# SI 10 Intercom User Manual

**Single button** **Dual button**

# Safety Notices

1. Please use the specified power adapter. If special circumstances need to use the power adapter provided by other manufacturers, please make sure the voltage and current provided in accordance

with the requirements of this product, meanwhile, please use the safety certificated products, otherwise may cause fire or get an electric shock.

2. When using this product, please do not damage the power cord, or forcefully twist it、Stretch pull or banding, and not to be under heavy pressure or between items, Otherwise may cause the power cord damage, thus lead to fire or get an electric shock.
3. Before use, please confirm the temperature and environment humidity suitable for the product work. (Move the product from air conditioning room to natural temperature, which may cause this product surface or internal components produce condense water vapor, please open power use it after waiting for this product is natural drying).
4. Non-technical staff not remove or repair, improper repair or may cause electric shock, fire or malfunction, etc， Which can lead to injury accident, and also can cause your product damage.
5. Do not use fingers, pins, wire and other metal objects, foreign body into the vents and gaps. It may cause current through the metal or foreign body, which even cause electric shock and injury accident. If any foreign body or objection falls into the product please stop usage.
6. Please do not discard the packing bags or stored in places where children could reach, if children trap his head with it, may cause nose and mouth blocked, and even lead to suffocation.
7. Please use this product with normal usage and operating, in bad posture for a long time to use this product may affect your health.
8. Please read the above safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

## Directory



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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:

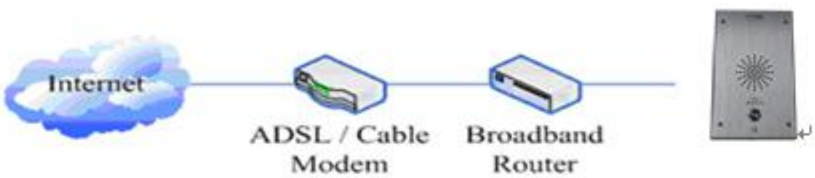
## 1. Connecting the power supply and the network

### (1) Connecting network

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

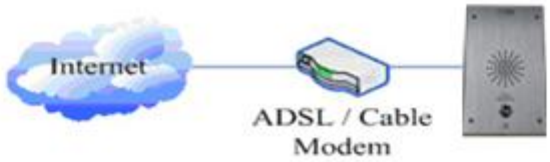
- **Broadband Router**

  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.**



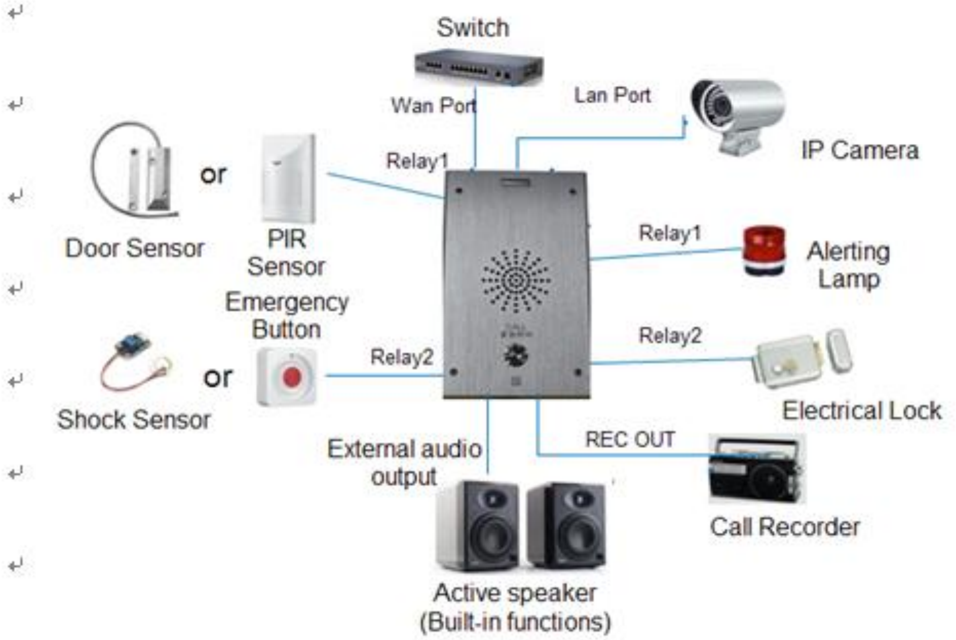

- **No Broadband Router**

  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.**



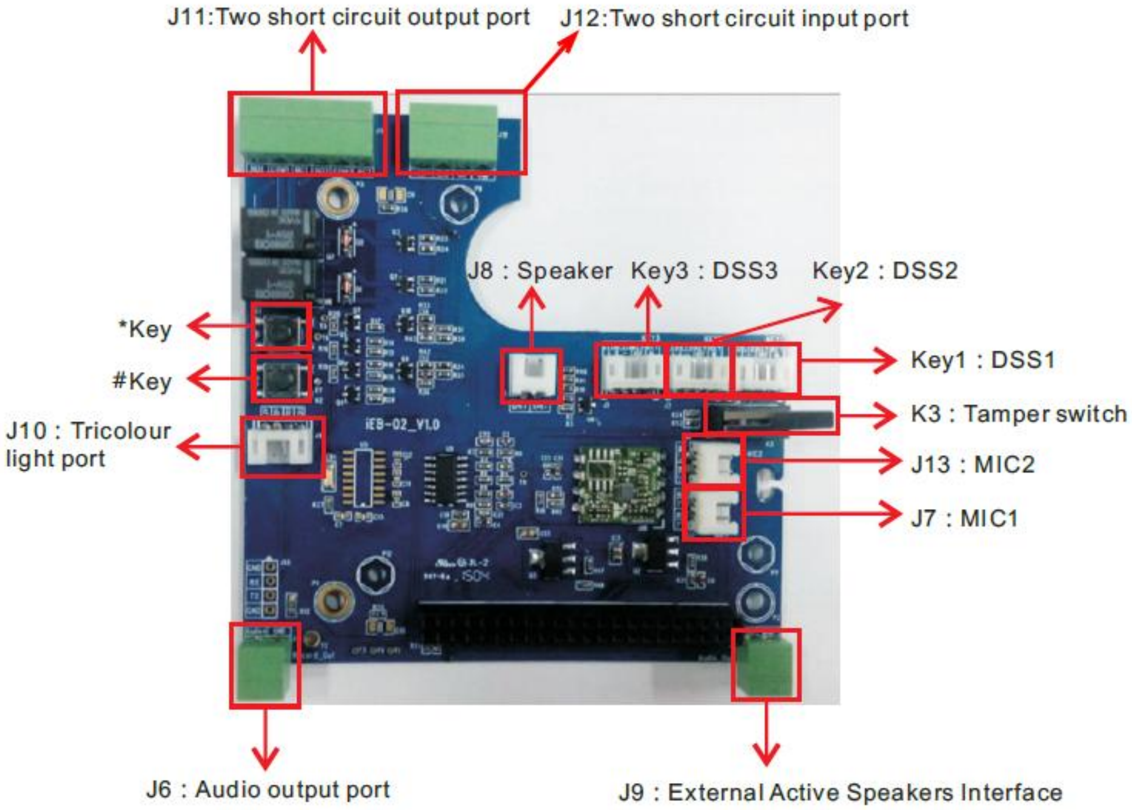

## (2)Interface specification

### a) Schematic diagram of peripherals



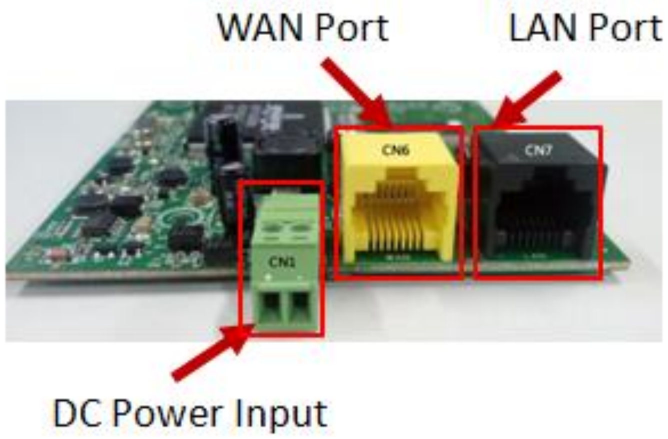

### b) Interface specification

- **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**

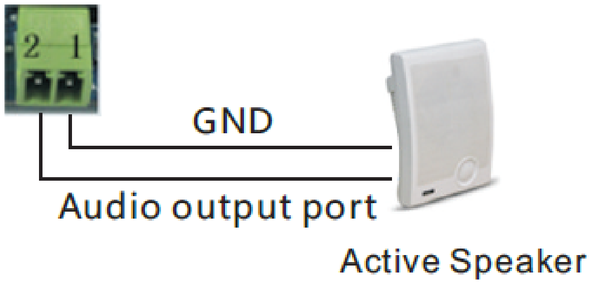
| Port | Description | Feature | Picture |
|---|---|---|---|
| CN1 | DC Power Input port | Input Range:+9~+16V DC<br><br>(Notice: Plus-n-Minus connection of the Power) |  |

| | | | |
|---|---|---|---|
| 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.<br>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.<br>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

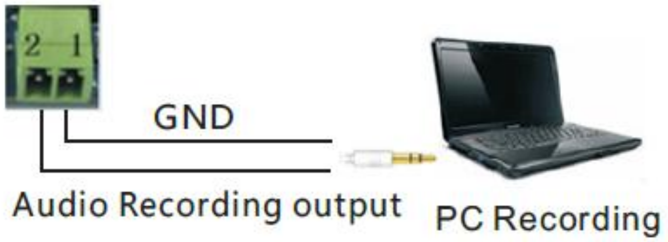
- **External Active Speakers**

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



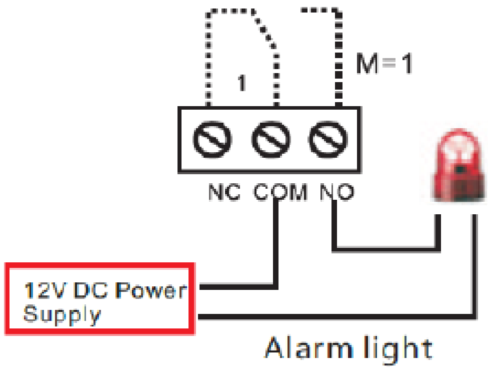

- **Audio Recording output port**

| 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);

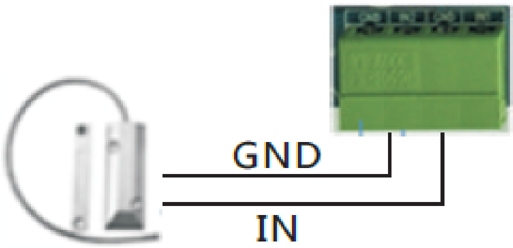
| J11: Short circuit output Port | | | | | |
|---|---|---|---|---|---|
| Output Port1(OUT2) | | | Output Port1(OUT1) | | |
| 6 | 5 | 4 | 3 | 2 | 1 |
| NC2 | COM2 | NO2 | NC1 | COM1 | NO1 |
| Normal close | Common terminal | Normal Open | Normal close | Common terminal | Normal Open |




- **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 |
|  | | | |



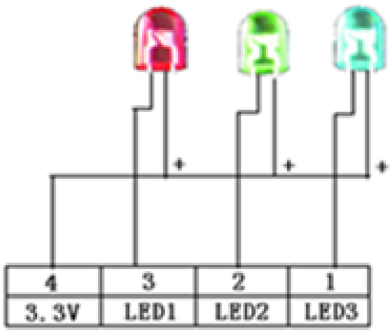



Door magnetic switch

- **Status lamp interface**

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



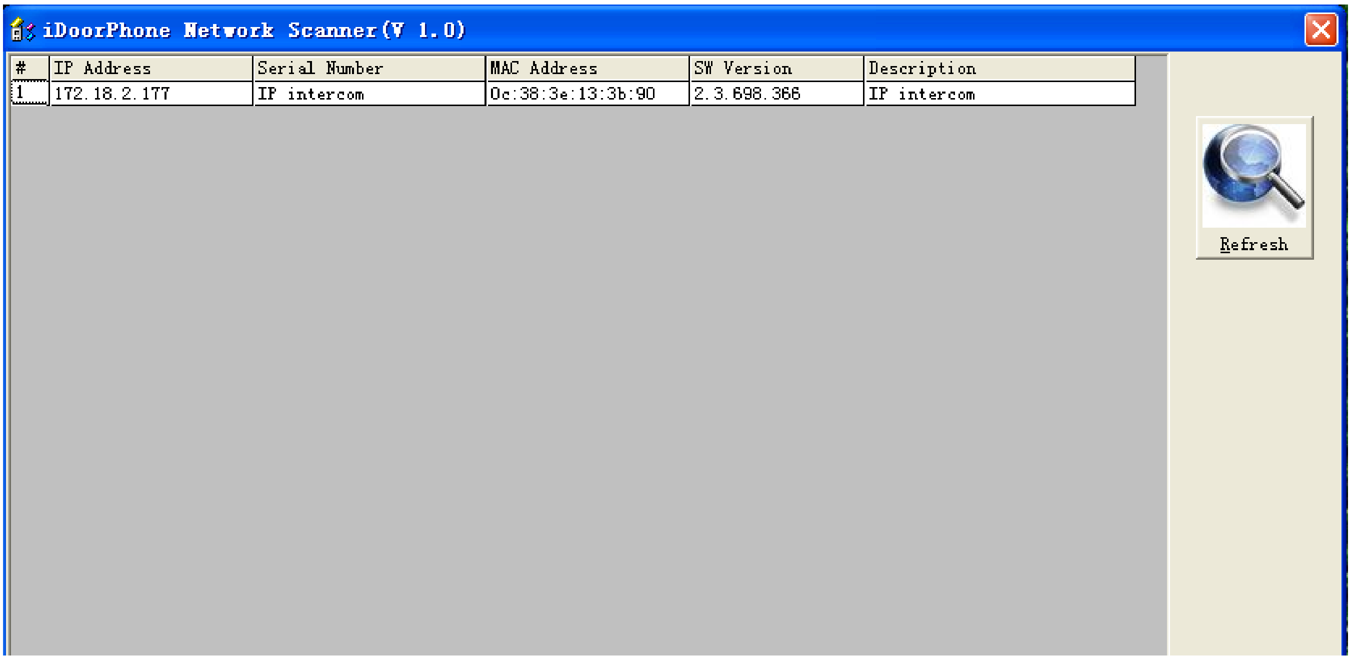



## 2. Quick Setting

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.

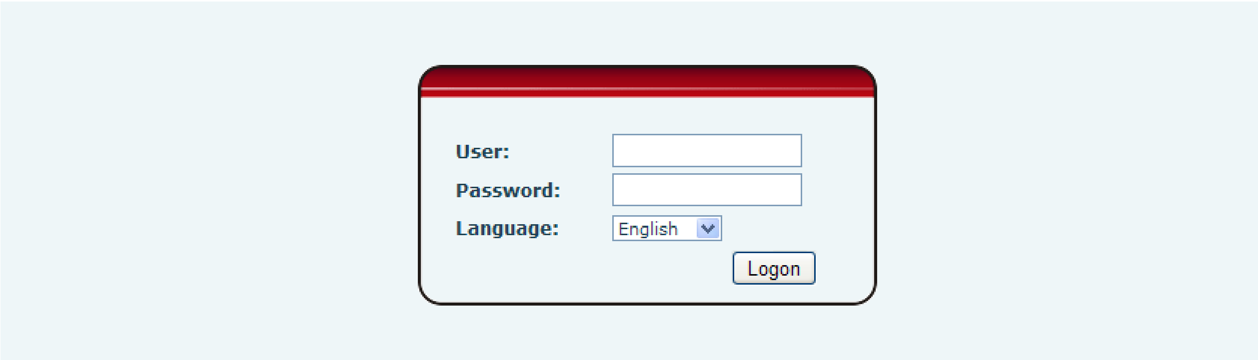
## 4. Call record

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

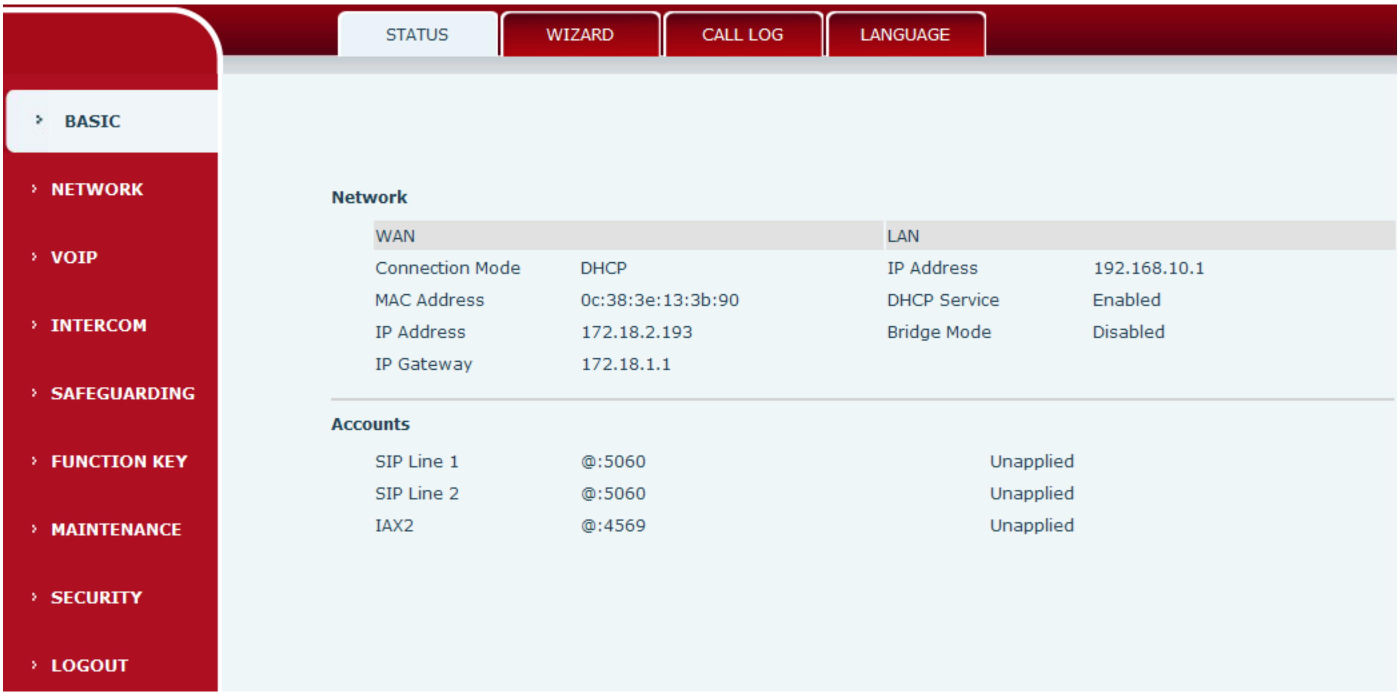
There are two levels of access: root level and general level. A user with root level access can browse and set all configuration parameters, while a user with general level can set all configuration parameters except server parameters for SIP or IAX2.

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

## 3. Configuration via WEB

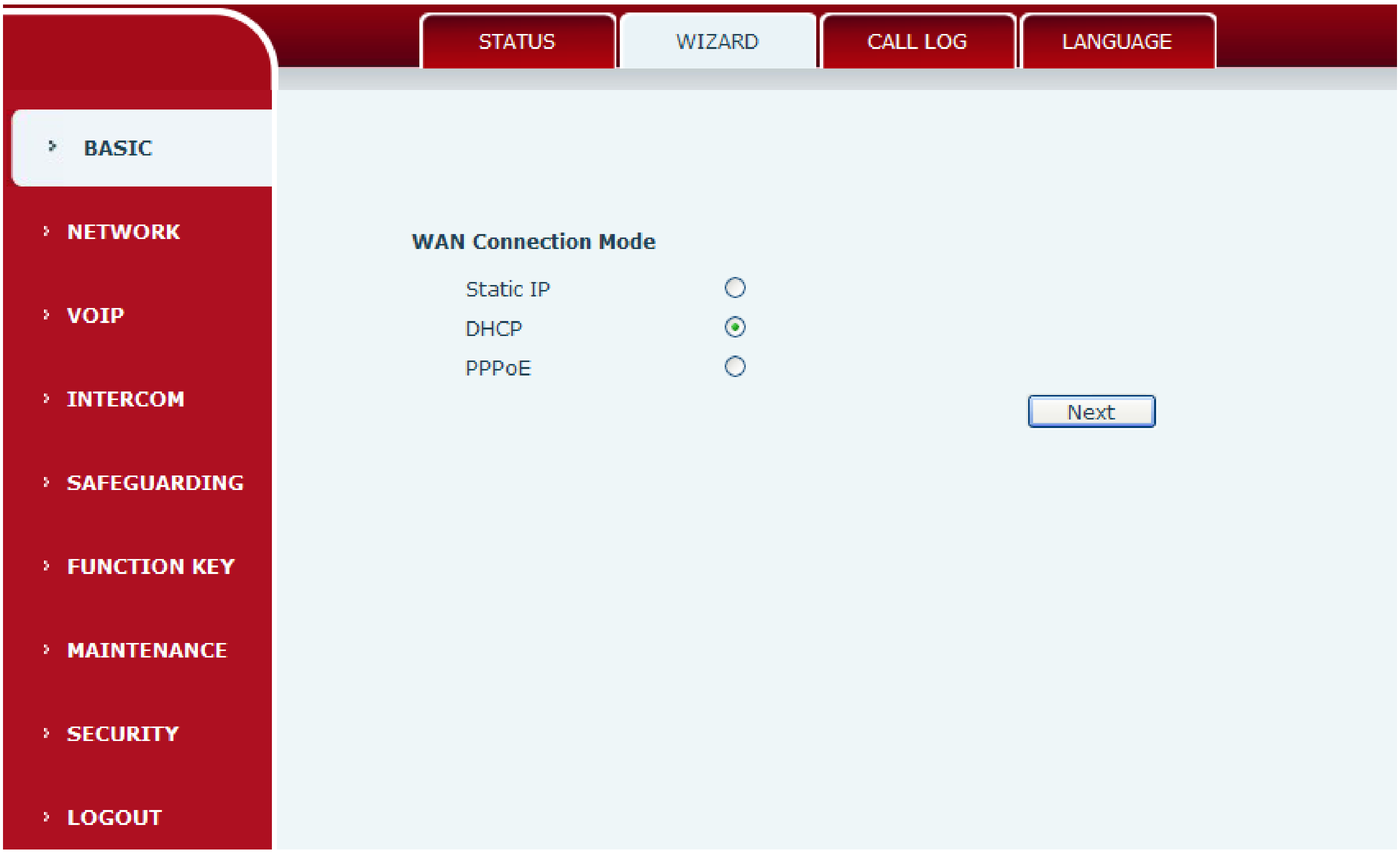
### (1)BASIC

#### a) STATUS



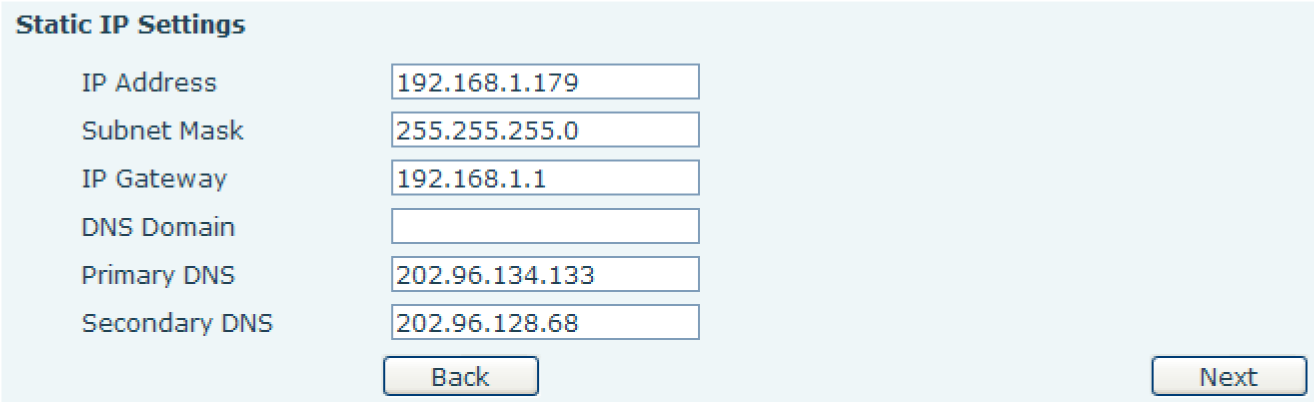

| Status | |
|---|---|
| **Field Name** | **Explanation** |
| Network | Shows the configuration information for WAN and LAN port, including connection mode of WAN port (Static, DHCP, PPPoE),MAC address, IP address of WAN port and LAN port, DHCP server, status for LAN port (ENABLED or DISABLED). |
| Accounts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

