User Manual

2N® IP Handset

Software Version: 1.4.1

Release Date：2020/11/01

Revision Date: 2022/03/11

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3 Safety Instructions

Please read the following safety notices before installing or using this unit. They are crucial for a safe and reliable operation of the device.

- Please use the external power supply that is included in the package. Another power supply may cause damage to the phone and affect the behavior or induce noise.
- Before using the external power supply in the package, please check the home power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If impaired, the power cord or plug must not be used because it may cause fire or electric shock.
- Do not drop, knock or shake the phone. Rough handling can break the internal circuit boards.
- This phone is designed for indoor use. Do not install the device in places exposed to direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposure of the phone to high temperatures or below 0℃ or high humidity.
- Avoid wetting the unit with any liquid.
- Do not attempt to open it. Any non-expert handling could damage the device. Consult your authorized dealer for help to avoid fire, electric shock and/or breakdown.
- Do not use harsh chemicals, cleaning solvents or strong detergents to clean the device. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- Do not touch the power plug during lightning storms to avoid an electric shock.
- Do not install this phone in ill-ventilated places to avoid bodily injuries. Before working on any equipment, be aware of the hazards associated with electrical circuitry and be familiar with the standard practices for preventing accidents.

4 Overview

4.1 Overview

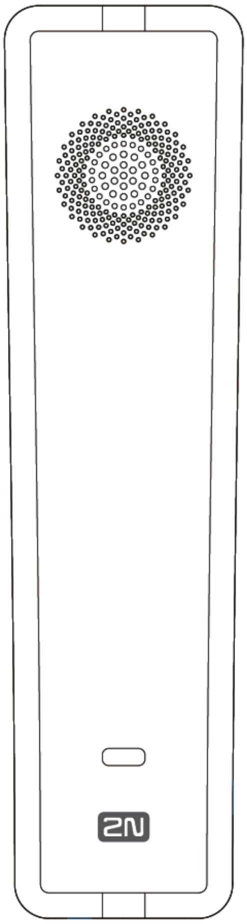
2N® IP Handset is a network telephone specifically designed for hotels. Its simple design brings an excellent user experience. The equipment is not only a telephone, but also a masterpiece placed in your living room or office.

H2U is the latest generation of the network telephone designed for hotels, which still keeps the excellent performance and specifications of the traditional equipment such as high-definition voice, high-performance echo cancelation, 100M Ethernet, QoS, encrypted transmission, automatic configuration, etc., innovation, smooth operation, flat interface setting and many other advantages.

For enterprise users, the device is a cost-effective, yet environmentally friendly piece of office equipment providing convenient operation. For home users, the device is a highly efficient communication device. Its users can flexibly configure and define the functions of one of the DSS keys, saving space and cost. It is an ideal choice for enterprise users and home users who pursue top quality and high efficiency.

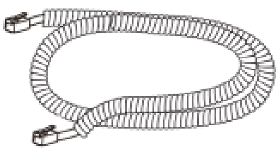
In order to help some interested users better understand the details of the product, this User Manual can be used as a reference guide for the use of X1S/X1SP. This document may not be applicable for the latest software version. Should you have any questions, you can use the help prompt interface of the X1S/X1SP phone or download and update your User Manual from the official website.

4.2 Packing Contents

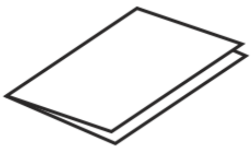


IP Phone

Handset



Handset Cord



Quick Guide

5 Desktop Installation

5.1 PoE and Use of External Power Adapters

The device supports two power supply modes: power supply from an external power adapter and via an Ethernet (PoE) complied switch.

The PoE power supply saves space and cost of additional power outlet provision. With a PoE switch, the device can be powered through a single Ethernet cable, which is also used for data transmission. By attaching a UPS system to the PoE switch, the device can keep working at power outages just like traditional PSTN telephones powered by the telephone line.

For the users who do not have PoE equipment, a traditional power adapter should be used. If the device is connected to a PoE switch and power adapter at the same time, the power adapter will be prioritized and the PoE power supply will be used as backup.

Please use the power adapter / PoE switch that meets the specifications to ensure that the device works properly.

5.2 Wall Mounting Method

The device supports the wall mounted installation.
Please follow the instructions in the figure below to install the phone：

1) Drill two holes in the wall with a vertical distance of 136.2 or 161.2 mm.
2) Insert two rubber plugs and screws in turn. Note that 5 mm is reserved between the nut and the wall, which is convenient for hanging the phone base.
3) Connect the cable, handset cable and power supply.
4) Align the wall hole on the base with the screws in step 2 and slide down to complete the installation.

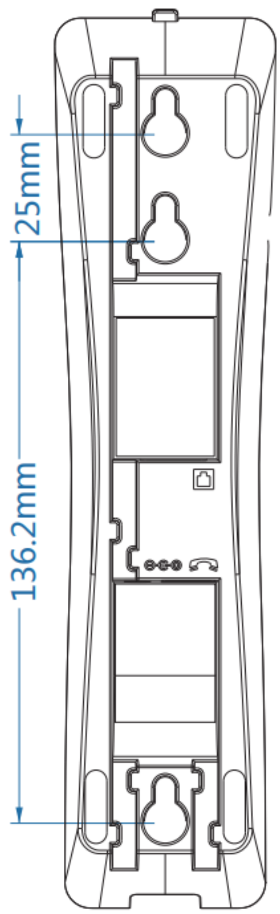


Figure 1 - Device Installation

Connect the power adapter, network, PC and phone to the appropriate ports as shown in the figure below.

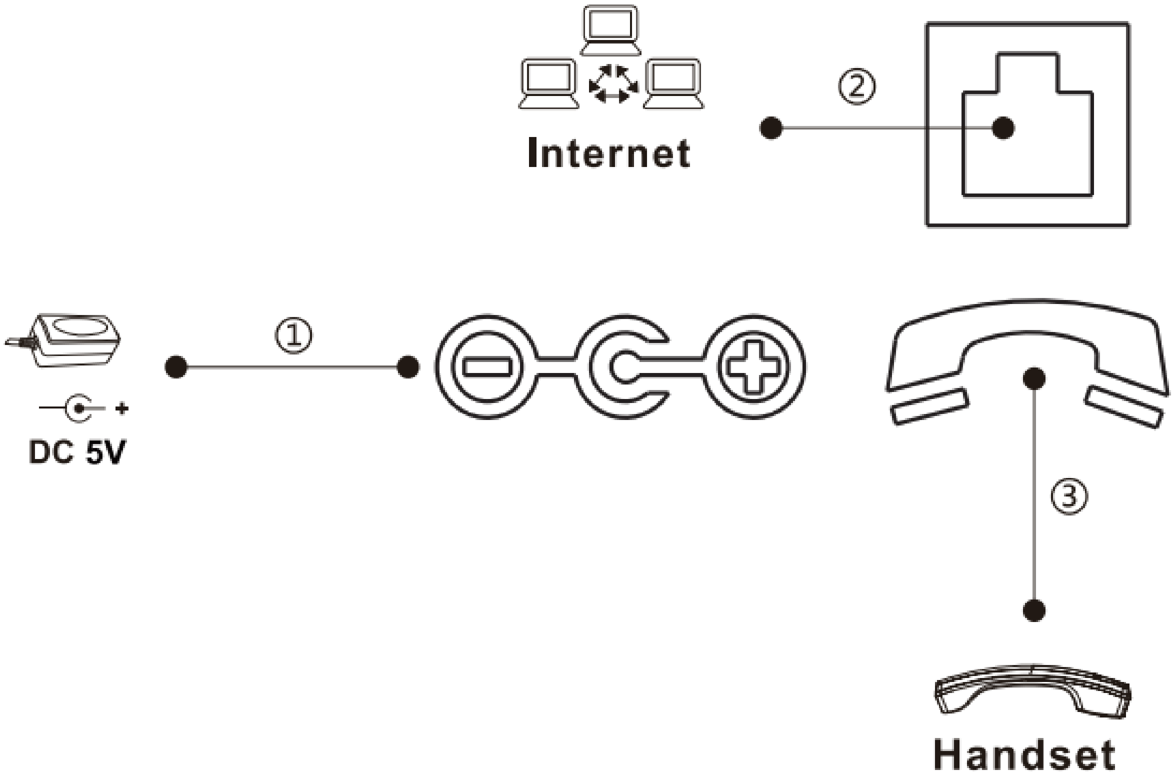


Figure 2 – Device Connection

6 Appendix Table

6.1 Appendix I – LED Definition

Table 1 - DSS KEY LED State

Type	LED Light	State
Default standby	Standby	Green on
	Mute	Green slow flash
	Line error (Registration failure)/Network disconnection	Red slow flash
Call	Calling/Pick up the handset	Red on
	Mute	Orange slow flash
	Hold/held	Orange slow flash
	Ringing	Red flash

7 Introduction to User

7.1 Keypad Instructions

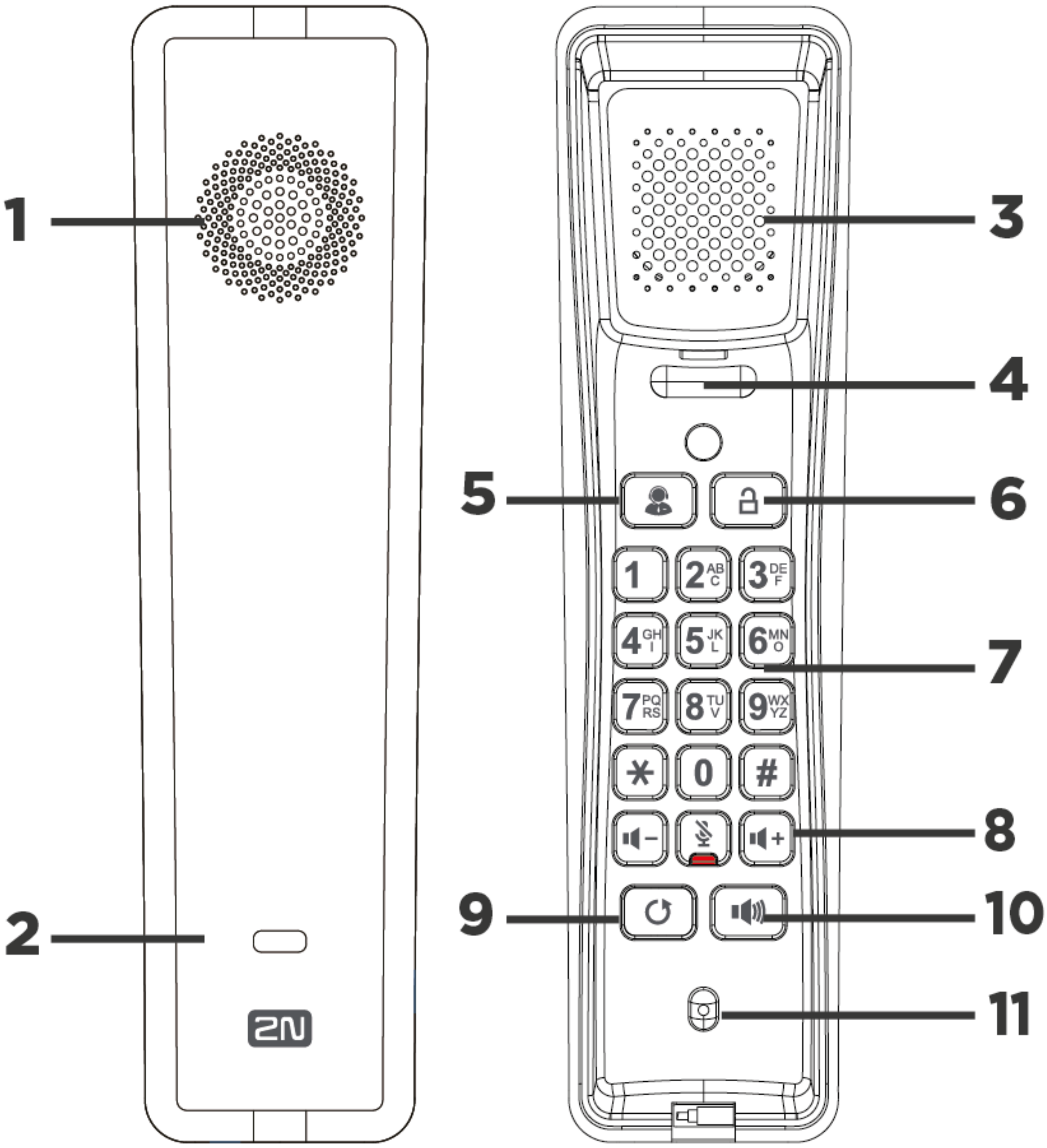


Figure 3 – Keypad Instructions

The figure above shows the keypad layout of the phone. Each button provides a function of its own. Refer to the instructions for the keys in the illustration in this subsection to operate the phone.

Table 2 – Keypad Instructions

Number	Keypad Names	Instruction
◯, 1	Hands-Free Speaker	Sound playing hands-free channel
◯,2	Status Indicator Lamp	Power indication/line status indication
◯,3	Horn Handset	Sound playing handset channel
◯,4	Hook	Handset and phone hang-up
◯,5	Function Key	User-defined functionality
◯,6	Door Unlock Button	Door lock control switch activation
◯,7	Standard Telephone Keys	12 standard telephone keys provide the same function as standard telephones, some keys also provide a special function dialed by a long key press. $\boxed{\#}$ - long press to broadcast IP（default English).
◯,8	Volume Key	Adding/subtracting volume - In the standby state, ring and ring configuration interface; press this button to increase/reduce the ring volume. Press this button to increase/lower the volume during a call. Mute key - Press this key to mute the microphone during a call.
◯,9	Redial	Press the Redial key to redial the last number dialed.
◯,10	Hands-Free Key	Press this key to open the speakerphone audio channel.
◯,11	Microphone	Listen whether the receiver is answering (do not listen when the phone is hands-free).

7.2 Using Handset / Hands-Free Speaker

- **Using Handset**

For the use of the handset, pick up the handset from the base and press the number and "#" to dial the number. You can switch the phone audio channels by pressing the hands-free button.

- **Using Hands-Free Speaker**

For the use of the speakerphone, press the speakerphone button to dial a number, or dial a number and then press the speakerphone button. When the handset voice channel is opened, you can switch the phone audio channels by pressing the hands-free speaker button.

8 Basic Functions

8.1 Making Phone Calls

- **Default Line**

The device provides two line services (1 main line and 1 standby line). If both the lines are configured successfully, use line 1 to make or receive calls by default.

- **Dialing Methods**

You can dial a number in the following ways:

- Device end
  - Dial directly: pick up the handset and input the number, then press "#" to call out
  - Redial the last dialed number（Redial）
- Web end
  - Dial the web filled-in number
  - Select a phone number from the call logs

- **Cancel Call**

You can cancel an active call by putting back the handset/pressing down the spring.

8.2 Answering Calls

You can answer a call by picking up the handset or pressing the speakerphone button to open the hands-free channel.

The telephone does not support multiple calls. When a call is in progress, the user needs to hang up the active call before answering another call.

8.3 End of Call

To end a call, put the handset back on the phone base and press the speakerphone button.

