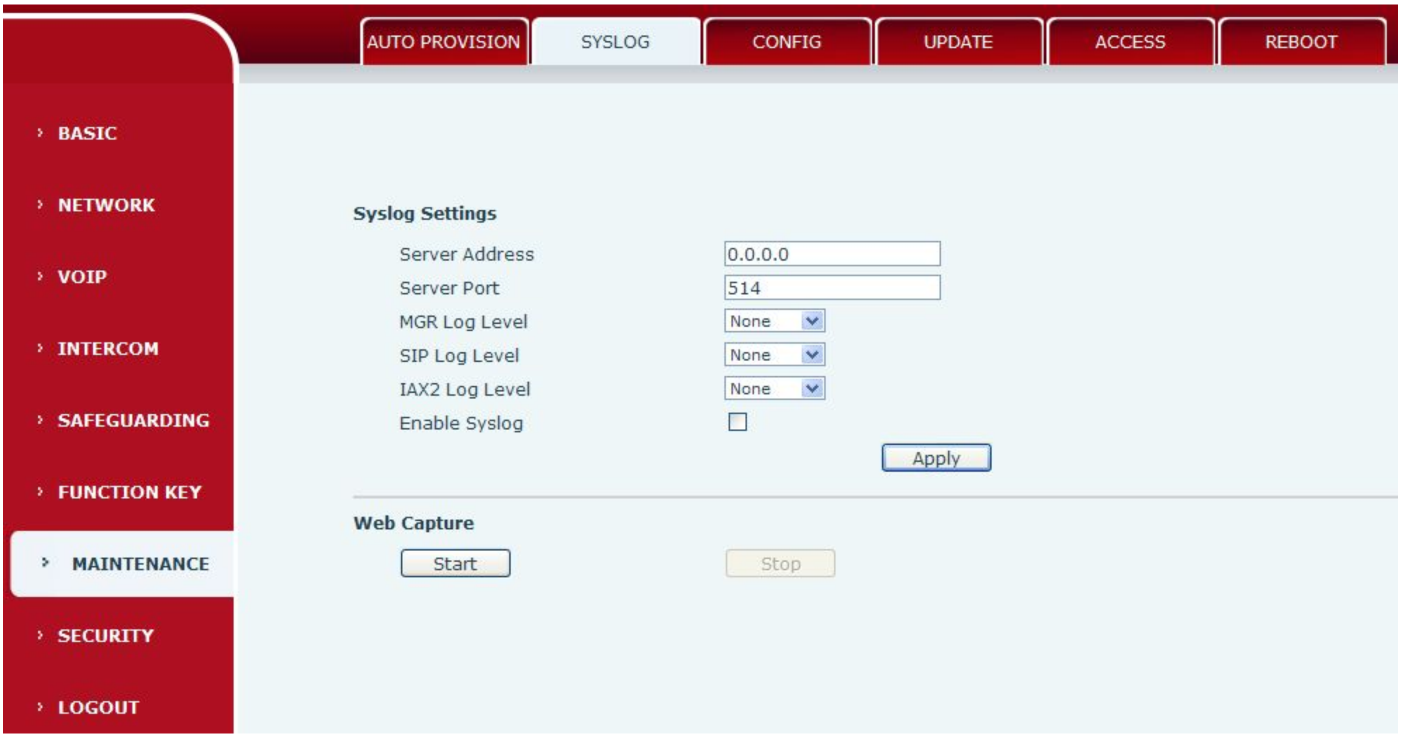
The equipment supports PnP, DHCP, and Phone Flash to obtain configuration parameters. They will be queried in the following order when the equipment boots.

DHCP option → PnP server → Phone Flash

Field Name	Explanation
Automatic update configuration	
Current Config Version	Show the current config file's version. If the version of configuration downloaded is higher than this, the configuration will be upgraded. If the endpoints confirm the configuration by the Digest method, the configuration will not be upgraded unless it differs from the current configuration
Common Config Version	Show the common config file's version. If the configuration downloaded and this configuration is the same, the auto provision will stop. If the endpoints confirm the configuration by the Digest method, the configuration will not be upgraded unless it differs from the current configuration.
CPE Serial Number	Serial number of the equipment
User	Username for configuration server. Used for FTP/HTTP/HTTPS. If this is blank the phone will use anonymous
Password	Password for configuration server. Used for FTP/HTTP/HTTPS.
Config Encryption Key	Encryption key for the configuration file
Common Config Encryption Key	Encryption key for common configuration file
Save Auto Provision Information	Save the auto provision username and password in the phone until the server url changes
DHCP Option Settings	

DHCP Option Setting	**The equipment supports configuration from Option 43, Option 66, or a Custom DHCP option. It may also be disabled.**
Custom DHCP Option	**Custom option number. Must be from 128 to 254.**
Plug and Play（PnP）Settings	
Enable PnP	**If this is enabled, the equipment will send SIP SUBSCRIBE messages to a 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.**
Field Name	**Explanation**
Phone Flash Settings	
Server Address	**Set FTP/TFTP/HTTP server IP address for auto update. The address can be an IP address or Domain name with subdirectory.**
Config File Name	**Specify configuration file name. The equipment will use its MAC ID as the 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.**
Update Mode	**1. Disable – no update** **2. Update after reboot – update only after reboot.** **3. Update at time interval – update at periodic update interval**

b) SYSLOG



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.

