H3 White_Hotel Phone
User Manual_V1.0

Document VER	Firmware VER	Explanation	Time
V1.0	2.0.2.2751	Initial issue	20161114

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1 Safety Instruction

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

- Please use the external power supply that is included in the package. Other power supply may cause damage to the phone, 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 power cord or plug is impaired, do not use it, it may cause fire or electric shock.
- Do not drop, knock or shake the phone. Rough handling can break internal circuit boards.
- This phone is design for indoor use. Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposure the phone to high temperature or below 0℃ or high humidity.
- Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling of the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock and breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

2 Overview

H3 is the newest series of phones designed for hotels. Its stylish, contemporary appearance, excellent voice quality and powerful functionality, along with matching integrated communications platforms can replace traditional phones and can become a new generation of intelligent terminal equipment.The H-Series hotel IP phone will look great in most hotel rooms and will support most application requirements.In addition, it has excellent call quality.

The H3 accomplished powerful telephony features by combining the communications platform and features such as call transfer, hotline,voice mail, call hold and more. The H3 IP phones support 6 programmable keys.They can be defined according to the hotel's needs.For example, they could be programmed with an equipment service hotline (housekeeping, ticketing, switchboard, food and beverage, etc.) or hotel special features (alarm clock, voice mail, etc.).In addition, it has a USB port to charge your mobile phone.

In order to help some users who are interested to read every detail of the product, this user manual is provided as a user’s reference guide. Still, the document might not be up to date with the newly release software, so please kindly download updated the latest user manual from website, or contact with support if you have any question using H3.

3 Installation

3.1 Use PoE or external power adapter

H3, called as ‘the device’ hereafter, supports two power supply modes, power supply from external power adapter and supports 802.3af Class 2 Power over Ethernet (PoE) complied switch.

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

For users who do not have PoE equipment, the 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 used in priority and will switch to PoE power supply at power failure on the power adapter.

Please use the power adapter supplied and the PoE switch met the specifications to ensure the device worked properly.

3.2 Connection methods

Please connect power adapter, network, PC, and handset to the corresponding ports as described in below picture.