8.4 Redial

- Redial the last outgoing number
  When the phone is in the standby mode, press the redial button and the phone will call out the last number dialed.
- Call out any number with the redial key
  Enter the number, press the redial key and the phone will call out the number on the dial.
- Redial record clearing
  After every use of the phone, Redial will default to the last used number; therefore, it is necessary to clear the records used by the last customer without affecting the use of other customers.

8.5 Auto-Answering

You can enable the auto-answering feature on the device and any incoming call will be automatically answered. The auto-answering function can be enabled on a line basis.

- **WEB interface:**
  Log into the phone page, enter [**Line**] >> [**SIP**], select [**SIP**] >> [**Basic settings**], start auto-answering and click Apply after setting the automatic answering time.

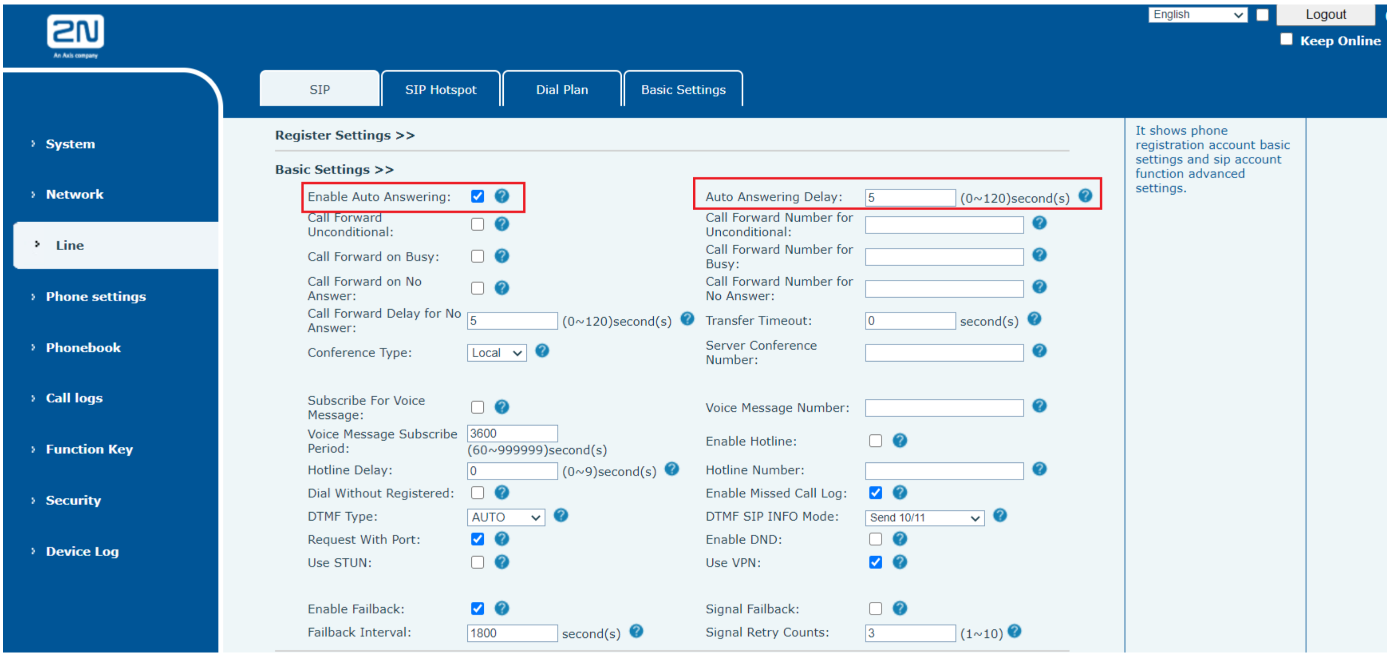


Figure 4 – Auto-Answering Web Page

8.6 Mute

You can turn on the mute mode during a call and turn off the microphone so that the local voice is not heard. Normally, the mute mode is automatically turned off at the end of a call.

You can also turn on mute on any screen (such as the free screen) and mute the ring tone automatically when there is an incoming call.

The mute mode can be turned on in all call modes (handset or hands-free).

8.6.1 Call Mute

During conversation, press the mute button on the phone, the mute LED goes red and the power LED goes orange.

Cancel mute: press again to cancel mute on the phone. When the mute LED goes out, the power LED returns to its original state

8.6.2 Ringing Mute

Mute ring tone: press the mute button when the phone is in the standby mode.
The mute LED is bright red, the power LED is flashing green. There is no ringer for incoming calls.

Cancel ring tone mute: On standby or the incoming call screen, press the mute button again or volume up to cancel the ring tone mute.

8.7 Call Hold/Resume

You can press the [**Hold**] button to maintain the current call. This button will become the [**Resume**] button and you can press "Resume" to restore the call.

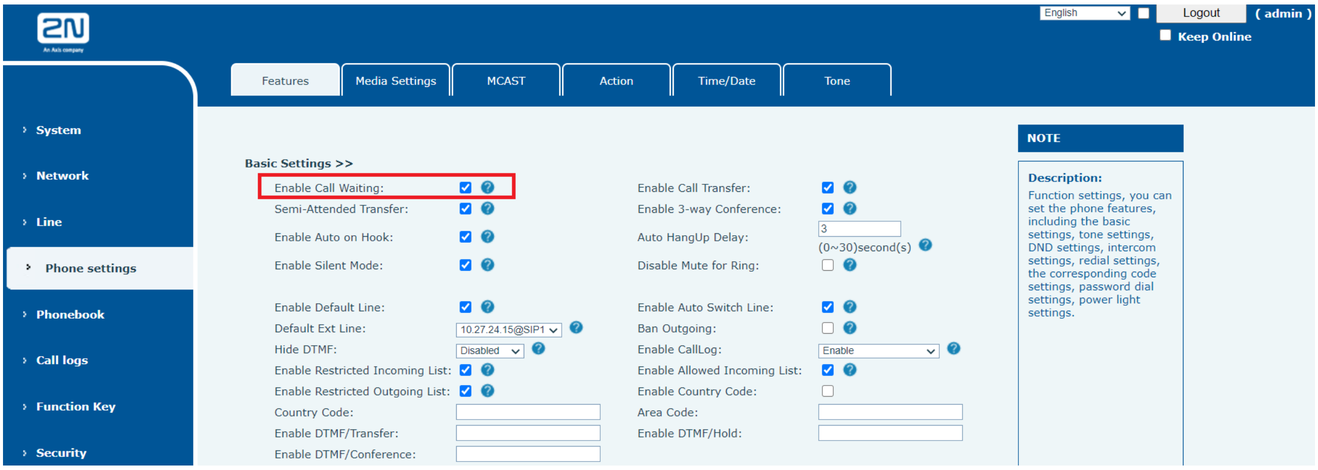
8.8 Call Waiting

- Enable call waiting: new calls can be accepted during a call.
- Disable call waiting: new calls will be automatically rejected and the busy tone will be prompted.

- Enable call waiting tone: when you receive a new call on the line, the device will beep.

Enable/disable the call waiting function via the web interface.

- WEB interface: Enter [**Phone Settings**] >> [**Features**] >> [**Basic Settings**], enable/disable call waiting and the call waiting tone.



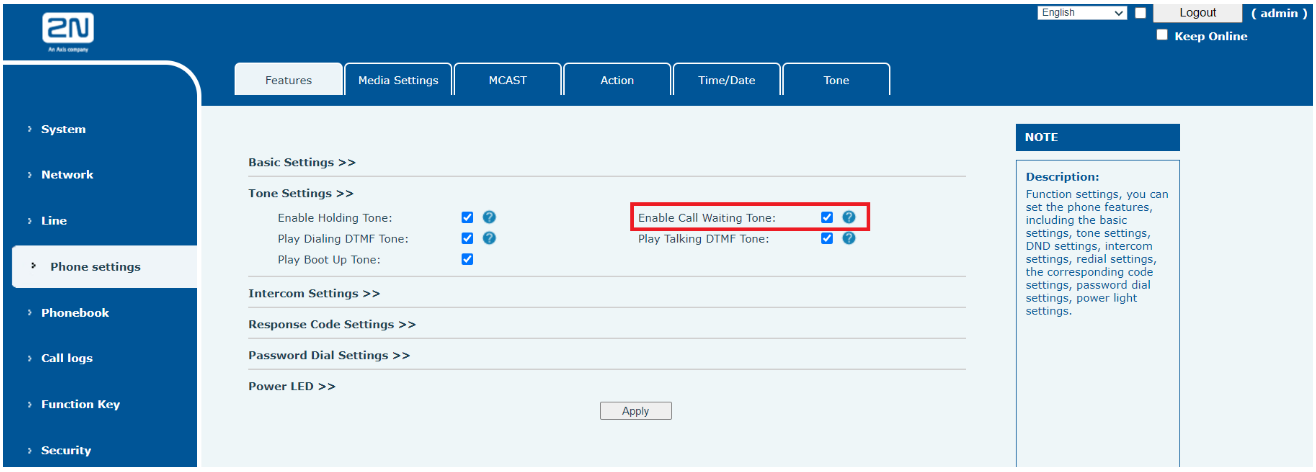


Figure 5 - Call Waiting Tone Setting Web Page

8.9 Conference

8.9.1 Local Conference

To conduct a local conference, you need to log into the web page and enter [**Line**] >> [**SIP**] >> [**Basic settings**]. The meeting mode is set as Local (Local is default), as shown in the figure below:

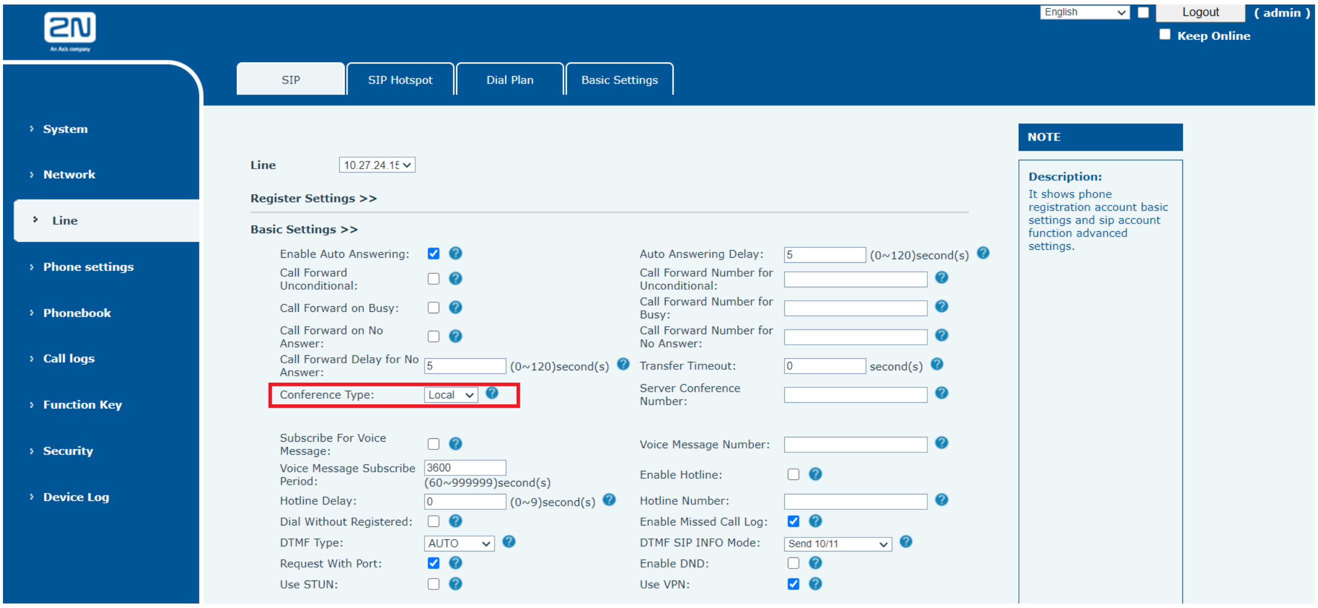


Figure 6 - Local Conference Setting

There are two ways how to create a local conference:

1） The device has two channels for communication. Press the conference button on the call interface. Having selected the conference number, select the other number that already exists.

2） If the device has a call in progress, press the conference key in the call interface, enter the number to join the meeting and press the call. After the opposite end is answered, press the conference button again to set up a local tripartite conference.

Note: During the meeting, press the separate key to separate the meeting and press the end key to end the call.

8.9.2 Network Conference

The users need server support for network conferences.

Log in to the web page, enter [**Line**] >> [**SIP**] >> [**Basic settings**], set the conference mode as server (default is local), set the server conference room number (please consult your system administrator), as shown in the figure below：

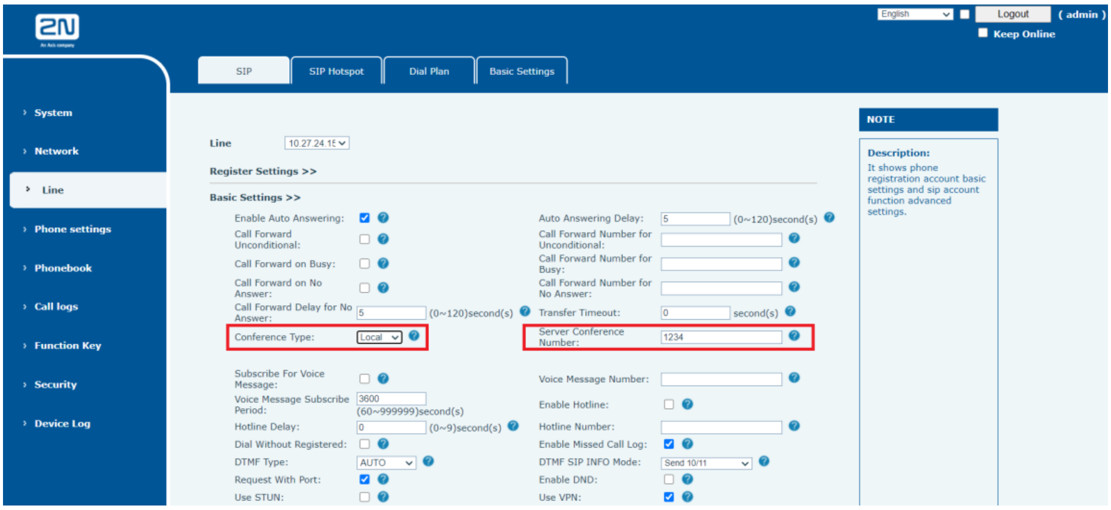


Figure 7 - Network Conference

How to join a network conference：

- Enter the multi-party call number of the network conference room and the password, then all enter the conference room.
- The two phones have established common calls. Press the conference button to invite new members to the conference. Follow the voice prompt to operate.

Note: The upper limit of the number of participants in the network conference varies according to the server.

8.10 Hotline

The device supports hotline dialing. After setting up the hotline dialing, directly pick up the handset, hands-free, earphone, etc. and the phone will automatically call according to the hotline delay time.

- You can also set up a hotline on the website [**Line**] >> [**SIP**] >> [**Basic Settings**].
- The setup hotline also corresponds to the SIP line. That means that the hotline set on the SIP1 webpage can only be activated on the SIP1 line.

