# i18

# IP Video Intercom User Manual



**Wall mounted** **In-wall**

# Safety Notices

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

# A. Product introduction

i18 is a full digital natwork Video Intercom, with its core part adopts mature VoIP solution, stable and reliable performance, hands-free adopting digital full-duplex mode, voice loud, video clear, generous appearance, solid durable, easy for installation, comfortable keypad and low power consumption.

## 1. Appearance of the product

Single button

Dual button

## 2. Description

| Picture | Description | Function |
|---|---|---|
|   | DSS Key LED | Network error: Blink with 2s<br>Network running: Off<br>Registration failed: Blink with 6s<br>Registration succeeded: On |

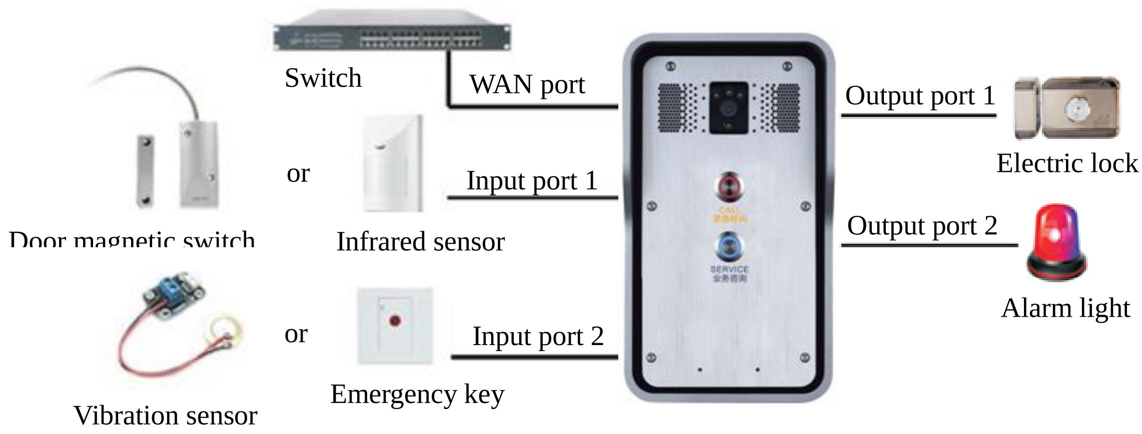
## B. Start Using

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

### 1. Confirm the connection

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

#### 1) Power port




#### 2) Power port

Power supply ways: 12v/DC or POE.

| CN16 | |  |
|---|---|---|
| **1** | **2** | |
| +12V | GND | |
| 12V 1A/DC | | |

#### 3) Security functions Input port

| CN10 | | | |  |
|---|---|---|---|---|
| **4** | **3** | **2** | **1** | |
| GND | IN2 | GND | IN2 | |
| **Input port 2** | | **Input port 1** | | |

4) **Security functions Output port**

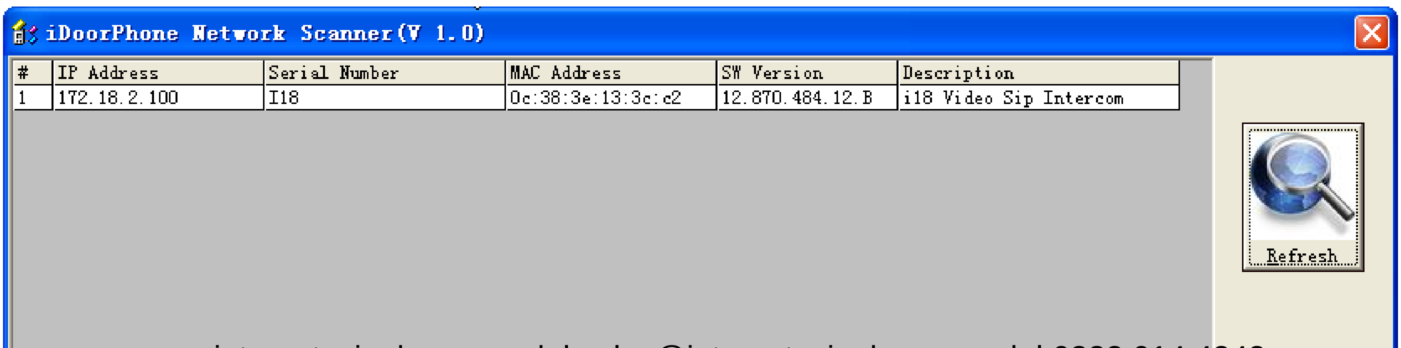
| CN11 | | | | | |
|---|---|---|---|---|---|
| **6** | **5** | **4** | **3** | **2** | **1** |
| **NC2** | **COM** | **NO2** | **NC1** | **COM** | **NO1** |
| **Normally close** | **common port** | **Normally open** | **Normally close** | **common port** | **Normally open** |
| **Output port 2** | | | **Output port 1** | | |


## 2. Quick Setting

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

In prior to this step, please make sure your broadband Internet online can be normal operated, and complete the connection of the network hardware. The product factory default network mode is DHCP. Thus, only connect equipment with DHCP network environment that network can be automatically connected.

- Press and hold "#" key for 3 seconds and the door phone will report the IP address by voice, or use the "iDoorPhoneNetworkScanner.exe " software to find the IP address of the device.
  (Download address http://download.fanvil.com/tool/iDoorPhoneNetworkScanner.exe )
  **Note:** when power on, 30s waiting is needed for device running.
- 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 set DSS key in the Webpage (Intercom -> function key).
- You can set function parameters in the Webpage (Safeguarding).



# C. Basic operation

## 1. Answer a call

When a call comes in, the device will answer automatically. If you cancel auto answer feature and set auto answer time, you will hear the bell ring at the set time and the device will auto answer after a timeout.

## 2. Call

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

## 3. End Call

Enable DSS key hang up to end call.

## 4. Call record

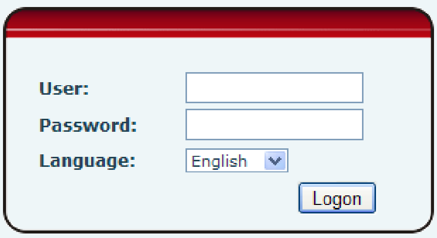
The device provides three call records, missed call, received call, dialed call. You can see call records of the webpage.

# D. Page settings

## 1. Browser configuration

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

Enter the user name and password and click the [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 has been rebooted.

## 2. Password Configuration

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

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

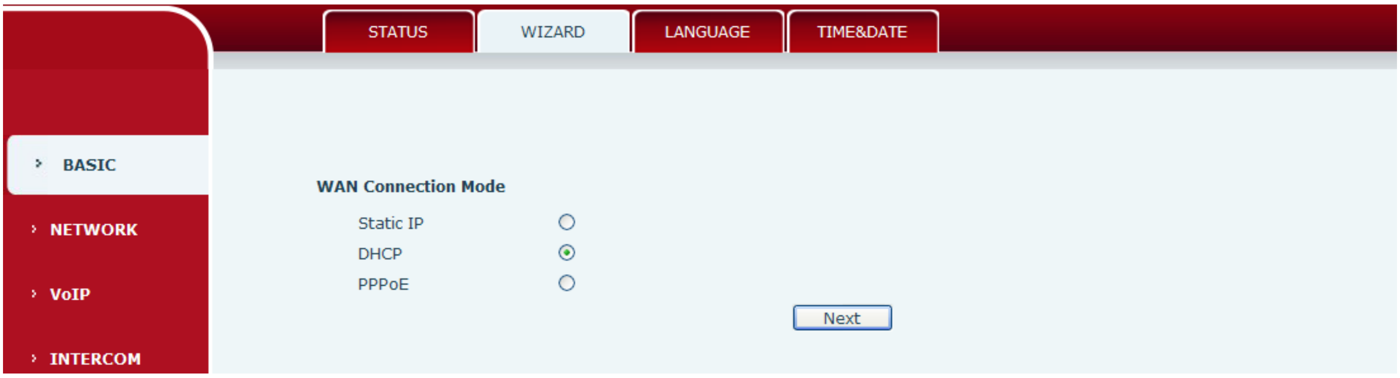
## 3. Configuration via WEB

### (1) BASIC

### a) STATUS


| **Status** | |
|---|---|
| **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. |
| Accounts | Shows the phone numbers and registration status for the 2 SIP LINES. |

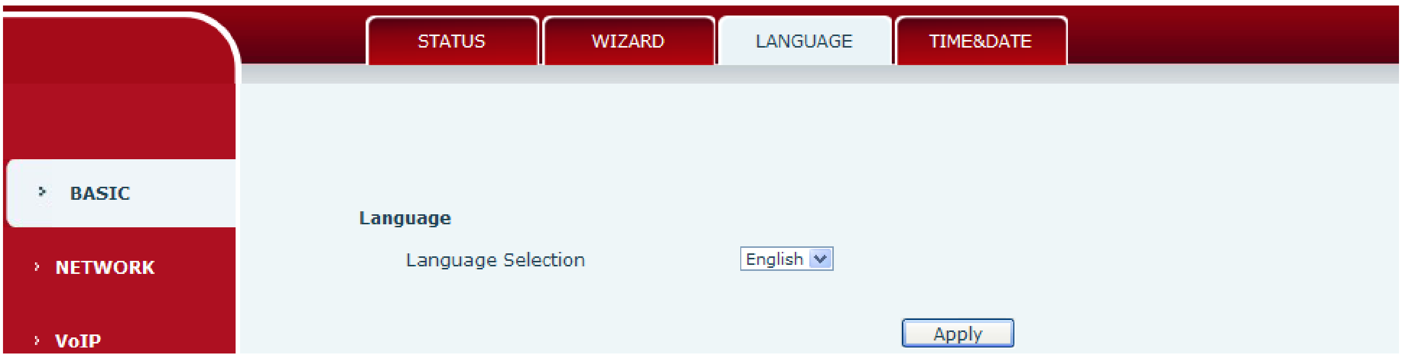
## b) WIZARD




| 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. | |
| 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. | |

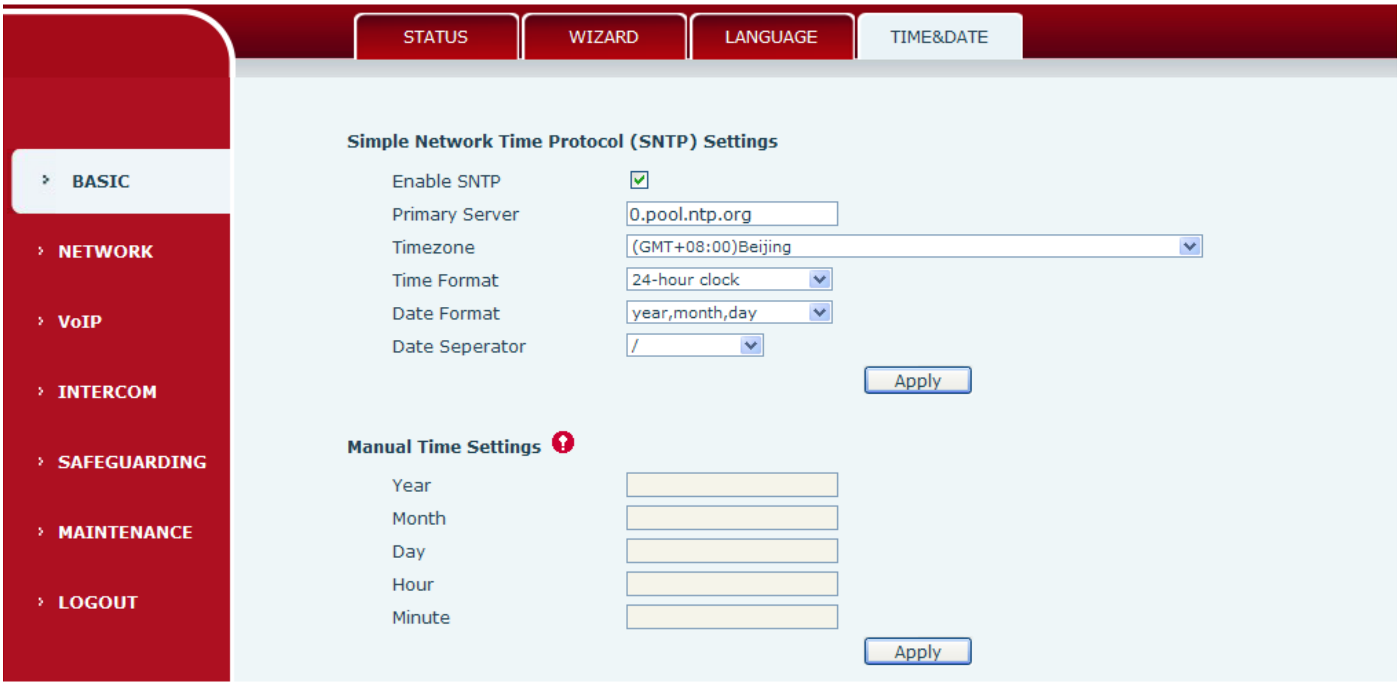
## c) LANGUAGE

Set the current language.