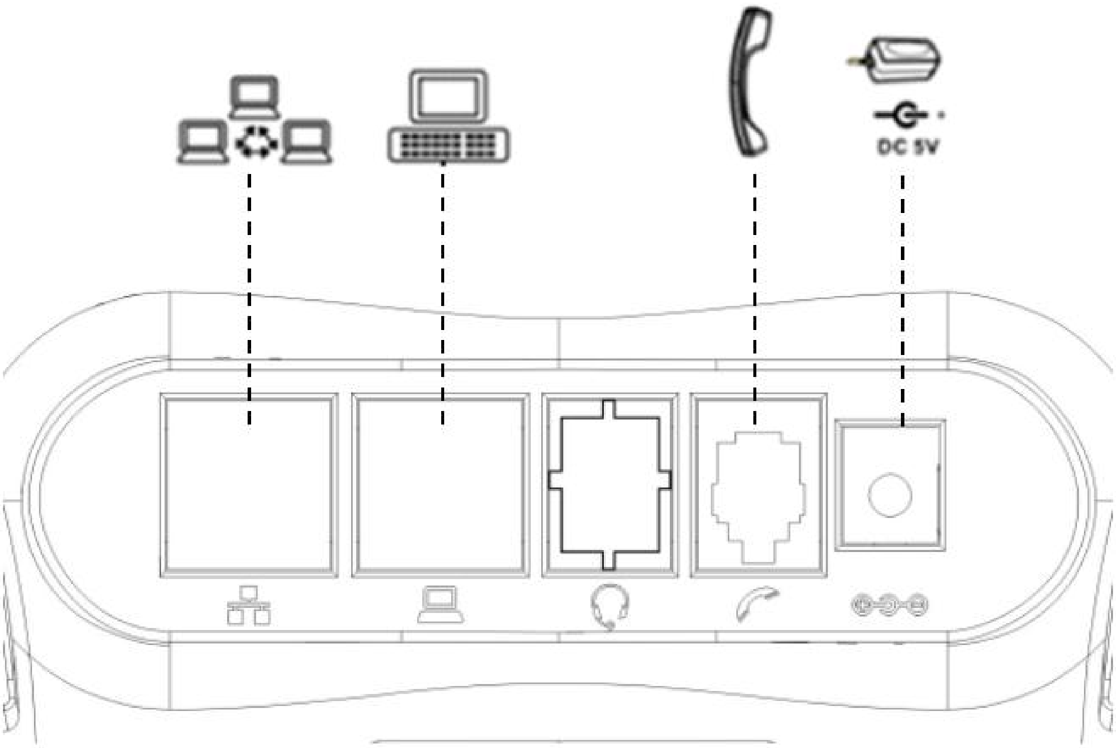

Figure 1 - Connecting to the device

4 Introduction to the Phone User Interface

4.1 Keypad

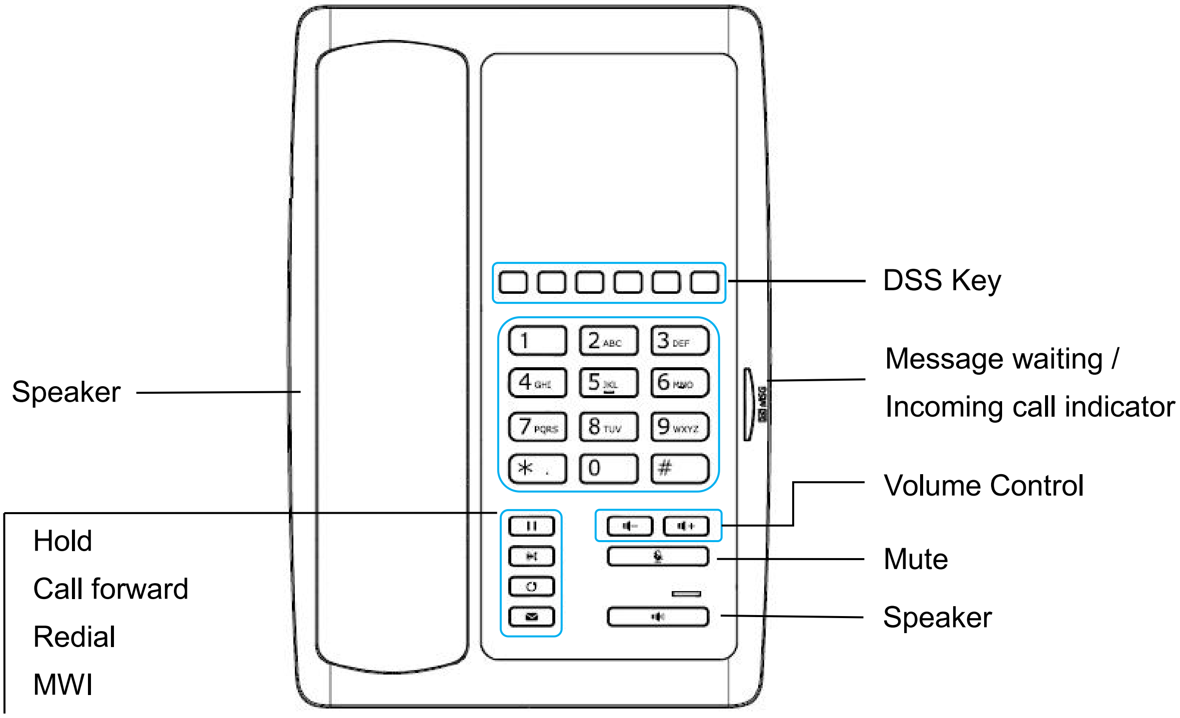

Figure 2 - Keypad

The above picture shows the keypad layout of the device. Each key provides its own specific function. User should refer to the illustration in this section about the usage of each key and the description in this document about each function.

Message waiting / incoming call indicator - The light flashes when the telephone rings for incoming calls. When the telephone system supports Messages Waiting Indication (MWI) function and there are some voice messages, the light will also flash.

- Standard telephone keys - The 12 standard telephone keys provide the same function as standard telephones
- Redial - By pressing 'Redial' button, user can redial the last dialed number.
- MWI - When have a voice message, press "information" key, you can consult the message.
- Hands-free - By pressing this button once, user can turn on the audio channel of hands-free
- Microphone Mute - User can mute the microphone with this button during talking mode.
- Volume -/+ - In standby, ringing, ring configuration screen, user can press the 2 buttons to lower/increase the ringtone volume, in talking and audio volume adjustment screen, user can press these buttons to lower/increase the audio volume.