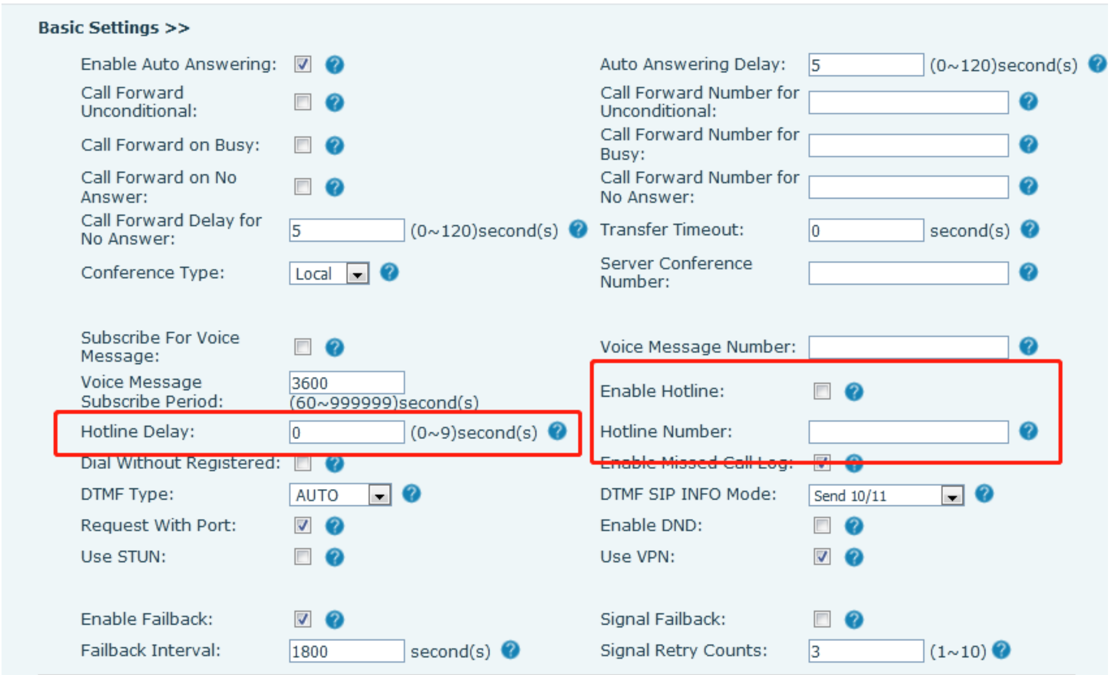


Figure 8 - Hotline Setup Web Page

9 Advanced Functions

9.1 Intercom

Once enabled, the Intercom function can automatically receive calls from an intercom.

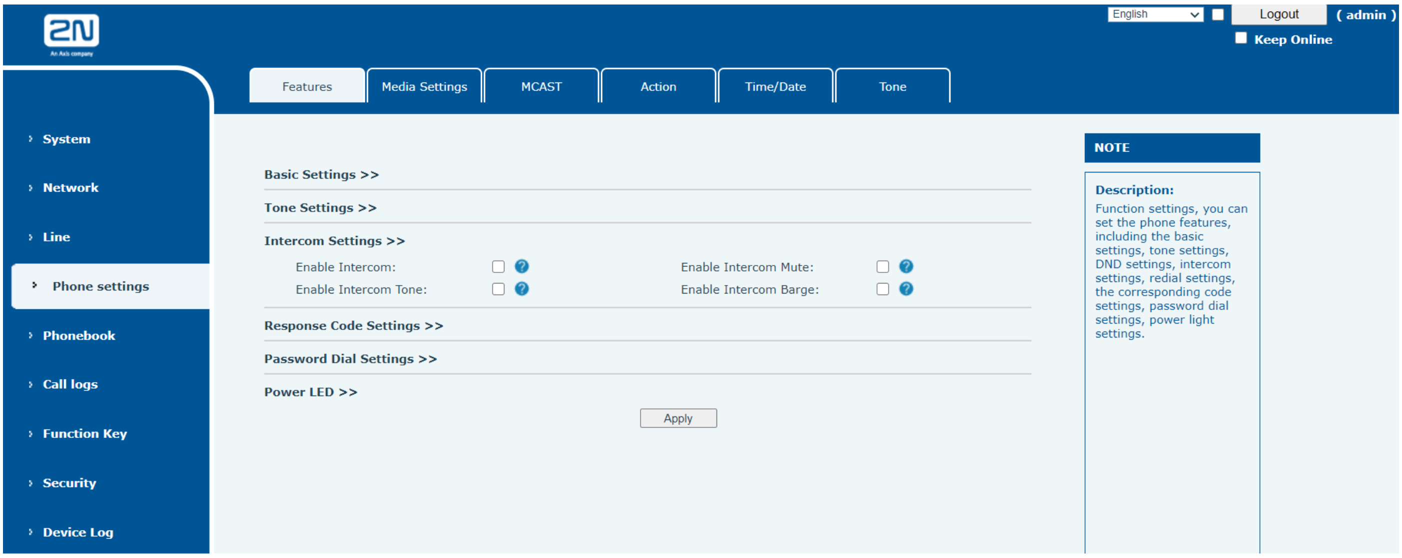


Figure 9 - Web Intercom Configuration

Table 3 - Intercom Configuration

Parameter	Description
Enable Intercom	When Intercom is enabled, the device will accept the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after a specific delay.
Enable Intercom Mute	Enable the mute mode during the intercom call.
Enable Intercom Tone	If the incoming call is an intercom call, the phone plays the intercom tone.
Enable Intercom Barge	Enable Intercom Barge by selecting it; the phone auto answers an intercom call during a call. If the current call is an intercom call, the phone will reject the other intercom call.

9.2 MCAST

This feature allows you to make some kind of broadcast call to the people who are in the multicast group. You can configure a multicast DSS Key on the phone, which allows you to send a Real Time Transport Protocol (RTP) stream to the pre-configured multicast address without involving SIP signaling. You can also configure the phone to receive an RTP stream from a pre-configured multicast listening address without involving SIP signaling. You can specify up to 10 multicast listening addresses.

9.3 Messages

9.3.1 MWI（Message Waiting Indicator）

If the line service supports the voice message feature in case the user fails to answer the call, the caller can leave a voice message on the server to the user.
The user will be notified of the server voice message and the power LED status.

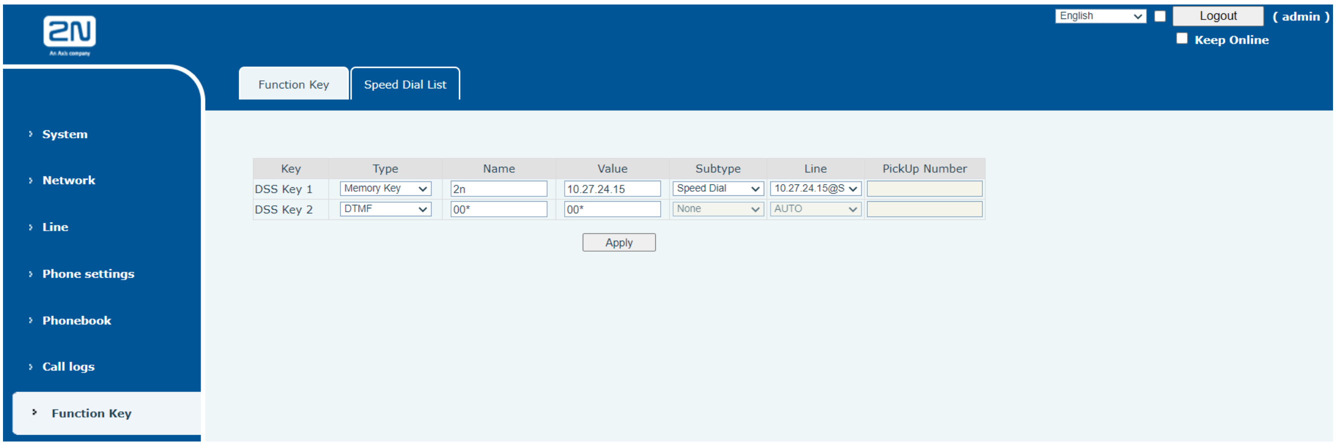


Figure 10 - New Voice Message Notification

To listen to a voice message, you must first configure the voicemail number. Having done so, you can retrieve the voicemail on the default line.

10 Phone Settings

10.1 Basic Settings

10.1.1 Language

You can set the phone language through the web interface.

- Web interface: Log in to the phone web page and set the language in the drop-down box in the right-hand upper corner of the page, as shown in the figure below:

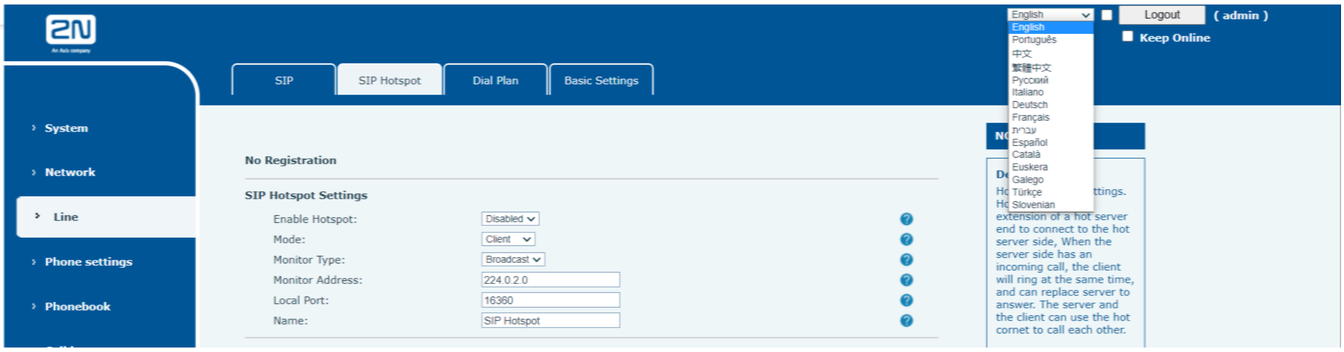


Figure 11 - Language Setting Web Page

- The function box on the right-hand side of the web interface language setting box is "Synchronize language to phone". If it is selected, the phone language will be synchronized with the web page language. If it is not selected, it will not be synchronized.

10.2 Function Keys

The device has a total of 11 configurable custom function keys: one direct call foreground key and 10 custom digital speed dial keys:

Device direct call key - default configuration as a fixed number (customizable replacement)

0~9 numeric keys - can be used as custom shortcut keys and customized via the web page. The users can quickly dial the corresponding number by a long press of each shortcut key.

The DSS Key can be configured as follows:

- Memory Key
  - Speed Dial/Intercom/BLF/Presence/Call Park/Call Forward (to someone)
- DTMF
- Action URL
- MCAST Paging

Webpage interface: [**Function key**] >> [**Function key**].

Moreover, you can also add a user-defined title to the DSS Keys, which is configured as Memory Key / Line / URL / Multicast / Prefix.

NOTICE!   The user-defined title is up to 10 characters.

Refer to *11.26 Function Key* and 6.3 Appendix I – LED Definition for more details.

11 Web Configurations

11.1 Web Page Authentication

The user can log into the web page of the phone to manage the user's phone information and operate the phone. The user must provide the correct user name and password to log in.

For security reasons, we recommend that you change the default administrator password upon the first login to the web interface by pressing the Modify button in the User Management section.

Default access data:

User: admin

Password: 2n

11.2 System >> Information

The user can get device system information on this page including:

- Model
- Hardware Version
- Software Version
- Uptime

And summarization of network status:

- Network Mode
- MAC Address
- IP
- Subnet Mask
- Default Gateway

Besides, summarization of SIP account status:

- SIP User
- SIP Account Status (Registered / Unapplied / Trying / Timeout)

11.3 System >> Account

On this page, you can change the password for the login page.

The users with administrator rights can also add or delete users, manage users as well as set permissions and passwords for new users.

11.4 System >> Configurations

On this page, the users with administrator privileges can view, export or import the phone configuration, or reset the phone to factory settings.

- **Clear Configurations**

Select the module in the configuration file to be cleared.

SIP: account configuration

AUTOPROVISION: automatic configuration upgrade

TR069: TR069 related configuration

MMI: MMI module including authentication user information, web access protocol, etc.

DSS Key: DSS Key configuration

BASIC NETWORK: NETWORK configuration

- **Clear Tables**

Select the local data table to be cleared, all selected by default.

- **Reset Phone**

The phone data will be cleared including the configuration and database tables.

11.5 System >> Upgrade

Web interface: log into the phone web page and enter the [system] >> [upgrade] page.

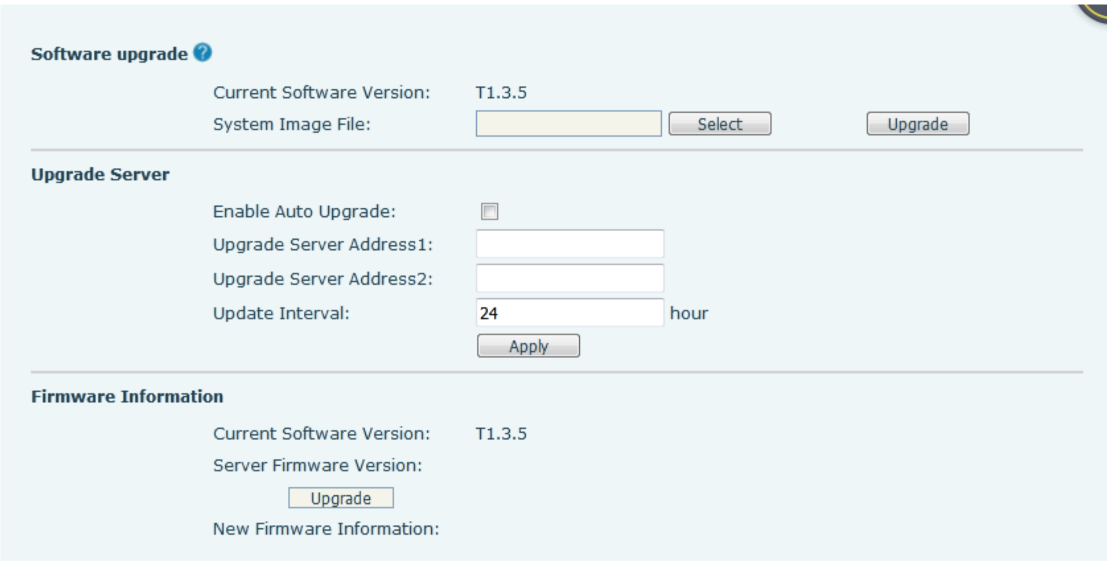


Figure 12 – Firmware Upgrade Web Page

Table 4 - Firmware Upgrade

Parameter	Description
Upgrade Server	
Enable Auto Upgrade	Enable automatic upgrade. If there is a new version, txt and new software firmware on the server, the phone will show a prompt upgrade message after the Update Interval.
Upgrade Server Address1	Set the available upgrade server address.
Upgrade Server Address2	Set the available upgrade server address.
Update Interval	Set the Update Interval.
Firmware Information	
Current Software Version	Show the Current Software Version.
Server Firmware Version	Show the Server Firmware Version.
[Upgrade] button	If there is a new version, txt and new software firmware on the server, the page will display the version information and the upgrade button will become available. Click [Upgrade] to upgrade the firmware.
New version of description information	When there is a corresponding TXT file and version on the server side, the TXT and version information will be displayed under the new version description information.