## d) TIME&DATE

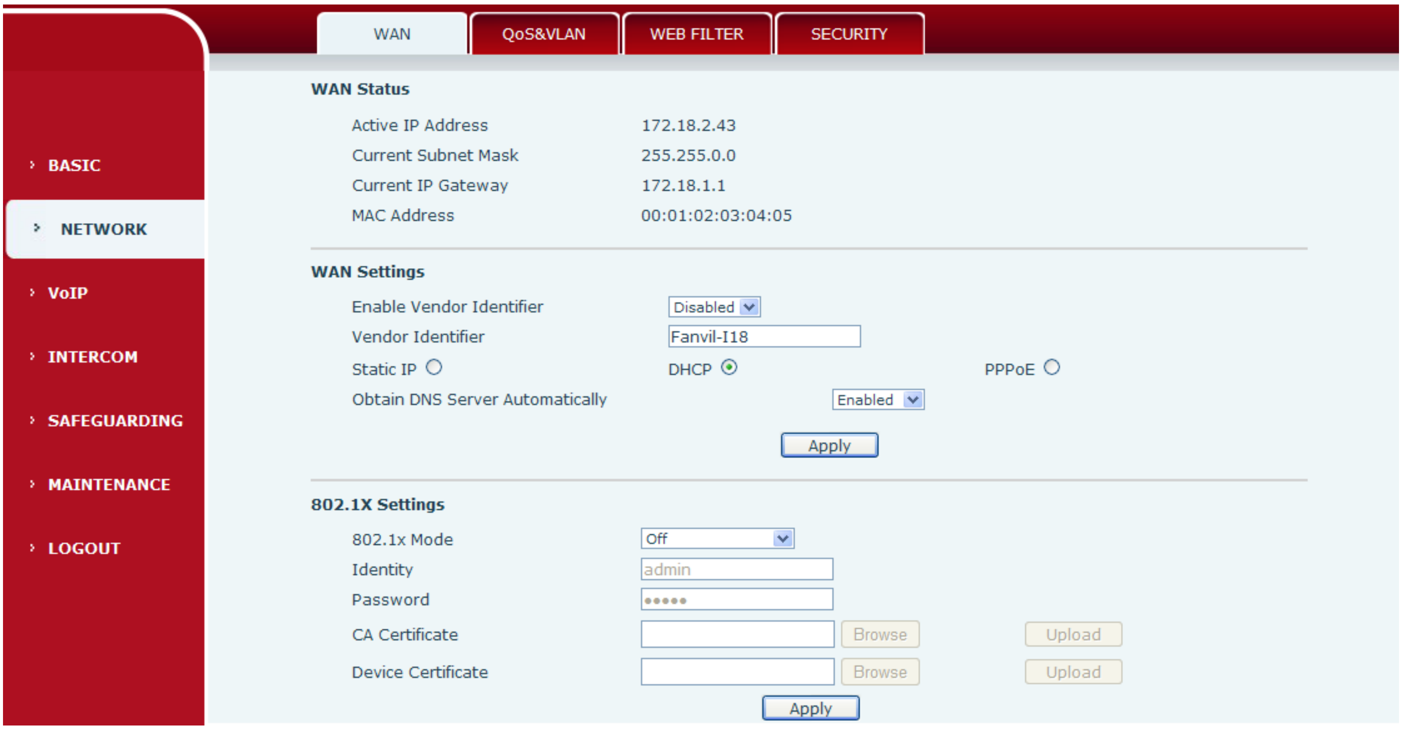
Set the time zone and SNTP (Simple Network Time Protocol) server on this page. Daylight Saving Time configuration and Manual Time and Date entry can also be done in this page.

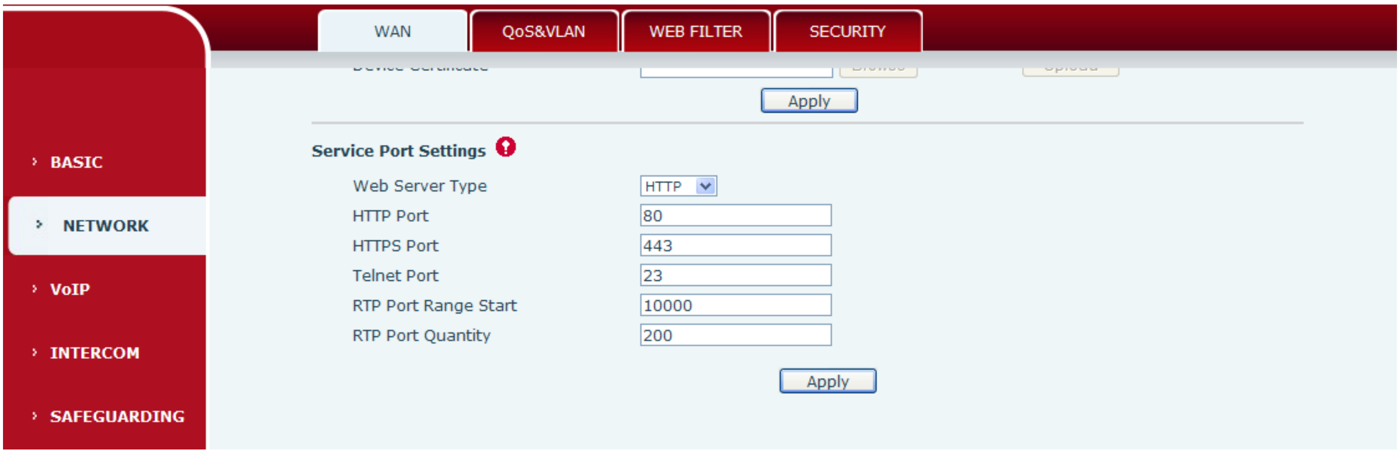



| Time&Date | |
|---|---|
| **Field Name** | **Explanation** |
| **System Current Time** | |
| Display the current time | |
| **Simple Network Time Protocol (SNTP) Settings** | |
| Enable SNTP | Enable or Disable SNTP |
| Primary Server | IP address of Primary SNTP Server |
| Time zone | Local Time Zone |
| Time Format | Configuration time format, the default is 24 hours. |
| Date Format | Configure date display format, the default is (date) (month) (year) |
| Date Seperator | Configure the date seperator |
| **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. | |

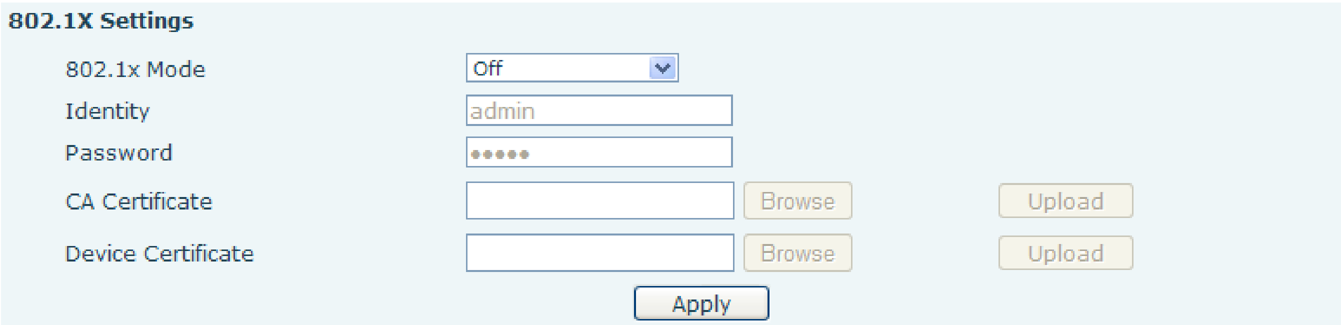
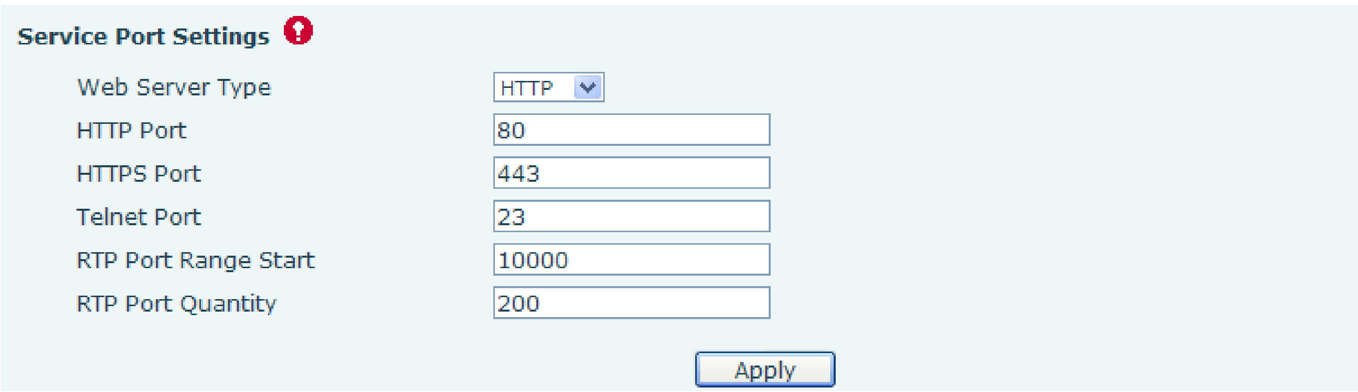
## (2) NETWORK

### a) WAN




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

<table>
<tr><td colspan="2">WAN Settings</td></tr>
<tr><td>Enable Vendor Identifier</td><td>Enable or disable Vendor Identifier</td></tr>
<tr><td>Vendor Identifier</td><td>Configure display Vendor Identifier</td></tr>
<tr><td colspan="2">Select the appropriate network mode. The equipment supports three network modes:</td></tr>
<tr><td>Static</td><td>Network parameters must be entered manually and will not change. All parameters are provided by the ISP.</td></tr>
<tr><td>DHCP</td><td>Network parameters are provided automatically by a DHCP server.</td></tr>
<tr><td>PPPoE</td><td>Account and Password must be input manually. These are provided by your ISP.</td></tr>
<tr><td colspan="2">If Static IP is chosen, the screen below will appear. Enter values provided by the ISP.</td></tr>
<tr><td colspan="2">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.</td></tr>
<tr><td colspan="2">802.1X Settings</td></tr>
<tr><td colspan="2">
</td></tr>
<tr><td>User</td><td>802.1X user account</td></tr>
<tr><td>Password</td><td>802.1X password</td></tr>
<tr><td>Enable 812.1X</td><td>Enable or Disable 812.1X</td></tr>
<tr><td>CA Certificate</td><td>Choose the CA Certificate and then click upload to upgrade</td></tr>
<tr><td>Device Certificate</td><td>Choose the Device Certificate and then click upload to upgrade</td></tr>
<tr><td colspan="2">Service port Settings</td></tr>
<tr><td colspan="2">
</td></tr>
</table>

| 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. | |

### b) 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, frames broadcast from Port 1 will be sent to Ports 2, 3, and 4.