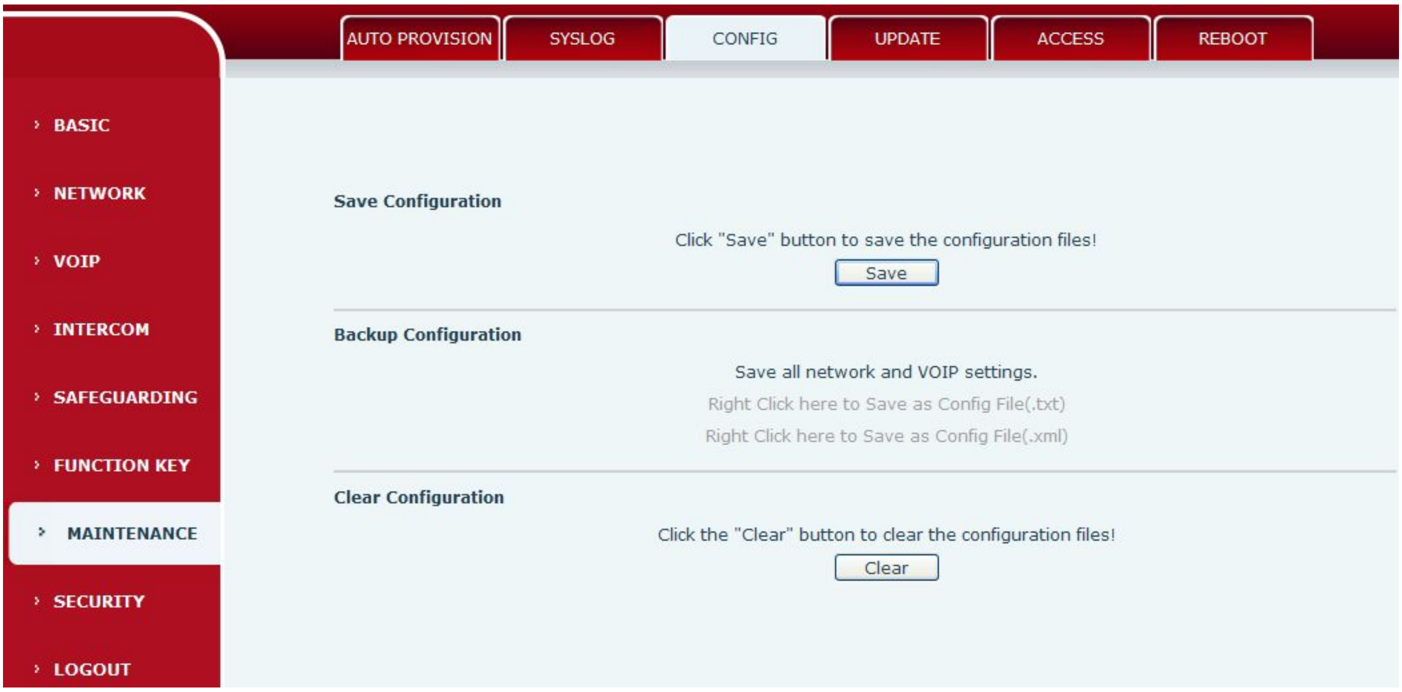
Level 5: notice; It is the normal but significant condition.

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
System log 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.
IAX2 log level	Set the level of IAX2 log.
Enable system log	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