- The file requested from the server is a TXT file called vendor_model_hw10.txt.Hw followed by the hardware version number. It is written as hw10 if there is no difference in hardware. All Spaces in the filename are replaced with underline.
- The URL requested by the phone is HTTP:// server address/vendor_Model_hw10 .txt. The new version and the requested file should be placed in the download directory of the HTTP server.
- The TXT file format must be UTF-8.
- vendor_model_hw10.TXT The file format is as follows:
  Version=1.6.3 #Firmware
  Firmware=xxx/xxx.z #UR Relative paths are supported and absolute paths are possible, distinguished by the presence of protocol headers.
  BuildTime=2018.09.11 20:00
  Info=TXT|XML
  Xxxxx
  Xxxxx
  Xxxxx
  Xxxxx
- After the update cycle interval passes and if the server has available files and

versions, the phone will prompt as shown below. Click [View] to check the version information and upgrade.

11.6 System >> Auto Provision

Page interface: log into the phone page and enter the [system] >> [automatic deployment] page.

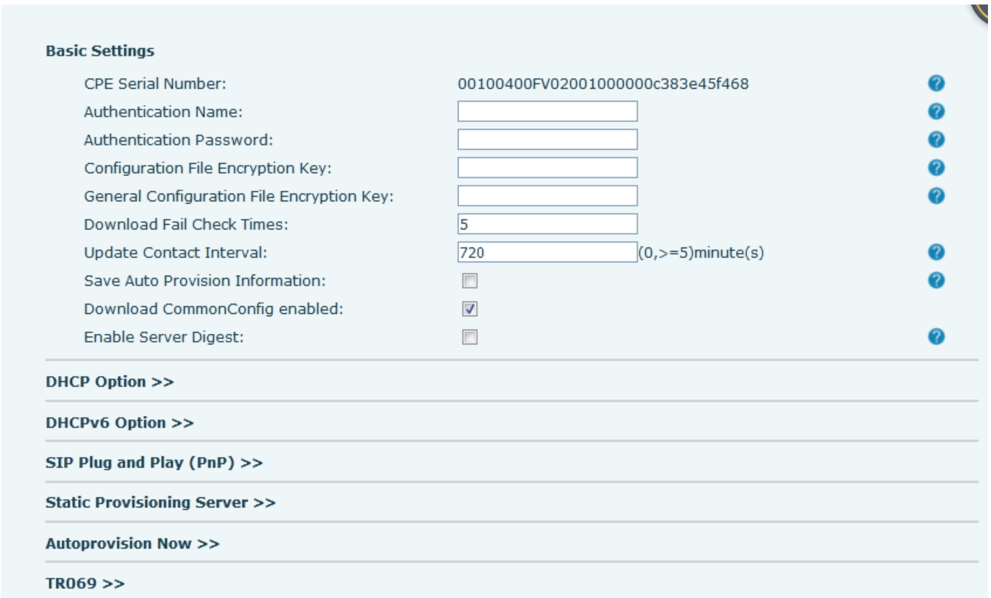


Figure 13 - Auto Provision Settings

2N® IP Handset supports SIP PnP, DHCP options, Static provision, TR069. If all of the 4 methods are enabled, the priority from high to low is as follows:

PNP>DHCP>TR069> Static Provisioning

Transferring protocol: FTP, TFTP, HTTP, HTTPS

For details refer to **2N® IP Handset Auto Provision**.

Parameters	Description
Basic Settings	
CPE Serial Number	Display the device SN.
Authentication Name	Provision server username
Authentication Password	Provision server password
Configuration File Encryption Key	If the device configuration file is encrypted, you should add the encryption key here.
General Configuration File Encryption Key	If the common configuration file is encrypted, you should add the encryption key here.

Download Fail Check Times	If the download fails, the phone will retry with the configured times.
Update Contact Interval	The phone will update the phonebook with the configured interval. If it is 0, the feature is disabled.
Save Auto Provision Information	Save the HTTP/HTTPS/FTP username and password. If the provision URL is kept, the information will be kept.
Download Common Config enabled	The phone will download the common configuration file.
Enable Server Digest	When the feature is enabled and if the server configuration has been changed, the phone will download and update.
DHCP Option	
Option Value	Configure DHCP option: DHCP option supports DHCP custom option | DHCP option 66 | DHCP option 43 - 3 methods to get the provision URL. The default is Option 66.
Custom Option Value	Custom Option value is allowed from 128 to 254. The option value must be the same as defined on the server.
Enable DHCP Option 120	Use Option120 to get the SIP server address from the DHCP server.
SIP Plug and Play (PnP)	
Enable SIP PnP	Enable/disable PnP. If PnP is enabled, the phone will send a SIP SUBSCRIBE message with a broadcast method. Any server that supports the feature will respond and send a Notify with URL to the phone. The phone could get the configuration file with the URL.
Server Address	Broadcast address. It is 224.0.0.0 by default.
Server Port	PnP port
Transport Protocol	PnP protocol: TCP or UDP
Update Interval	PnP message interval
Static Provisioning Server	
Server Address	Server address provisioning. Supports both IP address and Domain address.
Configuration File Name	Configuration file name. If it is empty, the phone will request the common file and device file which is named as its MAC address. The file name could be a common name, $mac.cfg, $input.cfg. The file format supports CFG/TXT/XML.
Protocol Type	Transferring protocol type: FTP, TFTP, HTTP and HTTPS.
Update Interval	Configuration file update interval time. It is 1 by default, which means that the phone will check the update every 1 hour.
Update Mode	Provision Mode

	1. Disabled. 2. Update after reboot 3. Update after interval
TR069	
Enable TR069	Enable TR069 after selection.
ACS Server Type	There are 2 server type options: common and CTC.
ACS Server URL	ACS server address
ACS User	ACS server username (up to 59 characters)
ACS Password	ACS server password (up to 59 characters)
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone while connecting.
TLS Version	TLS version (TLS 1.0, TLS 1.1, TLS 1.2)
INFORM Sending Period	INFORM signal interval time. It ranges from 1 to 999 s.
STUN Server Address	Configure the STUN server address.
STUN Enable	Enable the STUN server for TR069.

11.7 System >> Tools

This page provides tools for users to resolve problems.

- Syslog
  You can choose the log level, export the system log in order to analyze the problem in case of failure.
- Web Capture
  Grab packets from the network data to analyze problems in case of failure
- Watch Dog
  When stuck while in use, the device will automatically restart and recover.
- Ping
  Check the destination IP address to be reached and record the result, showing whether the destination is responding and how long it takes to receive the reply.

11.8 System >> Reboot Phone

This page can restart the phone.

11.9 Network >> Basic

The phone only supports wired network connections. The phone USES an IP network connection to provide services. Unlike traditional telephony based on circuit technology, IP telephony exchanges packets and data over a network based on the IP address of the telephony.

To enable the phone, the network configuration must be performed correctly. The default network mode of the device is DHCP/IPv4. To modify the mode, go to the device web configuration interface.
Web interface: [network] >> [basic] select network mode

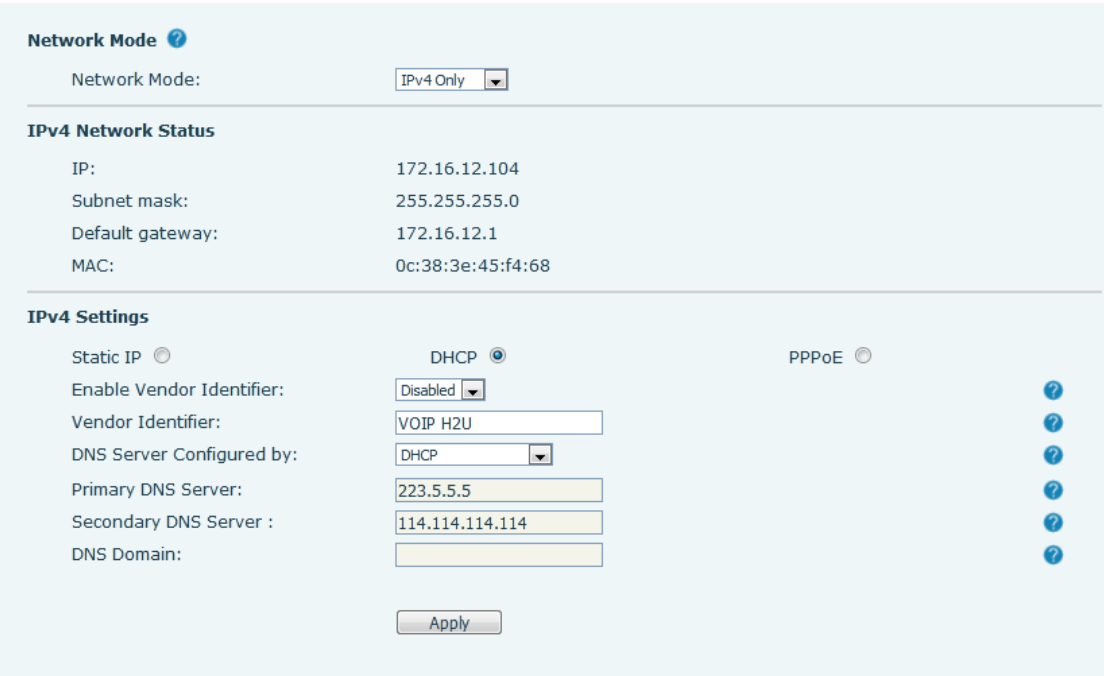


Figure 14 - Network Mode Settings

- **IP Mode**

There are 3 network protocol mode options: IPv4, IPv6 and IPv4 & IPv6.

- **IPv4**

In the IPv4 mode, there are 3 connection mode options: DHCP, PPPoE and Static IP.

In the DHCP mode, the phone will get the IP address from the DHCP server (router).

- Use DHCP DNS: It is enabled by default. “Enable” means that the phone gets the DNS address from the DHCP server and “disable” means it does not.
- Use DHCP time: It is disabled by default. “Enable” means to manage the time of getting the DNS address from the DHCP server and “disable” means not to.

With PPPoE, the phone will get the IP address from the PPPoE server.

- Username: PPPoE user name
- Password: PPPoE password

In the Static IP mode, you must configure the IP address manually.

- IP Address: Phone IP address
- Mask: sub mask of your LAN
- Gateway: gateway IP address. The phone can access the other network via it.
- Primary DNS: primary DNS address. The default is 8.8.8.8, Google DNS server address.
- Secondary DNS: secondary DNS. Available when the primary DNS is unavailable.

■ **Pv6**

In IPv6, there are 2 connection mode options: DHCP and Static IP.

- DHCP configuration refers to IPv4 introduction on the preceding page.
- Static IP configuration is almost the same as IPv4, except for the IPv6 Prefix.
- IPv6 Prefix: similar to the IPv4 mask.

11.10 Network >> Service Port

This page provides settings for the web page login protocol, protocol port and RTP port.

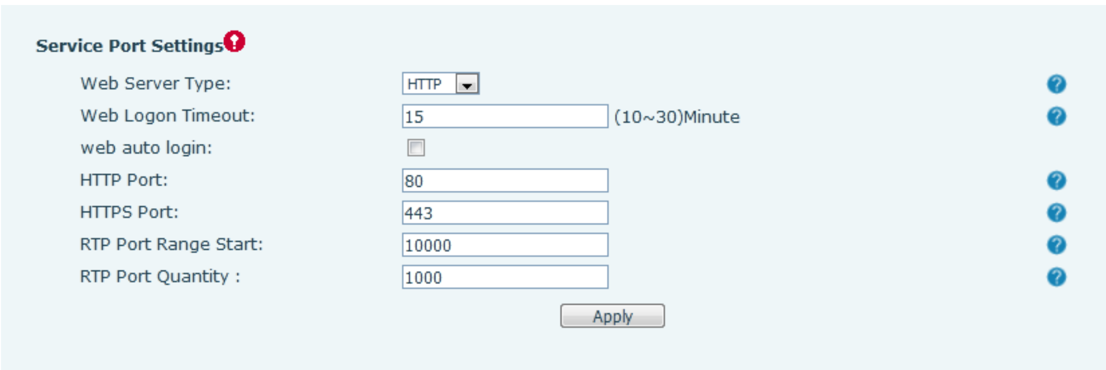


Figure 15 - Service Port Settings

Table 5 - Service Port

Parameter	Description
Web Server Type	Reboot to take effect after settings. Optionally, the web page login is HTTP/HTTPS.
Web Logon Timeout	The default is 15 minutes, you are logged out of the login page after the timeout and need to relog in.
Web auto login	After the timeout, you do not have enter your user name and password, you will automatically log in to the web page.
HTTP Port	The default is 80. If you want system security, you can set ports other than 80, such as 8080, webpage login: HTTP://ip:8080.
HTTPS Port	The default is 443, same as the HTTP port.
RTP Port Range Start	The value range is 1025 to 65535. The RTP port value starts from the initial value set. For each call, for the voice and video port value, 2 is added.
RTP Port Quantity	Number of calls

11.11 Network >> VPN

Virtual Private Network (VPN) is a technology allowing devices to create a tunneling connection to a server and become part of the server's network. The network transmission of the device may be routed through the VPN server.

For some users, especially enterprise users, a VPN connection might be required before their line registration is activated. The device supports two VPN modes: Layer 2 Transportation Protocol (L2TP) and OpenVPN.

The VPN connection must be configured and started (or stopped) from the device web portal.

- **L2TP**

NOTICE! The device only supports non-encrypted basic authentication and non-encrypted data tunneling. For the users who need data encryption, please use OpenVPN instead.

To establish an L2TP connection, log in to the device web portal and open webpage

[**Network**] >> [**VPN**]. In VPN Mode, check the “Enable VPN” option and select “L2TP”, then fill in the L2TP server address, Authentication Username and Authentication Password in the L2TP section. Press “Apply” to connect the device to the L2TP server. Once the VPN connection is established, the VPN IP Address should be displayed in the VPN status. There may be a delay of the connection establishment. You may need to refresh the page to update the status.

Once the VPN configuration is complete, the device will try to connect to the VPN automatically whenever the device boots up until the user disables it. Sometimes, if the VPN connection is not established immediately, the user may try to reboot the device and check if the VPN connection gets established after reboot.

- **OpenVPN**

To establish an OpenVPN connection, you should get the following authentication and configuration files from the OpenVPN hosting provider and name them as follows:

OpenVPN Configuration file:	client.ovpn
CA Root Certification:	ca.crt
Client Certification:	client.crt
Client Key:	client.key

Then upload these files to the device on the web page [**Network**] >> [**VPN**], select OpenVPN Files. Then check “Enable VPN” and select “OpenVPN” in VPN Mode and click “Apply” to enable the OpenVPN connection.

Like L2TP connection, the OpenVPN connection will be established whenever the system reboots until the user disables it manually.

11.12 Network >> Advanced

- **LLDP**

Link Layer Discovery Protocol (LLDP) is a vendor independent link layer protocol used by network devices for advertising their identities and capabilities to neighbors on a LAN segment.

The phone can use LLDP to find the VLAN switch or other VLAN devices and use the LLDP learn feature to apply the VLAN ID from the VLAN switch to the phone itself.