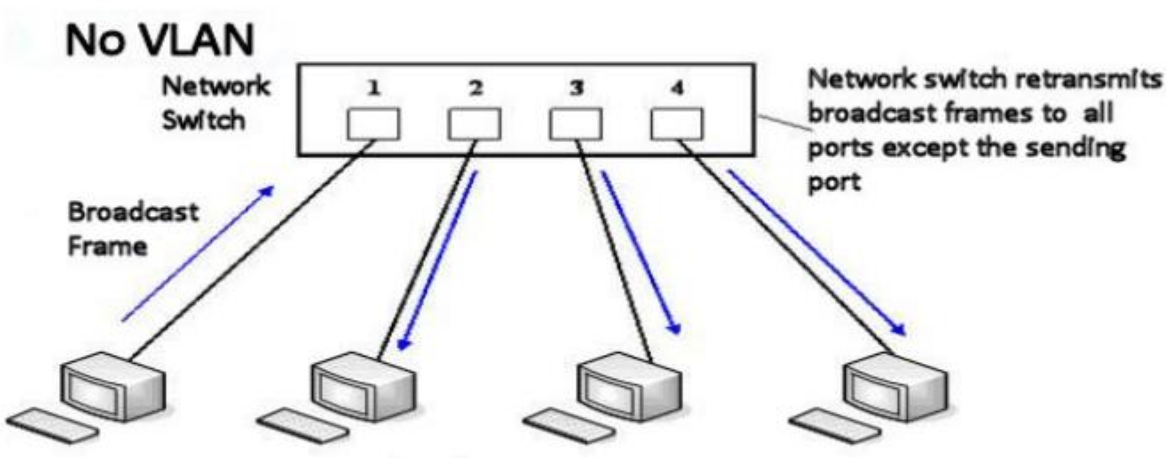
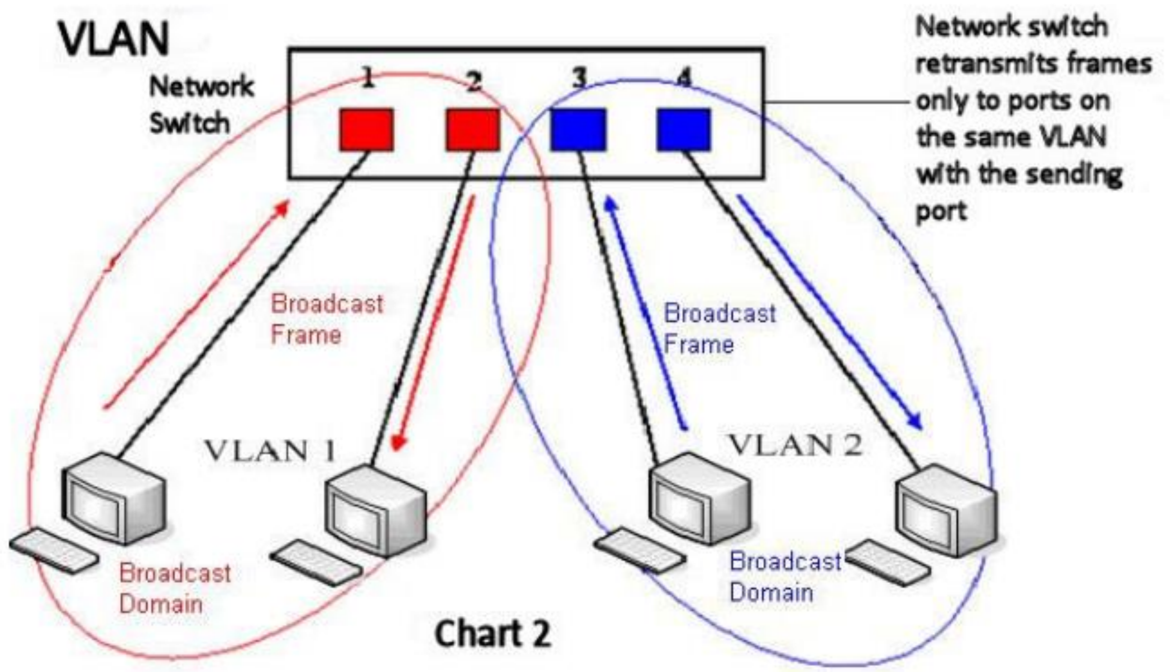
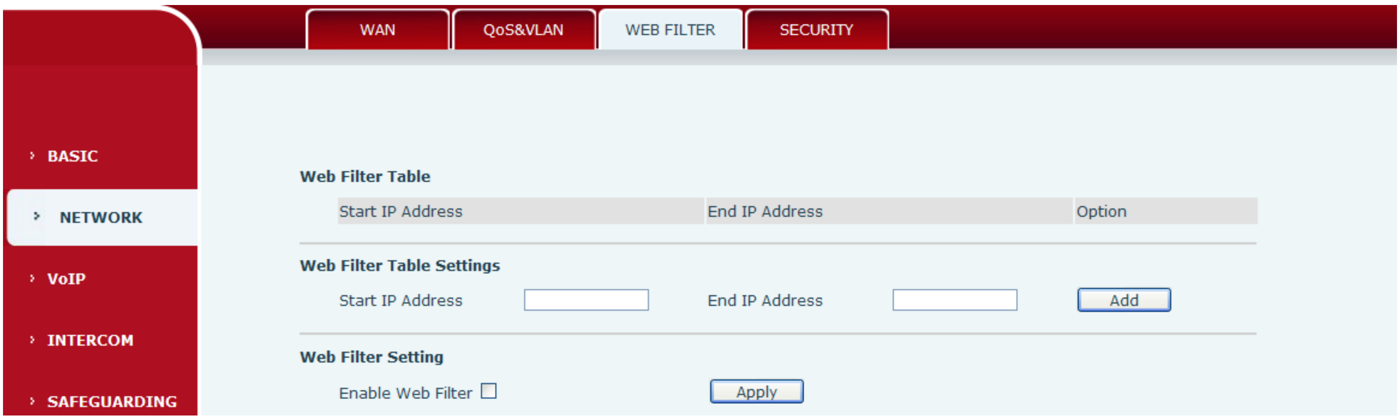



Chart 1

- 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.



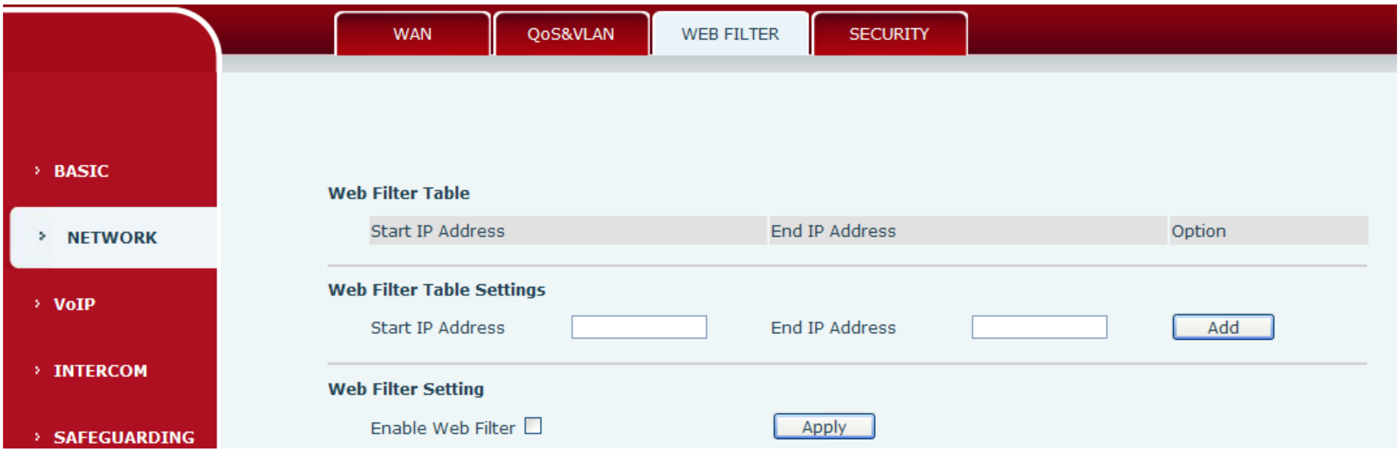

Note: In practice, VLANs are distinguished by the use of VLAN IDs.




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

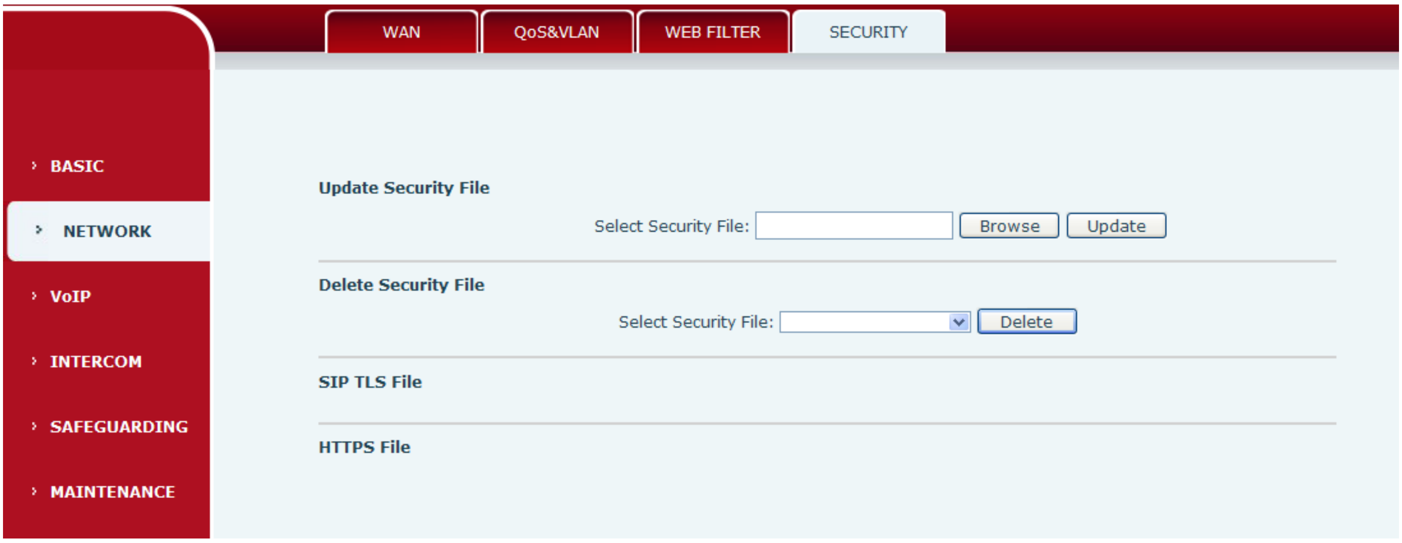
| Field Name | Explanation |
|---|---|
| **Quality of Service (QoS) Settings** | |
| Enable DSCP | Enable or Disable Differentiated Services Code Point (DSCP) |
| SIP DSCP | Specify the value of the SIP DSCP in decimal |
| Audio RTP DSCP | Specify the value of the Audio DSCP in decimal |
| Video RTP DSCP | Specify the value of the Video 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 |
| 802.1P Priority | Specify the value of the 802.1p priority. Range is 0-7 |

## c) 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. |
| **Web Filter Table** |
| Web page 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. |