## b) WIZARD

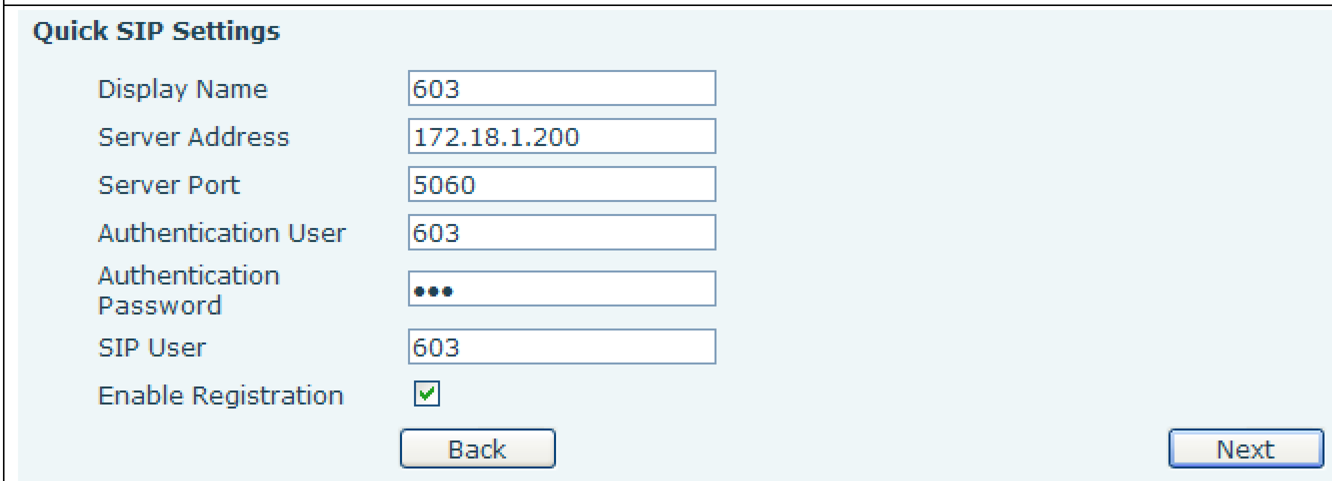



| Wizard | |
|---|---|
| **Field Name** | **Explanation** |
| Select the appropriate network mode. The equipment supports three network modes: | |
| Static IP mode | The parameters of a Static IP connection must be provided by your ISP. |
| DHCP mode： | In this mode, network parameter information will 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.** | |

**WAN**

| | |
|---|---|
| Connection Mode | Static IP |
| Static IP Address | 192.168.1.179 |
| IP Gateway | 192.168.1.1 |

**SIP**

| | |
|---|---|
| Server Address | 172.18.1.200 |
| Account | 603 |
| Phone Number | 603 |
| Registration | Enabled |

Back Finish

**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.**

**PPPoE Settings**

| | |
|---|---|
| Service Name | ANY |
| User | admin |
| Password | ●●●●● |

Back Next

| | |
|---|---|
| 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 Information**

| Start Time | Duration | Dialed Calls |
|---|---|---|
| April 22 11:22 | 1 second(s) | 172.18.2.193 |
| April 22 11:22 | 1 second(s) | 172.18.2.193 |

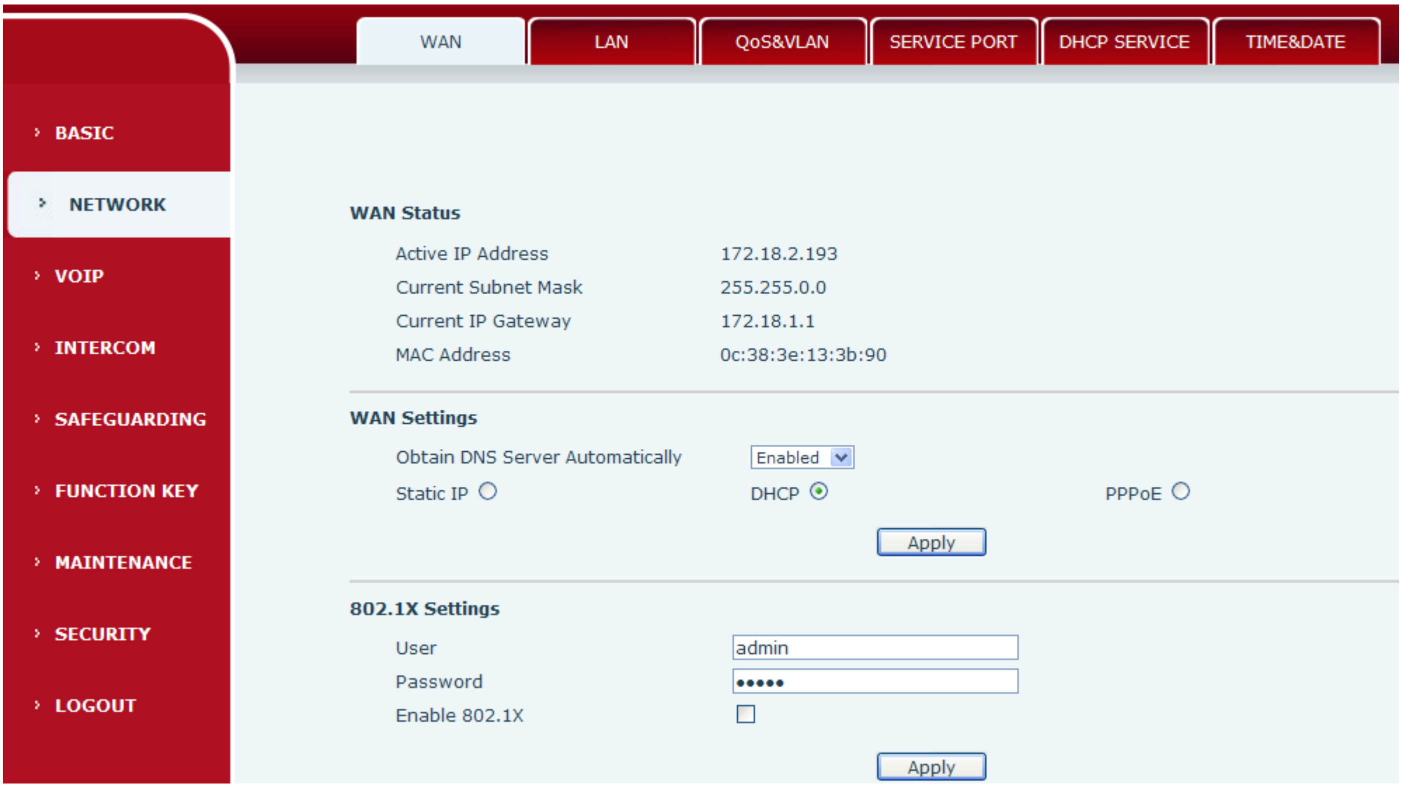
**Call log**

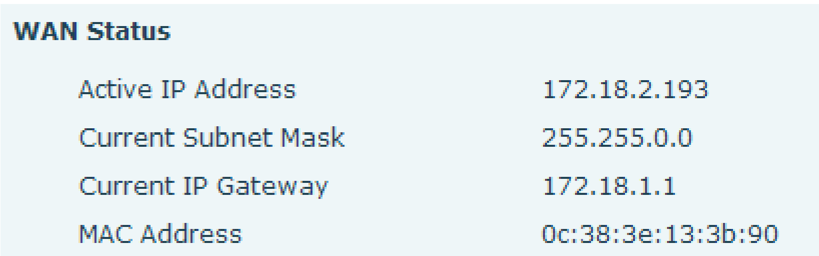
| Field Name | Explanation |
|---|---|
| Start time | Start time of the outgoing call |
| Duration | Duration of the outgoing call |

| | |
|---|---|
| Dialed calls | Account, protocol, and line of the outgoing call |

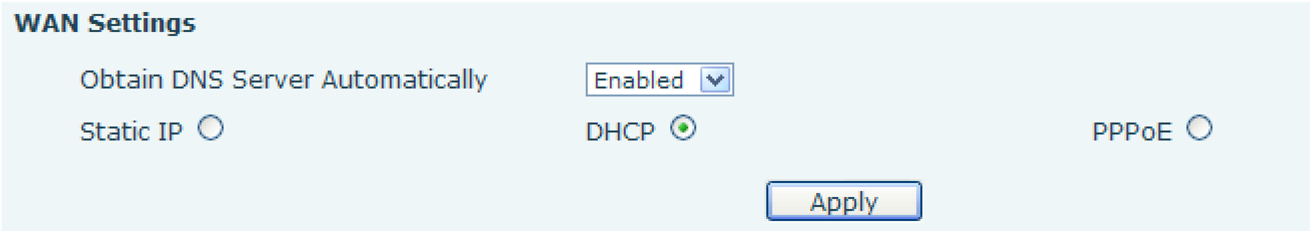
## (2)NETWORK

### a) WAN

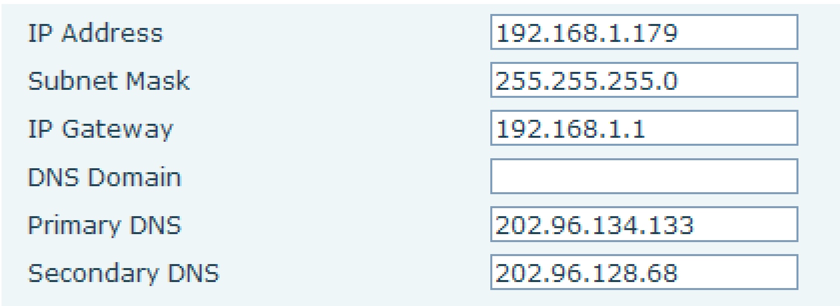



| WAN | |
|---|---|
| **Field Name** | **Explanation** |
|  | |

| 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 |

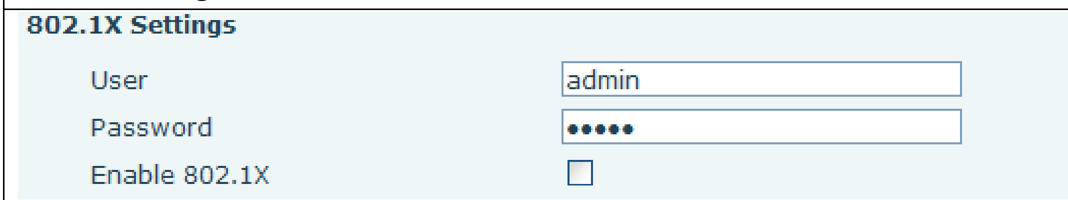



| | |
|---|---|
| Select the appropriate network mode. The equipment supports three network modes: | |
| Static | Network parameters must be entered manually and will not change. All parameters are provided by the ISP. |
| DHCP | Network parameters are provided automatically by a DHCP server. |
| PPPoE | 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 |

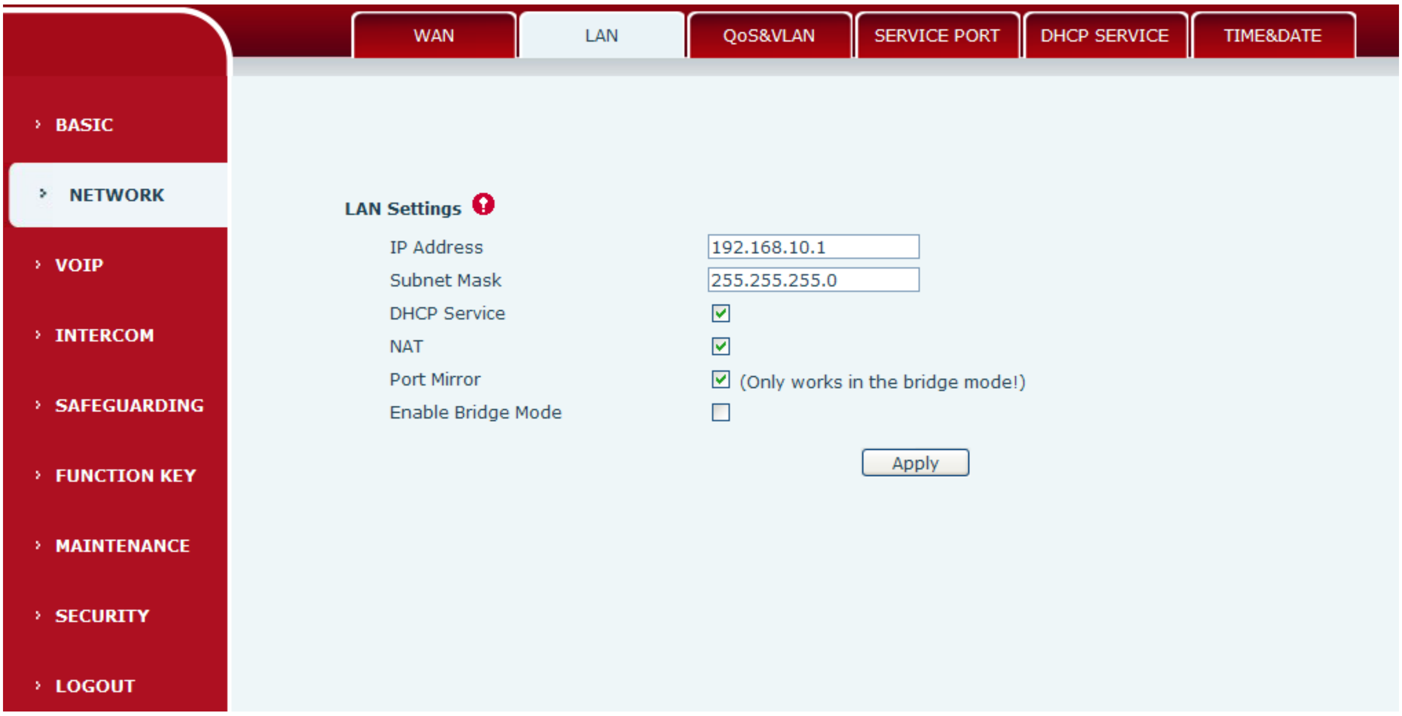
| Field Name | Explanation |
|---|---|
| **802.1X Settings** | |




| | |
|---|---|
| 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.

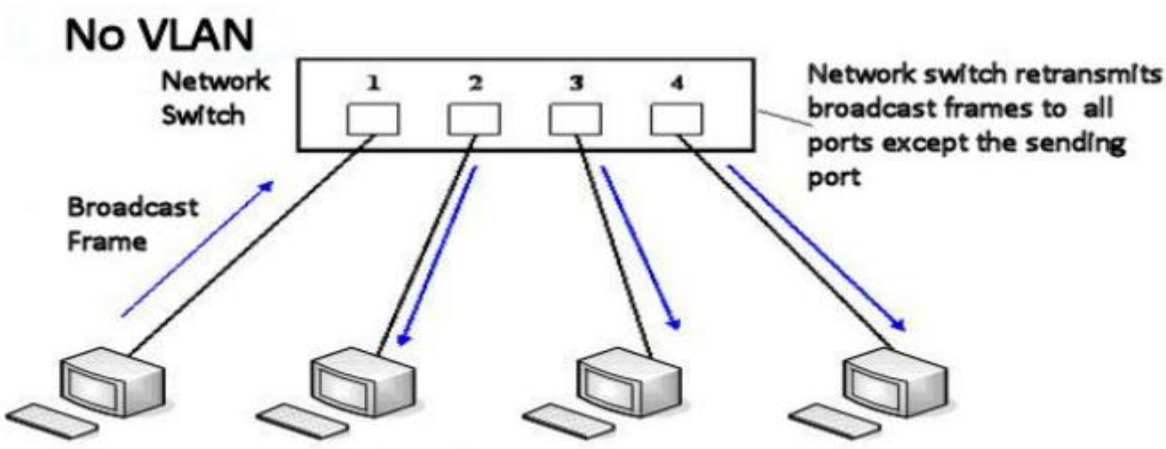
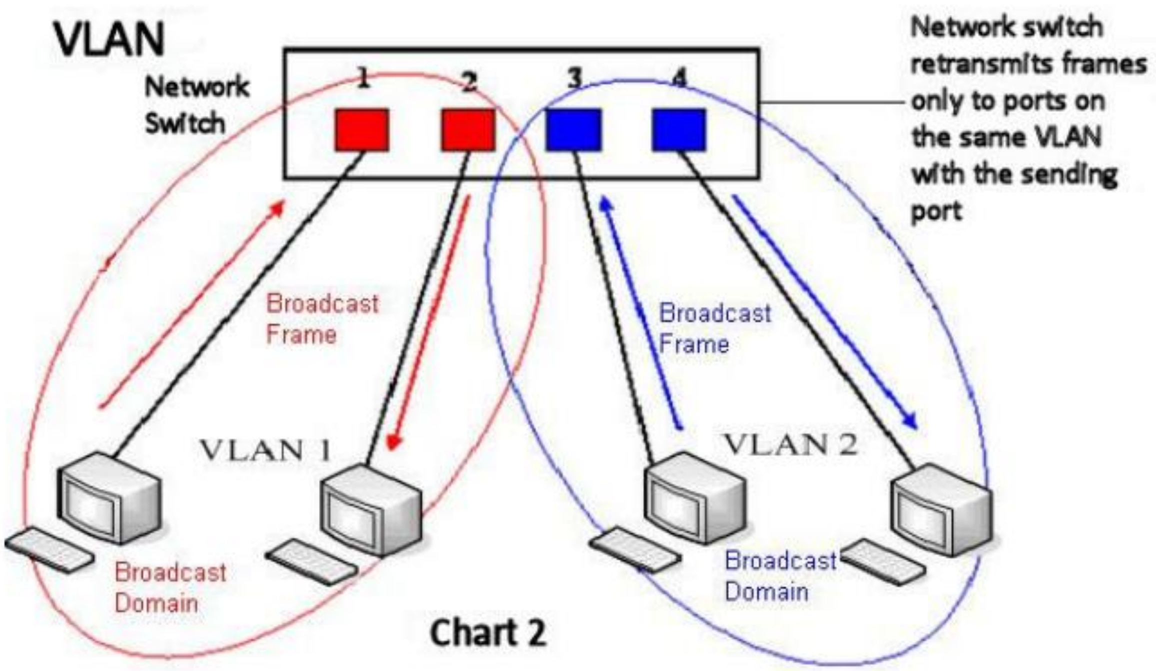
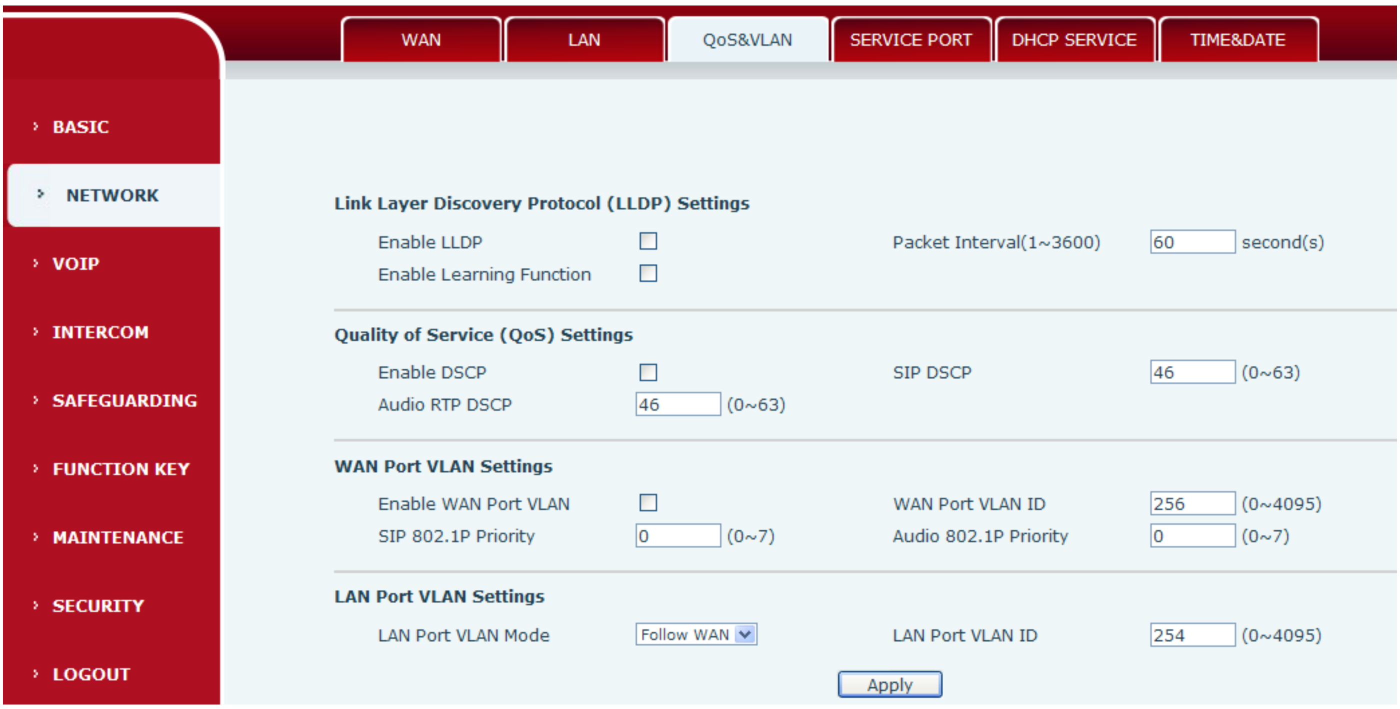



Chart 1

➢ 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.