- **CDP**

Cisco Discovery Protocol (CDP) is a nonprofit charity that runs the global disclosure system for investors, companies, cities, states and regions to manage their environmental impacts. According to the CDP, Cisco devices can share the OS version, IP address, hardware version and so on.

Table 6 - QoS & VLAN

Parameters	Description
LLDP Setting	
Report	Enable LLDP.
Interval	LLDP requests interval time.
Learning	Apply the learned VLAN ID to the phone configuration.
QoS	
QoS Mode	SIP DSCP and audio DSCP configuration
WAN VLAN	
WAN VLAN	WAN port VLAN configuration
LAN VLAN	
LAN VLAN	LAN port VLAN configuration
CDP	
CDP	CDP enable/disable, CDP interval time

11.13 Line >> SIP

Configure the Line service on this page.

Table 7 - Line Configuring Web Page

Parameters	Description
Register Settings	
Line Status	Display the current line status at page loading. To get the up-to-date line status, refresh the page manually.
Activate	Activate the line service.
Username	Enter the service account username.
Authentication User	Enter the authentication user of the service account.
Display Name	Enter the display name to be sent in a call request.
Authentication Password	Enter the authentication password of the service account.
Realm	Enter the SIP domain if requested by the service

	provider.
Server Name	Input the server name.
SIP Server 1	
Server Address	Enter the IP or FQDN address of the SIP server.
Server Port	Enter the SIP server port, 5060 by default.
Transport Protocol	Set up the SIP transport line using TCP or UDP or TLS.
Registration Expiration	Set the SIP expiration date.
SIP Server 2	
Server Address	Enter the IP or FQDN address of the SIP server.
Server Port	Enter the SIP server port, 5060 by default.
Transport Protocol	Set up the SIP transport line using TCP or UDP or TLS.
Registration Expiration	Set the SIP expiration date.
SIP Proxy Server Address	Enter the IP or FQDN address of the SIP Proxy server.
Proxy Server Port	Enter the SIP Proxy server port, 5060 by default.
Proxy User	Enter the SIP Proxy user.
Proxy Password	Enter the SIP Proxy password.
Backup Proxy Server Address	Enter the IP or FQDN address of the backup Proxy server.
Backup Proxy Server Port	Enter the backup Proxy server port, 5060 by default.
Basic Settings	
Enable Auto Answering	Enable auto-answering; incoming calls will be answered automatically after a delay.
Auto Answering Delay	Set the delay for incoming calls before the system automatically answers them.
Call Forward Unconditional	Enable unconditional call forwarding; all incoming calls will be forwarded to the number specified in the next field.
Call Forward Number for Unconditional	Set the number for unconditional call forwarding.
Call Forward on Busy	Enable call forwarding whenever the phone is busy; any incoming call will be forwarded to the number specified in the next field.
Call Forward Number for Busy	Set the number for call forwarding on busy.
Call Forward on No Answer	Enable call forwarding whenever an incoming

	call is not answered within the configured timeout; the call will be forwarded to the number specified in the next field.
Call Forward Number for No Answer	Set the number for call forwarding on no answer.
Call Forward Delay for No Answer	Set the no answer timeout before the call is forwarded.
Transfer Timeout	Set the timeout for call transfer process.
Conference Type	Set the call conference type: Local=call conference set up by the device, support of up to two remote parties, Server=call conference set up by dialing a conference room on the server.
Server Conference Number	Set the conference room number when the conference type is set to Server.
Subscribe for Voice Message	Enable the device to subscribe for voice message waiting notifications. If enabled, the device will receive a notification from the server if there is a voice message waiting on the server.
Voice Message Number	Set the number for retrieving voice messages.
Voice Message Subscribe Period	Set the interval of voice message notification subscription.
Enable Hotline	Enable hotline configuration; the device will dial a specific number immediately via the audio channel opened by the handset off-hook or turned-on hands-free speaker or headphone.
Hotline Delay	Set the hotline delay before the system automatically dials it.
Hotline Number	Set the hotline dialing number.
Dial Without Registered	Set a call by Proxy without registration.
Enable Missed Call Log	If enabled, the phone will save missed calls into the missed call log.
DTMF Type	Set the DTMF type to be used for the line.
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'.
Enable DND	Enable Do Not Disturb; any incoming call to this line will be rejected automatically.
Subscribe for Voice Message	Enable the device to subscribe for voice message waiting notifications. If enabled, the

	device will receive a notification from the server if there is a voice message waiting on the server.
Use VPN	Set the line to use the VPN restrict route.
Use STUN	Set the line to use STUN for NAT traversal.
Enable Failback	Enable switching to the primary server if available.
Failback Interval	A Register message is used to periodically detect the time interval for the availability of the main Proxy.
Signal Failback	Multiple Proxy cases, whether to allow the invite/register request to also execute failback.
Signal Retry Counts	The number of attempts that the SIP Request considers Proxy unavailable under multiple Proxy scenarios.
Codec Settings	Set the priority and availability of the codecs by adding to or removing from the list.
Video Codecs	Select the video code to preview video.
Advanced Settings	
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send a feature code to the server by dialing the number specified in each feature code field.
Enable DND	Set the feature code to dial the server.
Disable DND	Set the feature code to dial the server.
Enable Call Forward Unconditional	Set the feature code to dial the server.
Disable Call Forward Unconditional	Set the feature code to dial the server.
Enable Call Forward on Busy	Set the feature code to dial the server.
Disable Call Forward on Busy	Set the feature code to dial the server.
Enable Call Forward on No Answer	Set the feature code to dial the server.
Disable Call Forward on No Answer	Set the feature code to dial the server.
Enable Blocking Anonymous Call	Set the feature code to dial the server.
Disable Blocking Anonymous Call	Set the feature code to dial the server.
Call Waiting On Code	Set the feature code to dial the server.
Call Waiting Off Code	Set the feature code to dial the server.
Send Anonymous On Code	Set the feature code to dial the server.

Send Anonymous Off Code	Set the feature code to dial the server.
SIP Encryption	Enable SIP encryption to encrypt SIP transmissions.
RTP Encryption	Enable RTP encryption to encrypt RTP transmissions.
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session will be ended if no new session timer event update has been received after the timeout period.
Session Timeout	Set the session timer timeout.
Enable BLF List	Enable/disable the BLF list.
BLF List Number	The BLF list allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
Response Single Codec	If the setting is enabled, the device will use a single codec in response to an incoming call request.
BLF Server	The registered server will receive the subscription package from an ordinary application of the BLF phone. Please enter the BLF server: if the sever does not support the subscription package, the registered server and subscription server will be separated.
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep the NAT pinhole opened.
Keep Alive Interval	Set the keep alive packet transmitting interval.
Keep Authentication	Keep the authentication parameters from previous authentication.
Blocking Anonymous Call	Reject any incoming call that fails to present the caller ID.
User Agent	Set the user agent; the default is Model with Software Version.
Specific Server Type	Set the line to collaborate with a specific server type.
SIP Version	Set the SIP version.
Anonymous Call Standard	Set the standard to be used for anonymous.

Local Port	Set the local port.
Ring Type	Set the ring tone type for the line.
Enable user=phone	Set user=phone in SIP messages.
Use Tel Call	Set the use of the tel call.
Auto TCP	Set using TCP to guarantee usability of transport for SIP messages above 1500 bytes.
Enable Rport	Set the line to add rport in SIP headers.
Enable PRACK	Set the line to support PRACK SIP message.
DNS Mode	Select the DNS mode: A, SRV, NAPTR.
Enable Long Contact	Allow more parameters in the contact field per RFC 3840.
Enable Strict Proxy	Enable the use of strict routing. Having received packets from the server, the phone will use the source IP address instead of the address in the via field.
Convert URI	Convert the non-digit and alphabet characters to %hh hex code.
Use Quote in Display Name	Add a quote to the display name, i.e. “2N” vs 2N.
Enable GRUU	Support Globally Routable User-Agent URI (GRUU).
Sync Clock Time	Time Sync with server
Enable Inactive Hold	With the post-call hold capture package enabled, you can see that in the INVITE package, SDP is inactive.
Caller ID Header	Set the Caller ID Header.
Use 182 Response for Call waiting	Set the device to use 182 response code at the call waiting response.
Enable Feature Sync	Feature Sync with server
Enable SCA	Enable/disable SCA (Shared Call Appearance).
CallPark Number	Set the CallPark number.
Server Expire	Set the server using timeout.
TLS Version	Choose TLS Version.
uaCSTA Number	Set uaCSTA Number.
Enable Click To Talk	With the use of a special server, click to call out directly after enabling.
Flash mode	Choose the Flash mode: normal or SIP info.
Flash Info Content-Type	Set the SIP info content type.

Flash Info Content-Body	Set the SIP info content body.
PickUp Number	Set the scramble number when Pickup is enabled.
JoinCall Number	Set JoinCall Number.
Intercom Number	Set Intercom Number.
Unregister On Boot	Enable logout function.
Enable MAC Header	Open the SIP package registration with user agent with MAC.
Enable Register MAC Header	Open the registration with user agent with MAC.
BLF Dialog Strict Match	Enable/disable accurate matching of BLF sessions.
PTime(ms)	Enable/disable bringing of ptime field, the default is no.
SIP Global Settings	
Strict Branch	Set to strictly match the Branch field.
Enable Group	Set an open group.
Enable RFC4475	Set to enable RFC4475.
Enable Strict UA Match	Enable strict UA matching.
Registration Failure Retry Time	Set the registration failure retry time.
Local SIP Port	Modify the phone SIP port.

11.14 Line >> SIP Hotspot

SIP hotspot is a simple but practical function. With simple configurations, the SIP hotspot function can implement group ringing. SIP accounts can be expanded.

The phone set functions as a SIP hotspot and the other phone sets (B and C) function as SIP hotspot clients. When somebody calls phone set A, phone sets A, B, and C start ringing. When any phone set answers the call, the other phone sets stop ringing. The call can be answered by only one phone set. When B or C initiates a call, the SIP number registered by phone set A is the calling number.

To set a SIP hotspot, register one SIP account at least.

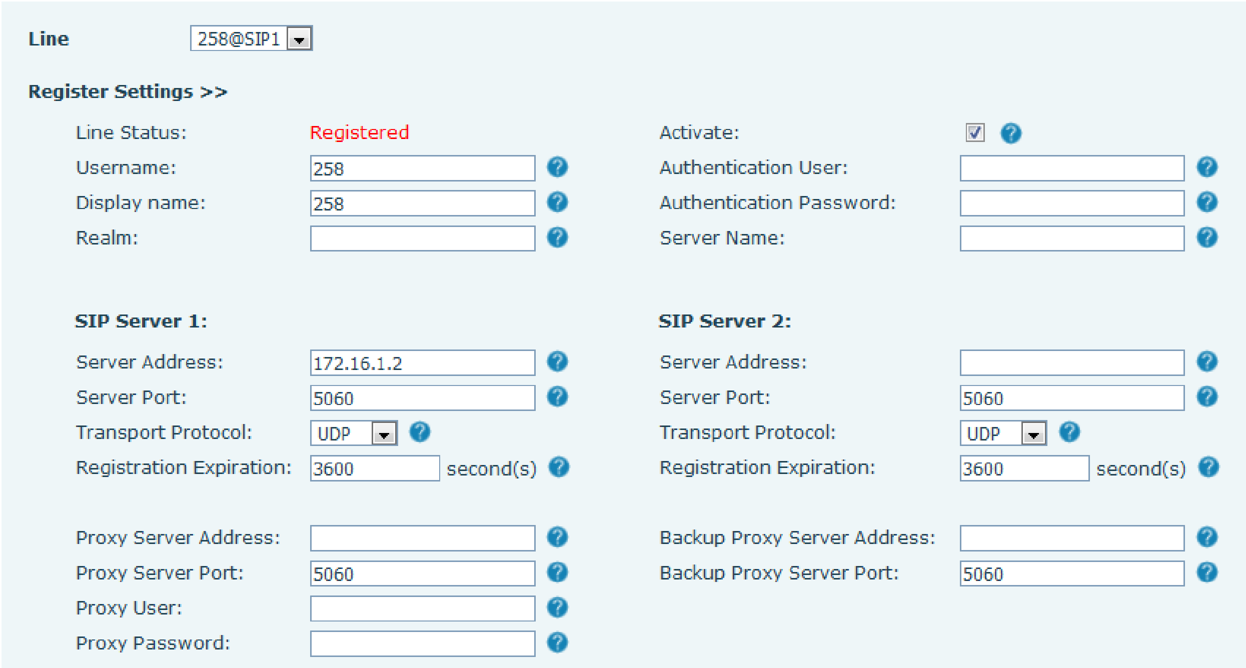


Figure 16 - Registering SIP Account

Table 8 - SIP Hotspot Parameters

Parameters	Description
Device Table	If your phone is set to “SIP hotspot server”, the Device Table connected to your phone will display as Client. If your phone is set to “SIP hotspot client”, the Device Table you can connect to will display as Server Device.
SIP Hotspot	
Enable Hotspot	Enable the Hotspot feature.
Mode	If you choose Hotspot, the phone will be a “SIP hotspot server”, if you choose Client, the phone will be a “SIP hotspot client”.
Monitor Type	Either Multicast or Broadcast is ok. If you want to limit the broadcast packets, you should better use Broadcast. But, if the client chooses Broadcast, the SIP hotspot phone must be Broadcast.
Monitor Address	The broadcast, hotspot server and hotspot client addresses must be same.
Remote Port	Type the Remote port number.

Configure SIP hotspot server:

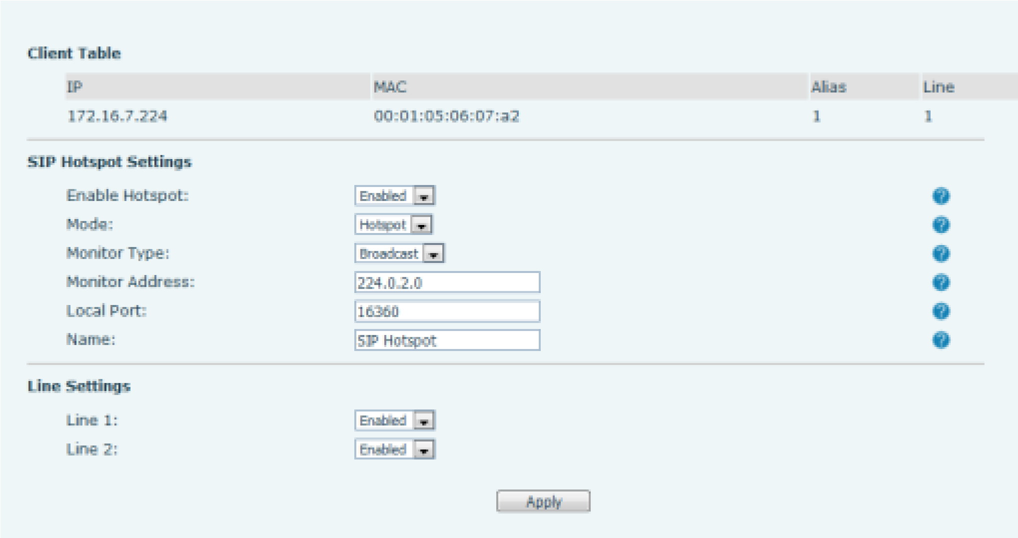


Figure 17 - SIP Hotspot Server Configuration

Configure SIP hotspot client:

To configure a SIP hotspot client, no SIP account needs to be set. The Phone set will automatically obtain and configure a SIP account. On the SIP Hotspot tab page, set Mode to Client. The values of the other options are the same as those of the hotspot.