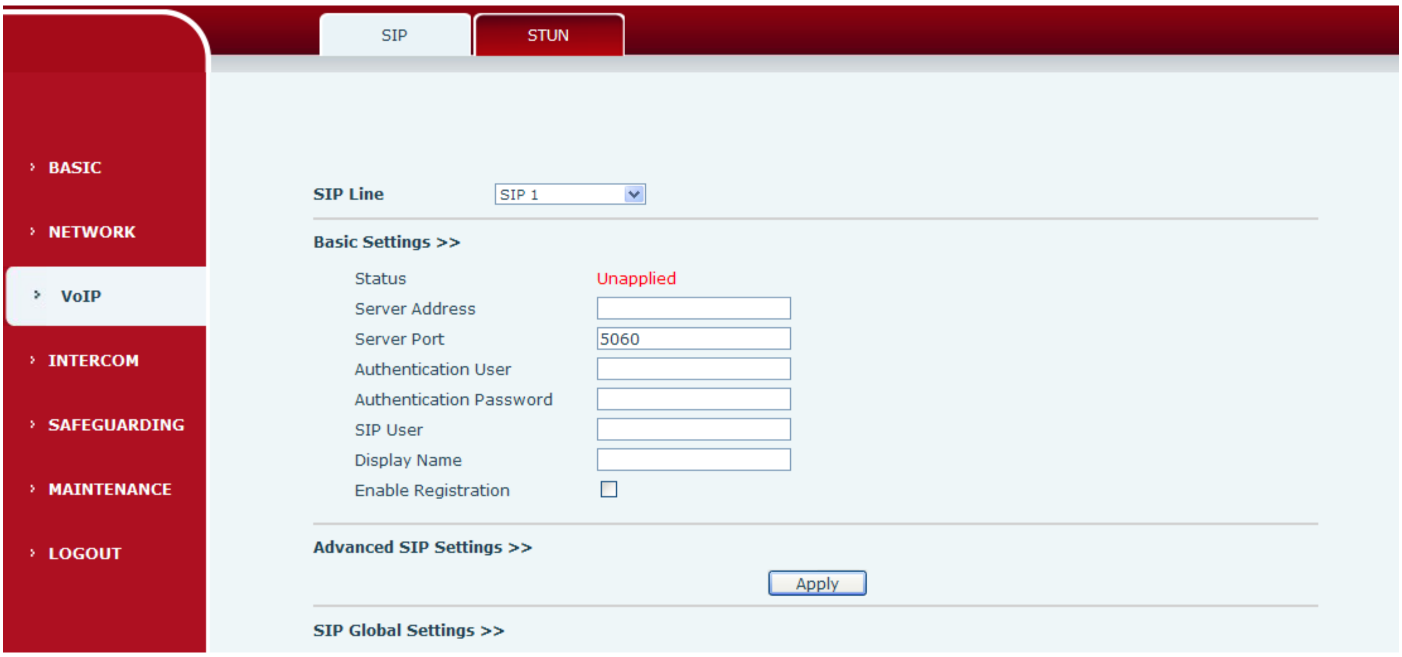
## d) SECURITY

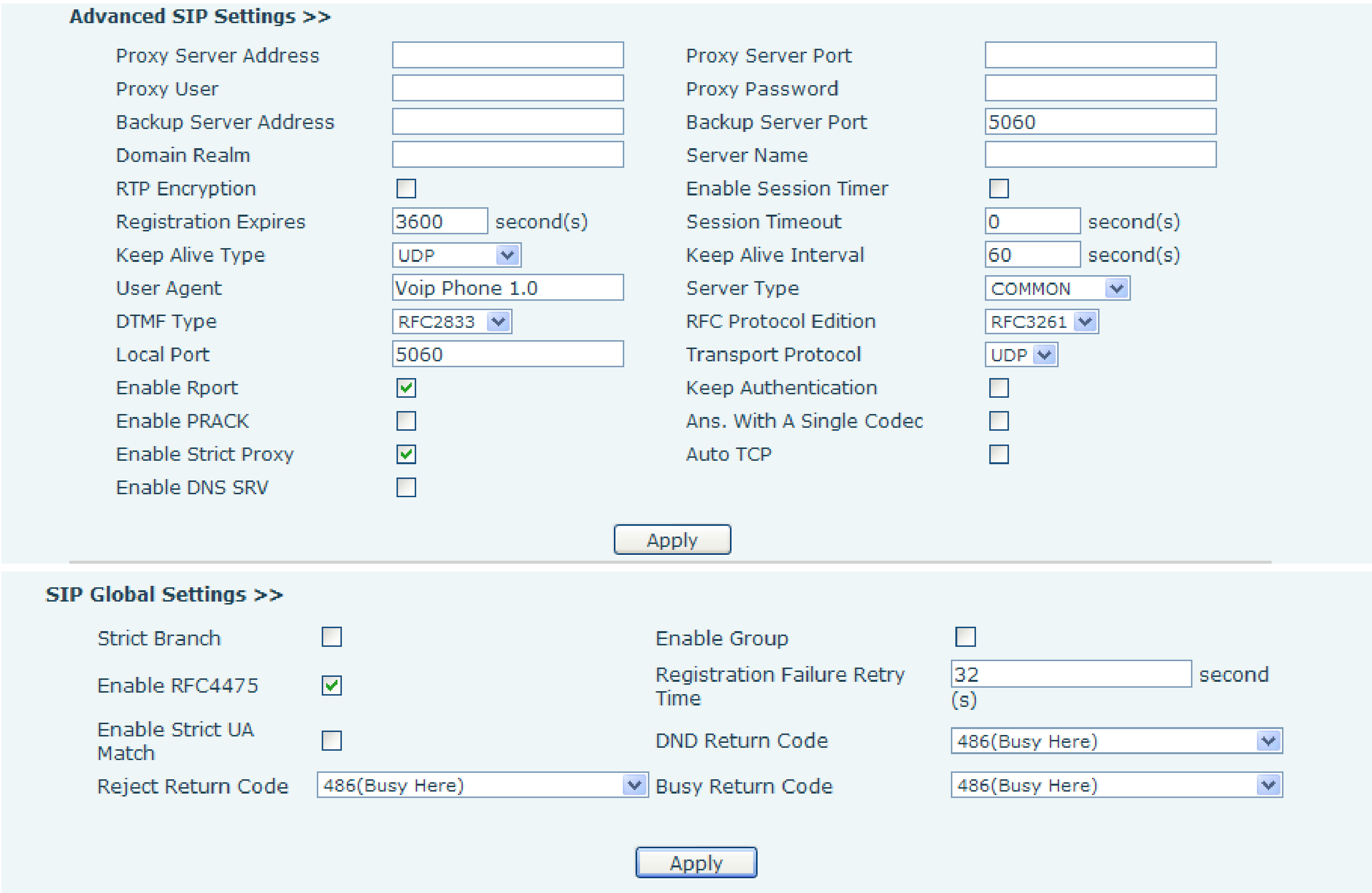



| Field Name | Explanation |
| --- | --- |
| Update Security File | Select the security file to be updated. Click the Update button to update. |
| Delete Security File | Select the security file to be deleted. Click the Delete button to Delete. |
| SIP TLS Files | Show SIP TLS authentication certificate. |
| HTTPS Files | Show HTTPS authentication certificate. |

# (3) VOIP

## a) SIP





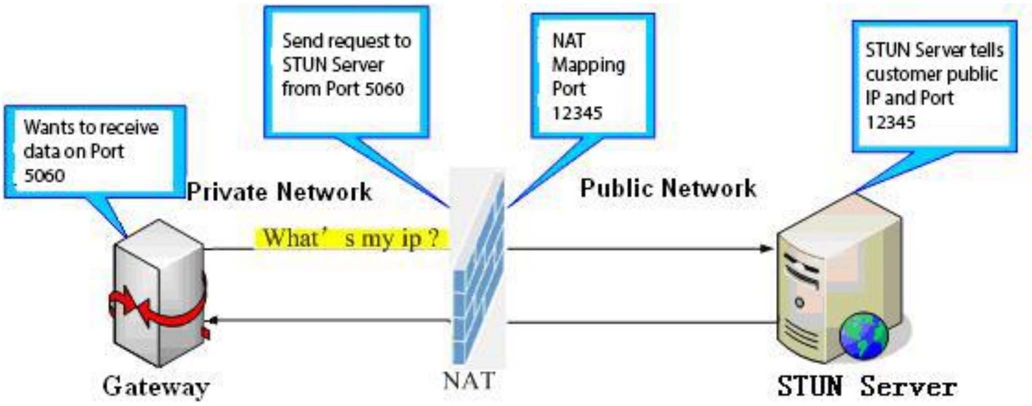

| SIP | |
|---|---|
| **Field Name** | **Explanation** |
| **Basic Settings** (Choose the SIP line to configured) | |
| Status | Shows registration status. If the registration is successful done, it will display “has been registered”, otherwise will display “not registered”. The wrong password will display “403 errors” and account number failure will display “timeout”. |
| Server Address | SIP server IP address or URI. |
| Server Port | SIP server port. Default is 5060. |
| Authentication User | SIP account name (Login ID). |
| Authentication Password | SIP registration password. |
| SIP User | Phone number assigned by VoIP service provider. Equipment will not register if there is no phone number configured. |
| Display Name | Set the display name. This name is shown on Caller ID. |
| Enable Registration | Check to submit registration information. |

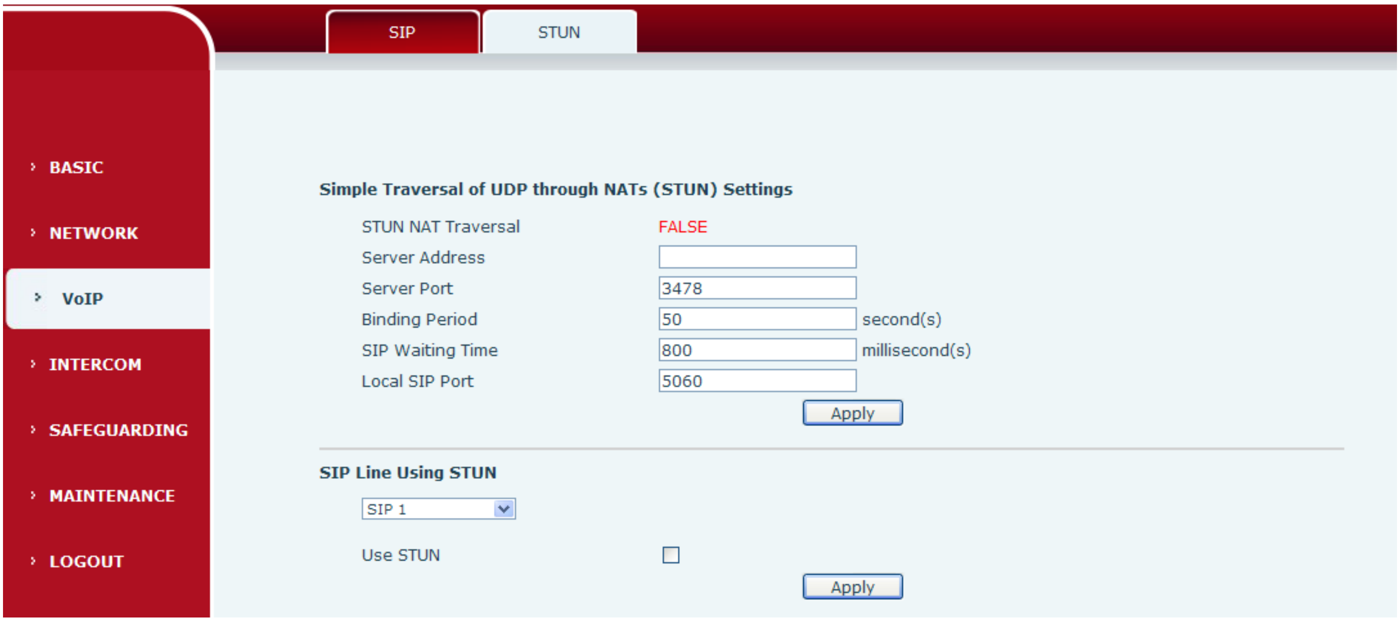
| Field Name | Explanation |
| --- | --- |
| **Advanced SIP Settings** | |
| Proxy Server Address | SIP proxy server IP address or URI, (This is normally the same as the SIP Registrar Server) |
| Proxy Server Port | SIP Proxy server port. Normally 5060. |
| Proxy User | SIP Proxy server account. |
|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

| Field Name | Explanation |
|---|---|
| Transport Protocol | Configuration using the transport protocol, TCP, TLS or UDP, the default is UDP. |
| Enable Rport | Enable/Disable support for NAT traversal via RFC3581 (Rport). |
| 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 |
| Enable PRACK | Enable or disable SIP PRACK function. Default is OFF. It is suggested this be used. |
| Ans. With a Single Codec | If enabled phone will respond to incoming calls with only one codec. |
| Enable Strict Proxy | Enables the use of strict routing. When the phone receives packets from the server it will use the source IP address, not the address in via field. |
| Auto TCP | Force the use of TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes |
| Enable DNS SRV | Enables use of DNS SRV records |
| **SIP Global Settings** | |
| Strict Branch | Enable Strict Branch - The value of the branch must be after"z9hG4bK" in the VIA field of the INVITE message received, or the phone will not respond to the INVITE. Note: This will affect all lines |
| Enable Group | Enable SIP Group Backup. This will affect all lines |
| Enable RFC4475 | Enable or disable RFC4475,default is enable。 |
| 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 |
| Enable Strict UA Match | Enable or disable Strict UA Match |
| DND Return Code | Specify SIP Code returned for DND. Default is 480 - Temporarily Not Available. |
| Reject Return Code | Specify SIP Code returned for Rejected call. Default is 603 – Decline. |
| Busy Return Code | Specify SIP Code returned for Busy. Default is 486 – Busy Here. |

## b) STUN

STUN – Simple Traversal of UDP through NAT –A STUN server allows a phone in a private network to know its public IP and port as well as the type of NAT being used. The equipment can then use this information to register itself to a SIP server so that it can make and receive calls while in a private network.



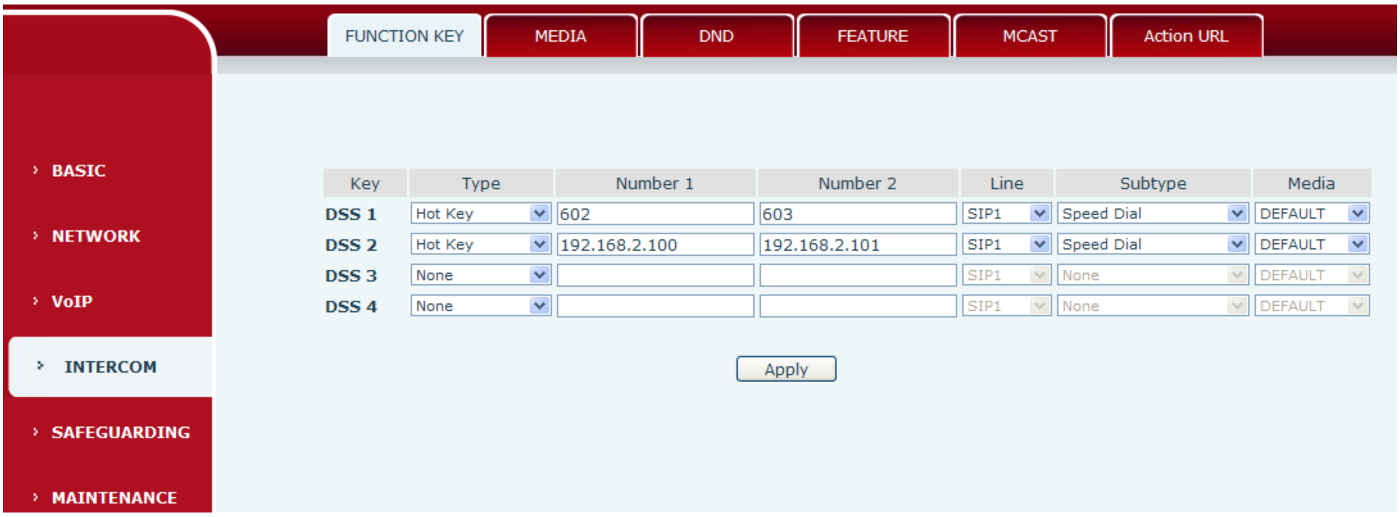

| STUN | |
|---|---|
| **Field Name** | **Explanation** |
| STUN NAT Traversal | Shows whether or not STUN NAT Traversal was successful. |
| Server Address | STUN Server IP address |
| Server Port | STUN Server Port – Default is 3478. |
| Binding Period | STUN blinding period – STUN packets are sent 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 |

| Field Name | Explanation |
|---|---|
| **SIP Line Using STUN** (SIP1 or SIP2) | |
| Use STUN | Use STUN |
| Note: the SIP STUN is used to achieve the SIP penetration of NAT, is the realization of a service, when the equipment configuration of the STUN server IP and port (usually the default is 3478), and select the Use Stun SIP server, the use of NAT equipment to achieve penetration. | |

## (4) INTERCOM

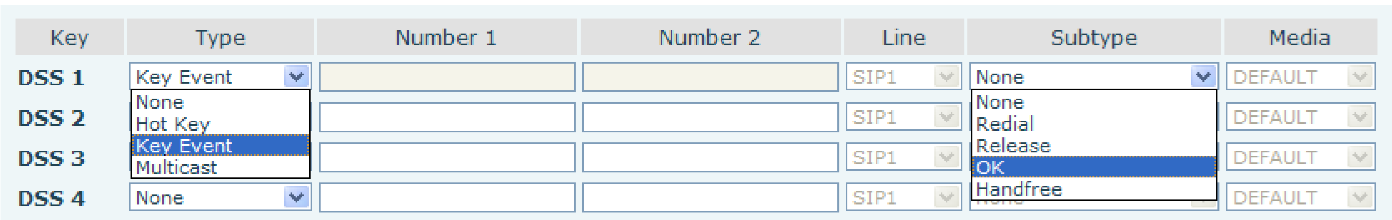
### a) FUNCTICON KEY

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




➢ **Key Event Settings**

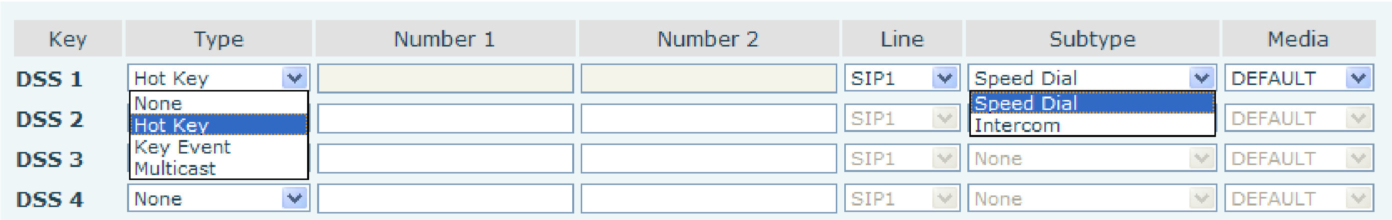
Set the key type to the Key Event.