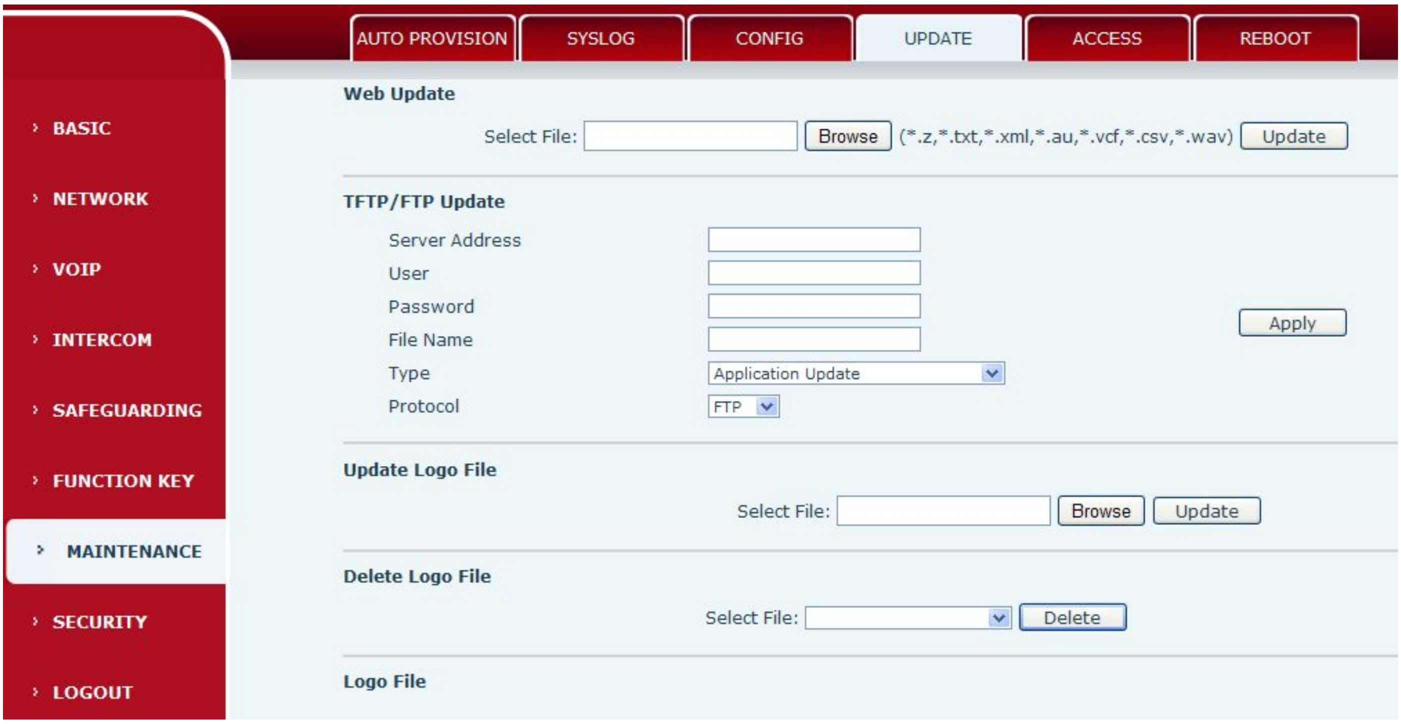
c) CONFIG



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

d) UPADTE

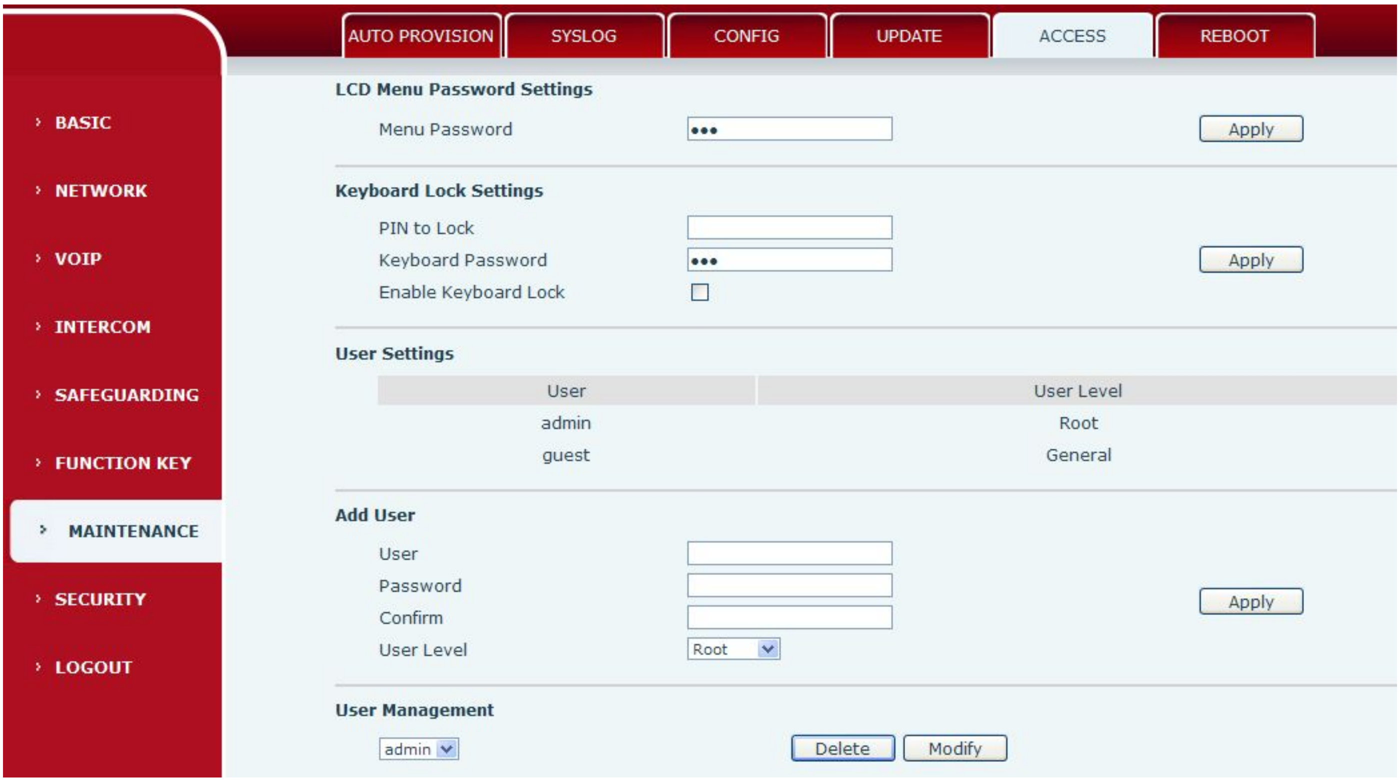
This page allows uploading configuration files to the equipment.



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.
TFTP/FTP Update	
Server	FTP/TFTP server address for download/upload. The address can be IP address or Domain name with subdirectory.
User	FTP server Username for download/upload.
Field Name	Explanation
Password	FTP server password for download/upload.
File Name	Name of update file or config file. The default name is the MAC of the equipment
Note: The exported config file can be modified. The config file is made up of modules. Modules which do not need changes may be deleted. For example, a config file can be downloaded and all modules removed except the SIP module. After rebooting, only the SIP settings will be changed	
Type	The system set type : 1. Application update: download system update file 2. Config file export: upload config file to FTP/TFTP server. It can then be named and saved. 3. Config fie import: Download the config file from FTP/TFTP server. The configuration will be effective after the equipment is reset.
Protocol	Select FTP/TFTP server.
UpdateLogoFile	You can update the device Logo file, click [Update] effect.
Delete Logo File	You can delete the device Logo file, click [Delete] effect.

e) ACCESS

Through this page, the user can according to need to add and remove users, can modify existing user permissions.



Field Name	Explanation
Menu Password	Sets the password for entering the setup menu from the equipment keypad. The password must be only digits
Keyboard Lock Settings	
PIN to Lock	Set of keyboard to fast locking the need to enter the password
Keyboard Password	Set of keyboard to unlock the need to enter the password
Enable Keyboard Lock	Open / Close keyboard lock
User Settings	
User	shows the current user name
User level	Show the user level; admin user can modify the configuration. General user can only read the configuration.
Add User	
User	Set User Account name
Password	Set the password
Confirm	Confirm the password
User level	There are two levels. Root user can modify the configuration. General user can 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.