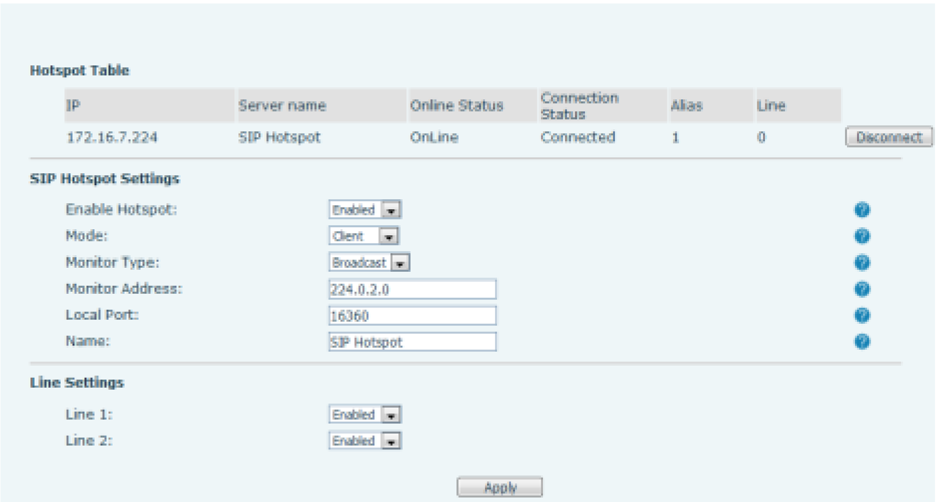


Figure 18 - SIP Hotspot Client Configuration

With the phone as the hotspot server, the default extension number is 0. When the phone is used as the client, the extension number is increased from 1; you can view the extension number via the [**SIP Hotspot**] page.

Call extension number:

- The hotspot server and client can dial each other through the extension number.
- For example, extension 1 dials extension 0.

11.15 Line >> Dial Plan

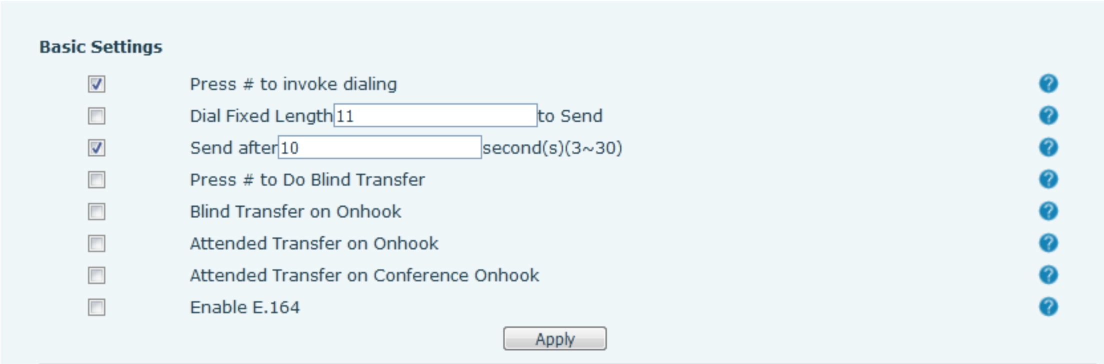


Figure 19 - Dial Plan Settings

Table 9 - Phone Dialing Methods

Parameters	Description
Press # to invoke dialing	The user dials the other party's number and then adds # to dial out.
Dial Fixed Length	The number entered by the user is automatically dialed out when it reaches a fixed length.
Timeout dial	The system dials automatically after timeout.
Press # to Do Blind Transfer	The user enters the number to be transferred and then presses the "#" key to transfer the current call to a third party.
Blind Transfer on Onhook	Having dialed the number, hang up the handset or turn off the hands-free function to transfer the current call to a third party.
Attended Transfer on Onhook	Hang up the handset or press the hands-free button to perform the attended transfer function, which can transfer the current call to a third party.
Attended Transfer on Conference Onhook	During a three-way call, hang up the handset and the remaining two parties will remain on the call.
Enable E.164	Please refer to the e.164 standard specification.

Add dialing rules:

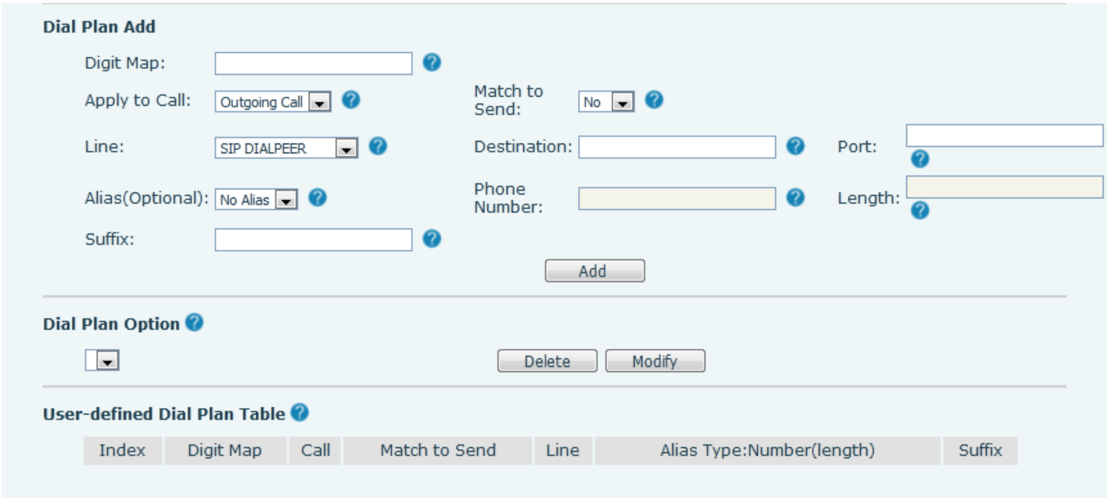


Figure 20 - Custom Setting of Dial-Up Rules

Table 10 - Dial-Up Rule Configuration Table

Parameters	Description
Dial rule	There are two types of matching: Full Matching and Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules. In Prefix matching, only a part of the number is entered followed by T. The mapping then takes place whenever these digits are dialed. The Prefix mode supports a maximum of 30 digits.
Note: 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.	
Destination	Set the Destination address. This is for IP direct.
Port	Set the Signal port, the default is 5060 for SIP.
Alias	Set the Alias. This is the text to be added, replaced or deleted. It is an optional item.
Note: There are four types of aliases. ■ all: xxx – xxx will replace the phone number.	

■ add: xxx – xxx will be dialed before any phone number. ■ del – The characters will be deleted from the phone number. ■ rep: xxx – xxx will be substituted for the specified characters.	
Suffix	Characters to be added to the phone number end. It is an optional item.
Length	Set 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. It is an optional item.

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

Example 1: All Substitution -- Assume that it is desired to place a direct IP call to IP address 172.168.2.208. Using this feature, 123 can be substituted for 172.168.2.208.

User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix	Media
1	"123"	Out	No	SIP DIALPEER(172.16.1.15:5560)			Default

Figure 21 - Dial Rules Table (1)

Example 2: Partial Substitution -- To dial a long-distance call to Beijing requires dialing the 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.

User-defined Dial Plan Table

Index	Digit Map	Call	Match to Send	Line	Alias Type:Number(length)	Suffix	Media
1	"1T"	Out	No	Fanvil@SIP1	rep:010(1)		Default

Figure 22 - Dial Rules Table (2)

Example 3: 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.

11.16 Line >> Basic Settings

Set up the register global configuration.

Table 11 - Line Global Configuration Web Page

Parameters	Description
STUN Settings	
Server Address	Set the STUN server address.
Server Port	Set the STUN server port, 3478 by default.
Binding Period	Set the STUN binding period which can be used to keep the NAT pinhole opened.
SIP Waiting Time	Set the timeout of STUN binding before SIP messages are sent.
TLS Authentication	
TLS Certification File	Upload/delete the TLS certification file used for encrypted SIP transmission.

11.17 Phone Settings >> Features

Configure the phone features.

Table 62 - General Function Settings

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow the user to take another incoming call during an active call. The default is enabled.
Enable Call Transfer	Enable Call Transfer.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it.
Enable 3-Way Conference	Enable 3-way conference by selecting it.
Enable Auto Onhook	The phone hangs up and returns to idle automatically in the hands-free mode.

Auto Onhook Time	Specify the Auto Onhook time; the phone hangs up and returns to idle automatically after the Auto Hand down time in the hands-free mode and plays the dial tone Auto Onhook time in the handset mode.
Ring for Headset	Enable Ring for Handset by selecting it; the phone plays the ring tone from the handset.
Auto Headset	Enable the headset plugged into the phone; press the ‘answer’ key or line key to answer a call with the headset automatically.
Enable Silent Mode	When this is enabled, the phone is muted, there is no call ringing and you can use the volume keys and mute key to unmute.
Disable Mute for Ring	When it is enabled, you cannot mute the phone.
Enable Default Line	If this is enabled, you can assign the default SIP line for dialing out rather than SIP1.
Enable Auto Switch Line	Enable the phone to select an available SIP line as default automatically.
Default Ext Line	Select the default line to be used for outgoing calls.
Ban Outgoing	If you select Ban Outgoing, you cannot dial out any number.
Hide DTMF	Configure the Hide DTMF mode.
Enable CallLog	Select whether to save the call log.
Enable Restricted Incoming List	Enable the restricted call list.
Enable Allowed Incoming List	Enable the allowed call list.
Enable Restricted Outgoing List	Enable the restricted allocation list.
Enable Country Code	Enable the country code.
Country Code	Fill in the country code.
Area Code	Fill in the area code.
Enable Number Privacy	Enable number privacy.
Match Direction	Matching direction options: from right to left and from left to right.
Start Position	Open number privacy after the start of the hidden location.
Hide Digits	Turn on number privacy to hide the number of digits.

Allow IP Call	If this is enabled, you can dial out with an IP address.
P2P IP Prefix	Prefix a point-to-point IP call
Caller Name Priority	Change the caller ID display priority.
Emergency Call Number	
Search path	Select the search path.
LDAP Search	Select LDAP for search.
Emergency Call Number	Configure the Emergency Call Number. You can dial the emergency call number even if the keypad is locked.
Restrict Active URI Source IP	Set the device to accept the Active URI command from the specific IP address. Refer to this **link** for details.
Push XML Server	Configure the Push XML Server; having received a request, the phone will determine whether or not to display the corresponding content on the phone which is sent by the specified server.
Enable Pre-Dial	If you disable this feature, the audio channel will be opened automatically by entering the number. If you enable the feature, you enter the number without opening the audio channel.
Enable Multi Line	If enabled, up to 10 simultaneous calls can exist on the phone, and if disabled, up to 2 simultaneous calls can exist on the phone.
Line Display Format	Custom line format：SIPn/SIPn：xxx/xxx@SIPn
Contact As White List Type	NONE/BOTH/DND White List/FWD White List
Block XML When Call	Disable XML push on call.
SIP notify	If this is enabled, the phone displays information when it has received the relevant notify content.
Tone Settings	
Enable Holding Tone	When this is enabled, a tone is played while the call is held.
Enable Call Waiting Tone	When this is enabled, a tone is played while the call is waiting.
Play Dialing DTMF Tone	Play the DTMF tone on the device when the

	user presses the phone digits at dialing, the default is Enabled.
Play Talking DTMF Tone	Play the DTMF tone on the device when the user presses the phone digits during talking, the default is Enabled.
DND Settings	
DND Option	Select the option to set DND on the line or on the phone.
Enable DND Timer	If the DND Timer is enabled, the DND is automatically turned on from the start to the end.
DND Start Time	Set the DND Start Time.
DND End Time	Set the DND End Time.
Intercom Settings	
Enable Intercom	When Intercom is enabled, the device will accept the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after a specific delay.
Enable Intercom Mute	Enable the mute mode during the intercom call.
Enable Intercom Tone	If the incoming call is an intercom call, the phone plays the intercom tone.
Enable Intercom Barge	Enable Intercom Barge by selecting it; the phone auto answers the intercom call during a call. If the current call is an intercom call, the phone will reject the other intercom call.
Response Code Settings	
DND Response Code	Set the SIP response code on call rejection in DND.
Busy Response Code	Set the SIP response code on line busy.
Reject Response Code	Set the SIP response code on call rejection.
Password Dial Settings	
Enable Password Dial	Enable Password Dial by selecting it. When the number entered begins with the password prefix, the N numbers following the password prefix will be hidden as *, N stands for the value that you enter in the Password Length field. For example: If you set the password prefix to 3 and the Password Length to 2 and then enter

	number 34567, 3**67 will be displayed on the phone.
Encryption Number Length	Configure the Encryption Number length.
Password Dial Prefix	Configure the prefix of the called number password.
Power LED	
Common	Standby power LED state: off when off, open is always bright red. Off by default.
SMS/MWI	The power LED status indicates that there is an unread short message/voice message, including off/on/slow flash/quick flash, the default is slow flash.
Missed	The power LED status indicates that there is a missed call, including off/on/slow flash/quick flash, the default is slow flash.
Talk/Dial	The power LED statuses in the talk/dial state: off is off, on is always red bright, the default is off.
Ringing	The power LED status indicates that there is an incoming call, including off/on/slow flash/quick flash, the default is flash.
Mute	The power LED status indicates the mute mode, including off/on/slow flash/quick flash, off by default.
Hold/Held	The power LED state, including off/on/slow flash/quick flash, is turned off by default when left/retained.

11.18 Phone Settings >> Media Settings

Change voice settings.

Table 13 - Voice Settings

Parameter	Description
Codecs Settings	Enable/disable voice encoding: G.711A/U, G.722, G.729, G.726-16, G726-24,

	G726-32, G.726-40, ILBC, Opus
Audio Settings	
Handset Volume	Set the Handset volume, the value must be 1~9.
Default Ring Type	Configure the default ring tones. If no special ringtone is set for the phone number, the default ringtone will be used.
Speakerphone Volume	Set the hands-free volume to 1-9.
Headset Ring Volume	Set the headset ring tone volume to 1~9.
Headset Volume	Set the headset volume to 1~9.
Speakerphone Ring Volume	Set the hands-free ring tone volume to 1~9.
G.723.1 Bit Rate	5.3kb/s or 6.3kb/s is available.
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
AMR Payload Type	Set the AMR load type, the range is 96~127.
Headset Mic Gain	Set the headset radio volume gain to fit different models of earphones.
Opus Payload Type	Set the Opus load type, the range is 96~127.
OPUS Sample Rate	Set the Opus sampling rate, including opus-nb (8kHz) and opus-wb (16kHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length.
Enable MWI Tone	When there is a new voice message, the phone plays a special dial tone.
Enable VAD	Enable the voice activity detection.
Onhook Time	Configure a minimum response time, which defaults to 200 ms.
EHS Type	The EHS headset is available after enabling.
RTP Control Protocol (RTCP) Settings	
CNAME user	Set the CNAME user.
CNAME host	Set the CNAME host.
RTP Settings	
RTP keep alive	Hold the call and send the packet after 30 s.
Alert Info Ring Settings	
Value	Set a value to specify the ring type.
Ring Type	Type1-Type9

11.19 Phone Settings >> MCAST

Using the multicast function, you can simply and conveniently send an announcement to each member of the multicast and send the multicast RTP stream to the preconfigured multicast address by setting the multicast key on the phone. Listen to and play the RTP stream sent from the multicast address by configuring the listening multicast address on the phone.