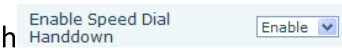



| DSS key type | Subtype | Usage |
|---|---|---|
| Key Event | None | Not responding |
| | Dial | Dial function |
| | Release | End calls |
| | OK | Identify key |
| | Handfree | The hand-free key(with hook dial, hang up) |

## 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.

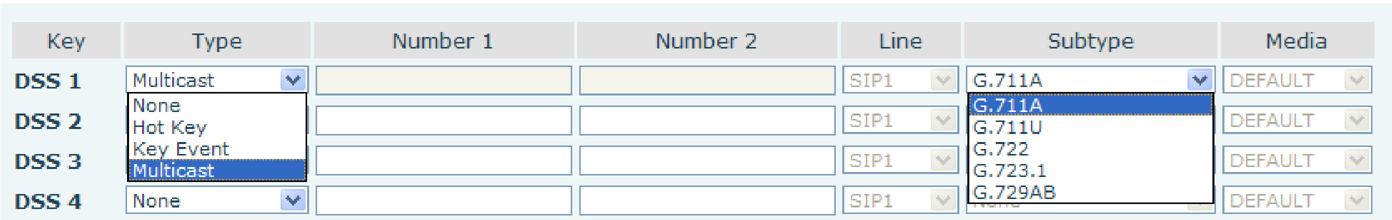



| DSS key type | Number | Line | Subtype | Usage |
|---|---|---|---|---|
| Hot Key | Fill the called party's SIP account or address | The SIP account corresponding lines | Speed Dial | In Speed dial mode, with   can define whether this call is allowed to be hang up by re-press the speed dial |
| | |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

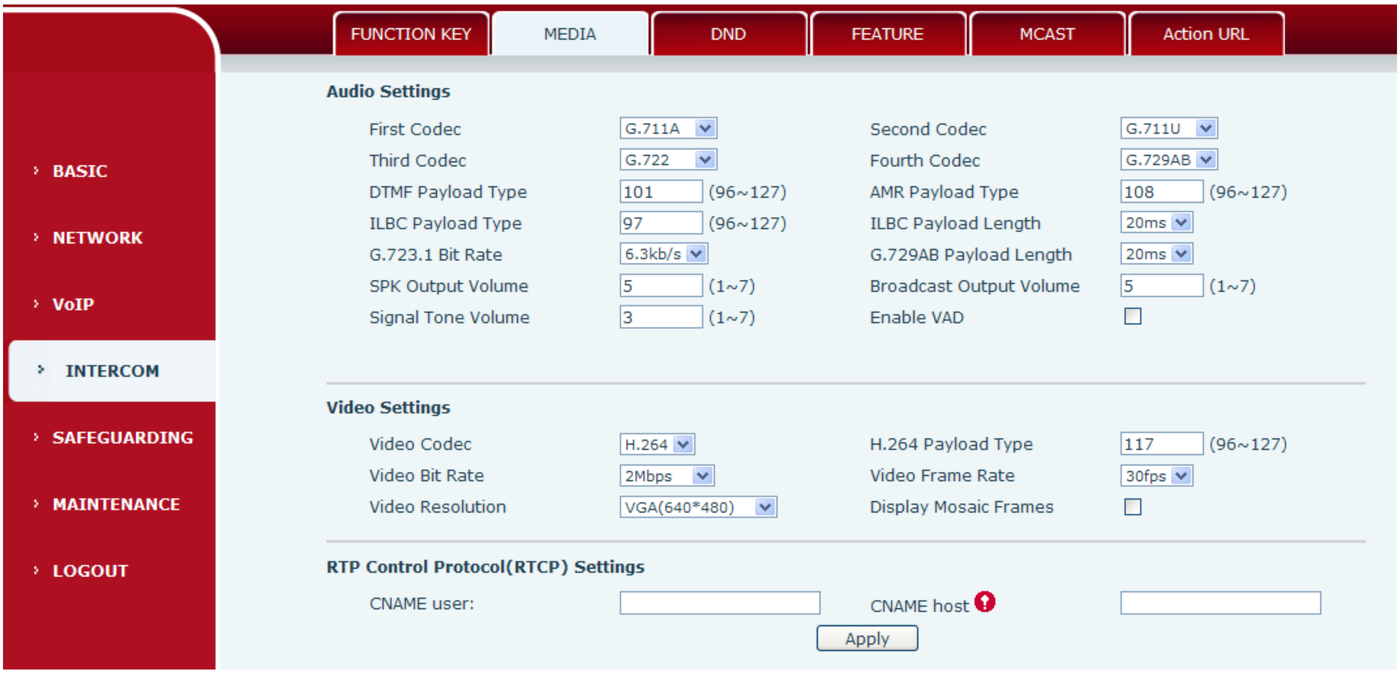
✧ operation mechanism

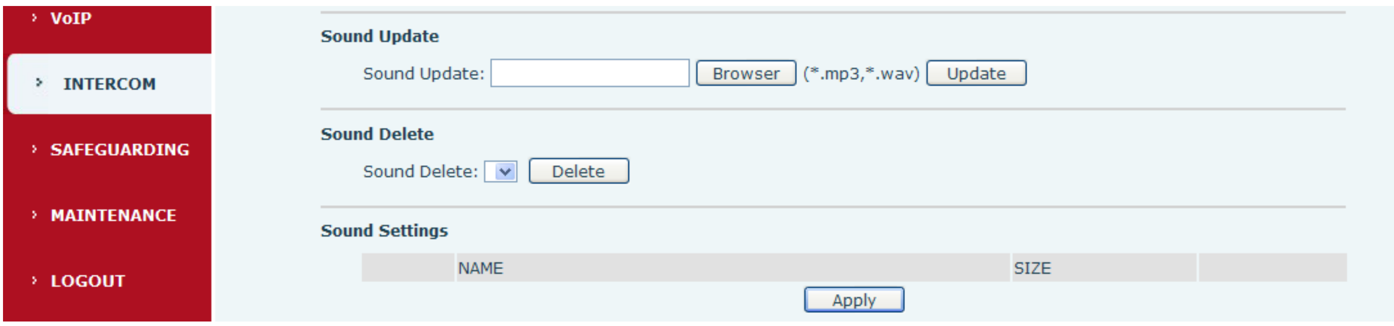
Device through the DSS Key configuration of multicast address and port and started coding; set by WEB to monitor the multicast address and port; device sends a multicast, listens to the address of the device can receive the multicast content.

✧ 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.

## b) MEDIA



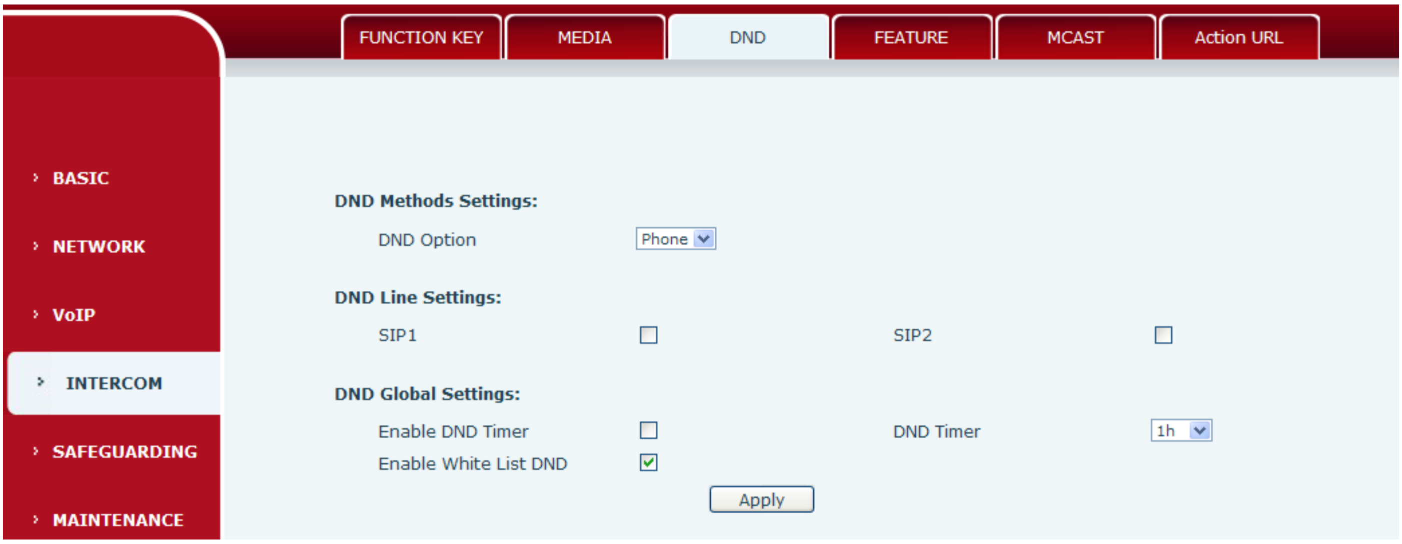




| Media Settings | |
|---|---|
| **Field Name** | **Explanation** |
| First Codec | The first codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, ILBC, AMR. |
| Second Codec | The second codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, ILBC, AMR. |
| Third Codec | The third codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, ILBC and AMR. |

| Field Name | Explanation |
| --- | --- |
| Fourth Codec | The forth codec choice: G.711A/U, G.722, G.723.1, G.726-32, G.729AB, ILBC and AMR. |
| DTMF Payload Type | The RTP Payload type that indicates DTMF. Default is 101 |
| AMR Payload Type | Set the AMR Payload type, Numerical based on between 96-127. |
| ILBC Payload Type | Set the ILBC Payload type, Numerical based on between 96-127. |
| ILBC Payload Length | Set the ILBC payload length. |
| G.723.1 Bit Rate | Choices are 5.3kb/s or 6.3kb/s. |
| G.729AB Payload Length | G.729AB Payload Length – Adjusts from 10 – 60 mSec. |
| SPK Output Volume | Set the speaker calls the volume level. |
| Broadcast Output Volume | Set the broadcast the output volume level. |
| Signal Tone Volume | Set the audio signal the output volume level. |
| Enable VAD | Enable or disable Voice Activity Detection (VAD). If VAD is enabled, G729 Payload length cannot be set greater than 20 mSec. |
| **Video Settings** | |
| Video Codec | Set the video codec used in video call (H.263, H.264) |
| H.264 Payload Type | Set the H.264 Payload type, Numerical based on between 96-127. |
| Video Bit Rate | Set the bandwidth of video call |
| Video Frame Rate | Set the video frame rate |
| Video Resolution | Set the video resolution; QCIF(176*144), CIF(352*288), VGA(640*480), 4CIF(704*576), 720P(1280x720).<br>Note: 720P only on the four nuclear phone support, And need to choose above 2M of the bandwidth |
| Display Mosaic Frames | Enable or Disable display mosaic |

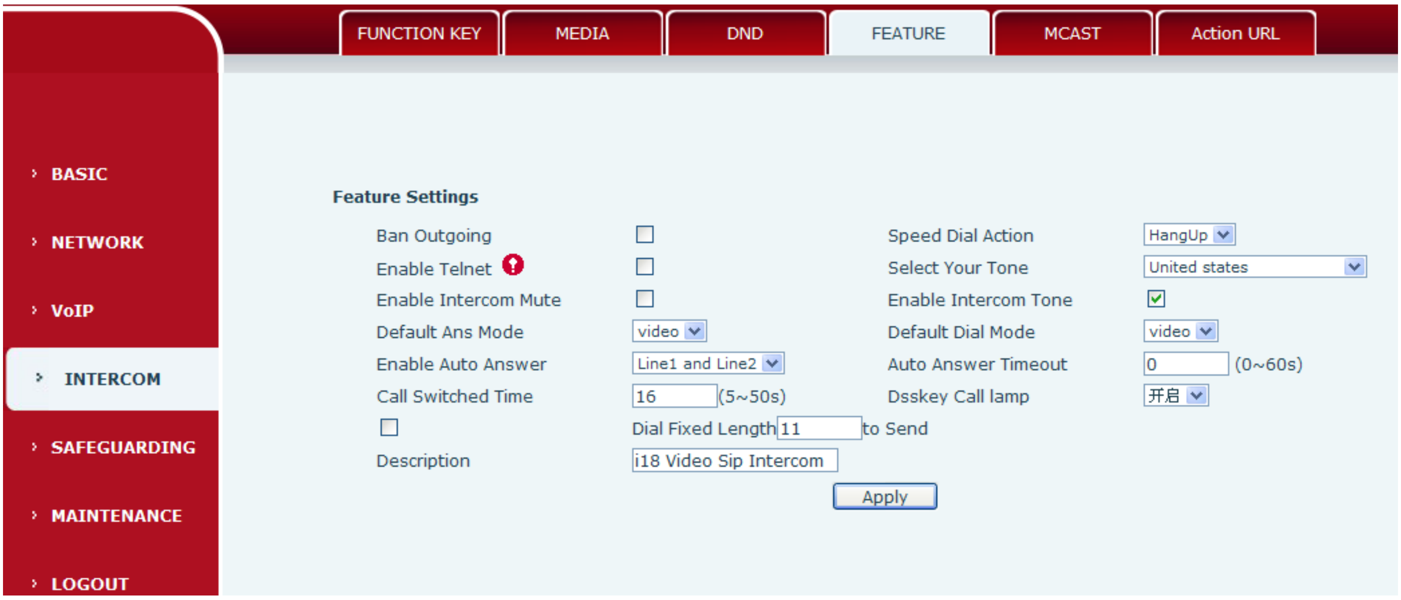
| Field Name | Explanation |
|---|---|
| **RTP Control Protocol(RTCP) Settings** | |
| CNAME user | Set CNAME user |
| CNAME host | Set CNAME host |
| **Sound Update** | |
| Choose the ring tone files and then click update to apply | |
| **Sound Delete** | |
| Delete the ring tone file | |
| **Sound Settings** | |
| Set the ring tong files, format is .mp3 and .wav | |