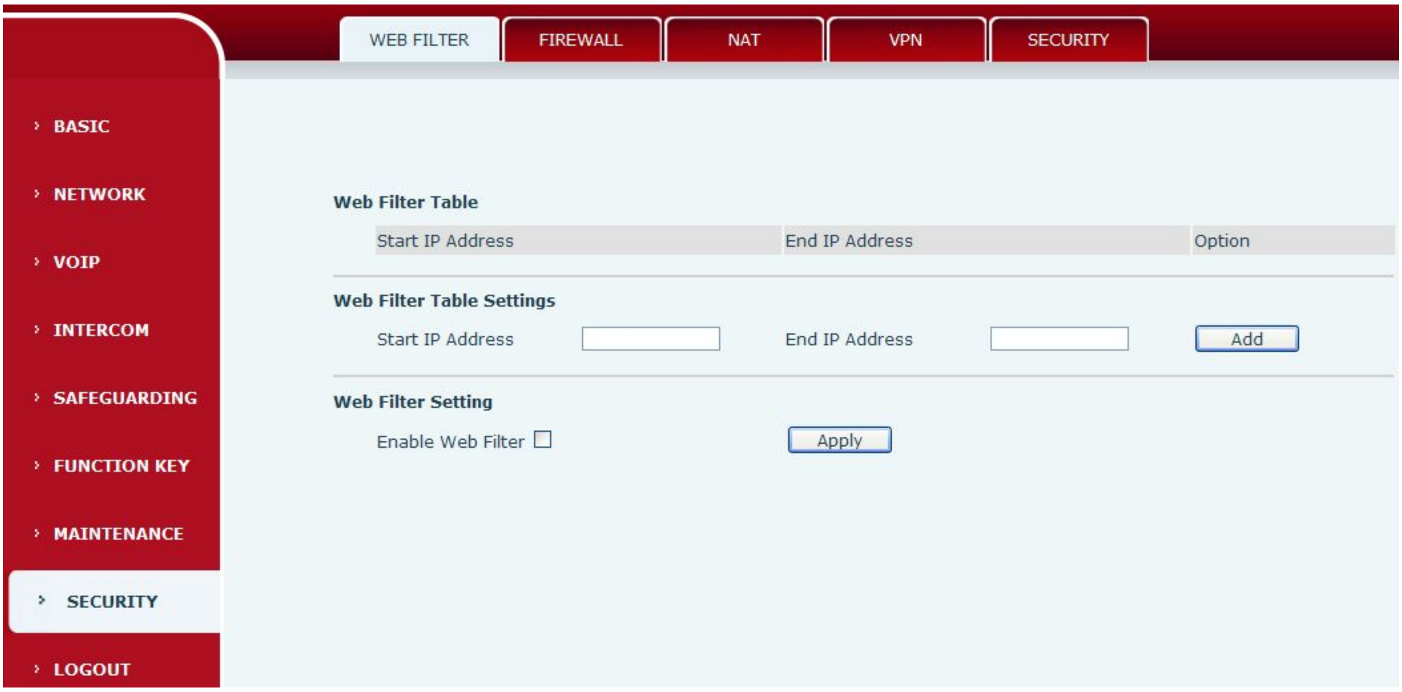
f) REBOOT

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

Note: Be sure to save the configuration before rebooting.

(8)SECURITY

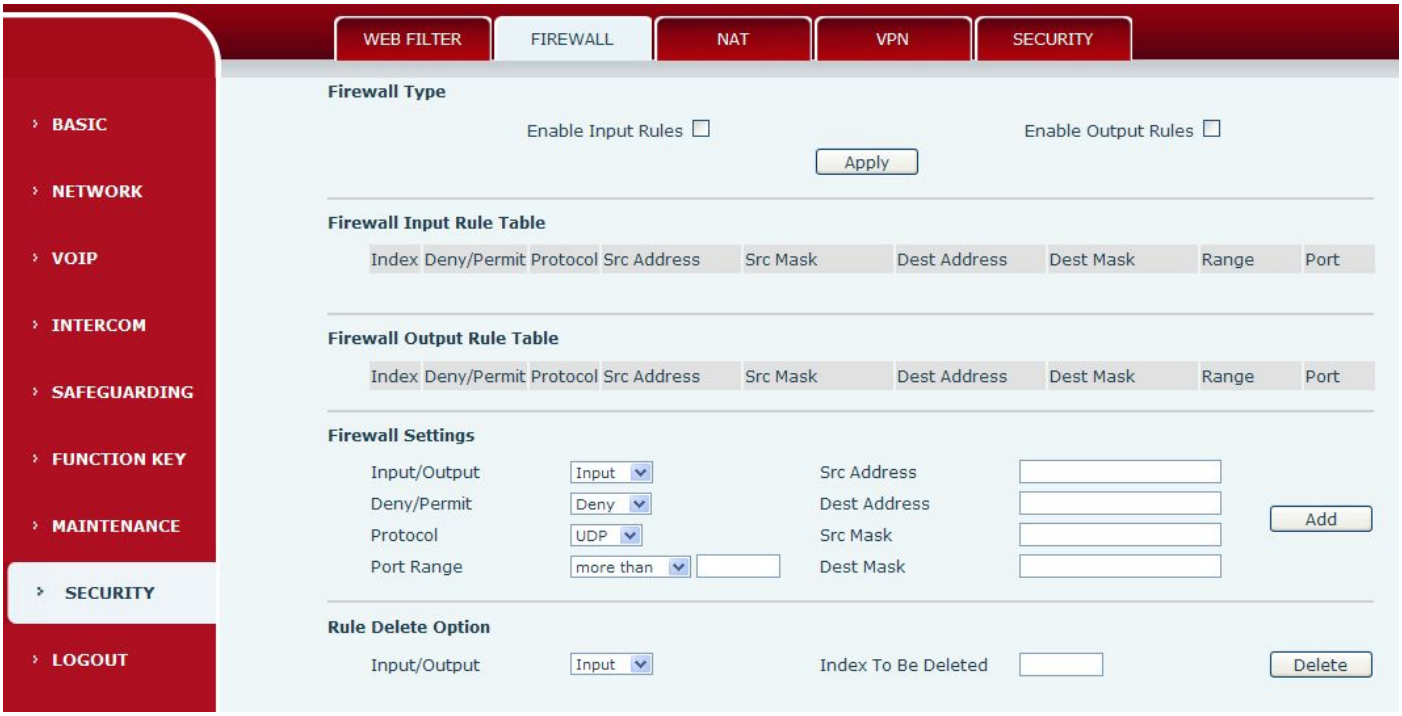
a) WEB FILTER



Web filter	
The Web filter is used to limit access to the equipment. When the web filter is enabled, only the IP addresses between the start IP and end IP can access the equipment.	
Field Name	**Explanation**
Web Filter Table	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.
Note: Be sure that the filter range includes the IP address of the configuration computer.

b) FIREWALL

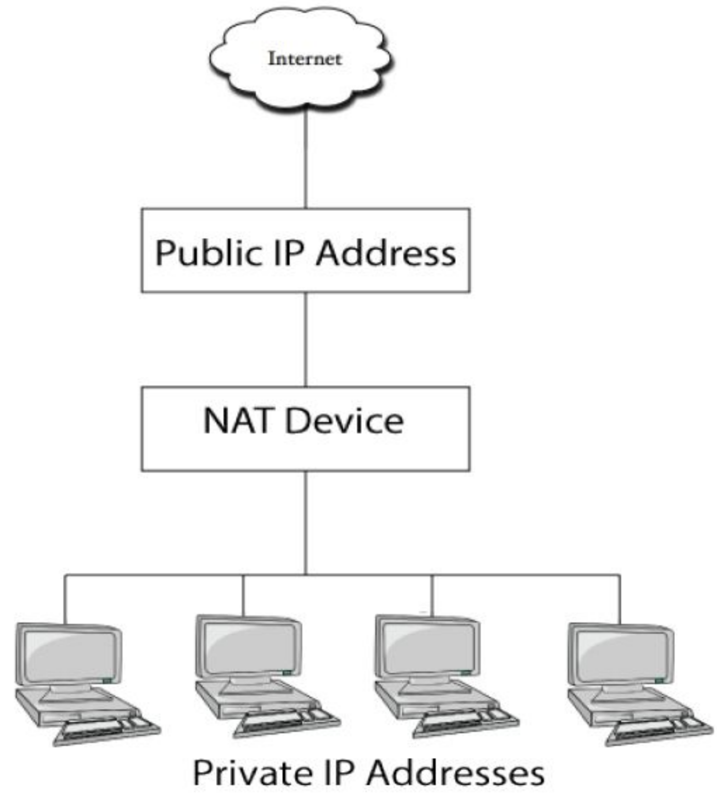


Firewall	
Firewall rules can be used to prevent unauthorized Internet users from accessing private networks connected to this phone (input rule), or prevent unauthorized devices connected to this phone from accessing the Internet (output rule). Each rule type supports a maximum of 10 items.	
Field Name	**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 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 Address	**Set destination address. It can be a single IP address or use * as a wild card. For example: 192.168.1.14 or *.*.*.14.**
Field Name	**Explanation**
Source Mask	**Set the source address mask. For example: 255.255.255.255 points to one host while 255.255.255.0 points to a C type network.**
Destination Mask	**Set the destination address mask. For example: 255.255.255.255 points to one host while 255.255.255.0 points to a C type network.**