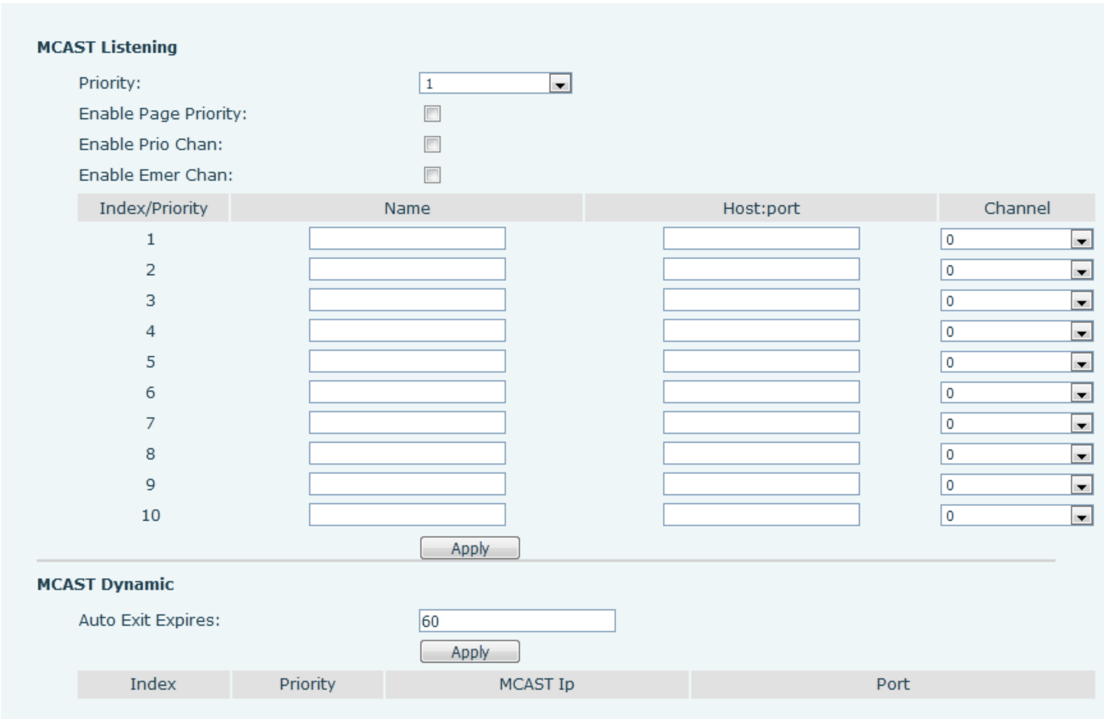


Figure 14 – MCAST

Table 14 - Multicast Parameters

Parameters	**Description**
Normal Call Priority	Define the priority of the active call, 1 is the highest priority, 10 is the lowest.
Enable Page Priority	The voice call in progress shall take precedence over all incoming paging calls.
Name	Listened multicast server name.
Host: port	Listened multicast server's multicast IP address and port.

11.20 Phone Settings >> Action

Action URL

Action URLs are used for IPPBX systems to submit phone events.

11.21 Phone Settings >> Time/Date

The user can configure the phone time settings on this page.

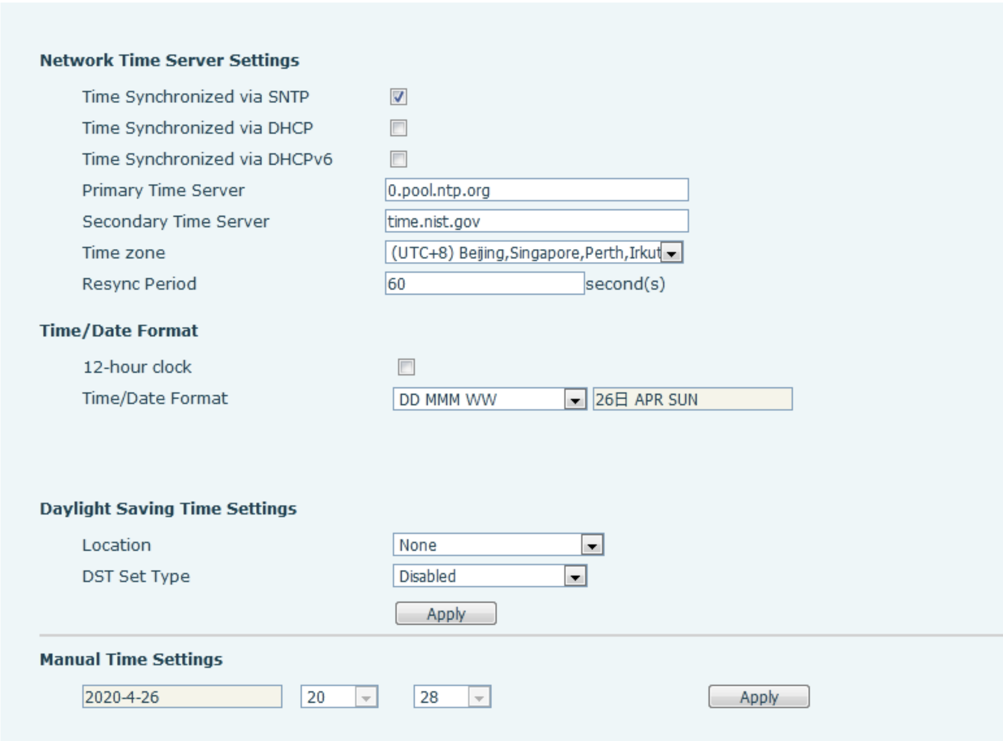


Figure 24 - Time/Date

Table 157 - Time&Date Settings

Parameters	**Description**
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP.
Time Synchronized via DHCP	Enable time-sync through DHCP.
Primary Time Server	Set the primary time server address.
Secondary Time Server	Set the secondary time server address if the primary server is unreachable; the device will try to connect to the secondary time server to get time synchronization.
Time Zone	Select the time zone.

Resync Period	Time of re-synchronization with time server
12-Hour Clock	Set the time display in the 12-hour mode.
Date Format	Select the time/date display format.
Daylight Saving Time Settings	
Location	Choose your Location; the phone will set the daylight saving time automatically based on the local conditions.
DST Set Type	Choose the DST Set Type; if Manual, you need to set the start time and end time.
Fixed Type	The daylight saving time rules are based on specific dates or relative rule dates for conversion. The display is in the read-only mode in the automatic mode.
Offset	Offset minutes when DST started
Month Start	DST start month
Week Start	DST start week
Weekday Start	DST start weekday
Hour Start	DST start hour
Minute Start	DST start minute
Month End	DST end month
Week End	DST end week
Weekday End	DST end weekday
Hour End	DST end hour
Minute End	DST end minute
Manual Time Settings	You can set your time manually.

11.22 Phone Settings >> Tone

This page allows you to configure a phone prompt.

You can either select the country area or customize the area. If the area is selected, the following information is brought up directly. If you choose to customize the area, you can modify the button tone, call back tone and other information.

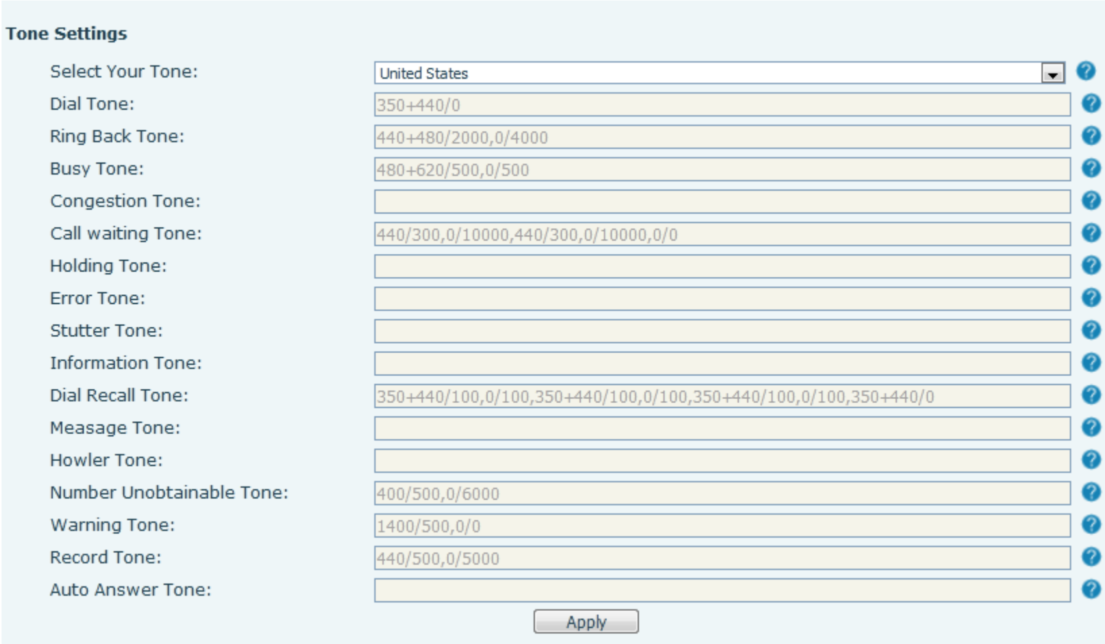


Figure 25 - Tone Setting Web Page

11.23 Phonebook >> Call List

■ Restricted Incoming Calls:

It is like a blacklist. Add the number to the blacklist and the user will no longer receive calls from the stored number until the user removes it from the list.

The user can add specific numbers/prefixes to the blacklist to block calls with the specified numbers/numbers with the specified prefix.

■ Restricted Outgoing Calls:

The user can add a number that restricts outgoing calls and cannot be called until the number is removed from the list.

11.24 Phonebook >> Web Dial

Use the web pages for call, reply and hang-up operations.

11.25 Call Logs

The phone can store up to 600 call records, the user can browse through the complete call record on this page. The call records can be sorted by time, call number, contact name or line and the call log can be screened by a call record type (incoming call, outgoing call, missed call, forward call).

The user can also save the number in the call log to his/her phonebook or add it to the blacklist/whitelist.

The users can also dial the web page by clicking the number in the call log.
The user can delete the call records by pressing the delete button or select all the call records by exporting.

11.26 Function Key >> Function Key

Table 16 - Function Key Configuration

Parameters	Description
Memory Key	**Speed Dial**: You can call the set number directly. This feature is convenient for dialing frequently used numbers. **Intercom**: This feature allows the operator or the secretary to connect the phone quickly; it is widely used in office environments.
DTMF	It allows the user to dial or edit a dial number easily.
Multicast	Configure the multicast address and audio codec. Press the key to initiate the multicast.
Action URL	The user can use a specific URL to make basic calls to the phone.

11.27 Function Key >> Speed Dial List

The user can configure the number buttons "0~9" as speed dial keys. Having completed the configuration as shown in the figure below, you can press the configured shortcut key for the phone to quickly dial the configuration number. Thus, you can make calls more quickly and conveniently, eliminating the need to dial and check the number.

11.28 Security >> Web Filter

The user can set up a configuration management phone that grants only machines with a certain network segment IP access.

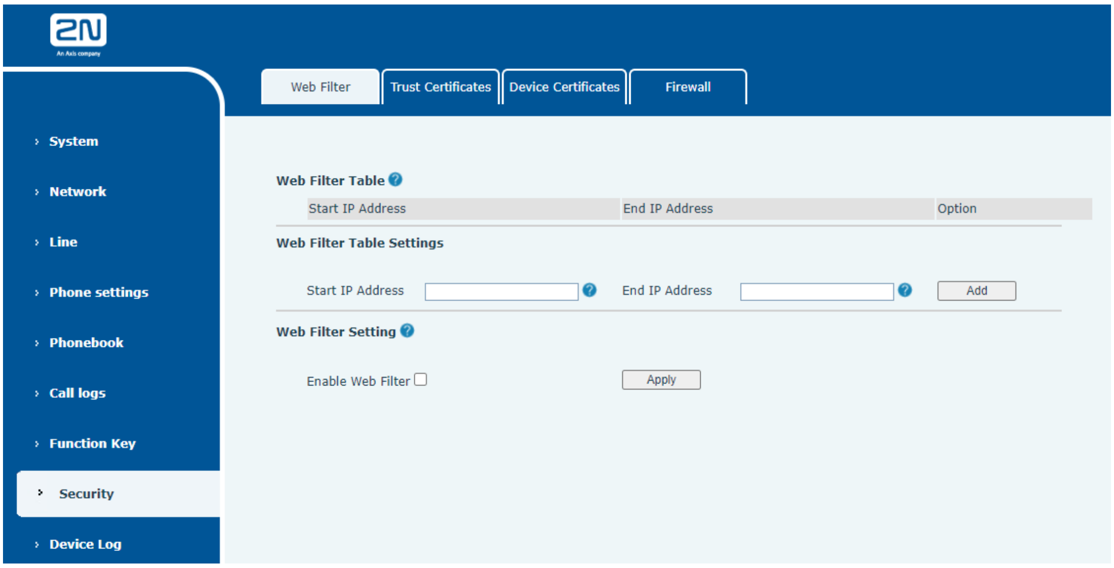


Figure 26 - Web Filter Settings

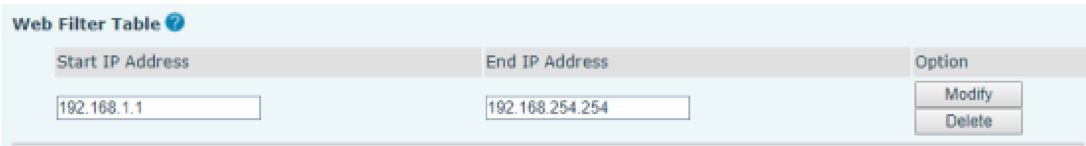


Figure 27 - Web Filter Table

Add and remove IP segments that are accessible. Configure the starting IP address within the start IP, end the IP address within the end IP and click [**Add**] for effect. A large network segment can be set or divided into several network segments for addition. When deleting, select the initial IP of the network segment to be deleted from the drop-down menu and then click [**Delete**] to take effect.

Enable web page filtering: configure enable/disable the web page access filtering. Click the "Apply" button to take effect.

Note: If the device you are accessing is in the same network segment as the phone, please do not configure the filter segment of the web page to be outside your own network segment, otherwise you will not be able to log in to the web page.

11.29 Security >> Trust Certificates

Set whether to open the license certificate and general name validation, select the certificate module.

You can upload and delete uploaded certificates.