## c) DND



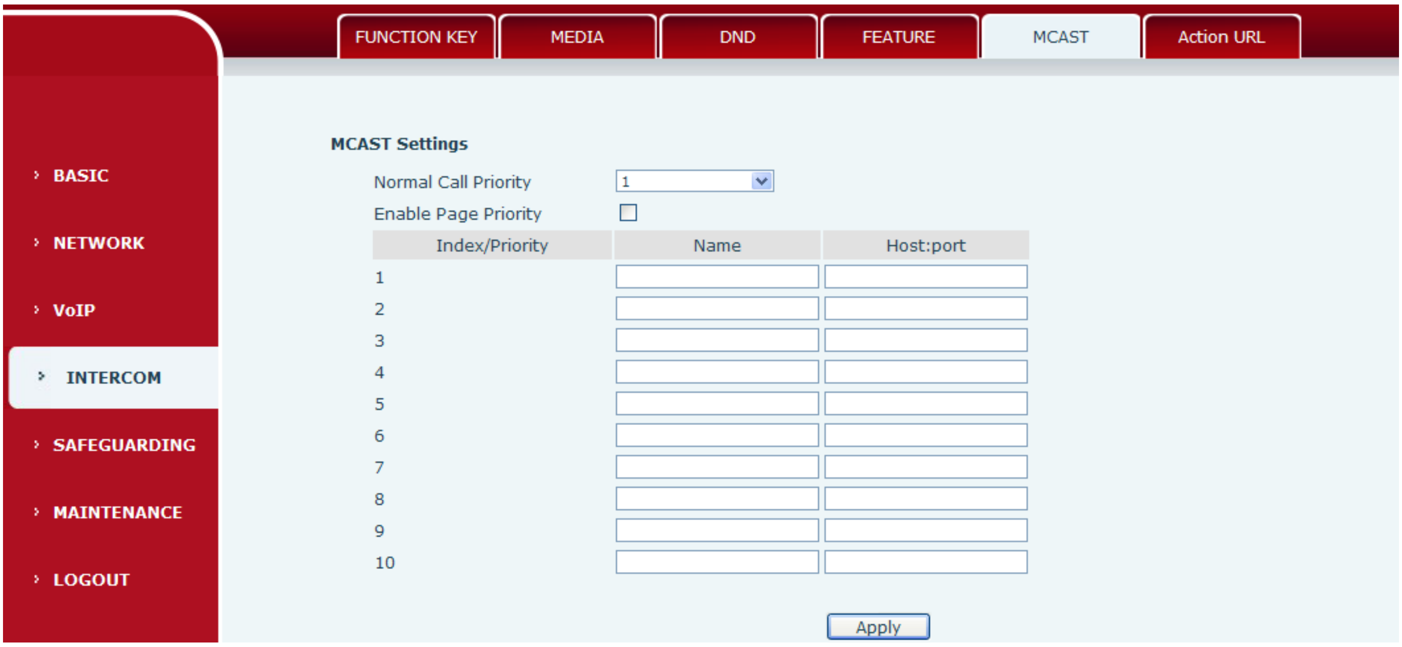

| Field Name | Explanation |
|---|---|
| **DND Methods Settings** | |
| DND Option | Set the DND option, default is phone. |
| **DND Line Settings** | |
| SIP1 | Enable or Disable sip1 DND |
| SIP2 | Enable or Disable sip2 DND |
| **DND Global Settings** | |
| Enable DND Timer | Enable or disable DND timer |
| DND Timer | Set the DND time |
| Enable White List DND | Enable or disable white list DND |

## d) FEATURE




| Feature | |
|---|---|
| **Field Name** | **Explanation** |
| **Feature Settings** | |
| Ban Outgoing | If enabled, no outgoing calls can be made. |
| Speed Dial Action | Default is Speed Dial Hand-down function |
| Enable Telnet | Enable or disable Telnet |
| Select Your Tone | Standard configuration signal sound. |
| 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. |
| Default Ans Mode | Set answer mode, default is video . |
| Default Dial Mode | Set dial mode, default is video. |
| Enable Auto Answer | Enable Auto Answer function |
| Auto Answer Timeout | Set Auto Answer Timeout |
| Call Switched Time | Set the call switched time. |
| Dsskey Call lamp | Configuration is enabled when the speed dial key to call light condition. |
| Dial Fixed Length to Send | The number will be sent to the server after the specified numbers of digits are dialed. |
| Description | device IP description |

## e) MCAST




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

**MCAST Settings**

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

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

- Priority:

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

- The options are as follows:
  - ✧ 1-10: To definite the priority of the common calls, 1 is the top level while 10 is the lowest
  - ✧ Disable: ignore all incoming multicast RTP stream
  - ✧ Enable the page priority:

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

- Web Settings:

**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, which is the highest priority.

Note: when pressing the multicast key for multicast session, both 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 | 224.0.0.2:1366 |
| 3 | group 3 | 224.0.0.6:3366 |
| 4 | | |
| 5 | | |
| 6 | | |
| 7 | | |
| 8 | | |
| 9 | | |
| 10 | | |

- **Blue part (name)**

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

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

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

- **Pink part (index / priority)**

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

- **Red part (priority)**

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

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

- **Green part (Enable Page priority)**

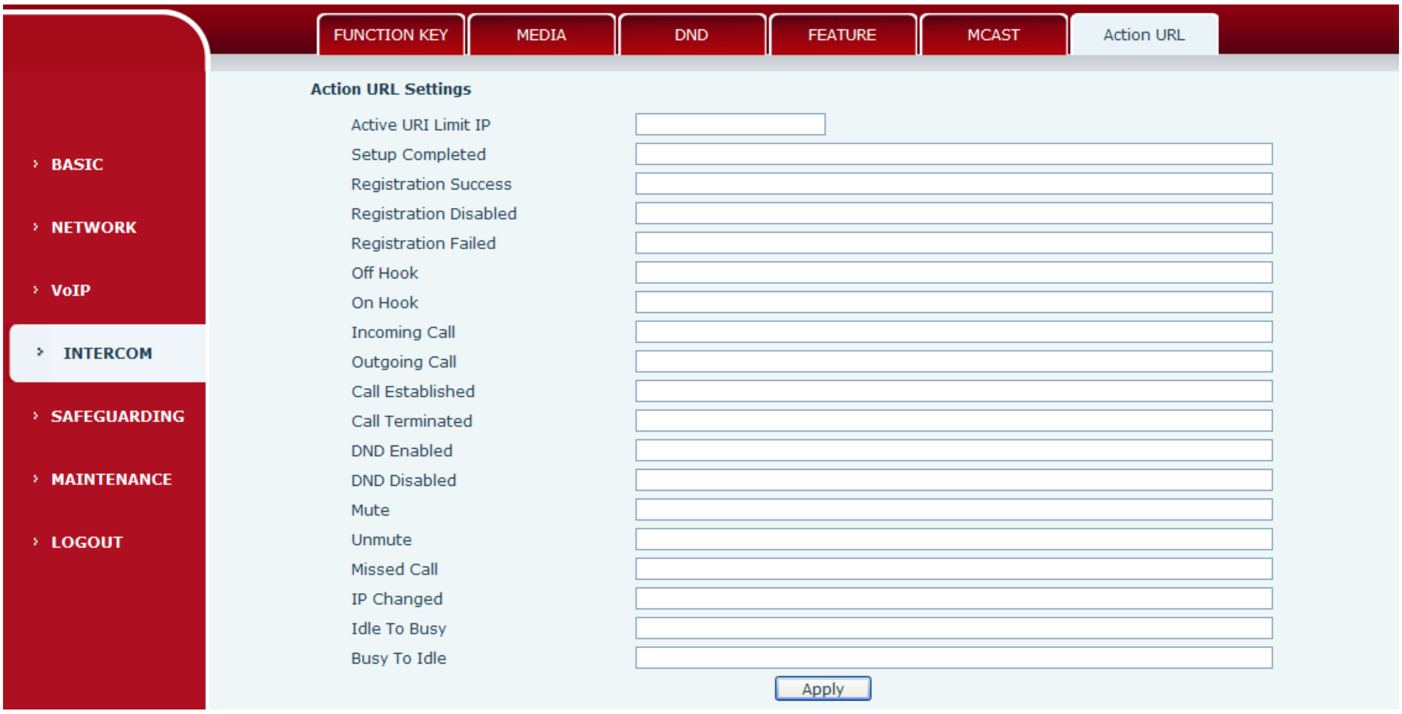
Set whether to open more priority is the priority of multicast, multicast is pink part number. Explain how to use:

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

**Multicast service**

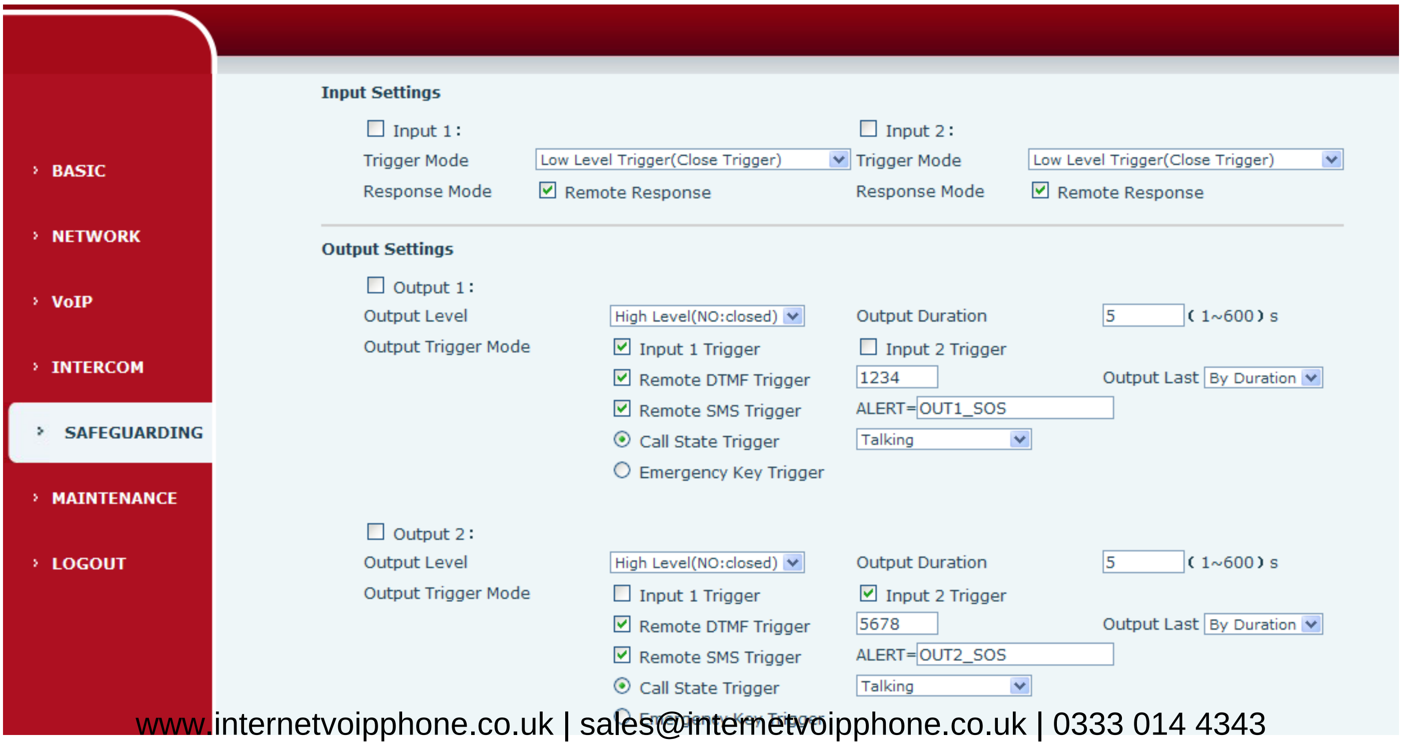
- **Send:** when configured ok, our key press shell on the corresponding equipment, equipment directly into the Talking interface, the premise is to ensure no current multicast call and 3-way of the case, the multicast can be established.
- **Lmonitor:** IP port and priority configuration monitoring device, when the call is initiated and incoming multicast, directly into the Talking interface equipment.