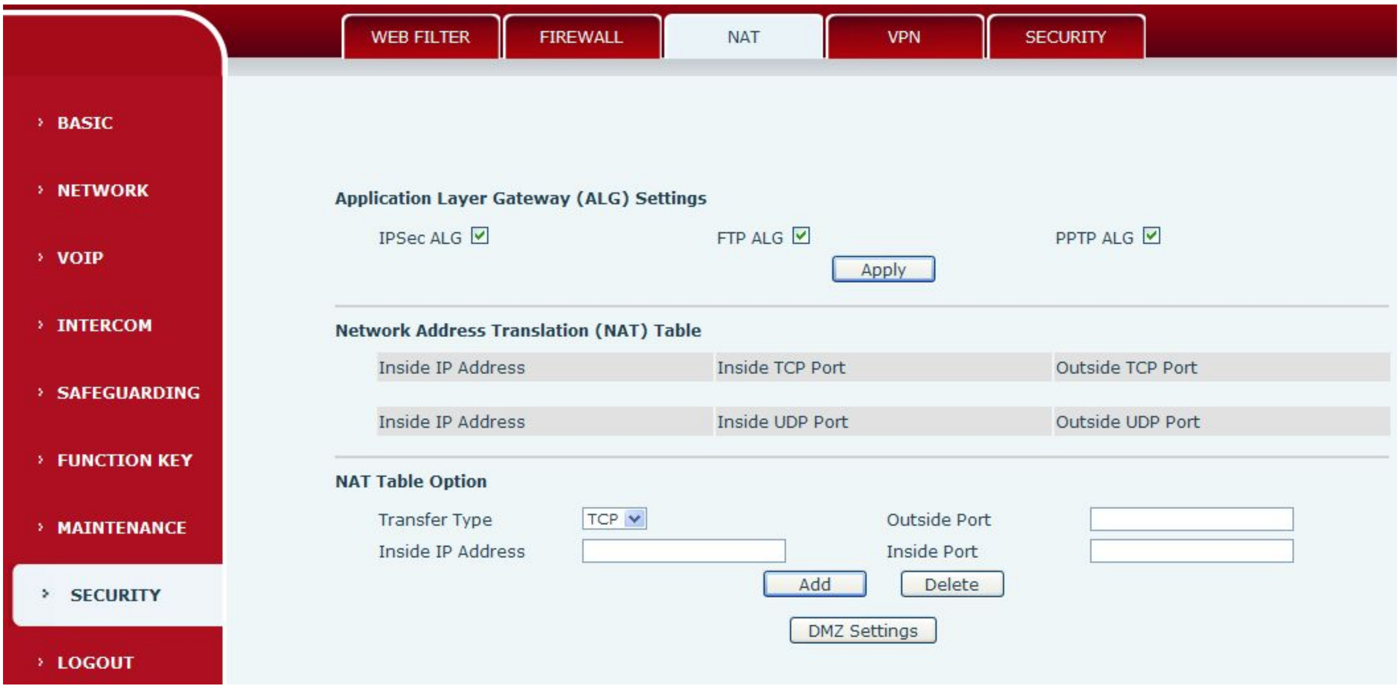
c) NAT

NAT is the process of modifying IP address and port information in transition from a private to a public network. NAT allows the use of one public address to support many private addresses.

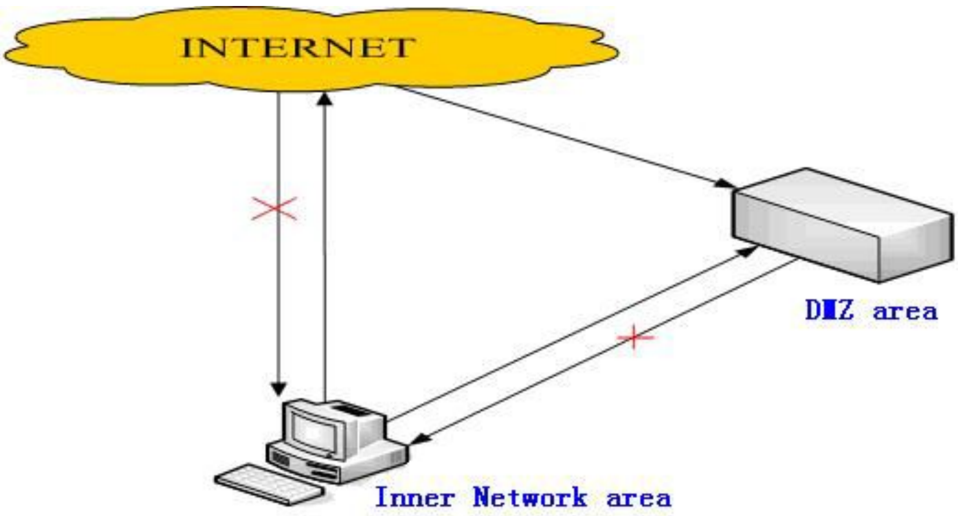


DMZ configuration：

Servers in a network most vulnerable to attack are those which provide services to users outside the local network. Many times these computers are placed into their own sub-network to provide more protection to the rest of the local network. This sub-network is called a DMZ (taken from "demilitarized zone"). Computers in the DMZ have limited connectivity to specific hosts in the internal network, although communication with other hosts in the DMZ and to the external network is allowed. This allows hosts in the DMZ to provide services to both the internal and external network, while a firewall controls the traffic between the DMZ servers and the internal network clients.



The following chart describes the network access control of DMZ.