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

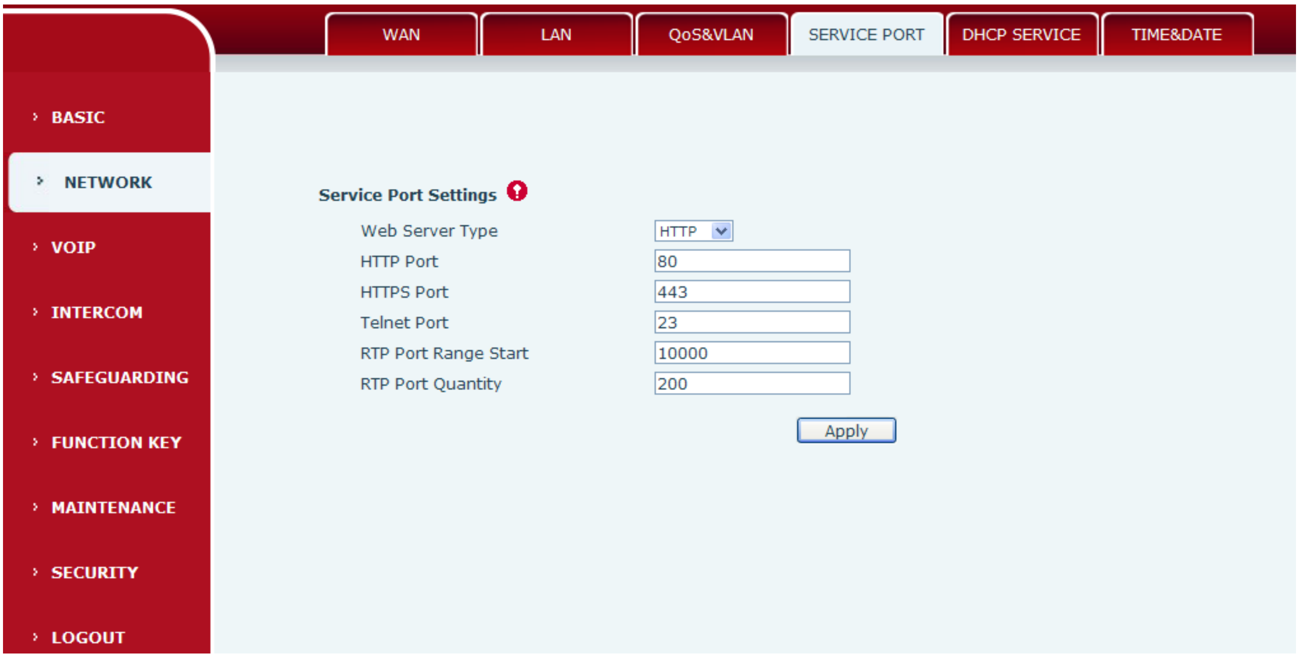



| QoS&VLAN | |
|---|---|
| **Field Name** | **Explanation** |
| **LLDP Settings** | |
| Enable LLDP | Enable or Disable Link Layer Discovery Protocol (LLDP) |

| | |
|---|---|
| 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.<br>Disable: Disable Port VALN<br>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 |

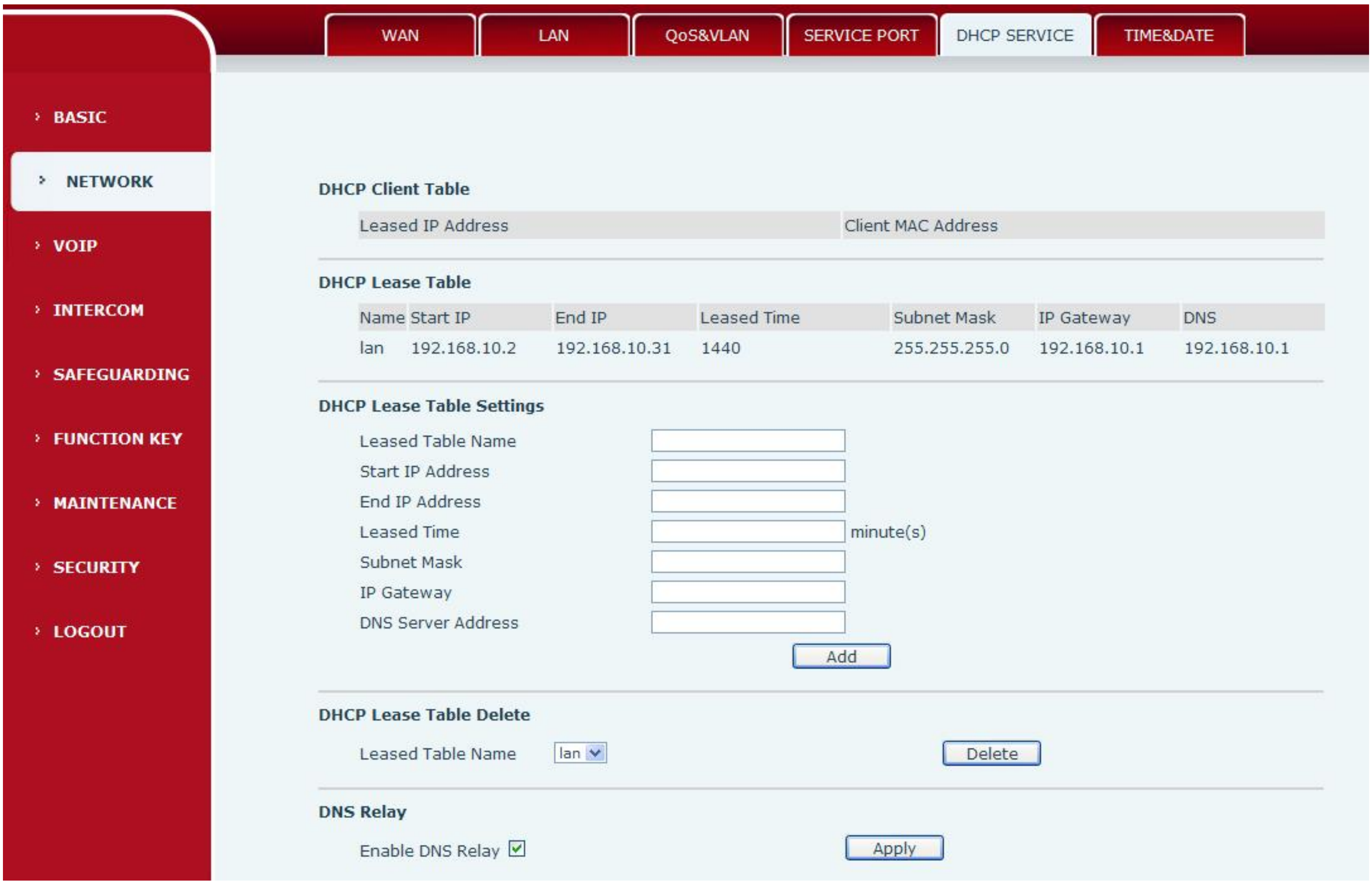
## d) SERVICE PORT

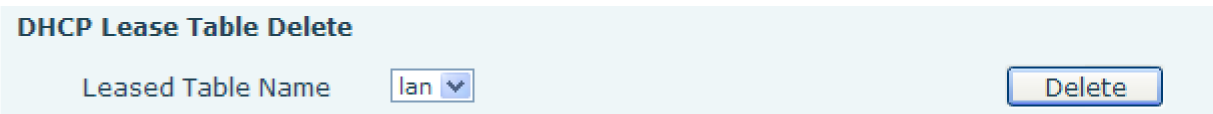
Set the port values for Telnet/HTTP/RTP on this page.

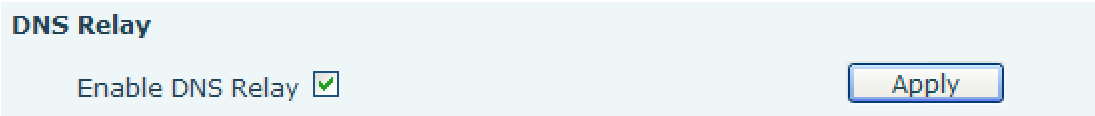



| **Service port** | |
|---|---|
| **Field Name** | **Explanation** |
| Web Server type | Specify Web Server Type – HTTP or HTTPS |
| HTTP port | Port for web browser access. Default value is 80. To enhance security, change this from the default. Setting this port to 0 will disable HTTP access.<br>Example: 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.<br>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:<br>1) Any changes made on this page require a reboot to become active.<br>2) It is suggested that changes to HTTP Port and Telnet ports be values greater than 1024.Values less than 1024 are reserved.<br>3) If the HTTP port is set to 0, HTTP service will be disabled. | |

## e) DHCP SERVICE

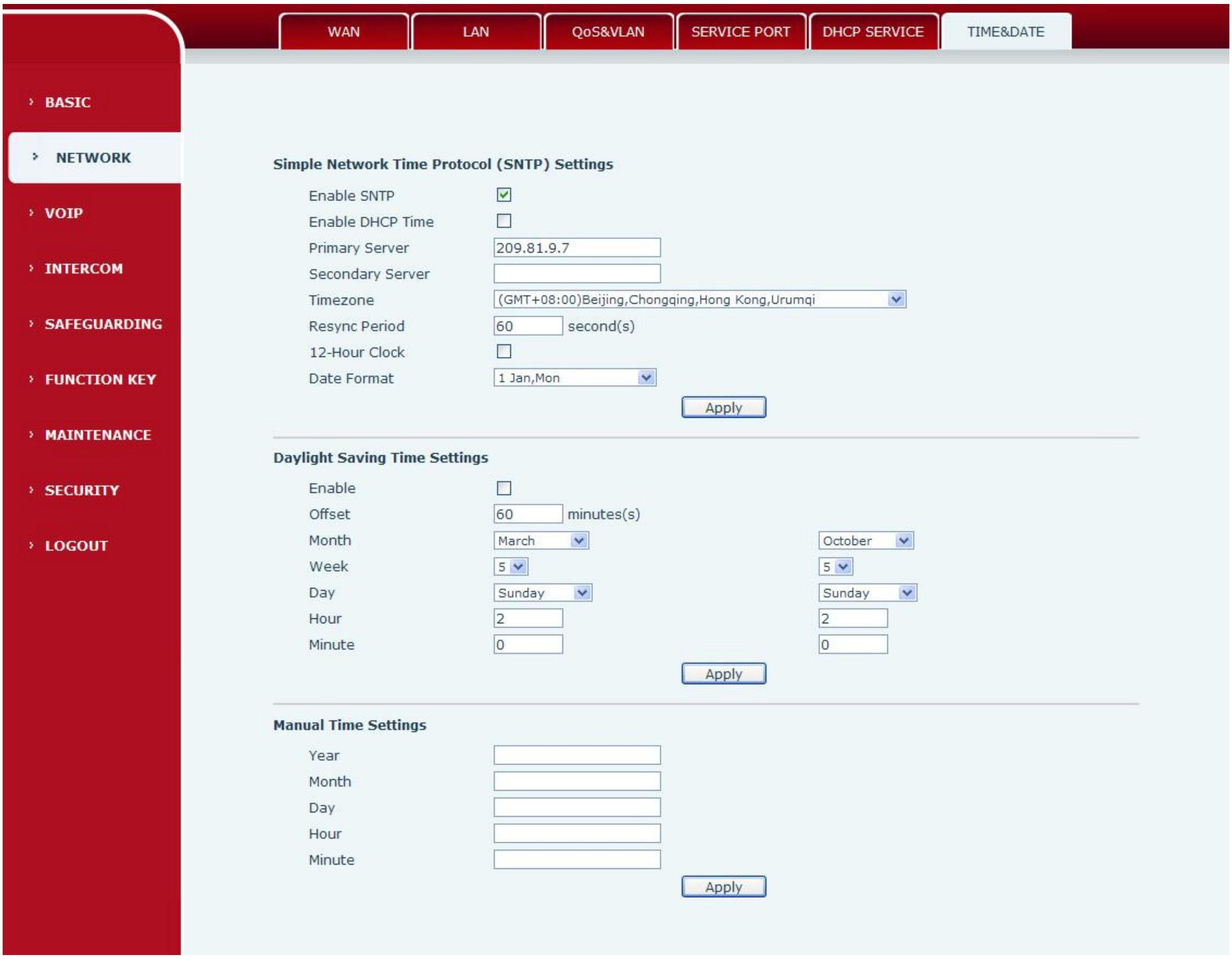



| **DHCP Server** | |
|---|---|
| **Field Name** | **Explanation** |
| DHCP Lease Table | IP-MAC mapping table. If the LAN port of the device connects to a device, this table will show its IP and MAC address. |
| **DHCP Lease Table Settings** | |
| Leased table name | Name of the lease table. |
| Start 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:<br>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<br>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.



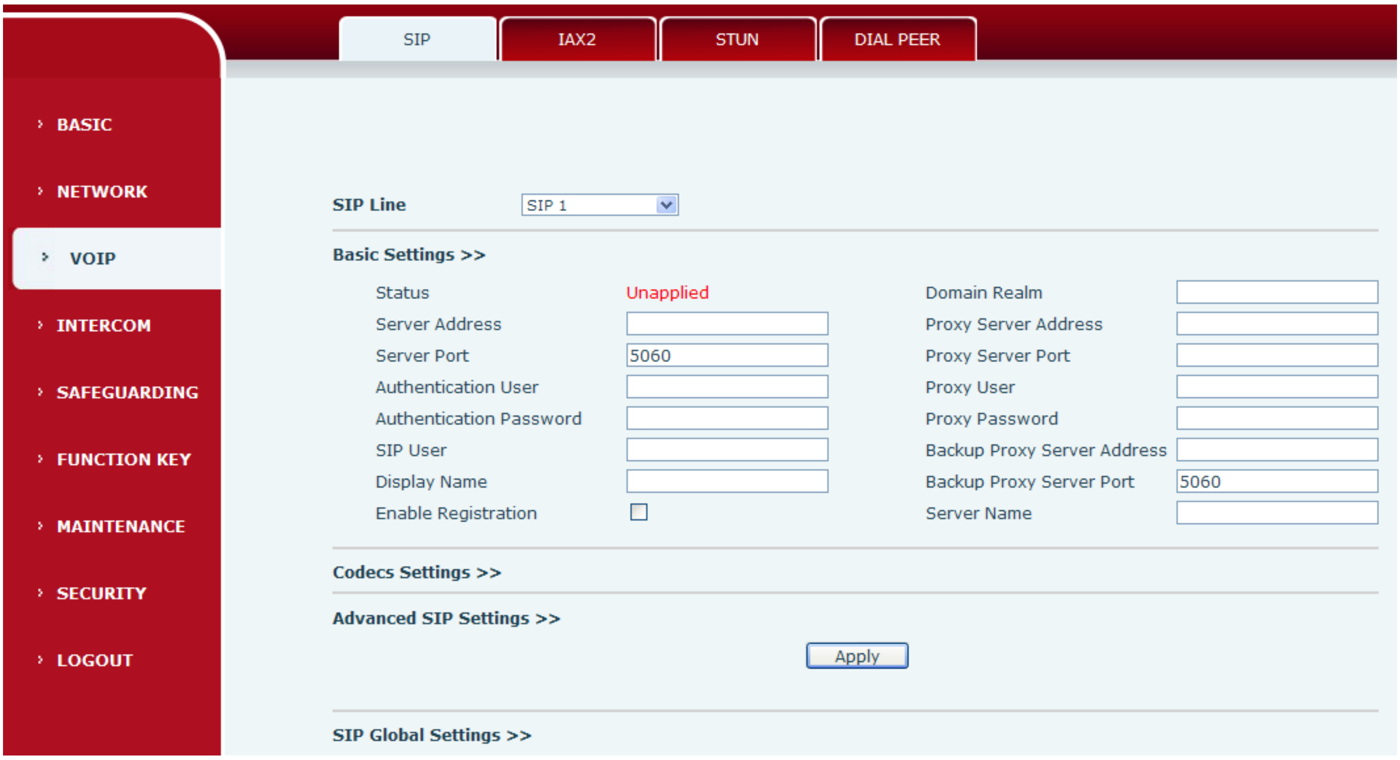

| TIME&DATE | |
|---|---|
| **Field Name** | **Explanation** |
| **SNTP Settings** | |
| Enable SNTP | Enable or Disable SNTP |
| DHCP Time | If this is enabled, equipment will synchronize time with DHCP server |
| Primary Server | IP address of Primary SNTP Server |
| Secondary Server | IP address of 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 |

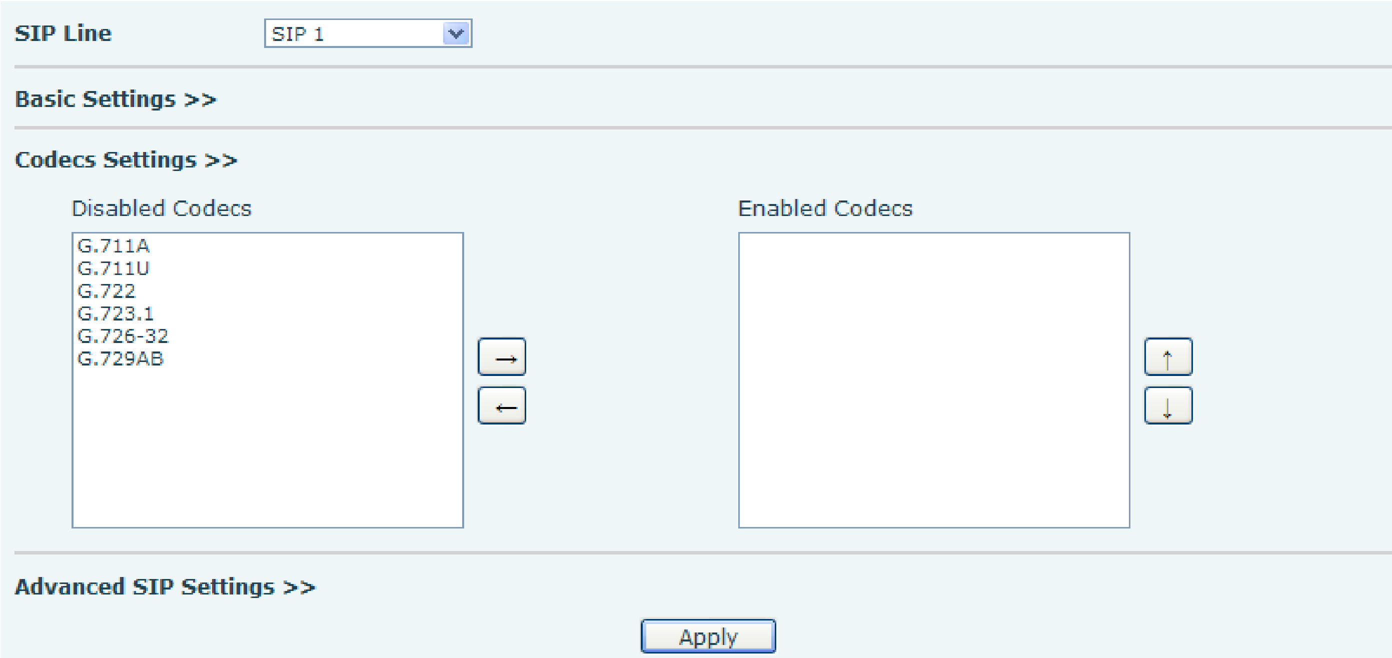
| | |
|---|---|
| Month | Start and end month for DST |
| Week | Start and end week for DST |
| Day | Start and end day for DST |
| Hour | Start and end hour for DST |
| Minute | Start and end minute for DST |
| **Manual Time Settings** | |
| Enter the values for the current year, month, day, hour and minute. All values are required.<br>Be sure to disable SNTP service before entering manual time and date. | |