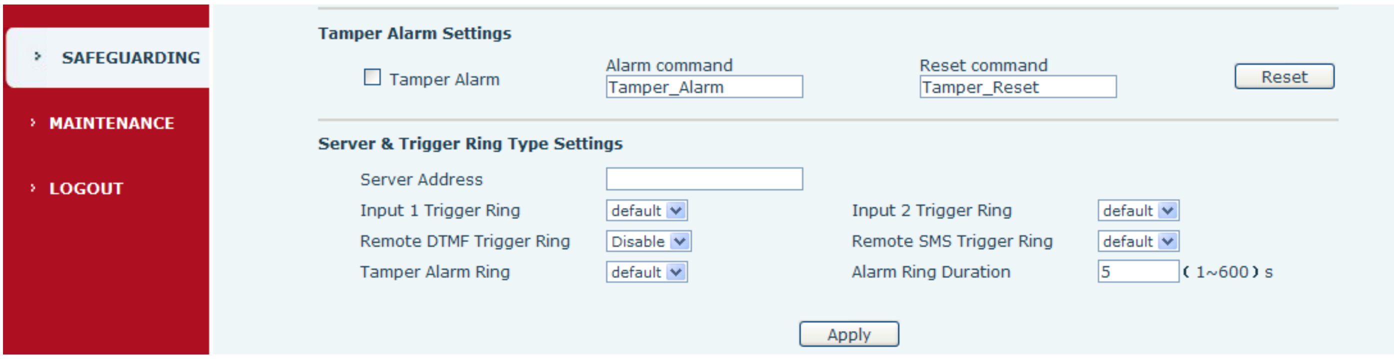
f) Action URL




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

## (5) SAFEGUARDING



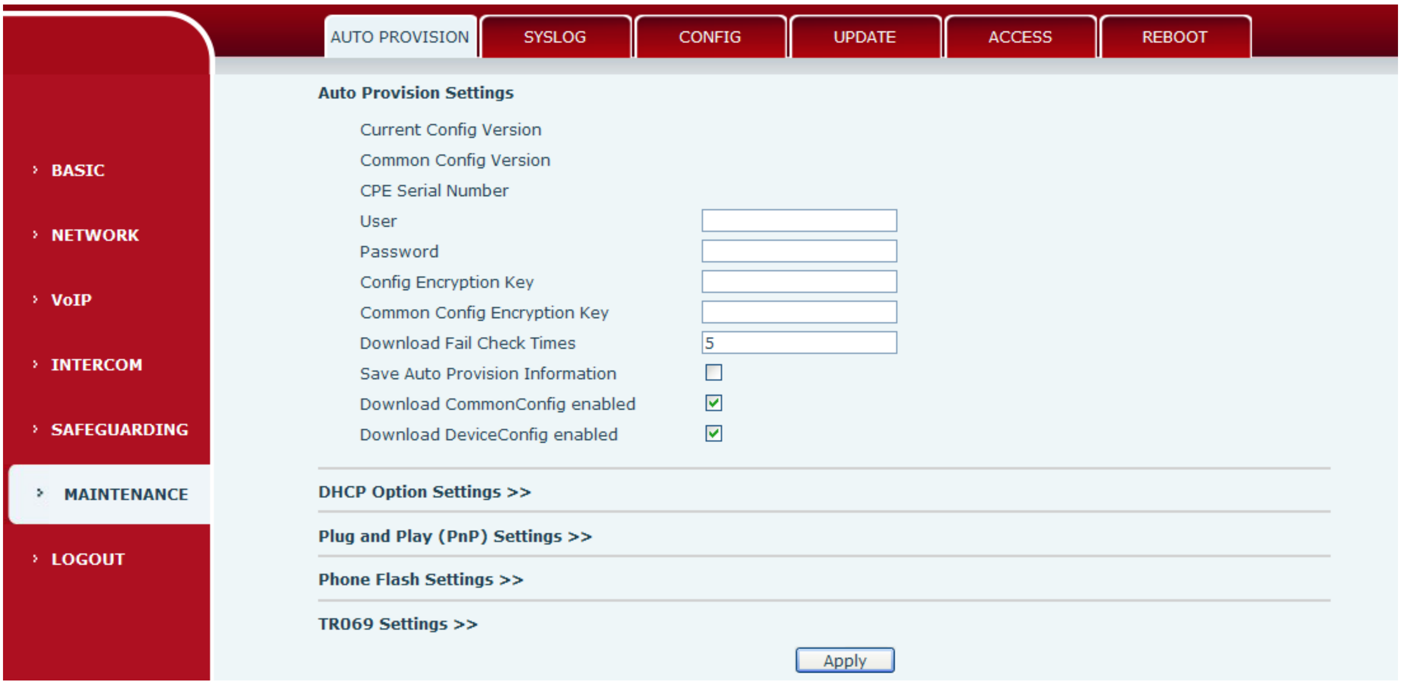



| **Safeguarding** | |
|---|---|
| **Field Name** | **Explanation** |
| **Input settings** | |
| Input 1 | Open /Close Input port1 |
| Trigger Mode | When choosing the low level trigger (closed trigger), detect the input port 1 (low level) closed trigger. |
| | When choosing the high level trigger (disconnected 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. |
| 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) |

| Field Name | Explanation | |
|---|---|---|
| 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 | |
| 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) MAINTENANCE

### 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.

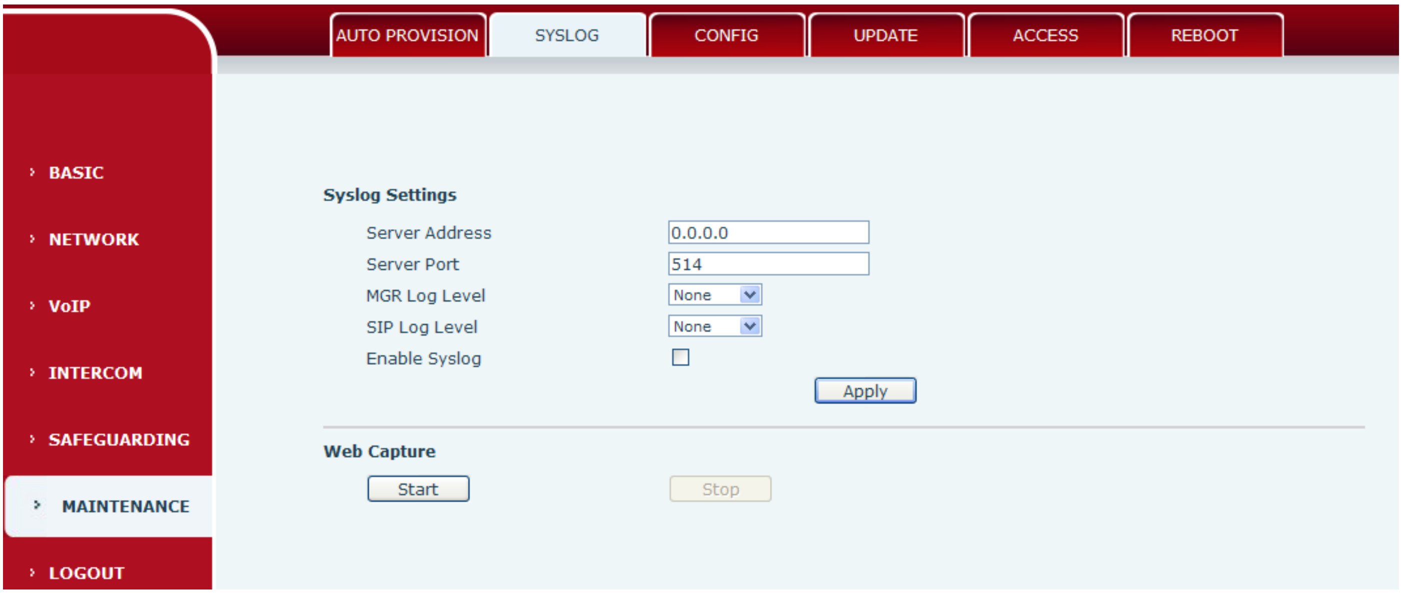
DHCP option → PnP server → Phone Flash

| Field Name | Explanation |
|---|---|
| **Auto Provision Settings** | |
| 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. |

| Field Name | Explanation |
| --- | --- |
| Config Encryption Key | Encryption key for the configuration file |
| Common Config Encryption Key | Encryption key for common configuration file |
| Download Fail Check Times | Download failed and check times |
| Save Auto Provision Information | Save the auto provision username and password in the phone until the server url changes |
| Download CommonConfig enabled | Enable or disable download commonconfig |
| Download DeviceConfig enabled | Enable or disable download deviceconfig |
| **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. |
| **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. |

| Field Name | Explanation |
|---|---|
| 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 |
| **TR069 Settings** | |
| Enable TR069 | Enable/Disable TR069 configuration |
| Enable TR069 Warning Tone | Enable or disable TR069 Warning Tone |
| ACS Server Type | Select Common or CTC ACS Server Type. |
| ACS Server URL | ACS Server URL. |
| ACS User | User name for ACS. |
| ACS Password | ACS Password. |
| TR069 Auto Login | Enable/Disable TR069 Auto Login. |

## 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.

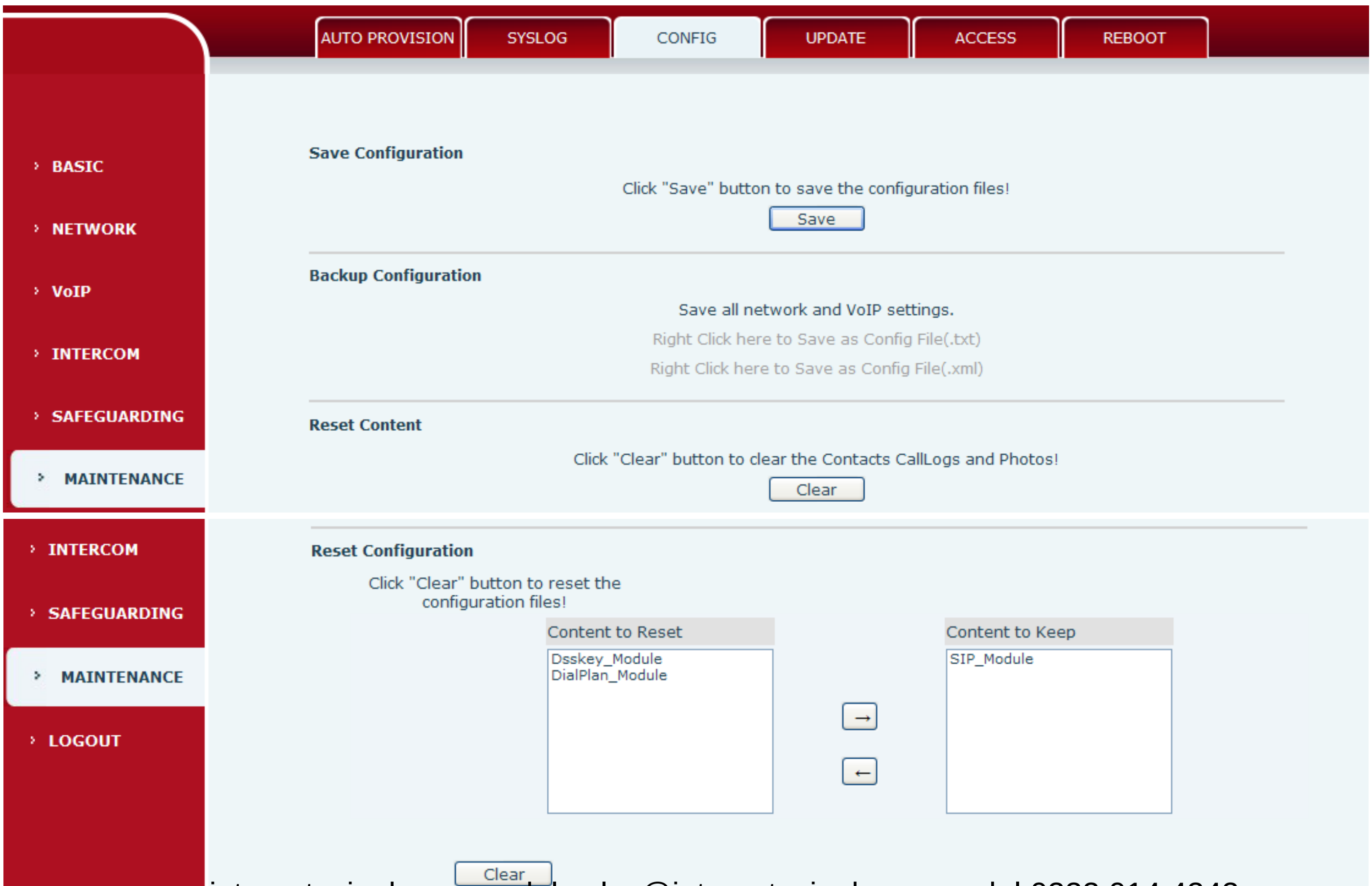
Level 5: notice; It is the normal but significant condition.