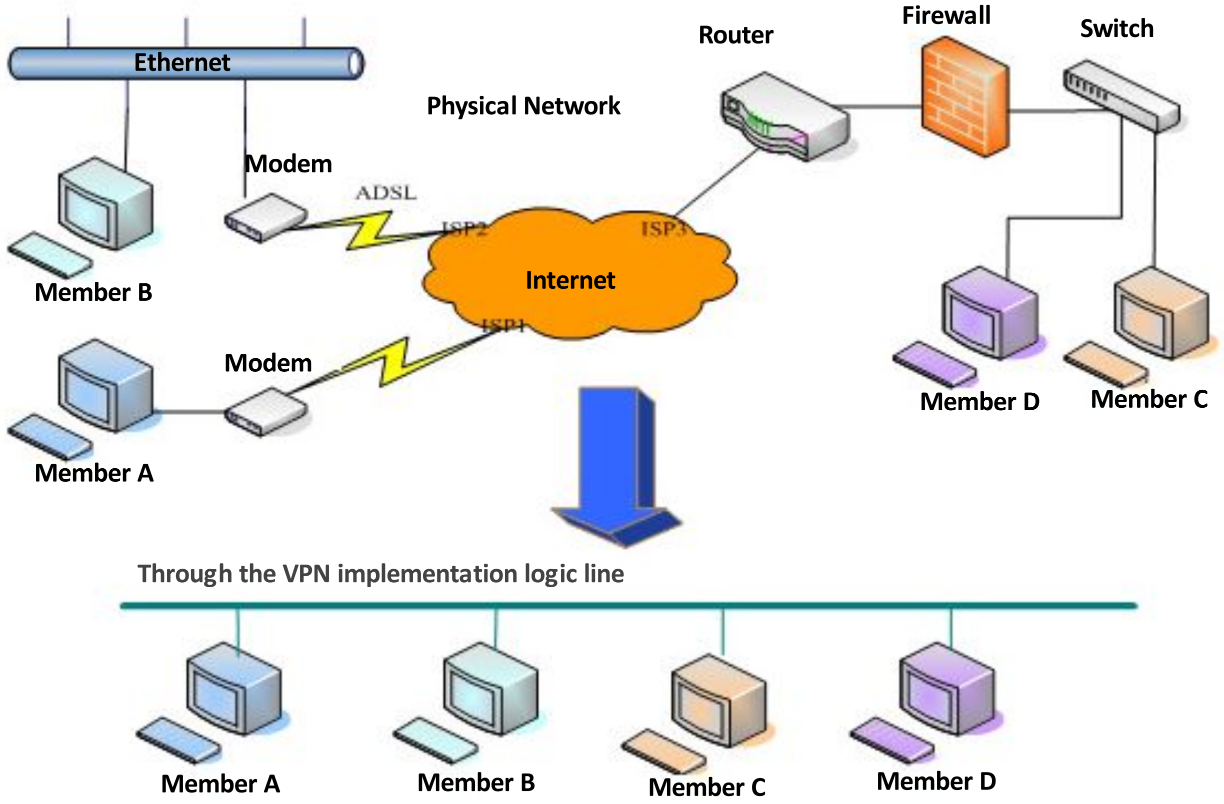


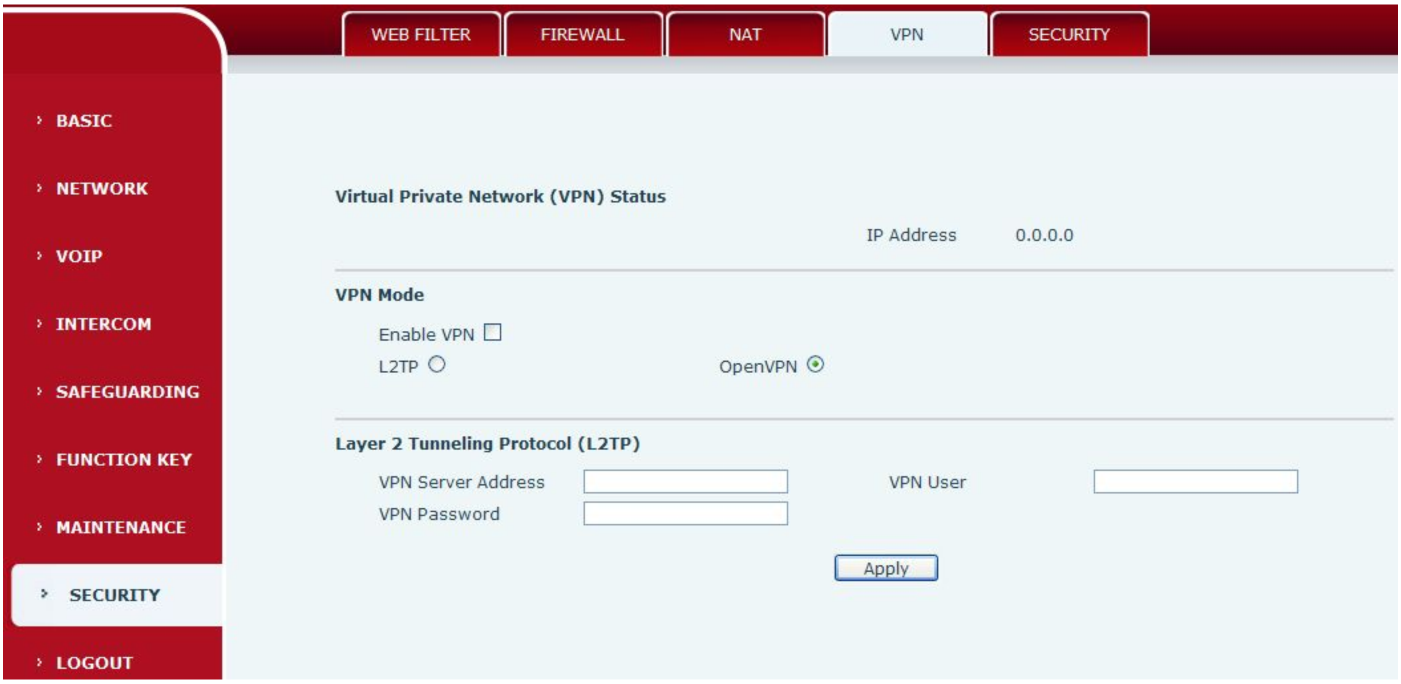
Field Name	**Explanation**
Protocol type Settings	
IPSec ALG	**Enable/Disable IPSec encryption. Default is enabled.**
FTP ALG	**Allow the ALG to securely pass FTP traffic. Default is enabled.**
PPTP ALG	**Allow the ALG to securely pass PPTP traffic. Default is enabled.**
Inside IP Address Inside TCP Port Outside TCP Port	
Shows the NAT TCP mapping tables	
Inside IP Address Inside UDP Port Outside UDP Port	
Shows the NAT UDP mapping tables	
Field Name	**Explanation**
NAT Table Option	
Transfer Type	**Select the TCP or UDP protocol.**
Inside IP	**Set the local IP address of device.**

Address	
Outside Port	Set the WAN (outside) port for NAT mapping
Inside Port	Set the LAN (inside) port for NAT mapping
Note: After entering settings, click the Add button to add new mapping table data. To delete an entry, enter its information and then click the Delete button.	

d) 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.