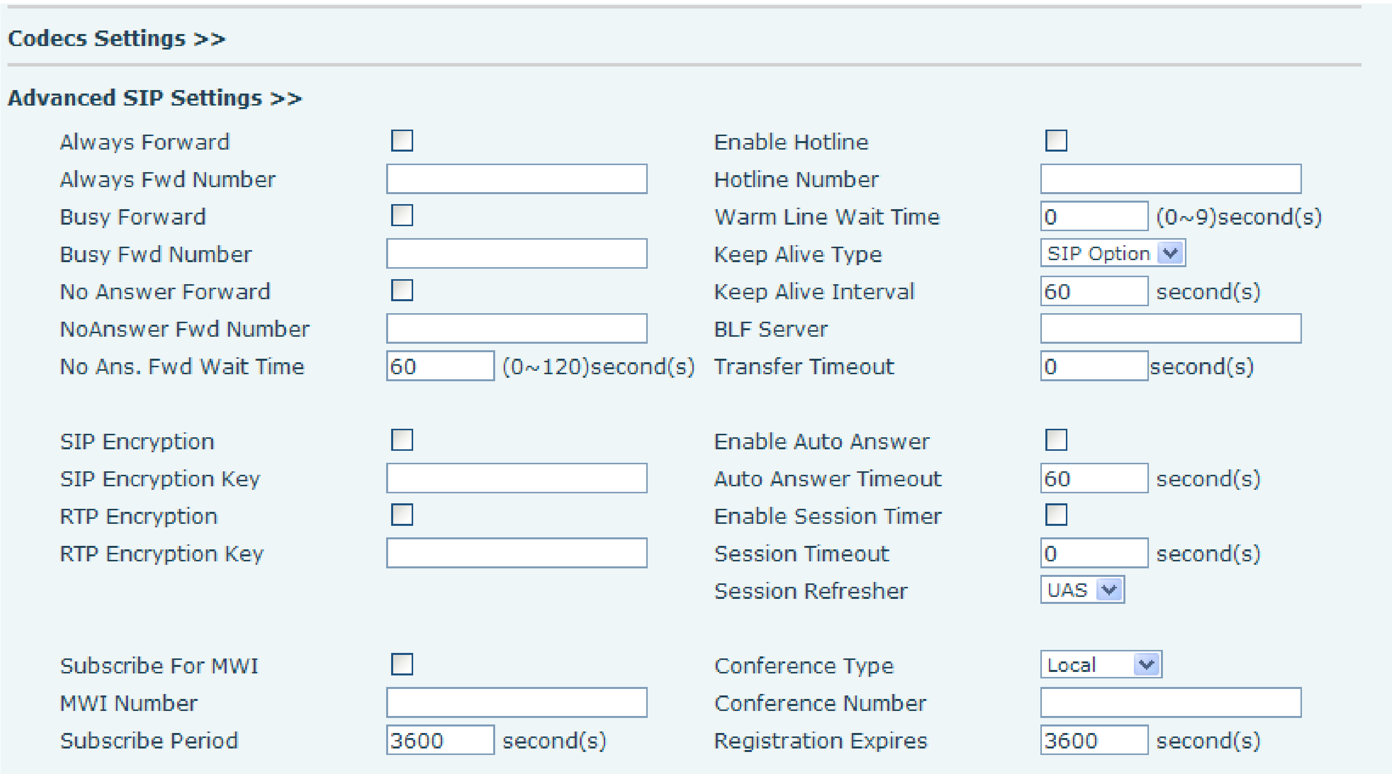
## (3)VOIP

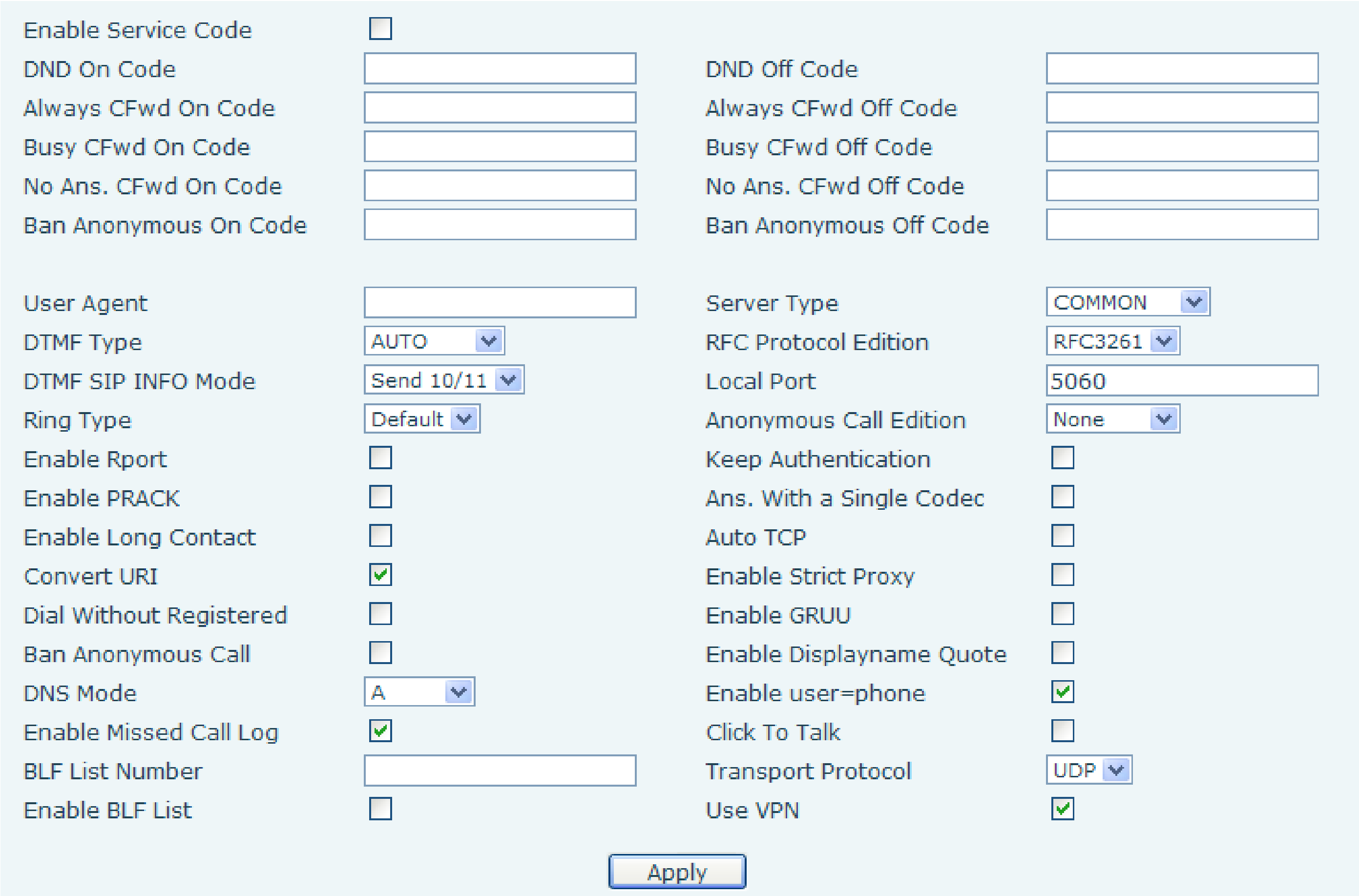
### a) SIP

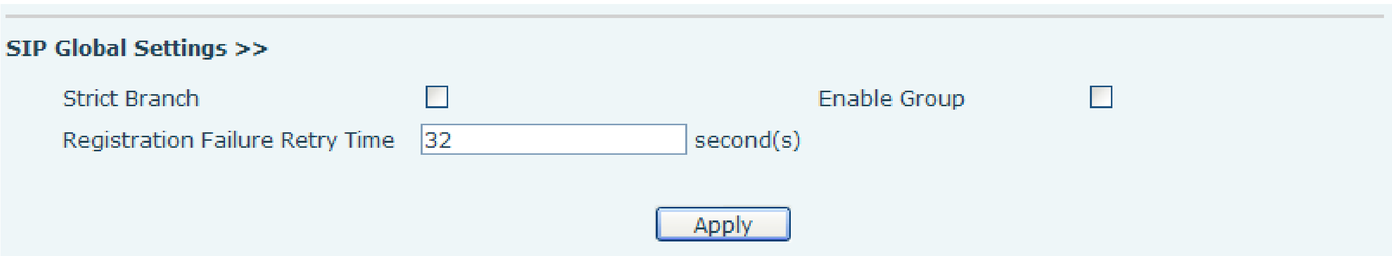
Configure a SIP server on this page




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




| 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. |

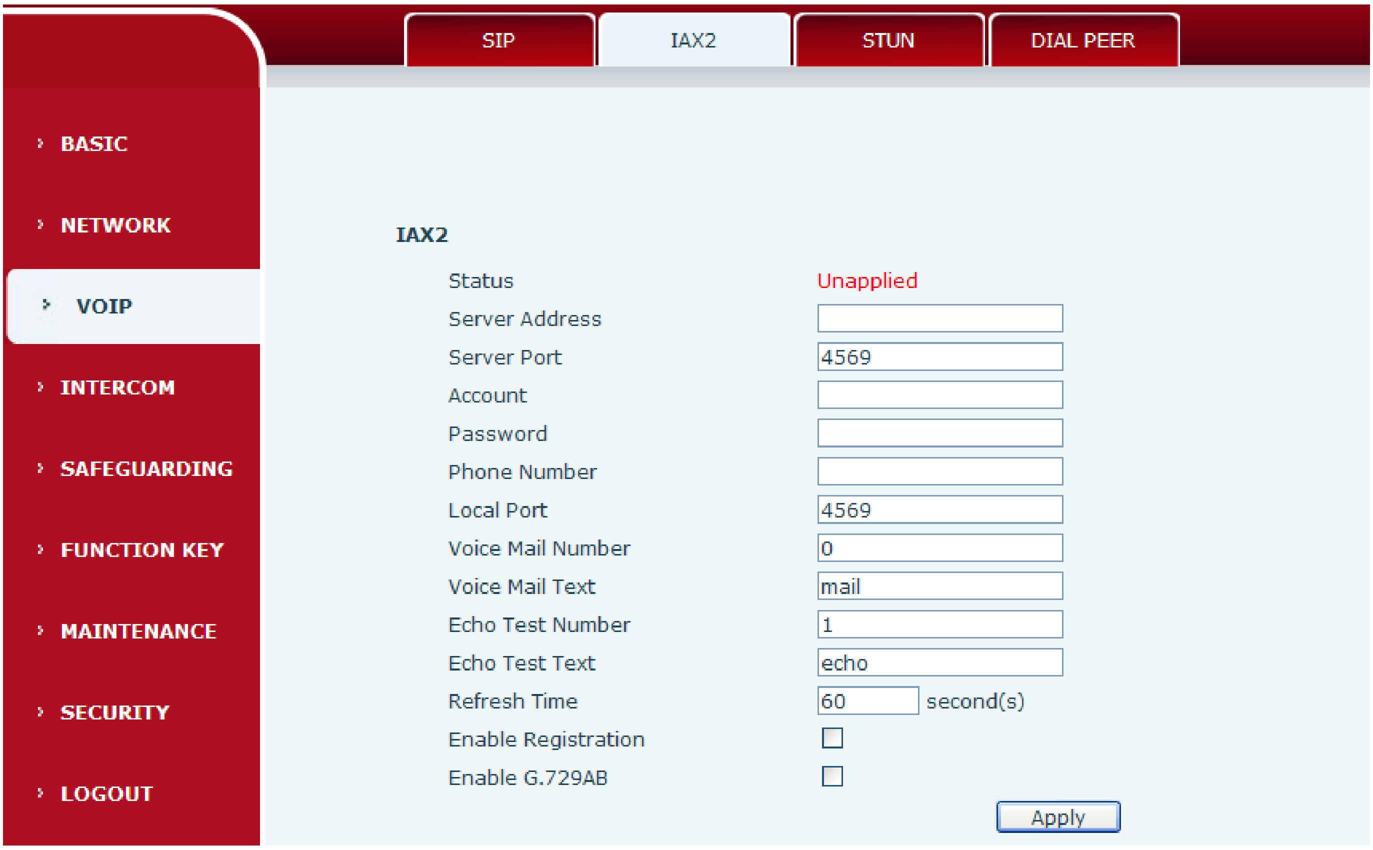
| **Field Name** | **Explanation** |
|---|---|
| Enable Service Code | Enables or disables the services described below. These codes will be sent to the SIP server to activate or deactivate the service. |
| DND On Code | Do Not Disturb (DND) – When this hot key is pressed, all calls to the extension 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 | Busy Call Forward On - When this function is enabled, the server will forward all calls |

| | |
|---|---|
| Code | 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:<br>● In-band<br>● RFC2833<br>● SIP_INFO<br>● AUTO<br>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.<br>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 Protocol | Configuration using the transport protocol, TCP, TLS or UDP, the default is UDP. |
| 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.<br>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

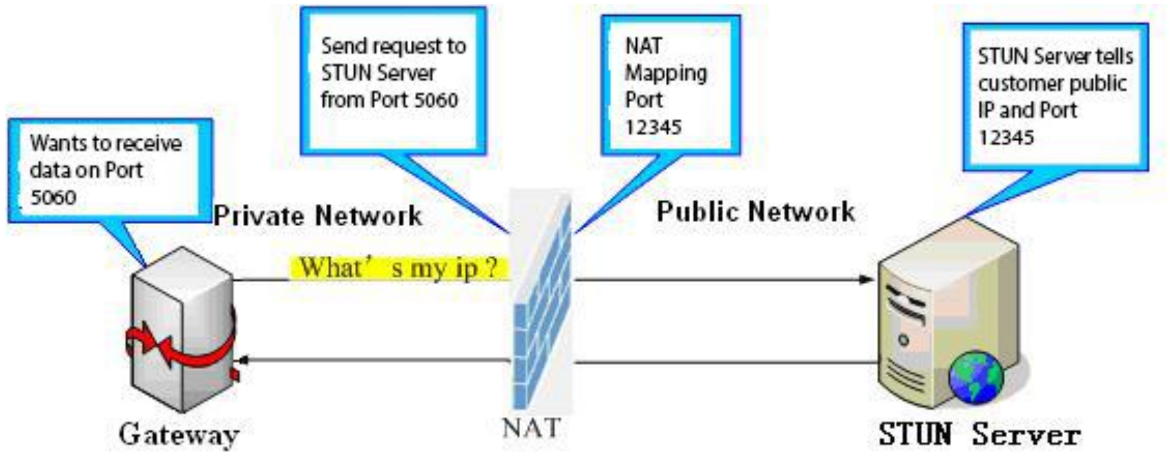


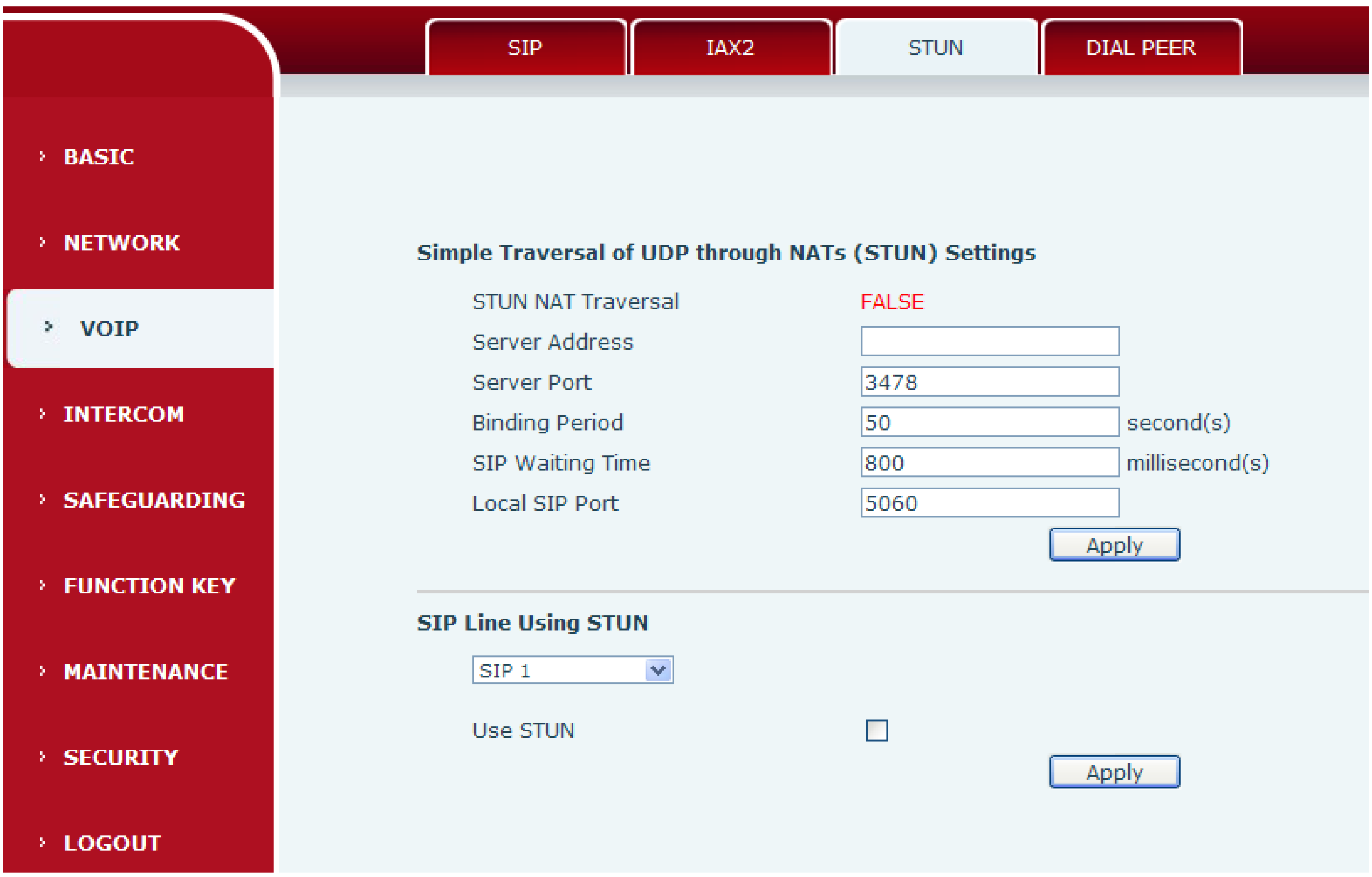

| **IAX2** | |
|---|---|
| **Field Name** | **Explanation** |
| Status | Shows registration status. Will show “Registered” if registered or “Unapplied” if not registered. |
| Server Address | IAX2 server address. |
| Server Port | IAX2 server port. Default is 4569. |
| Account | IAX2 account name for 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.

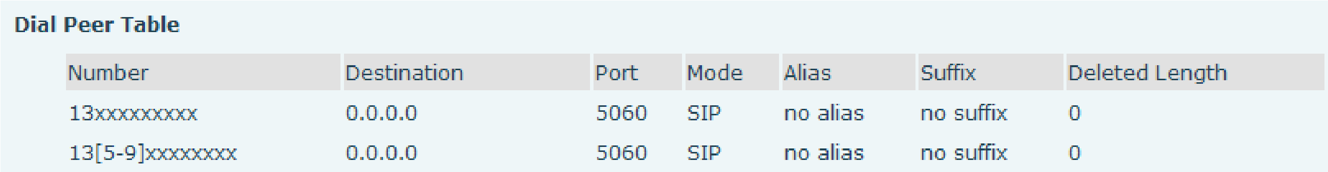
**Dial Peer Table**

| 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.

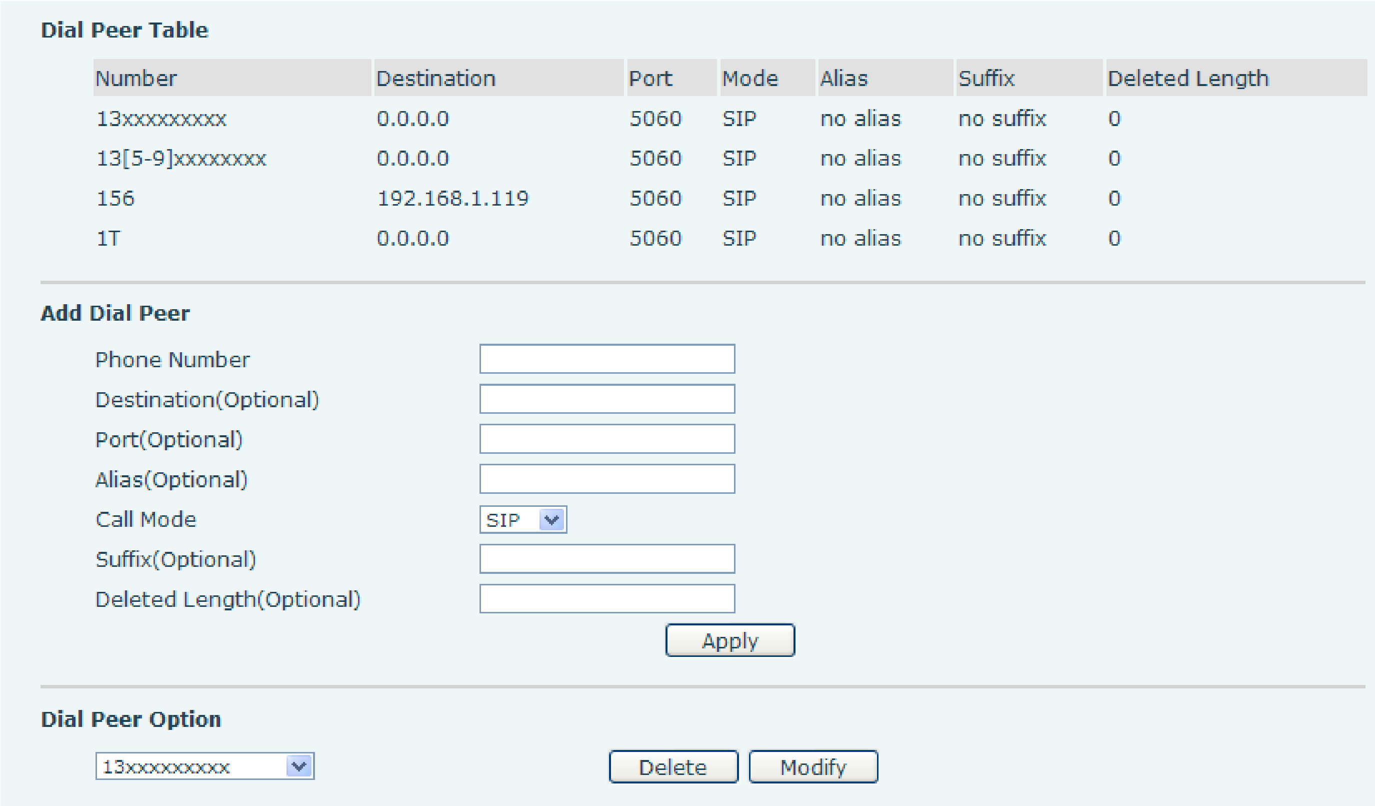
[] – Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.