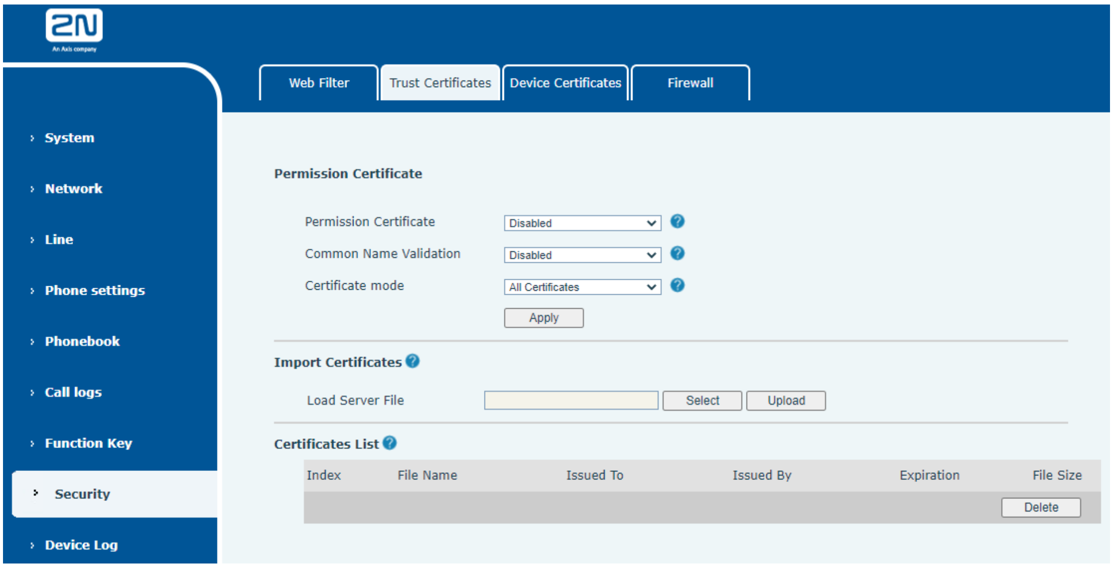


Figure 28 - Certificate Settings

11.30 Security >> Device Certificates

Select the device certificate as the default and custom certificate.

You can upload and delete uploaded certificates.

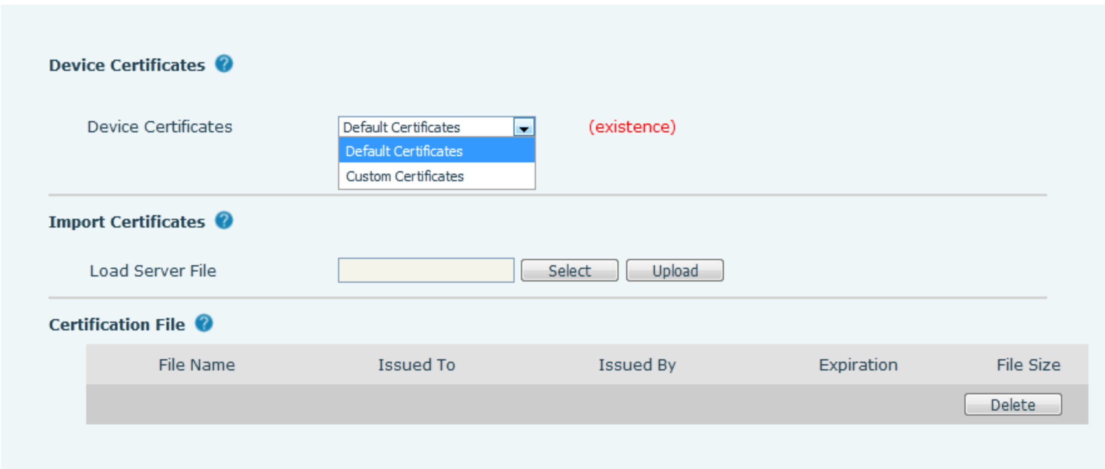


Figure 29 - Device Certificate Setting

11.31 Security >> Firewall

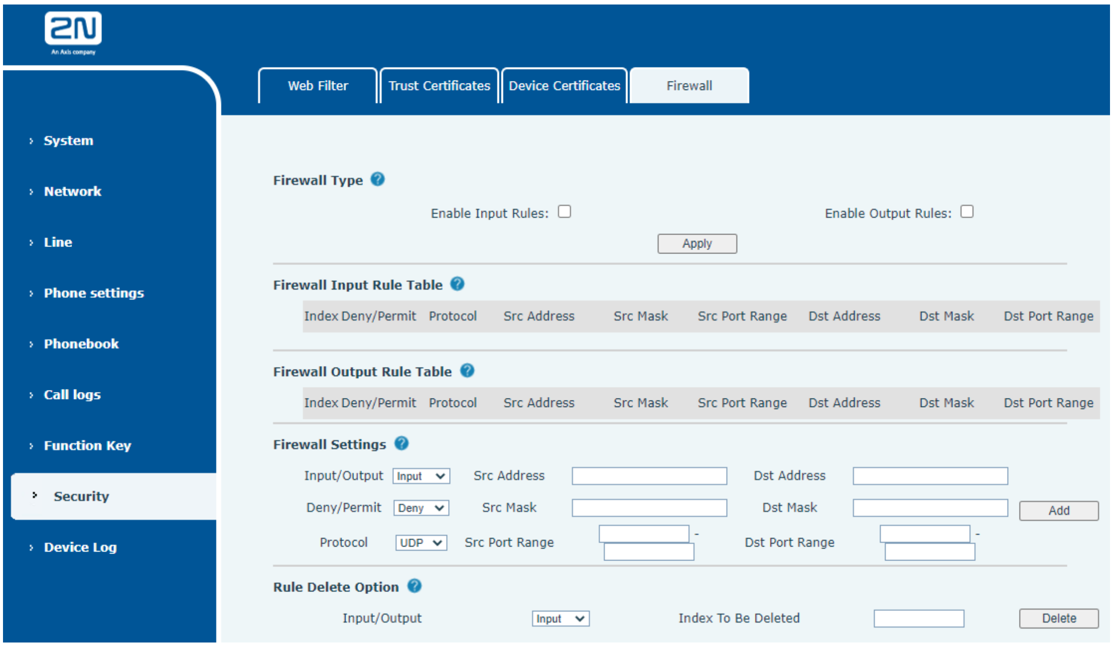


Figure 30 - Network Firewall Settings

On this page you can enable the input and output firewall, set the firewall input and output rules to prevent some malicious network access and/or restrict internal users' access to some external network resources, thus enhancing security.
The firewall rule set is a simple firewall module. This feature supports two types of rules: input rules and output rules. Each rule is assigned an ordinal number, allowing for up to 10 for each rule.
Considering the complexity of firewall settings, the following is an example to illustrate:

Table 17 - Network Firewall

Parameter	Description
Enable Input Rules	Indicates that the input rule application is enabled.
Enable Output Rules	Indicates that the output rule application is enabled.
Input/Output	Select whether the currently added rule is an input or output rule.
Deny/Permit	Select whether the current rule configuration is disabled or allowed.
Protocol	There are four types of filtering protocols: TCP | UDP | ICMP | IP.
Src Port Range	Filter port range

Src Address	The source address can be host address, network address or all addresses 0.0.0.0. It can also be a network address similar to *.*.*.0, such as 192.168.1.0.
Dst Address	The destination address can be either a specific IP address or the full address 0.0.0.0. It can also be a network address similar to *.*.*.0, such as 192.168.1.0.
Src Mask	Source address mask. When it is configured as 255.255.255.255, it means that the host is specific. When it is set to 255.255.255.0, it means that a network segment is filtered.
Dst Mask	Destination address mask. When it is configured as 255.255.255.255, it means a specific host. When it is set to 255.255.255.0, it means that a network segment is filtered.

After setting, click [**Add**] and a new item will be added to the firewall input rules.

Then select and click the [Apply] button.
In this way, when the device is running: ping 192.168.1.118, the packet cannot be sent to 192.168.1.118 because the output rule is forbidden. However, another IP of the ping 192.168.1.0 network segment can still receive the response packet from the destination host normally.

Select the list you want to delete and click [**Delete**] to delete the selected list.

11.32 Device Log >> Device Log

You can grab the device log and having encountered an abnormal trouble, you can send the log to the technician to locate the problem. See 12.5 Get Log Information.

12 Troubleshooting

When the phone is not in normal use, the user can try the following methods to restore the phone's normal operation or collect relevant information and send a problem report to the technical support mailbox.

12.1 Get Device System Information

Get information by pressing the [**Menu**] >> [**Status**] option in the phone. The following information will be provided:
Network information
Equipment information (model, software and hardware version), etc.

12.2 Reboot Device

Reboot the device from the soft-menu, [**Menu**] >> [**Basic**] >> [**Reboot System**], and confirm the action by [**OK**]. Or, simply remove the power supply and restore it again.

12.3 Reset Device to Factory Default

Reset Device to Factory Default will erase all the user configurations, preferences, databases and profiles on the device and restore the device back to the factory default state.

To perform a factory default reset, press [**Menu**] >> [**Advanced**] and then input the password to enter the interface. Then choose [**Factory Reset**] and press [**Enter**] and confirm the action by [**OK**]. The device will be rebooted into a clean factory default state.

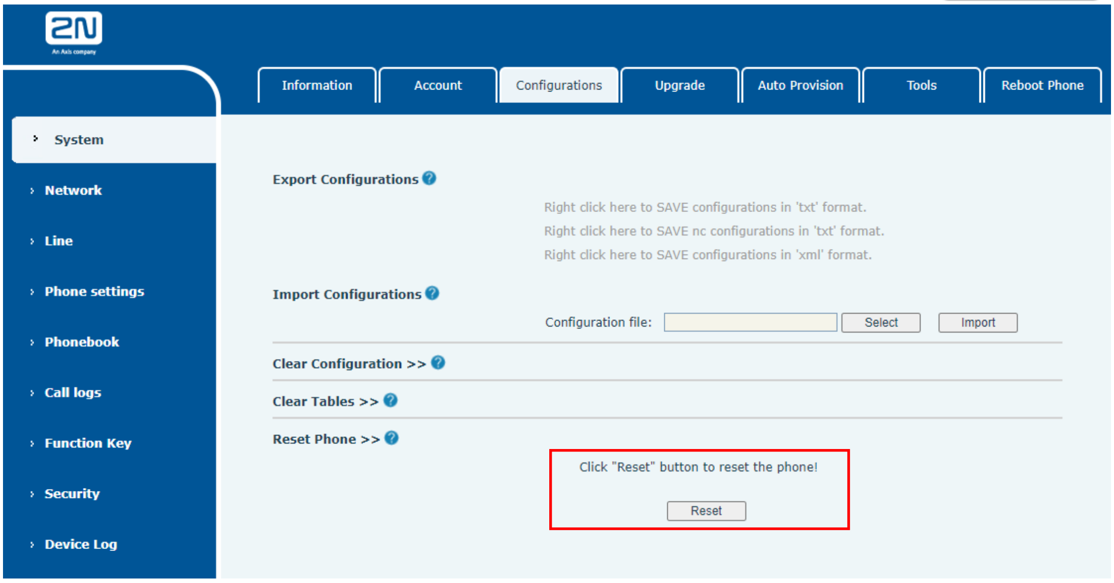


Figure 31 - Reset

12.4 Network Packet Capture

Sometimes it is helpful to dump the network packets of the device for issue identification. To get the packets dump of the device, log in to the device web portal, open the [**System**] >> [**Tools**] page and click [**Start**] in the “Network Packets Capture” section. A pop-up message will prompt you to save the capture file. Then perform such relevant operations as activate/deactivate the line or make phone calls and click [**Stop**] on the web page when the operation is finished. The network packets of the device during the period have been dumped to the saved file.

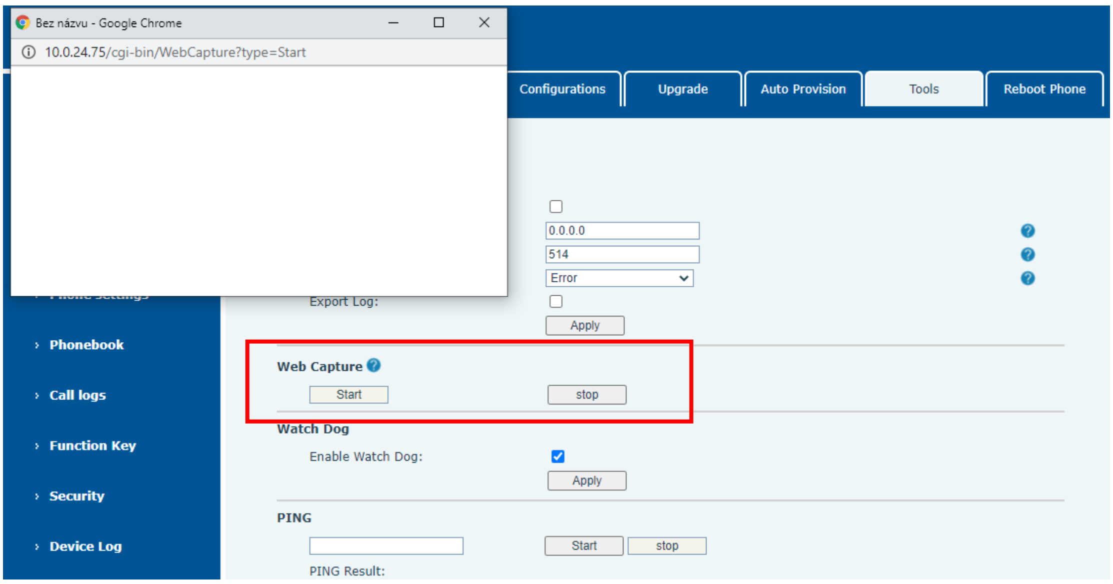


Figure 32 - Web Capture

You can examine the packets with a packet analyzer or send it to the technical support mailbox.

12.5 Get Log Information

Log information is helpful when encountering an exceptional problem. In order to get the phone log information, log in to the phone web page, open the [**Device log**] page and click the [**Start**] button. Follow the steps of the problem until the problem appears and then click the [**End**] button, [**Save**] for local analysis or send the log to the technician to locate the problem.

12.6 Common Trouble Cases

Table 88 - Trouble Cases

Trouble Case	Solution
Device cannot boot up	1. The device is powered by an external power supply via a power adapter or PoE switch. Please use a standard power adapter or PoE switch meeting the specification requirements and check if the device is properly connected to the power source.
Device cannot register to a service provider	1. Please check if the device is properly connected to the network. The network Ethernet cable should be connected to the [Network] port, NOT to the [PC] port. 2. Please check if the device has an IP address. Check the system information: if the IP displays "Negotiating…", the device does not have an IP address. Please check if the network configuration is correct. 3. If the network connection is fine, please check again your line configurations. If all the configurations are correct, please contact your service provider to get support or follow the instructions in "13.5 Network Packet Capture" to get the network packet capture of registration process and send it to the technical support for analysis.
No Audio or Poor Audio in Handset	1. Please check if the handset is connected to the correct handset () port, NOT the headphone () port. 2. The network bandwidth and delay may be not suitable for audio

	calls at the moment.
Poor Audio or Low Volume in Headphone	1. There are two headphone wire sequences in the market. Please use the headphone provided by 2N or consult 2N for the wire sequence if you wish to use a third-party headphone. 2. The network bandwidth and delay may be not suitable for audio calls at the moment.
Audio is chopping at far-end in Hands-free speaker mode	This is usually due to a loud volume feedback from the speaker to the microphone. Please lower down the speaker volume a little bit, the chopping will be gone.