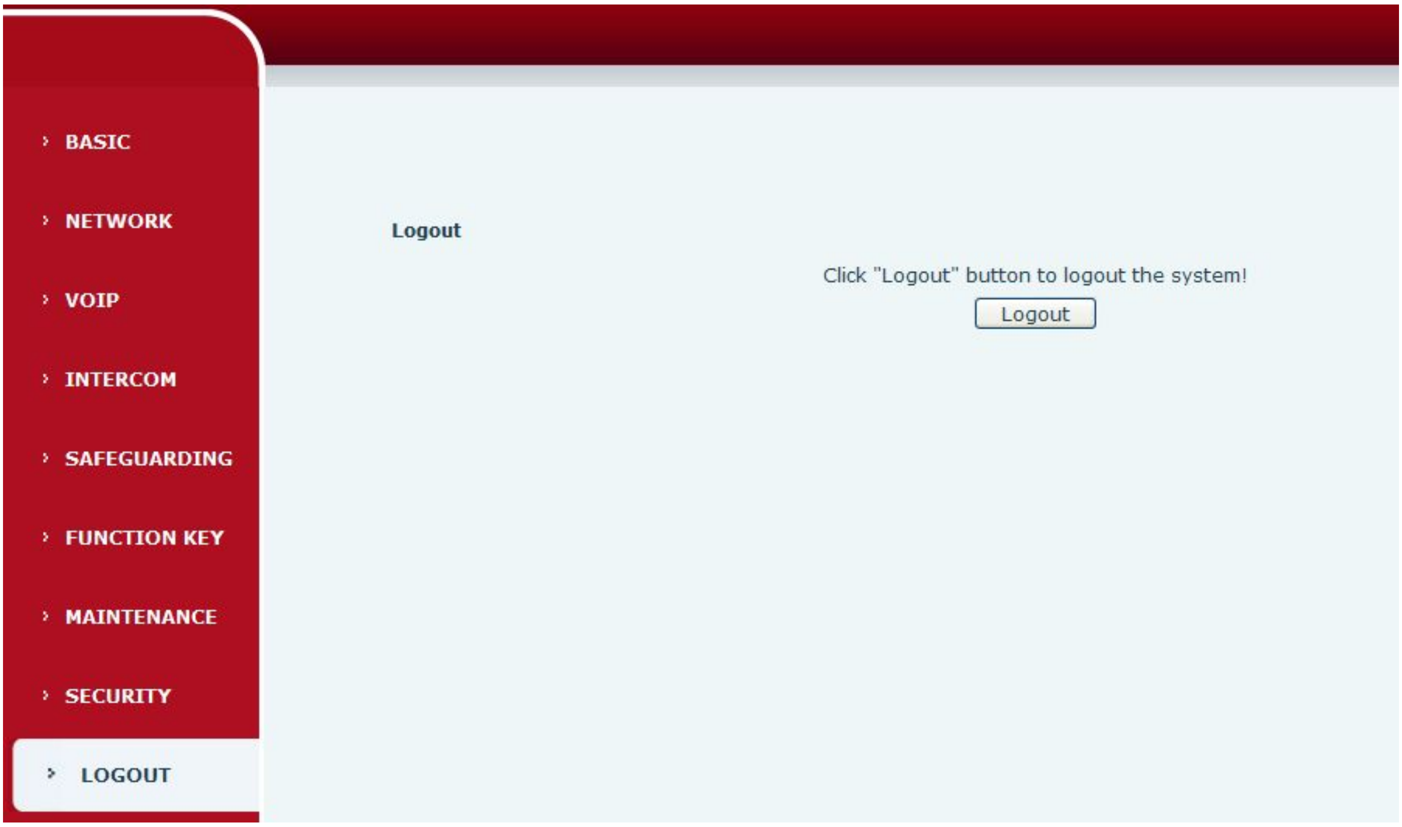


Field Name	Explanation
VPN IP	Shows the current VPN IP address.
VPN type	
Enable VPN	Enable/Disable VPN.
L2TP	Select Layer 2 Tunneling Protocol
Open VPN	Select OpenVPN Protocol. (Only one protocol may be activated. After the selection is made, the configuration should be saved and the phone rebooted.)
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.

e) SECURITY

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.

(9)LOGOUT



Click [Logout] from the web, visit next time when need to enter your user name and password.

E. Appendix

1. Technical parameters

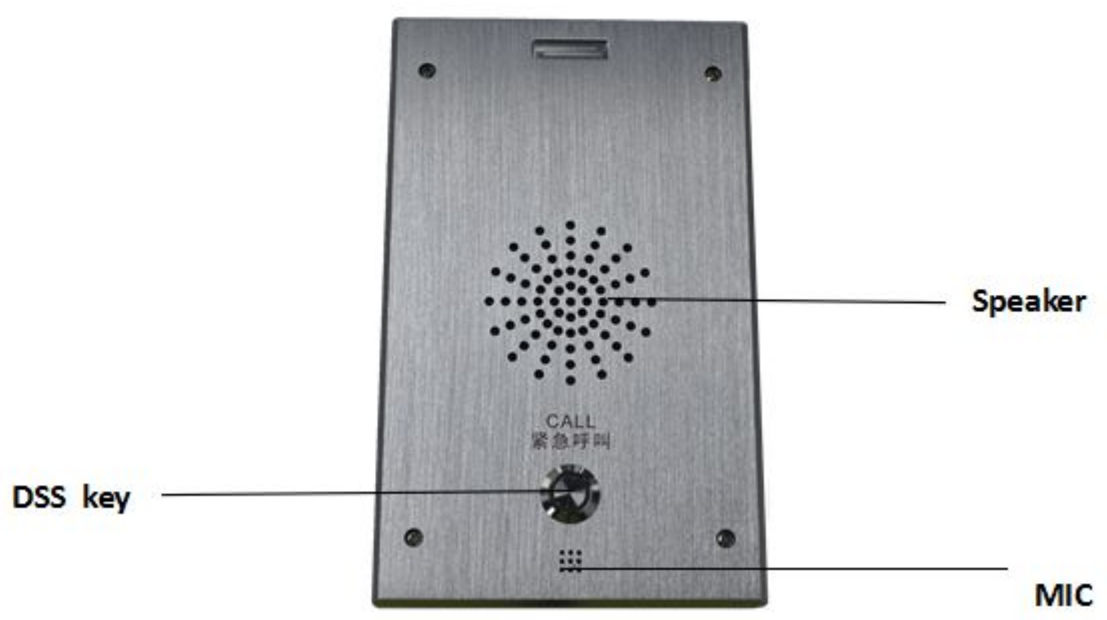
Communication protocol		SIP 2.0(RFC-3261)
Main chipset		Broadcom
Speech flow	Protocols	RTP/SRTP
	Decoding	G.729、G.723、G.711、G.722、G.726
	Audio amplifier	2.5W
	Volume control	Adjustable
	Full duplex speakerphone	Support (AEC)
Port	DSS key	One or Two (PH2.0 port)
	Indicating lamp	Three (PH2.0 port)
	MIC	Two (XH2.54 port)
	Speaker	One (XH2.54 port)
	An external active speaker	One (3.5mm port)
	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 mode		9V~16V/1A DC or POE
Cables		CAT5 or better
working temperature		-40°C to 70°C
working humidity		10% - 95%
storage temperature		-40°C to 70°C
overall dimension		195x120x39mm