**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 |



| DIAL PEER | |
|---|---|
| **Field Name** | **Explanation** |
| Phone Number | There are two types of matching: Full Matching or Prefix Matching.<br>In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules.<br>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.<br>1) Add: xxx – xxx will be dialed before any phone number.<br>2) All: xxx – xxx will replace the phone number. | |

| |
|---|
| 3) Del: The characters will be deleted from the phone number.<br>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:

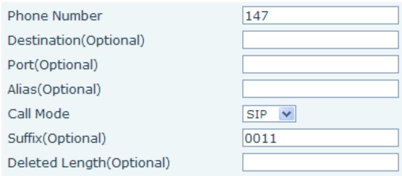
| Number | Destination | Port | Mode | Alias | Suffix | Deleted Length |
|---|---|---|---|---|---|---|
| 9T | 0.0.0.0 | 5060 | SIP | del | no suffix | 1 |
| 8T | 0.0.0.0 | 5060 | SIP | del | no suffix | 1 |

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:

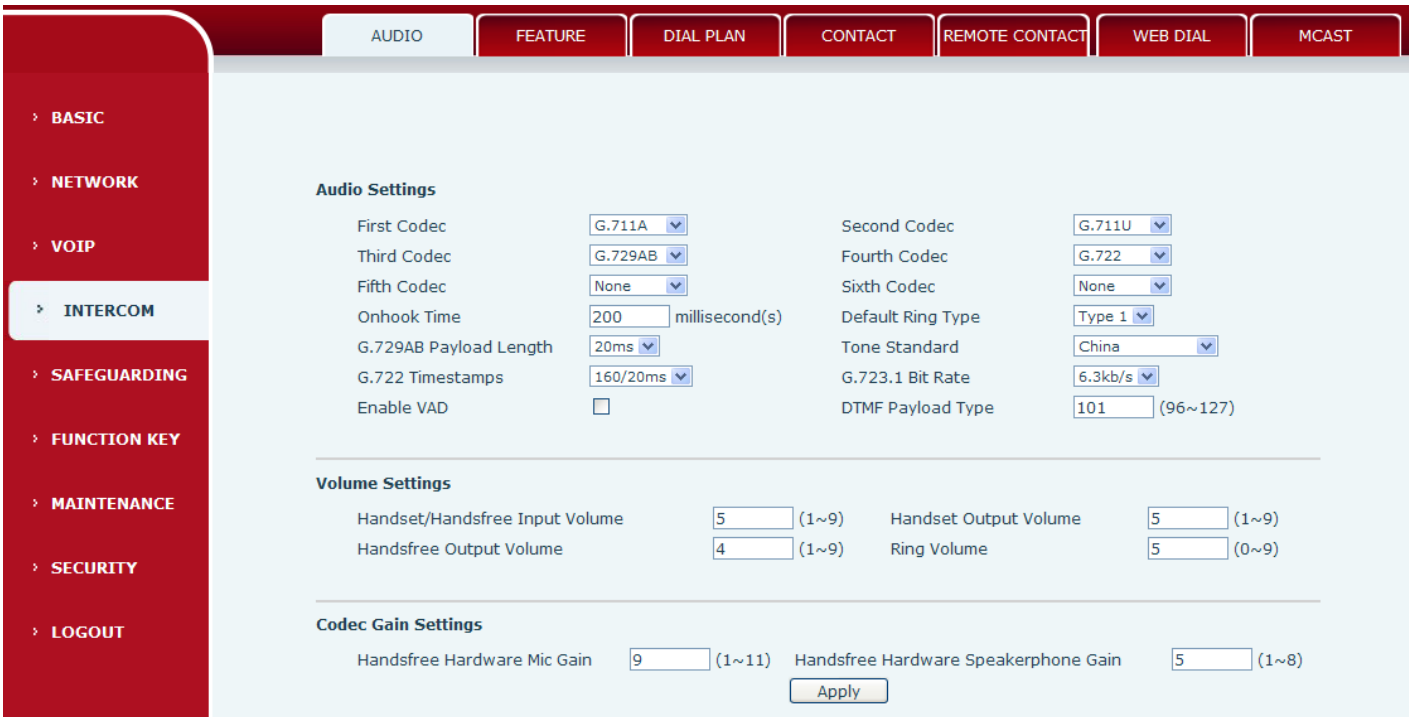
| Web Interface | Explanation | Example |
|---|---|---|
| Phone Number: 9T<br>Destination(Optional): 255.255.255.255<br>Port(Optional):<br>Alias(Optional): del<br>Call Mode: SIP<br>Suffix(Optional):<br>Deleted Length(Optional): 1 | Set phone number, Destination, Alias and Delete Length.<br>Phone number is XXXT; Destination is 255.255.255.255 (0.0.0.2) and Alias is del.<br>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"<br>The SIP2 server will receive "3333" |
| Phone Number: 2<br>Destination(Optional):<br>Port(Optional):<br>Alias(Optional): all:33334444<br>Call Mode: SIP<br>Suffix(Optional):<br>Deleted Length(Optional): 1 | This creates a speed dial function. Dialing "2", will cause the entire alias number to be sent out. | Dial "2"<br>The SIP1 server will receive 33334444 |
| Phone Number: 8T<br>Destination(Optional):<br>Port(Optional):<br>Alias(Optional): add:0755<br>Call Mode: SIP<br>Suffix(Optional):<br>Deleted Length(Optional): 1 | 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"<br>The SIP1 server will receive "07558309" |
| Phone Number: 010T<br>Destination(Optional):<br>Port(Optional):<br>Alias(Optional): rep:0866<br>Call Mode: SIP<br>Suffix(Optional):<br>Deleted Length(Optional): 3 | Set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is rep: xxx<br>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"<br>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.



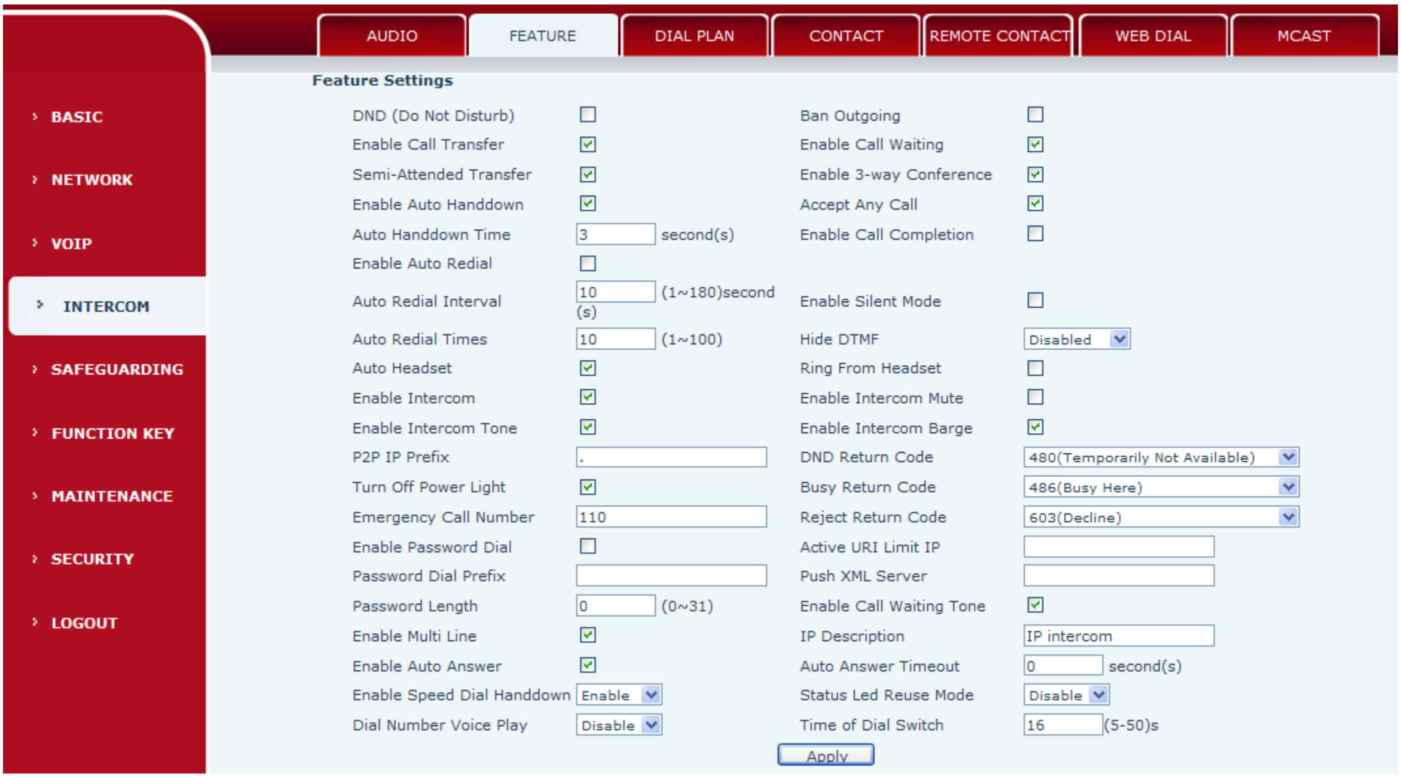

| Audio settings | |
|---|---|
| **Field Name** | **Explanation** |
| First Codec | The first codec choice: G.711A/U, G.722, G.723, G.729, G.726 |
| Second Codec | The second codec choice: G.711A/U, G.722, G.723, G.729, G.726, None |
| Third Codec | The third codec choice: G.711A/U, G.722, G.723, G.729, 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 |

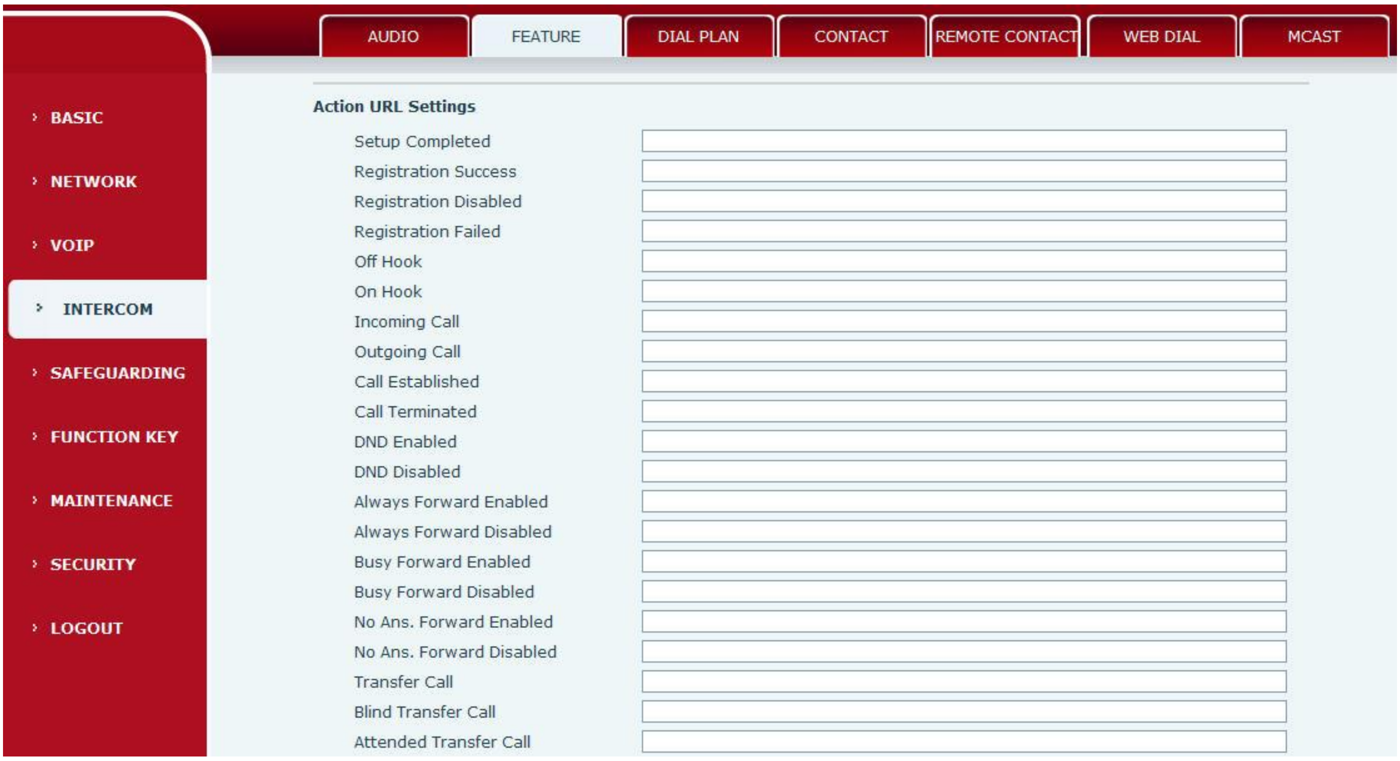
| Field Name | Explanation |
|---|---|
| G.729AB Payload Length | G.729 Payload Length – Adjusts from 10 – 60 mSec |

| | |
|---|---|
| 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.


AUDIO
FEATURE
DIAL PLAN
CONTACT
REMOTE CONTACT
WEB DIAL
MCAST
BASIC
NETWORK
VOIP
INTERCOM
SAFEGUARDING
FUNCTION KEY
MAINTENANCE
SECURITY
LOGOUT
Feature Settings
DND (Do Not Disturb)
Enable Call Transfer
Semi-Attended Transfer
Enable Auto Handdown
Auto Handdown Time
3
second(s)
Enable Auto Redial
Auto Redial Interval
10
(1~180)second(s)
Auto Redial Times
10
(1~100)
Auto Headset
Enable Intercom
Enable Intercom Tone
P2P IP Prefix
.
Turn Off Power Light
Emergency Call Number
110
Enable Password Dial
Password Dial Prefix
Password Length
0
(0~31)
Enable Multi Line
Enable Auto Answer
Enable Speed Dial Handdown
Enable
Dial Number Voice Play
Disable
Ban Outgoing
Enable Call Waiting
Enable 3-way Conference
Accept Any Call
Enable Call Completion
Enable Silent Mode
Hide DTMF
Disabled
Ring From Headset
Enable Intercom Mute
Enable Intercom Barge
DND Return Code
480(Temporarily Not Available)
Busy Return Code
486(Busy Here)
Reject Return Code
603(Decline)
Active URI Limit IP
Push XML Server
Enable Call Waiting Tone
IP Description
IP intercom
Auto Answer Timeout
0
second(s)
Status Led Reuse Mode
Disable
Time of Dial Switch
16
(5-50)s
Apply


AUDIO
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MCAST
BASIC
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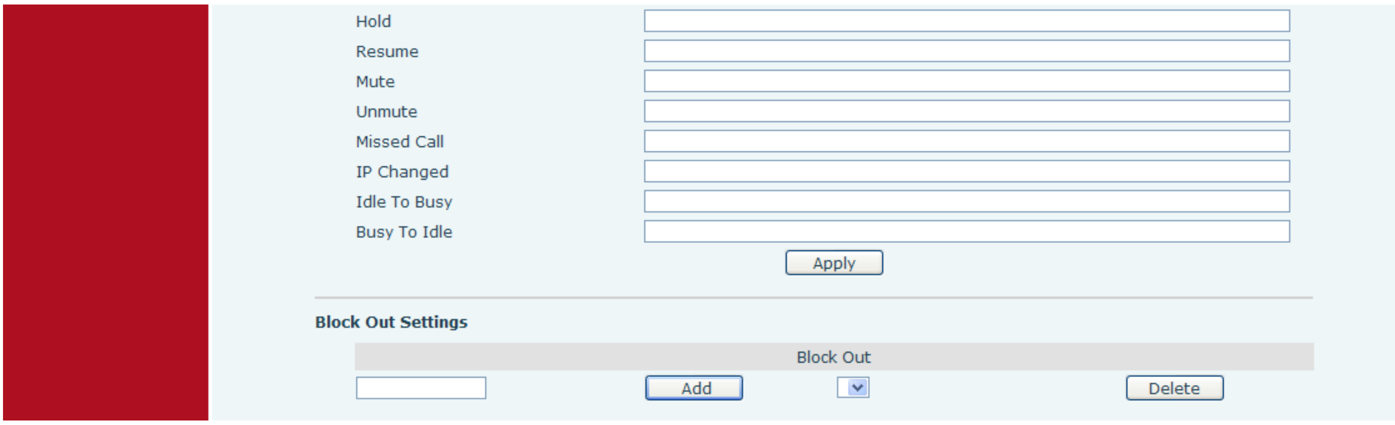
VOIP
INTERCOM
SAFEGUARDING
FUNCTION KEY
MAINTENANCE
SECURITY
LOGOUT
Action URL Settings
Setup Completed
Registration Success
Registration Disabled
Registration Failed
Off Hook
On Hook
Incoming Call
Outgoing Call
Call Established
Call Terminated
DND Enabled
DND Disabled
Always Forward Enabled
Always Forward Disabled
Busy Forward Enabled
Busy Forward Disabled
No Ans. Forward Enabled
No Ans. Forward Disabled
Transfer Call
Blind Transfer Call
Attended Transfer Call




| 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.<br>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 | Configuration Open / Close Dial Number Voice Play |

| | |
|---|---|
| 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.<br>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. | |

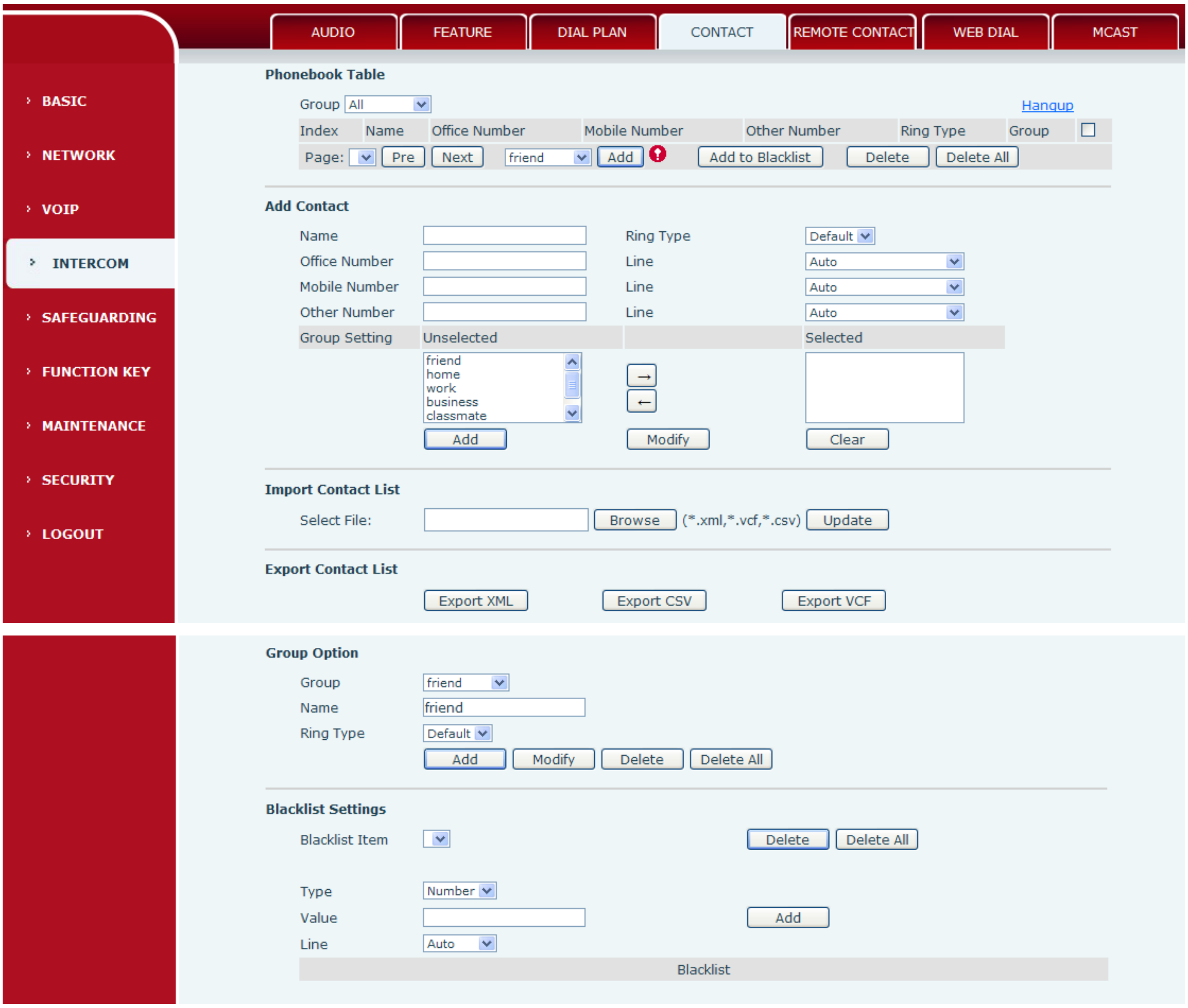
## c) DIAL PLAN


| **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 | Hang up after entering the target number for the transfer. The equipment will |

| | |
|---|---|
| on hook | 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

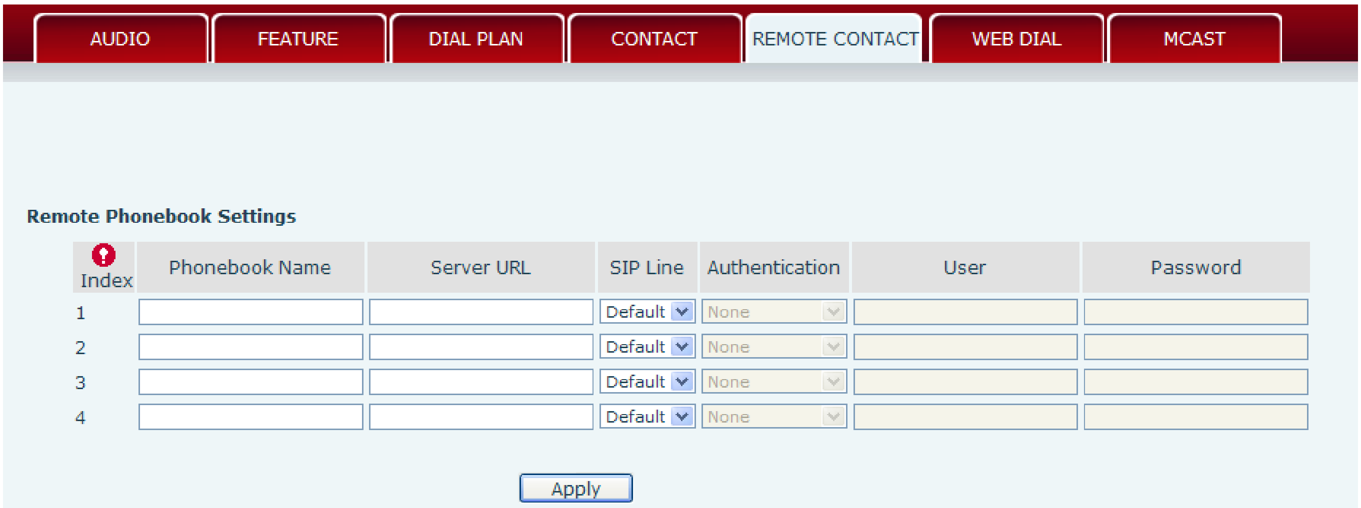
Enter the name, phone number and ring type for each contact here.




| 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 |

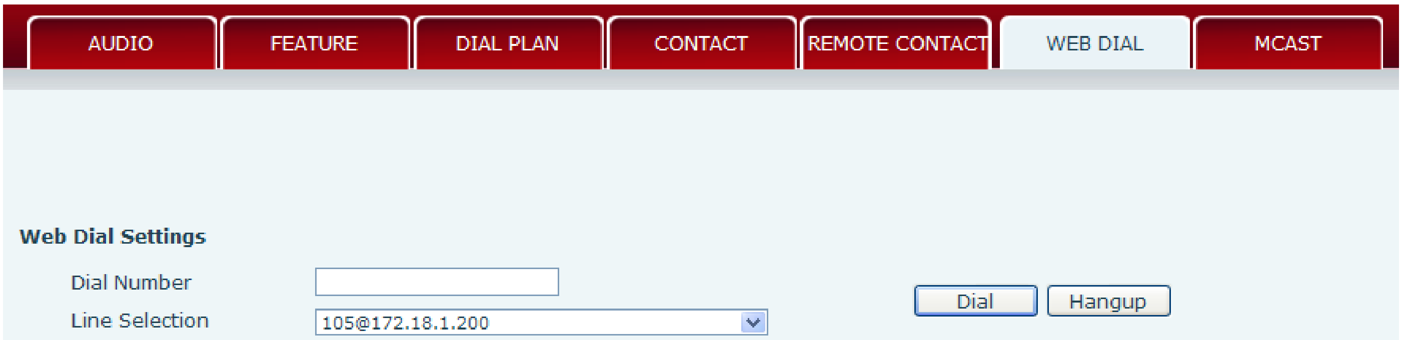
| | |
|---|---|
| Note: the capacity specified phone book is up to 500 records. You can add one or more add a contact to a group or a black list, click Delete to delete multiple contacts, click delete all delete all contacts have been added. | |
| **Add Contact** | |
| Name | Contact name |
| Office 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.<br>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.<br>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.<br>Allowed number lists must end with an entry which is only a “.”. | |