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

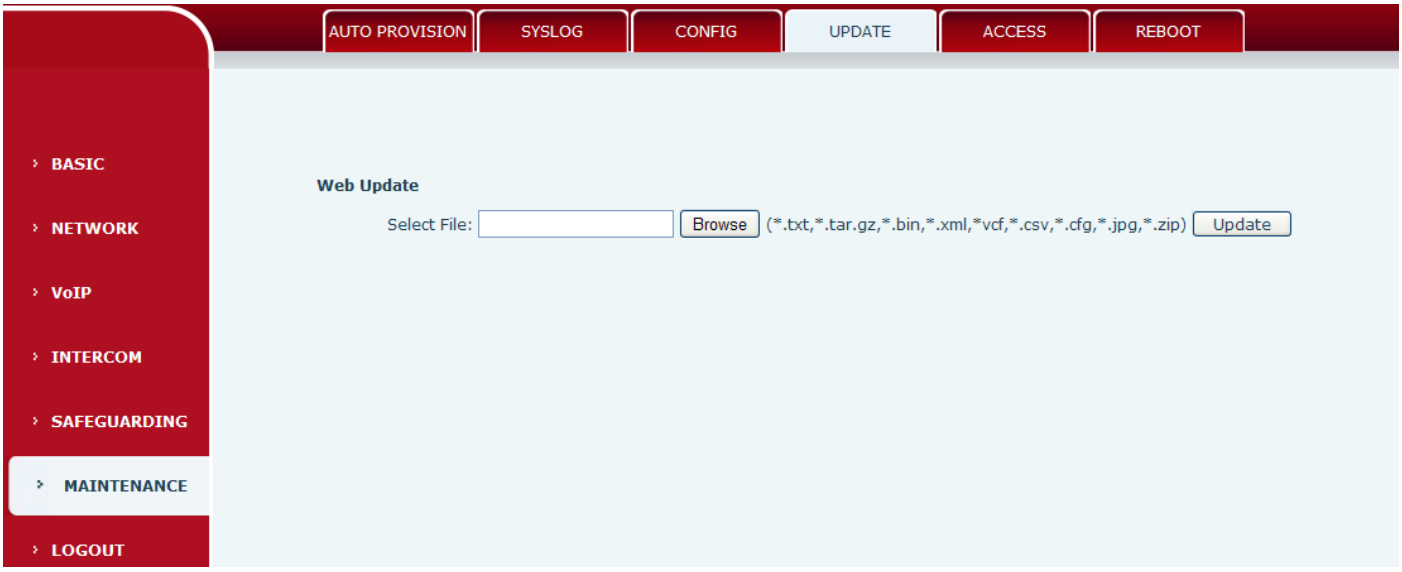
## c) CONFIG



| 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.” |
| Reset Content | Click the "clear" button can reset phone records and photos. |
| Reset Configuration | To reset the system and Automatic restart the equipment. |

## d) UPDATE

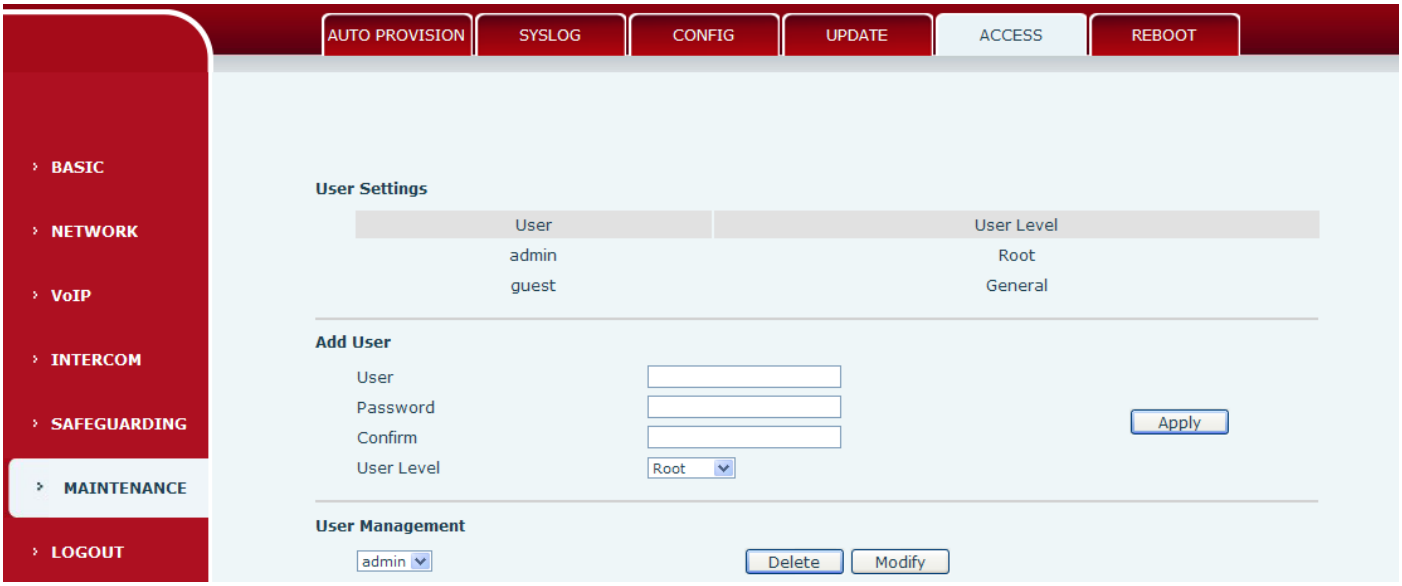
This page allows uploading configuration files to the equipment.




| 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. |

## e) ACCESS

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



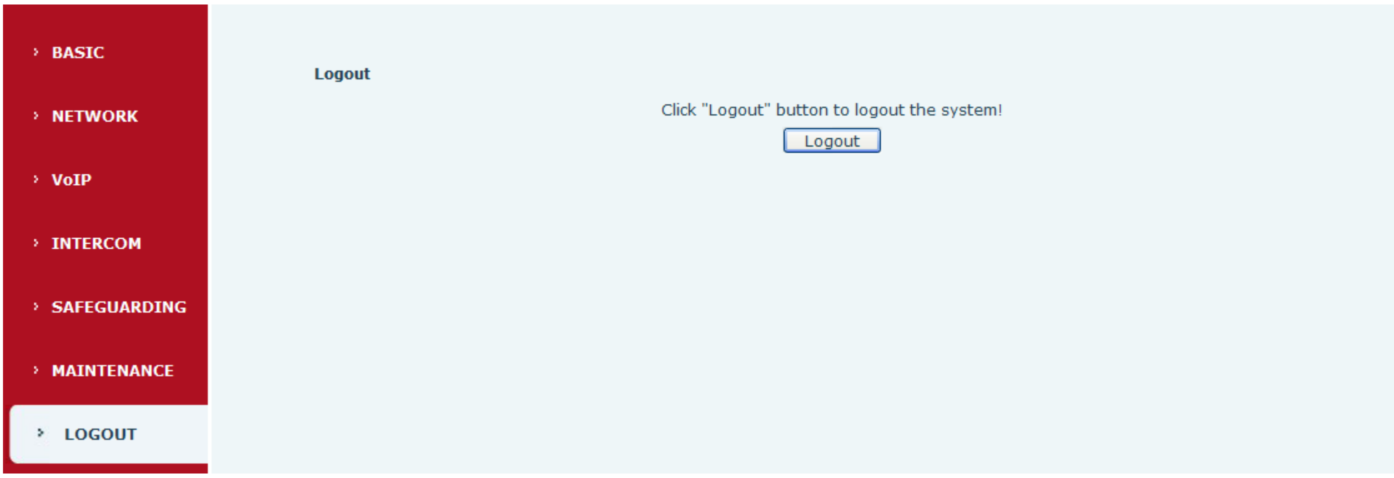

| Field Name | Explanation |
|---|---|
| **User Settings** | |
| User | shows the current user name |
| User level | Show the user level; admin user can modify the configuration. General user can only read the configuration. |
| **Add User** | |
| User | Set User Account name |
| Password | Set the password |
| 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 lead to reboot immediately.

Note: Be sure to save the configuration before rebooting.

## (7) LOGOUT




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

# E. Appendix

## 1. Technical parameters

| | | |
|---|---|---|
| **Communication protocol** | | SIP 2.0(RFC-3261) |
| **Main chipset** | | Freescale i.MX 6Quad |
| **Key** | **DSS key materials** | Stainless steel |
| | **DSS Key** | 1 or 2 |
| **Speech flow** | **Audio amplifier** | 3W |
| | **Volume control** | Adjustable |
| | **Full duplex speakerphone** | Support (AEC) |
| | **DTMF TYPE** | In-band, Out-of-band(RFC 2833), SIP INFO |
| | **wideband speech code** | G.722 |
| | **Narrowband speech code** | G711A/u, G.723.1, G.729AB, ILBC, AMR |
| **Video** | **Scope of broadband** | 64kbps~4Mbps |
| | **Video Framerate** | 10~30fps |
| | **resolution** | CIF, QCIF, VGA, 4CIF, 720P(HD) |
| | **Video Codec** | H.263, H.264 |
| **Port** | **Security linkage** | 2 embedded short circuit input interfaces |
| | | 2 embedded short circuit output interfaces |
| | **External speakers** | 1 embedded audio output interfaces |
| | **WAN** | 10/100BASE-TX s Auto-MDIX, RJ-45 |
| **Power supply mode** | | 12V / 1A DC or PoE |
| **Cables** | | CAT5 or better |
| **Shell Material** | | Cast aluminium panel, Cast aluminium back shell |
| **Working temperature** | | -40°C to 70°C |
| **Working humidity** | | 10% - 90% |
| **Storage temperature** | | -40°C to 70°C |
| **Installation way** | | Wall mounted or In-wall |
| **Dimension** | | Wall mounted: 223*130*74mm<br>In-wall: 270*150*61mm |

## 2. Basic functions

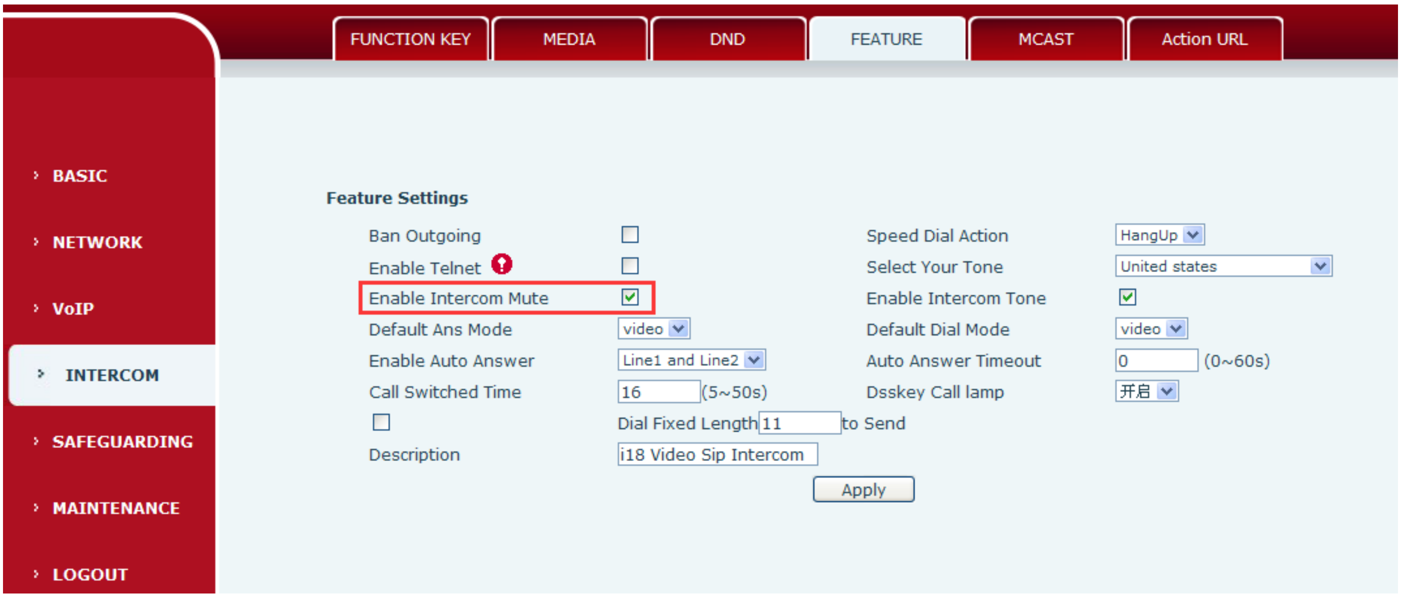
- 2 SIP Lines
- PoE Enabled
- Full-duplex speakerphone (HF)
- Intelligent DSS Keys (Speed Dial/intercom etc)
- Wall mounted / In-wall
- Special integrated noise reduction module
- Dual microphone Omnidirectional voice pickup
- 2 embedded short circuit input interfaces
- 2 embedded short circuit output interfaces. Support 4 controlled events: remote DTMF; remote server’s commands; interaction with short circuit input; talking status
- Output interface for active speaker
- Anti-tamper switch
- External power supply
- Record voice and video during calls (Optional)
- All in ONE: Radio and intercom, intelligent security function
- Industrial standard certifications: IP65， IK10， CE/FCC

## 3. Schematic diagram

## 4. The broadcast terminal configuration notice

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

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




✧ How to improve broadcasting tone quality?

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

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