2. Basic functions

- 2 SIP line
- POE enabled (Power over Ethernet)
- Full-duplex speakerphone
- Intelligent DSS Keys(Speed dial)
- Wall-mount installation
- Special integrated noise reduction module
- Dual microphone Omnidirectional voice pickup
- 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**

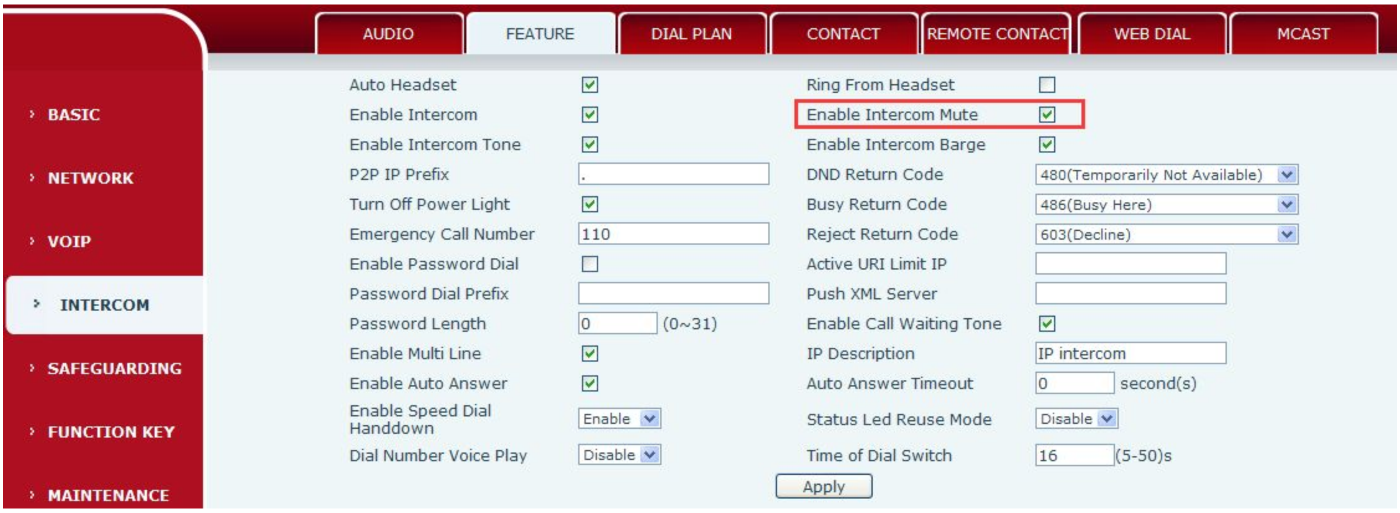
3. Schematic diagram



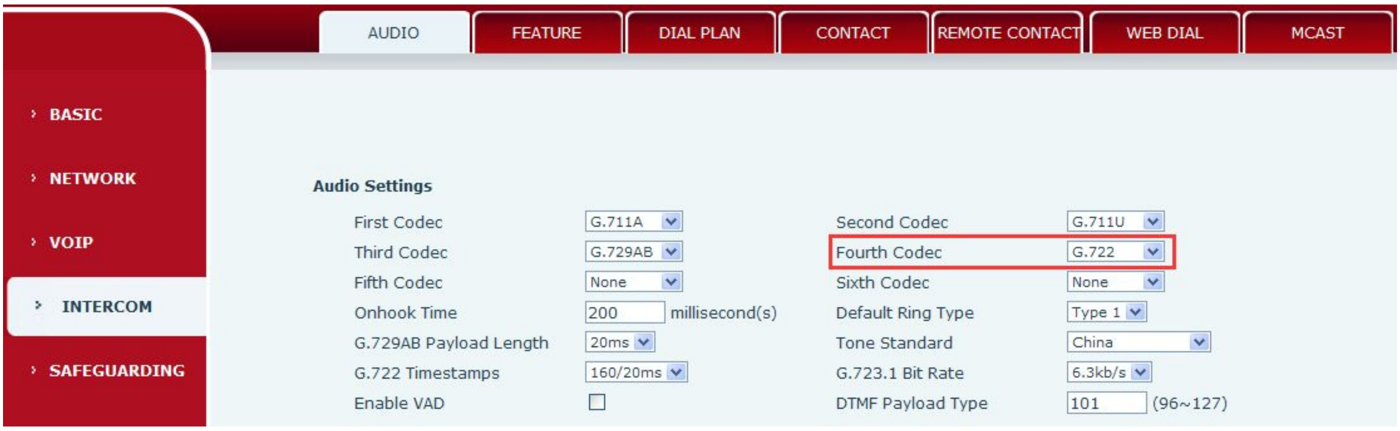
4. The radio terminal configuration notice

✧ **How to avoid an incoherency sound when the radio playing？**

When interrupt to use as radio, the sound of horn will be louder, 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 radio quality.



✧ How to improve broadcasting quality?



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

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.

5. The other function settings

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.