## e) REMOTE CONTACT



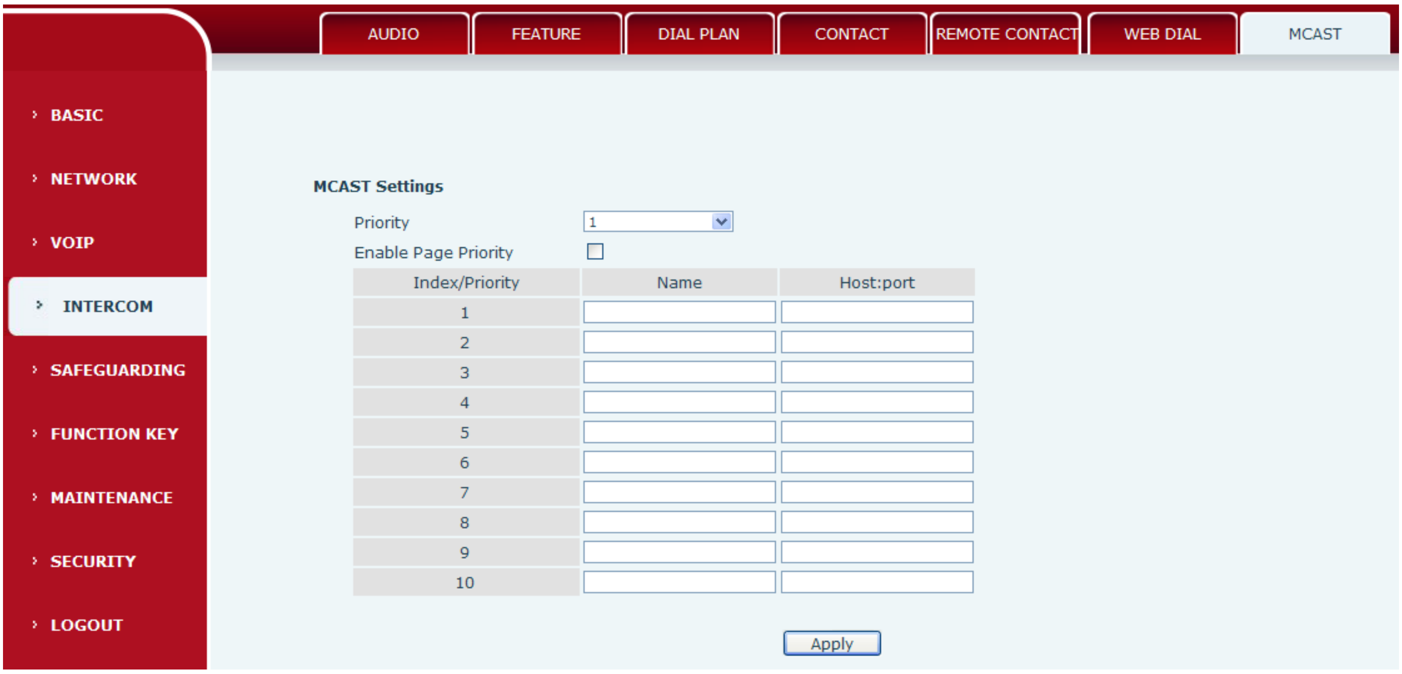

| Remote Phonebook Settings | |
|---|---|
| **Field Name** | **Explanation** |
| Phonebook name | Phonebook name displayed on the phone. |
| Server URL | Server URL of the remote phonebook. |
| SIP line | SIP line for the remote phonebook. |
| Authentication | Authentication mode for remote phonebook. |
| 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:

**MCAST 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, the highest priority;

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

**Listener configuration**

**MCAST Settings**

Priority: 3

Enable Page Priority: ☑

| Index/Priority | Name | Host:port |
|---|---|---|
| 1 | group 1 | 224.0.0.2:2366 |
| 2 | group 2 | 224.0.0.4:1366 |
| 3 | group 3 | 224.0.0.6:3366 |
| 4 | | |
| 5 | | |
| 6 | | |
| 7 | | |
| 8 | | |
| 9 | | |
| 10 | | |

- **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)**

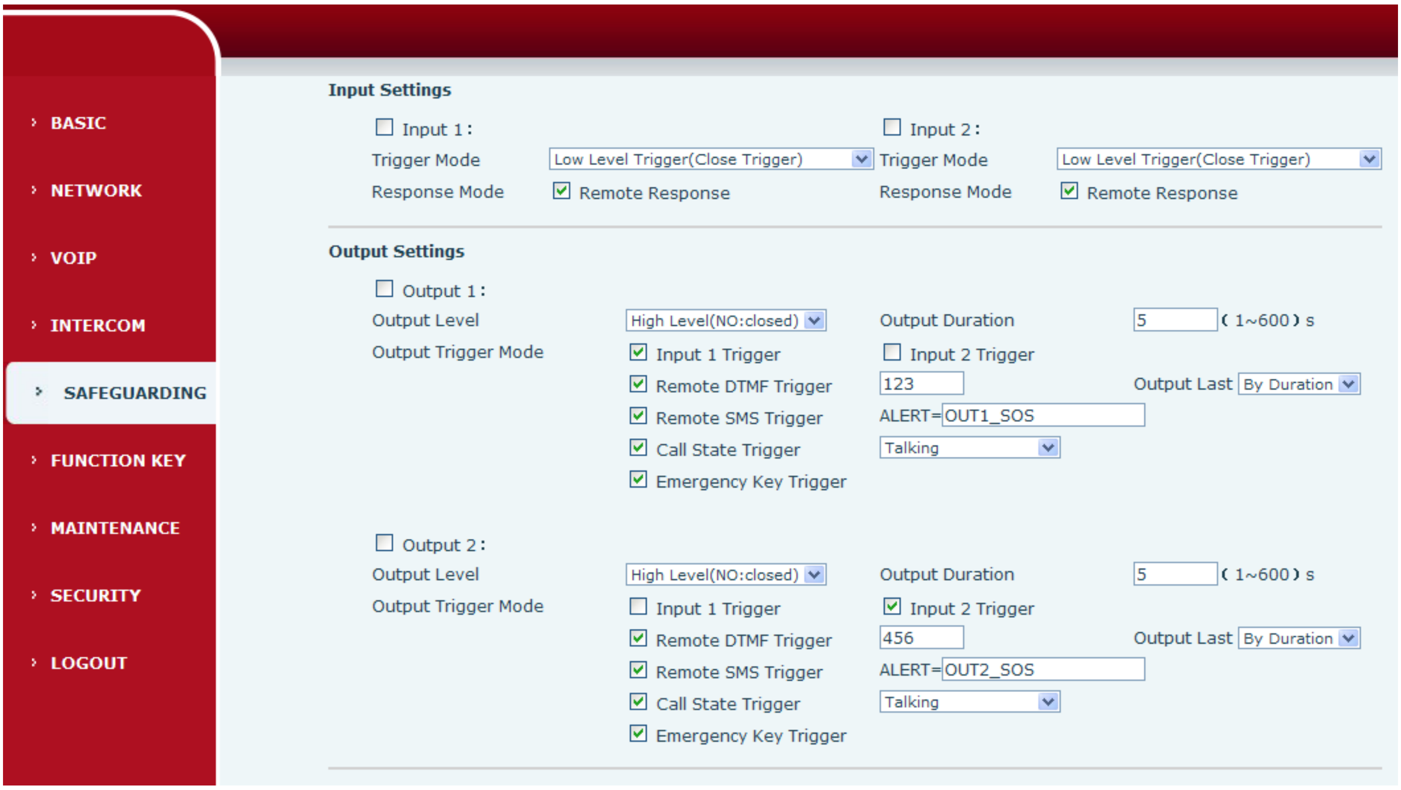
Set whether to open more priority is the priority of multicast, multicast is pink part number. Explain how to use:

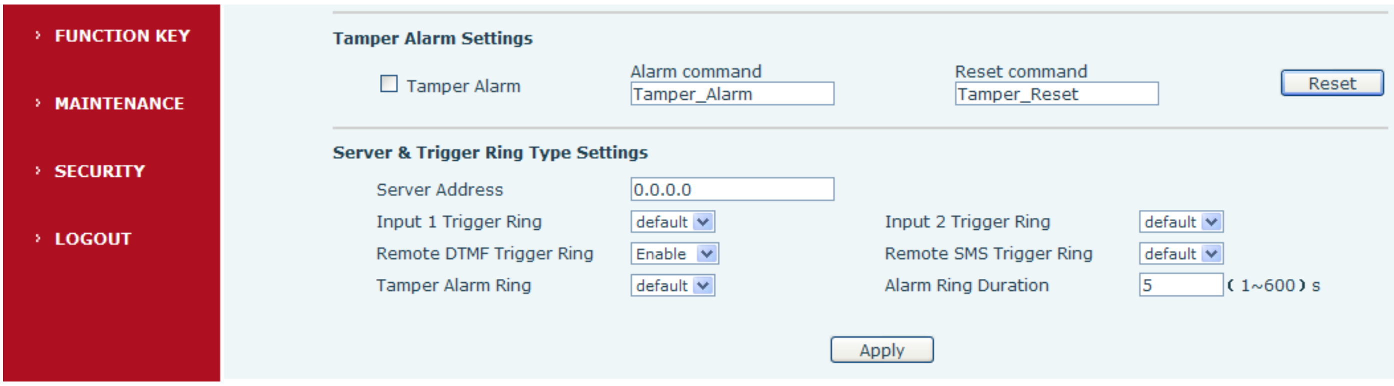
- ✧ The purpose of setting monitoring multicast "group 1" or "3" set up listening "group of 1" or "3" multicast address multicast call.
- ✧ All equipment has been a path or multi-path multicast phone, such as listening to "multicast information group 2".
- ✧ If multicast is a new "group of 1", because "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. |

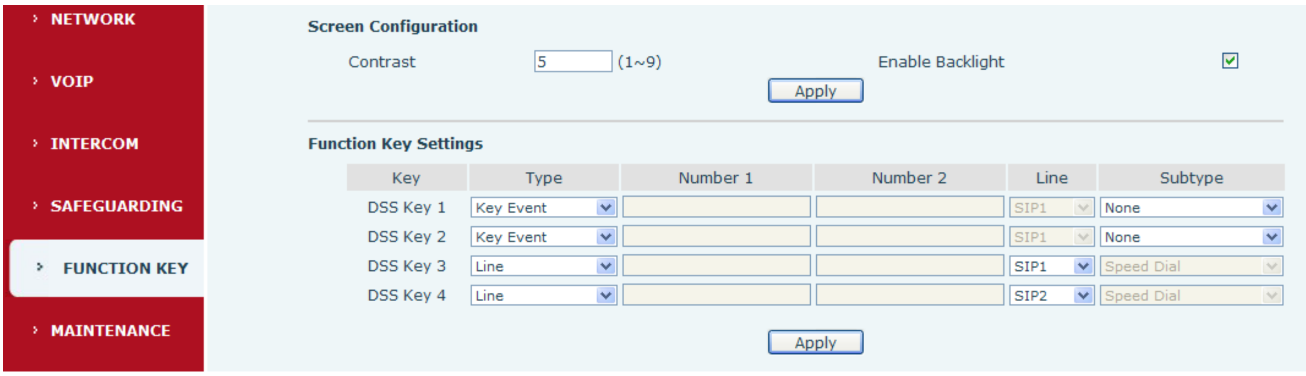
| Field Name | Explanation | |
|---|---|---|
| Output Trigger Mode: There are many kinds of trigger modes, multiple choices. | | |
| Input port1 trigger | When the input port1 meet to trigger condition, the output port1 will trigger(The Port level time change, By < Output Duration > control) | |
| Input 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.



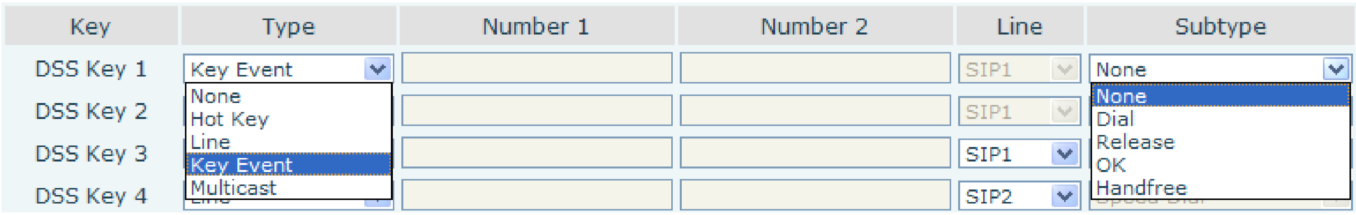

### a) Screen settings

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

## b) Function key settings

- **Key Event Settings**

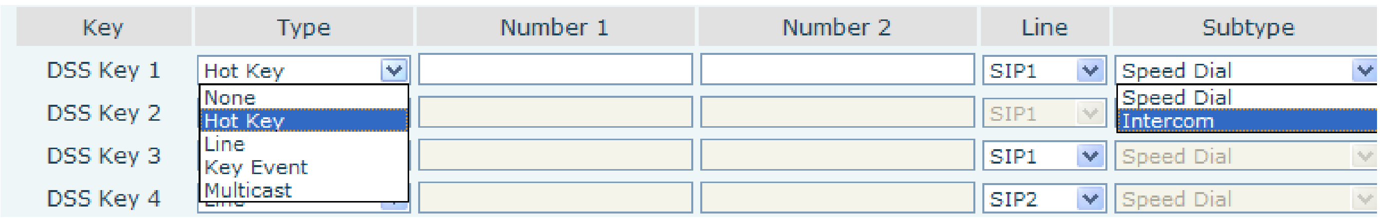
The Subtype configuration of Hot key.

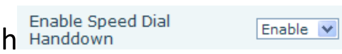



| DSS key type | Subtype | Usage |
|---|---|---|
| Key Event | None | Not responding |
| | Dial | Dial function |
| | Release | End calls |
| | OK | Identify key |
| | Handfree | The hand-free key(with hook dial, hang up) |

- **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.

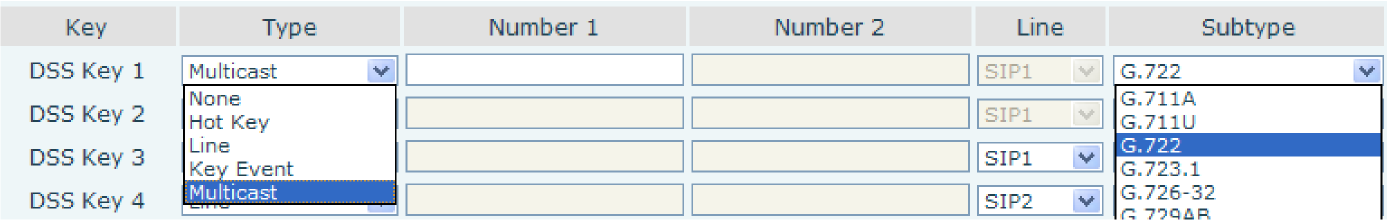



| DSS key type | Number | Line | Subtype | Usage |
|---|---|---|---|---|
| Hot key | Fill the called party's SIP account or address | The SIP account corresponding lines | Speed Dial | In Speed dial mode, with  can define whether this call is allowed to be hang up by re-press the speed dial |
| | | | 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

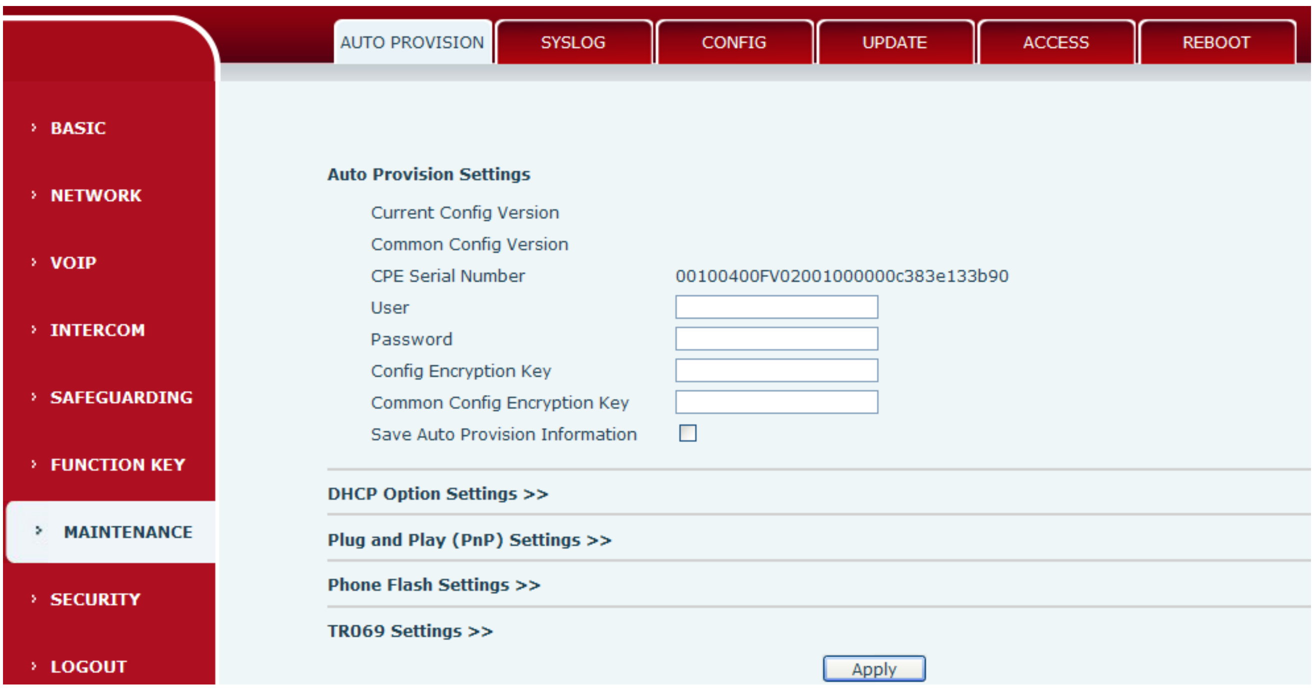
Device through the DSS Key configuration of multicast address and port and started coding; set by WEB to monitor the multicast address and port; device sends a multicast, listens to the address of the device can receive the multicast content.

✧ 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



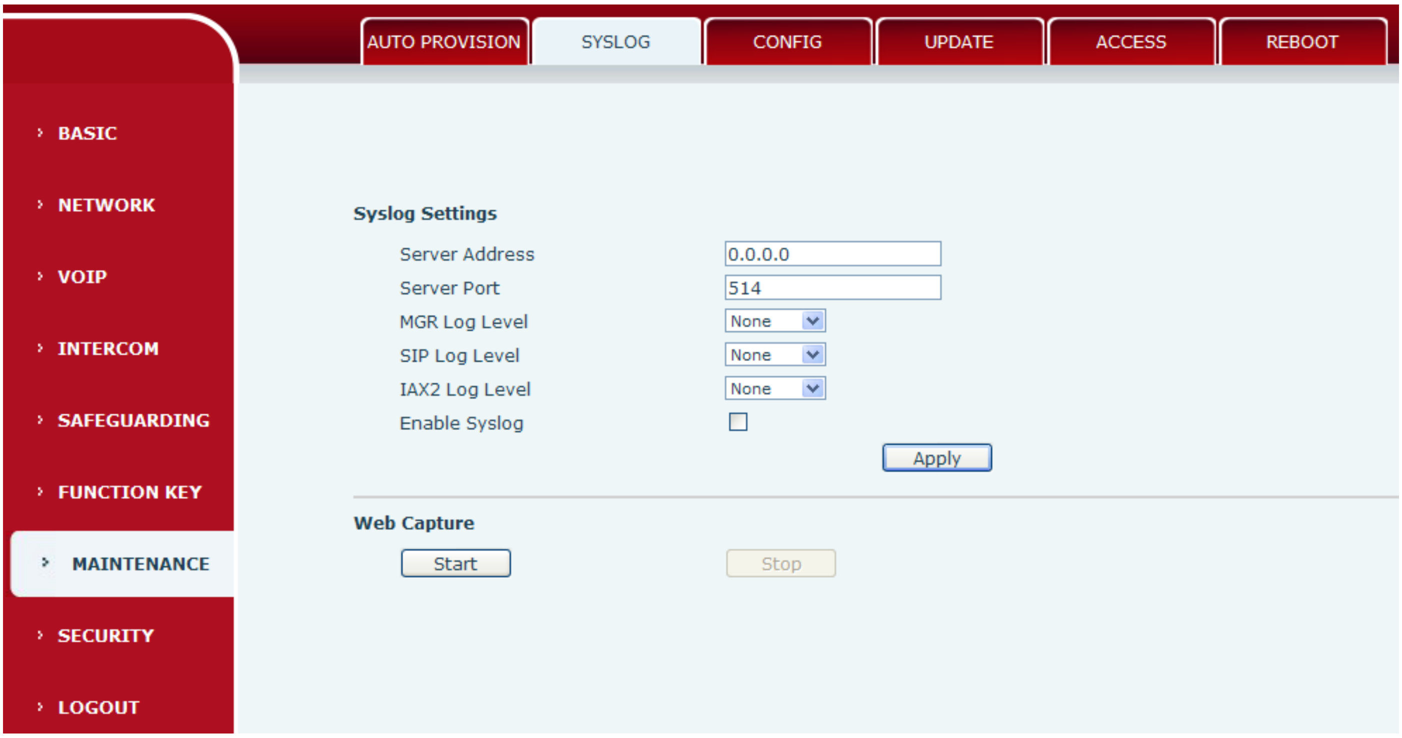
The equipment supports PnP, DHCP, and Phone Flash to obtain configuration parameters. They will be queried in the following order when the equipment boots.

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<br>2. Update after reboot – update only after reboot.<br>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.

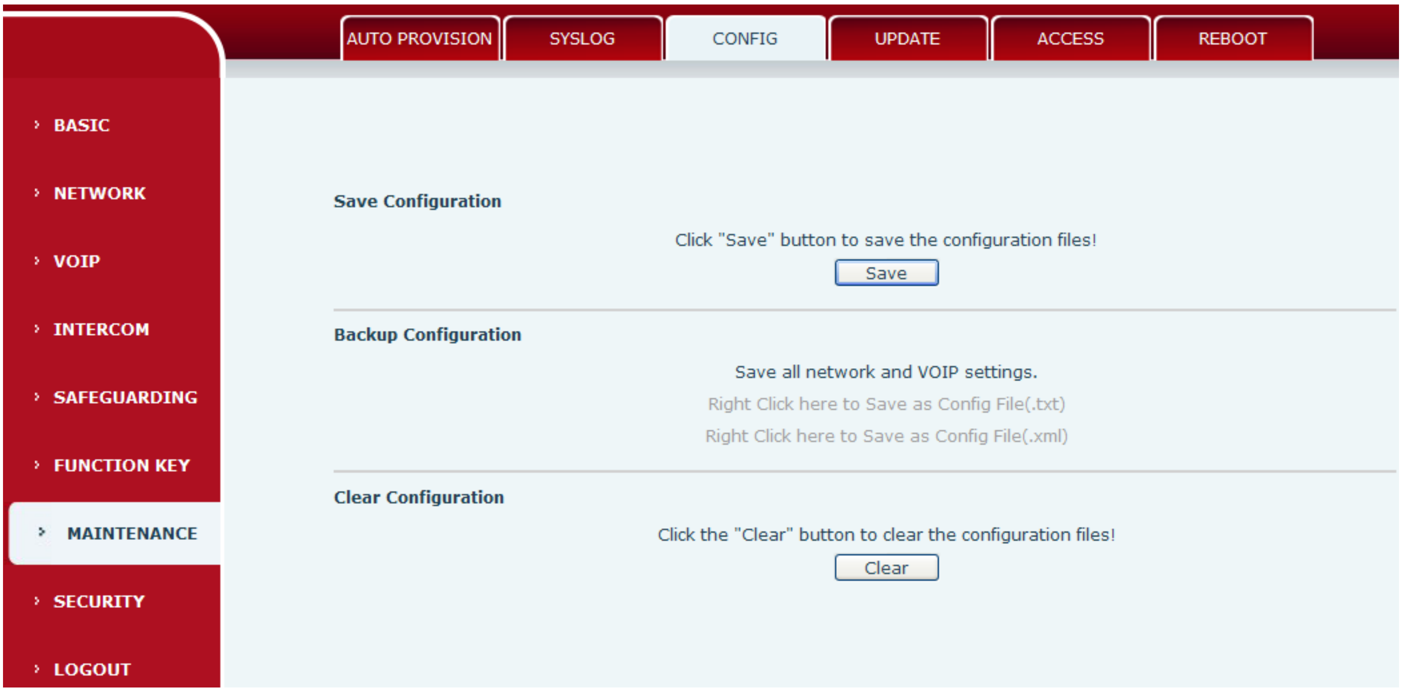
Level 5: notice; It is the normal but significant condition.

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

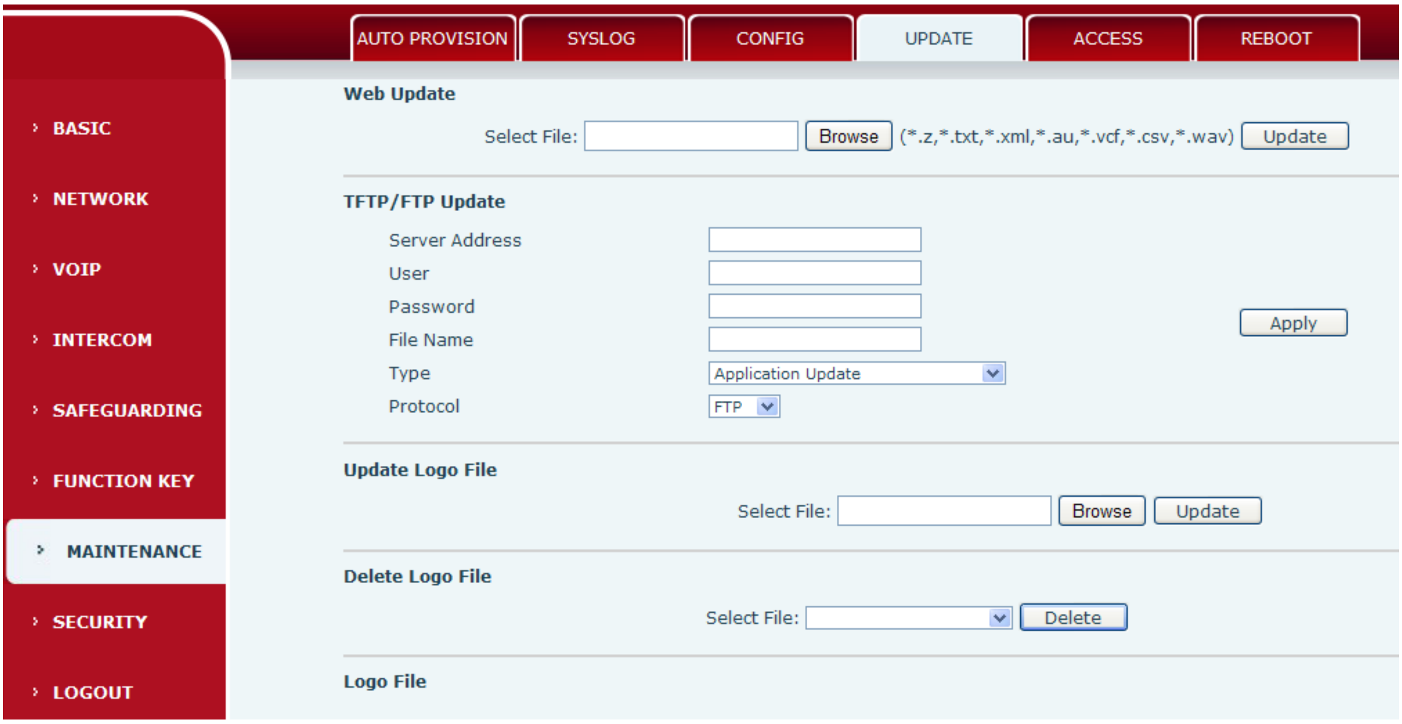



| Field Name | Explanation |
|---|---|
| Save Configuration | Save the current equipment configuration. Clicking this saves all configuration changes and makes them effective immediately. |
| Backup Configuration | Save the equipment configuration to a txt or xml file. Please note to Right click on the choice and then choose “Save Link As.” |

| | |
|---|---|
| Clear Configuration | Logged in as Admin, this will restore factory default and remove all configuration information.<br>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.

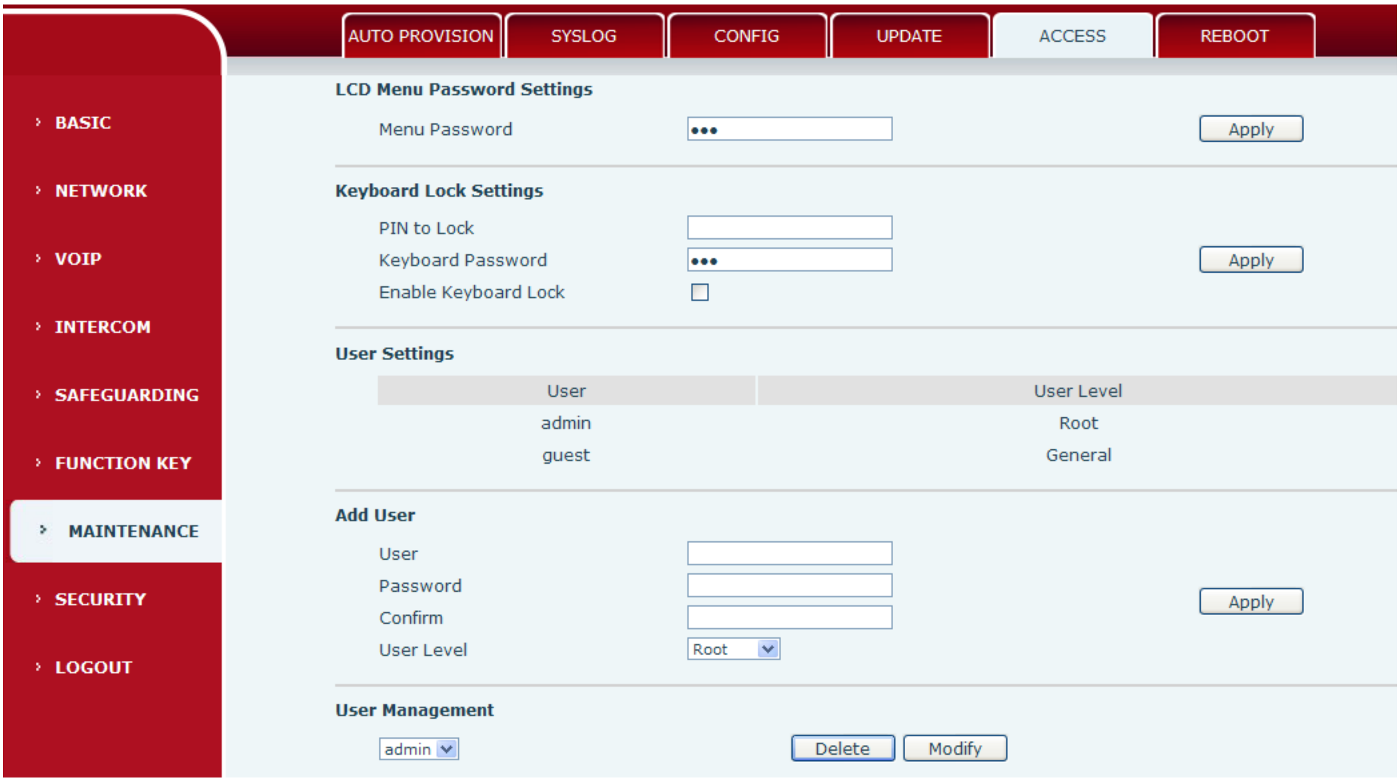



| Field Name | Explanation |
|---|---|
| **Web Update** | Browse to the config file, and press Update to load it to the equipment. Various types of files can be loaded here including firmware, ring tones, local phonebook and config files in either text or xml format. |
| **TFTP/FTP Update** | |
| Server | 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 :<br>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.<br>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. | |

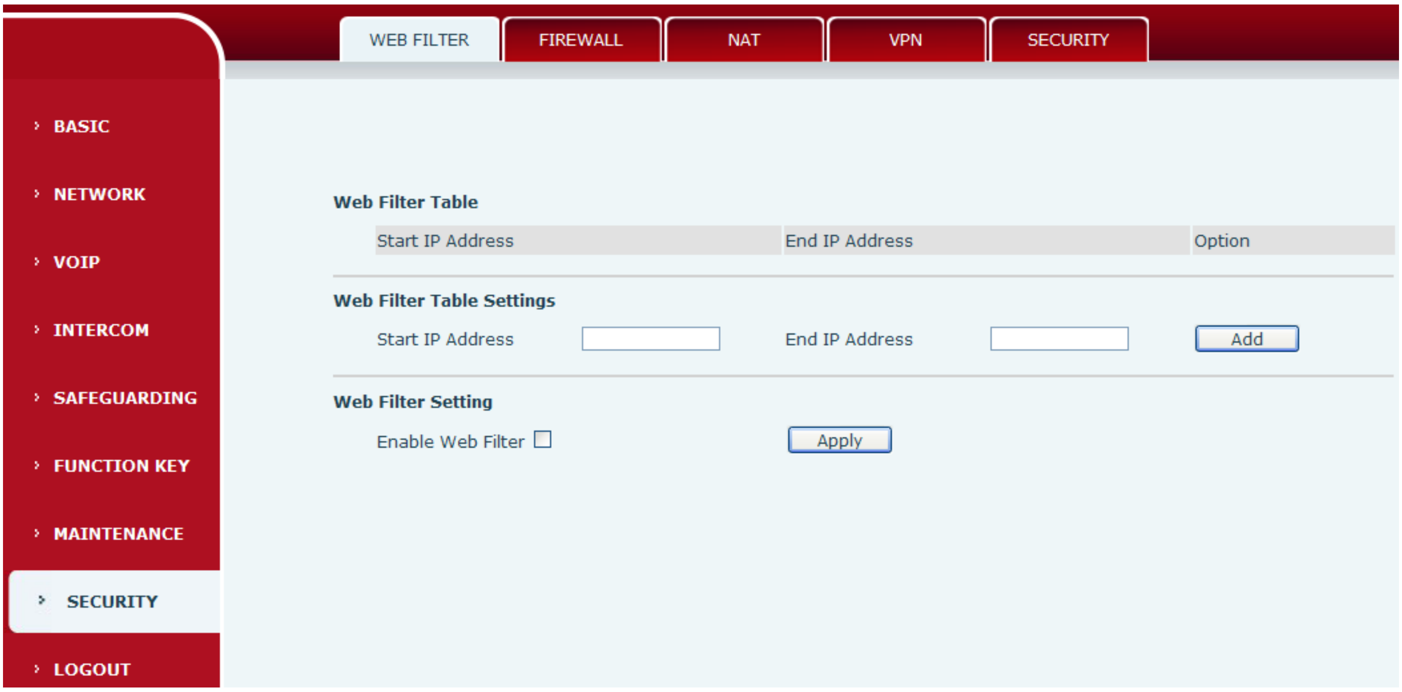
### f) REBOOT

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.

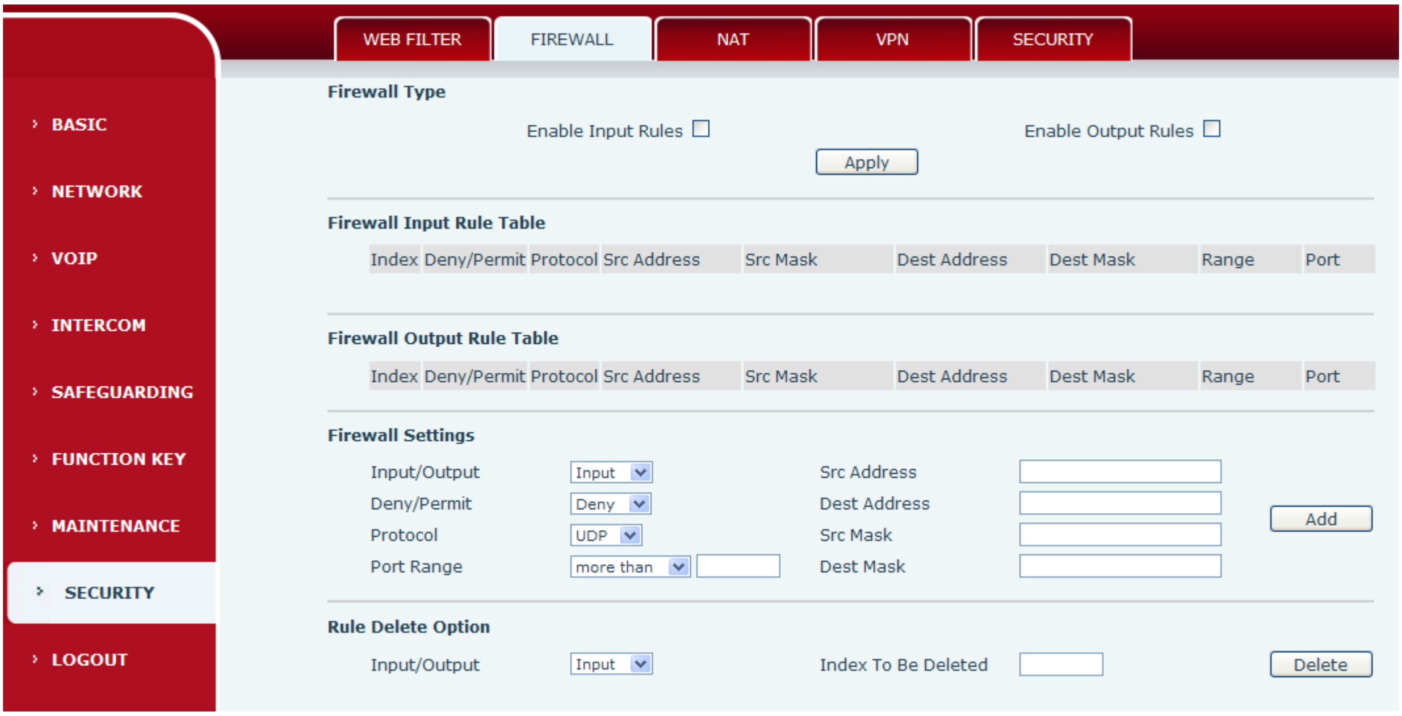
## (8)SECURITY

### 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

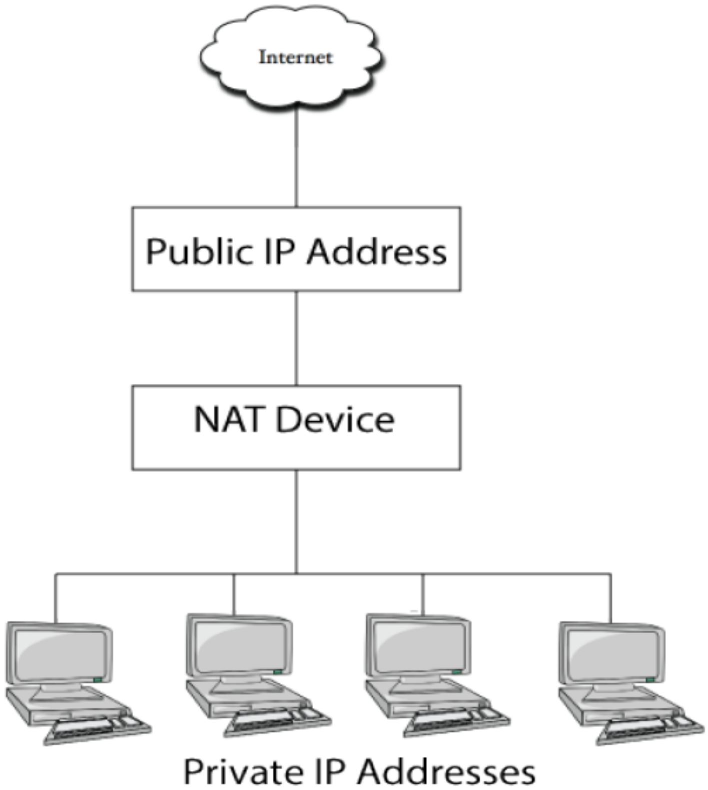



| **Firewall** | |
|---|---|
| Firewall rules can be used to prevent unauthorized Internet users from accessing private networks connected to this phone (input rule), or prevent unauthorized devices connected to this phone from accessing the Internet (output rule). Each rule type supports a maximum of 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 | Set the destination address mask. For example: 255.255.255.255 points to one host |

| | |
|---|---|
| Mask | while 255.255.255.0 points to a C type network. |

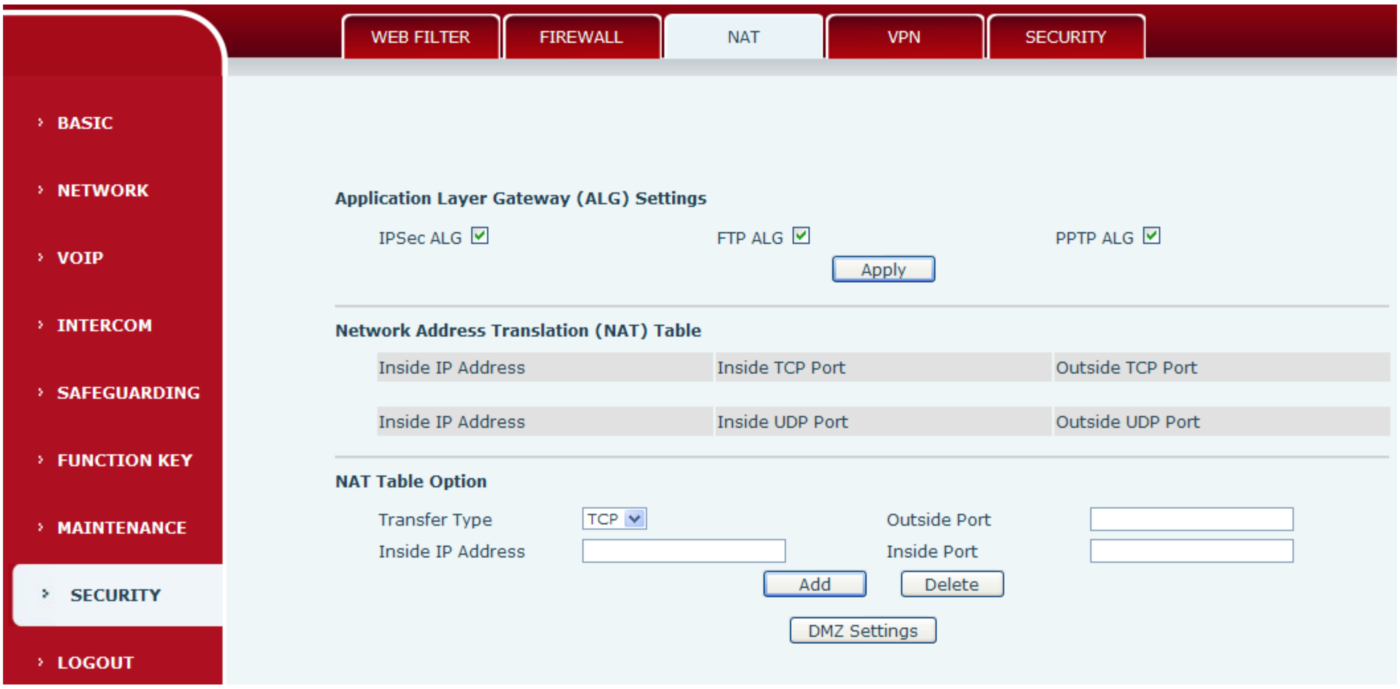
## c) NAT

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.

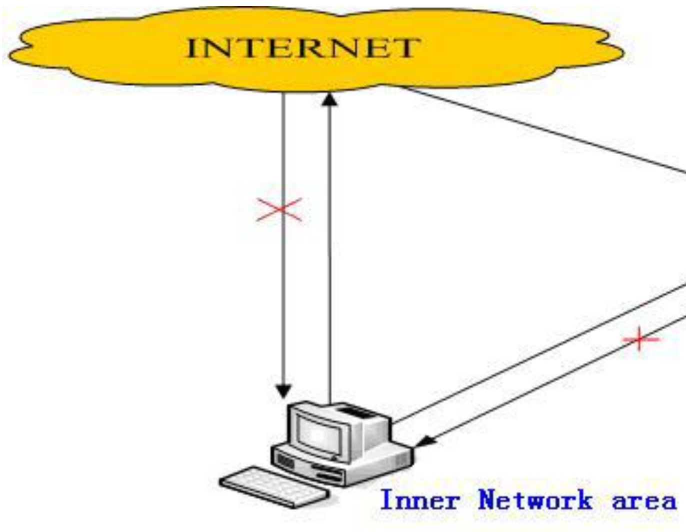



DMZ configuration：

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.

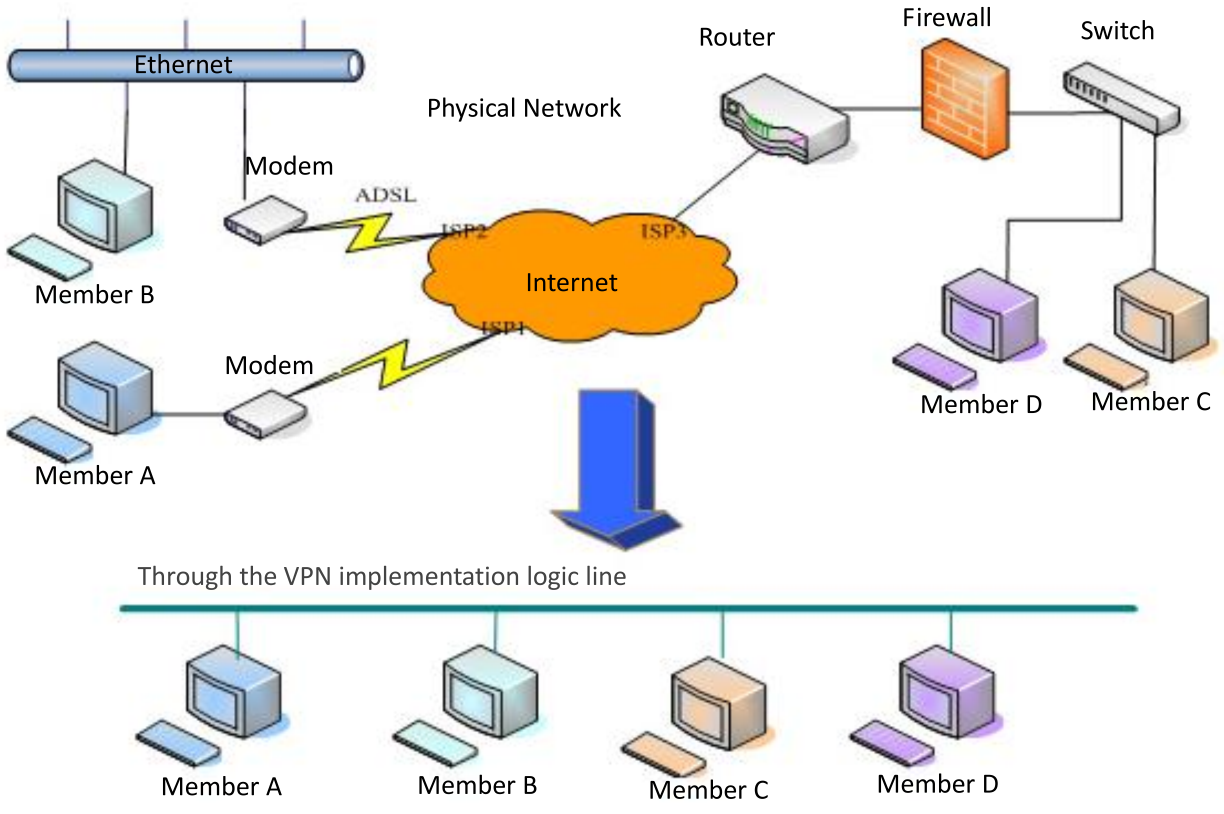


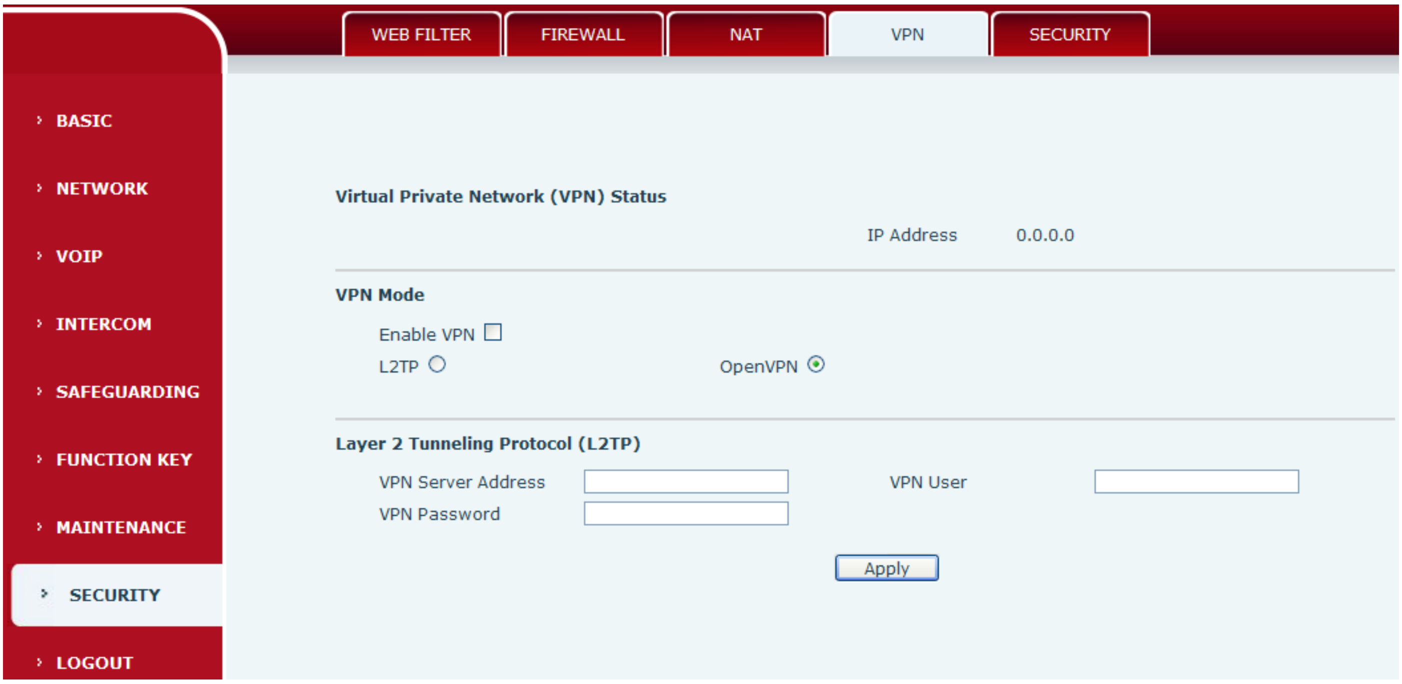

| **Field Name** | **Explanation** |
|---|---|
| **Protocol type Settings** | |
| IPSec ALG | Enable/Disable IPSec encryption. Default is enabled. |
| FTP ALG | Allow the ALG to securely pass FTP traffic. Default is enabled. |
| PPTP ALG | Allow the ALG to securely pass PPTP traffic. Default is enabled. |
| 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 Address | Set the local IP address of device. |
| 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

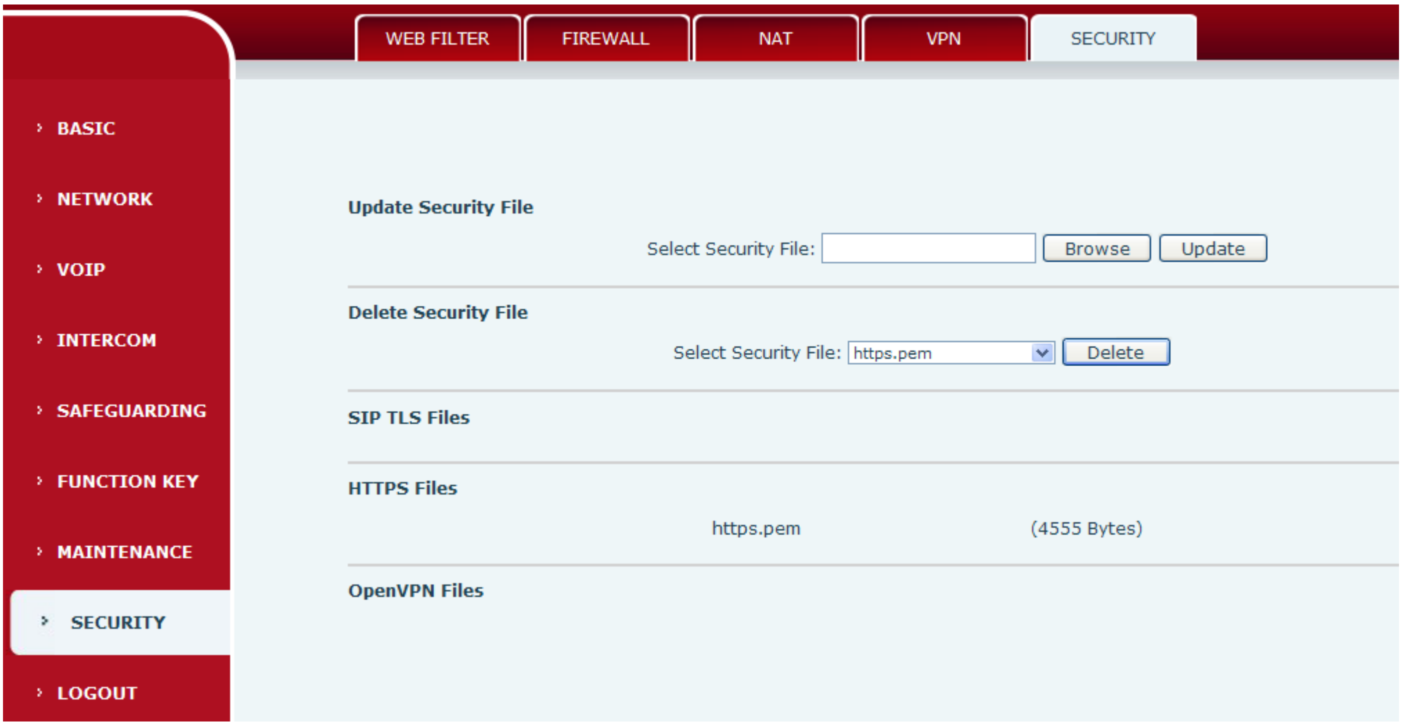
The device supports remote connection via VPN. It supports both Layer 2 Tunneling Protocol (L2TP) and OpenVPN protocol. This allows users at remote locations on the public network to make secure connections to local networks.





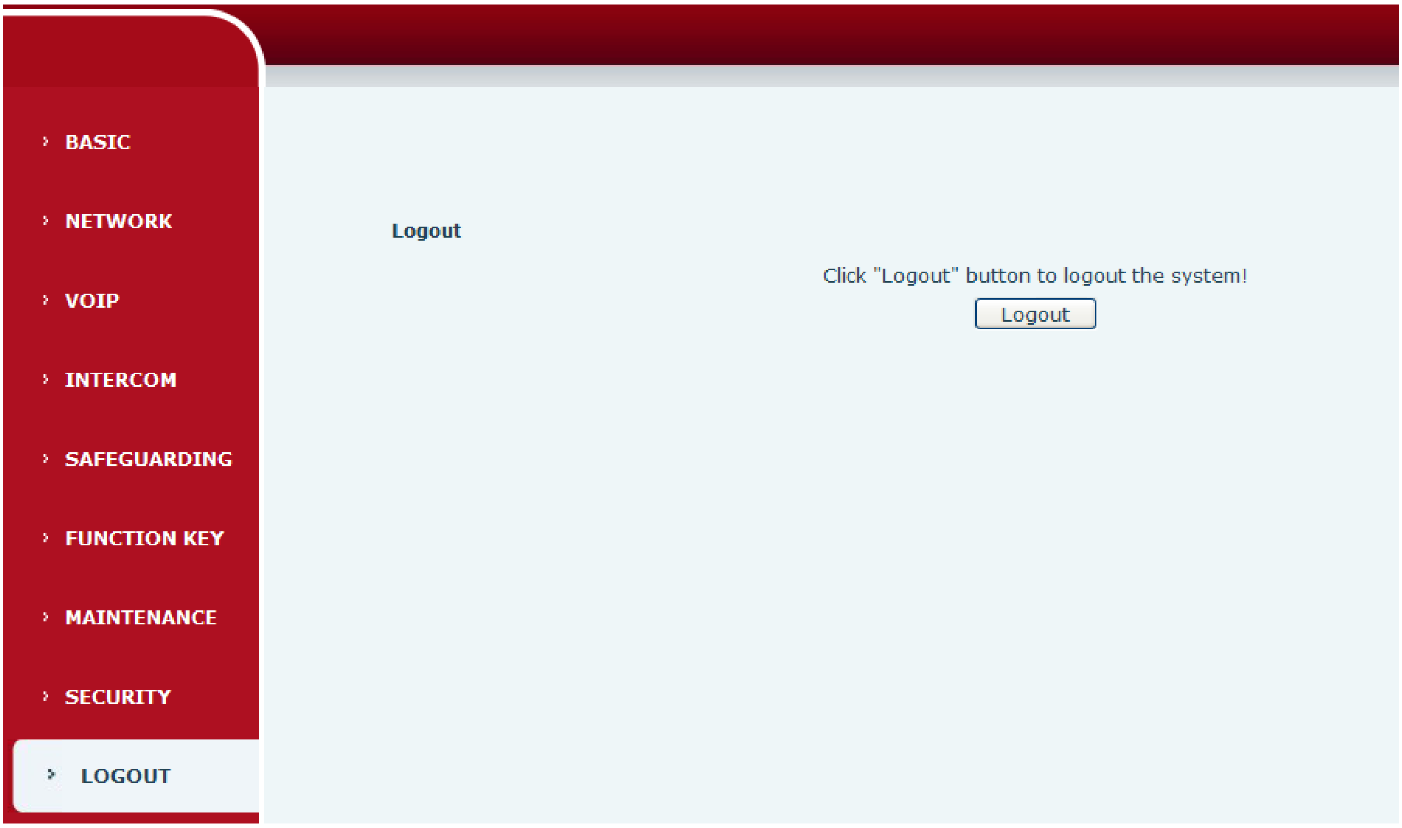

| Field Name | Explanation |
|---|---|
| VPN IP | Shows the current VPN IP address. |
| **VPN type** | |
| Enable VPN | Enable/Disable VPN. |
| L2TP | Select Layer 2 Tunneling Protocol |
| Open VPN | Select OpenVPN Protocol. (Only one protocol may be activated. After the selection is made, the configuration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

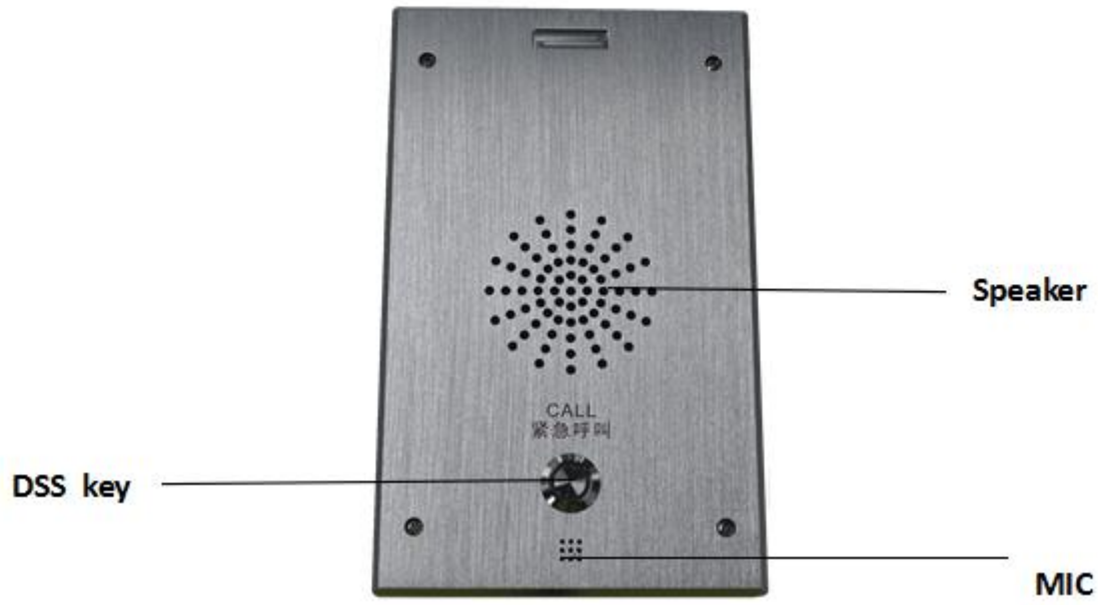
| | | |
|---|---|---|
| **Main chipset** | | Broadcom |
| **Speech flow** | **Protocols** | RTP/SRTP |
| | **Decoding** | G.729、G.723、G.711、G.722、G.726 |
| | **Audio amplifier** | 2.5W |
| | **Volume control** | Adjustable |
| | **Full duplex speakerphone** | Support (AEC) |
| **Port** | **DSS key** | 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 Omni directional 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

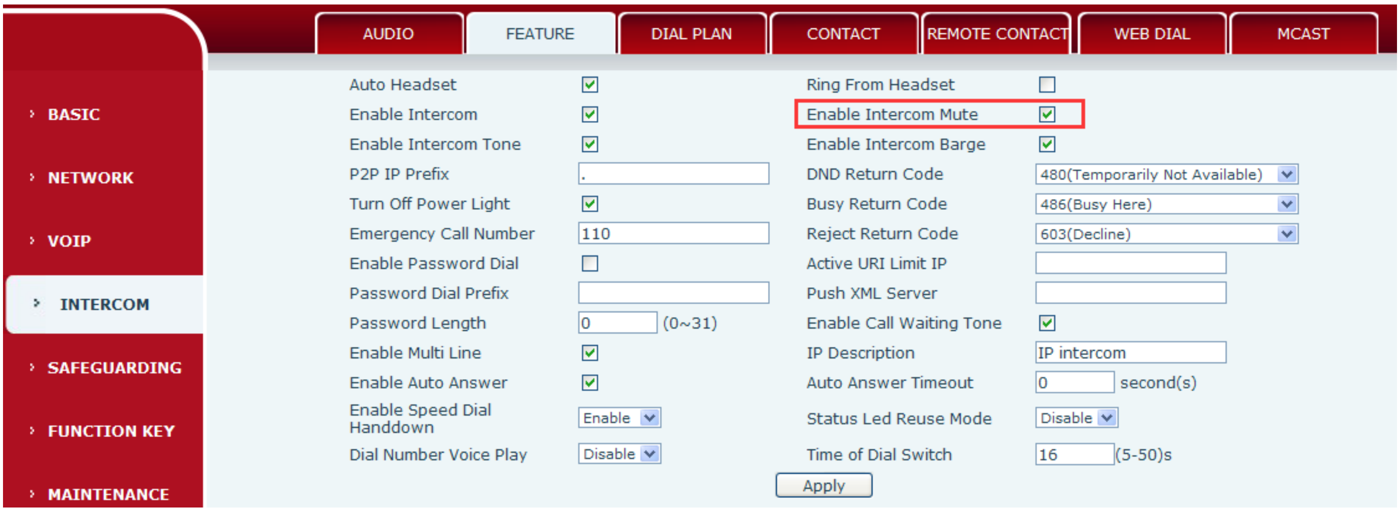



## 4. The radio terminal configuration notice

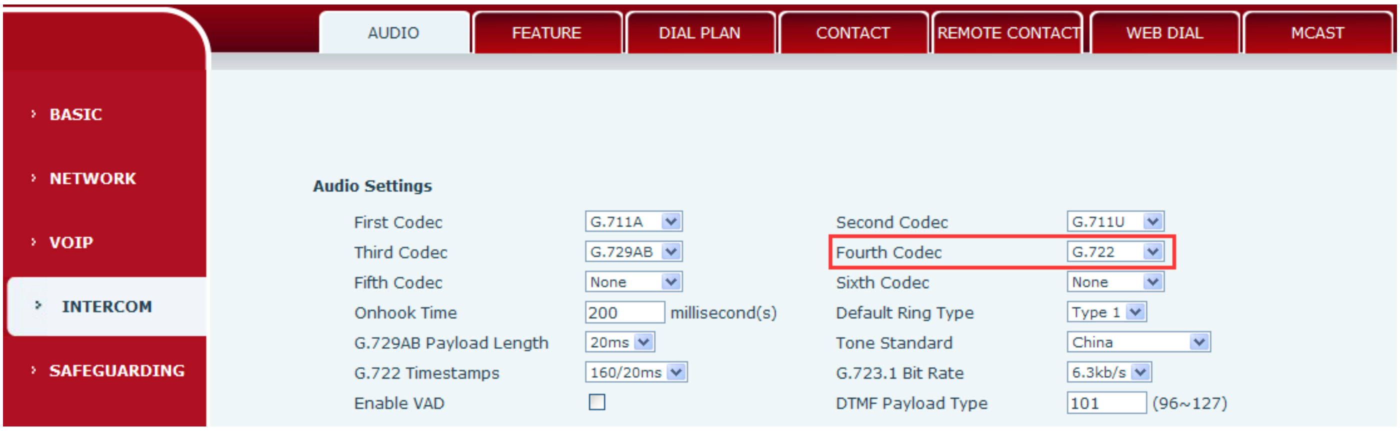
✧ **How to avoid an incoherency sound when the radio playing？**

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.

**Speedytel Technology Co., Ltd;** Add: Deyefeng Business Center, Gongyuan Road, Baoan District, Shenzhen, China
Tel: +86-755-29129392; Fax: +86-755-29124300; Email: sales@speedytel.com; support@speedytel.com.



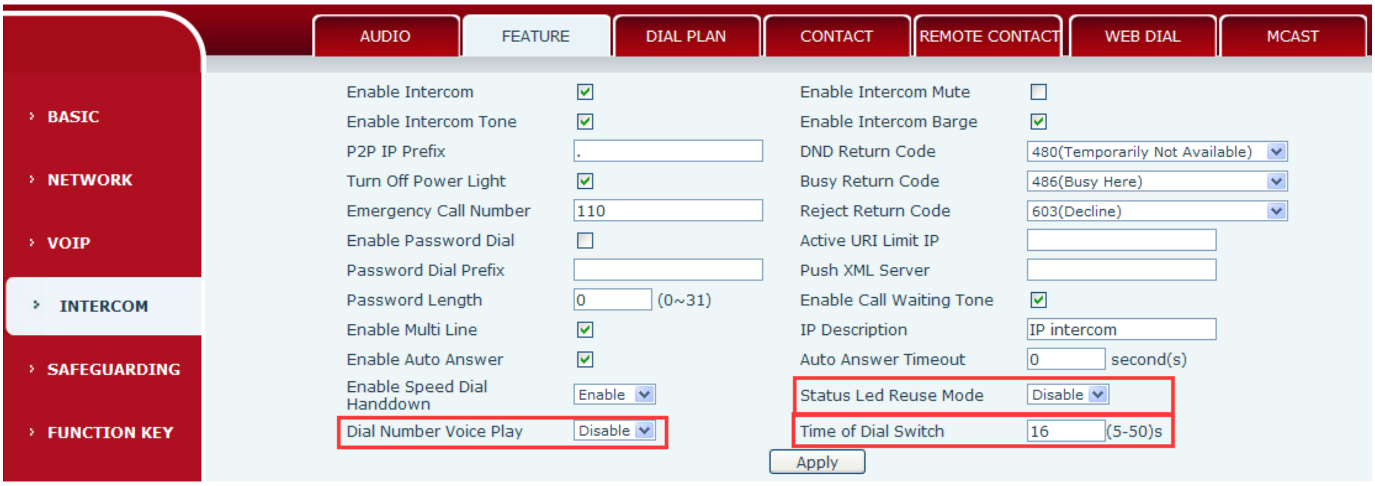

### ✧ How to improve broadcasting quality?




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.