5 Phone Settings

In order to get the device ready for making and receiving phone calls, the device must be configured with correct network configurations and the line must be configured with an SIP Service.

The SIP must be configured properly to be able to provide telephony service.

5.1 Getting IP address

DHCP is the default setting in network, and telephone will get the IP address from DHCP server(Router) after the line connected.

There are three common IP configuration modes.

- Dynamic Host Configuration Protocol (DHCP) – This is the automatic configuration mode by getting network configurations from a DHCP server. Users need not to configure any parameters manually. All configuration parameters will be getting from DHCP server and applied to the device. This is recommended for most users.
- Static IP Configuration – This option allows user to configure each IP parameters manually, including IP Address, Subnet Mask, Default Gateway, and DNS servers. This is usually used in an office environment or by power users.
- PPPoE – This option is often used by users who connect the device to a broadband modem or router. To establish a PPPoE connection, user should configure username and password provided by the service provider.

5.2 Checking IP address

Pick up the handset or press hands-free key, please input “# * 111” button, then you can hear the IP address voice information.

5.3 How to enter into web setting interface

Set the telephone through web interface.

- Connect the telephone and PC in the same LAN.
- Run the IE in the PC, and input the telephone IP in address bar.
- Input the user name and password, both of them are admin.
- Click Logon button to enter into the web setting interface.

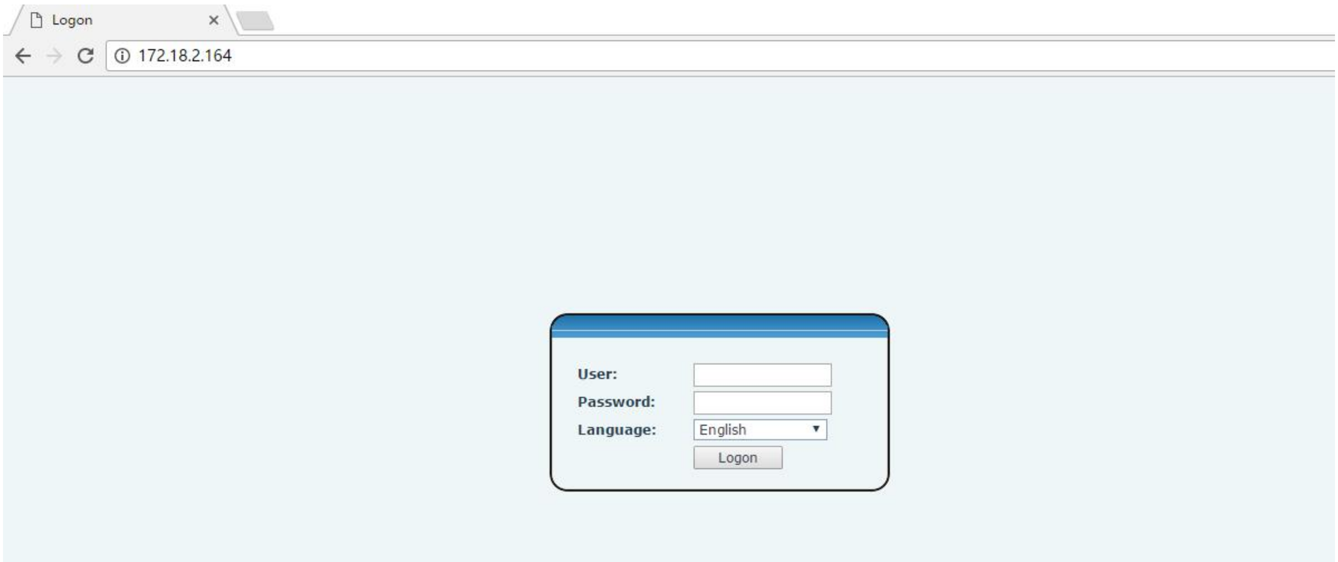

Figure 3 - The Web Login page

5.4 SIP setting

Enter into the web setting interface, select **Line->SIP,** and fill in the items below.

- Server address
- Account name
- Phone number
- Password

Click the “Apply” button to save the config, you can dial out after the register status is “Registered” with red color.

5.5 Memory key setting

Enter into the web setting interface, select **Function Key->Function Key.**

Select the function and fill in the number in the “value” items.

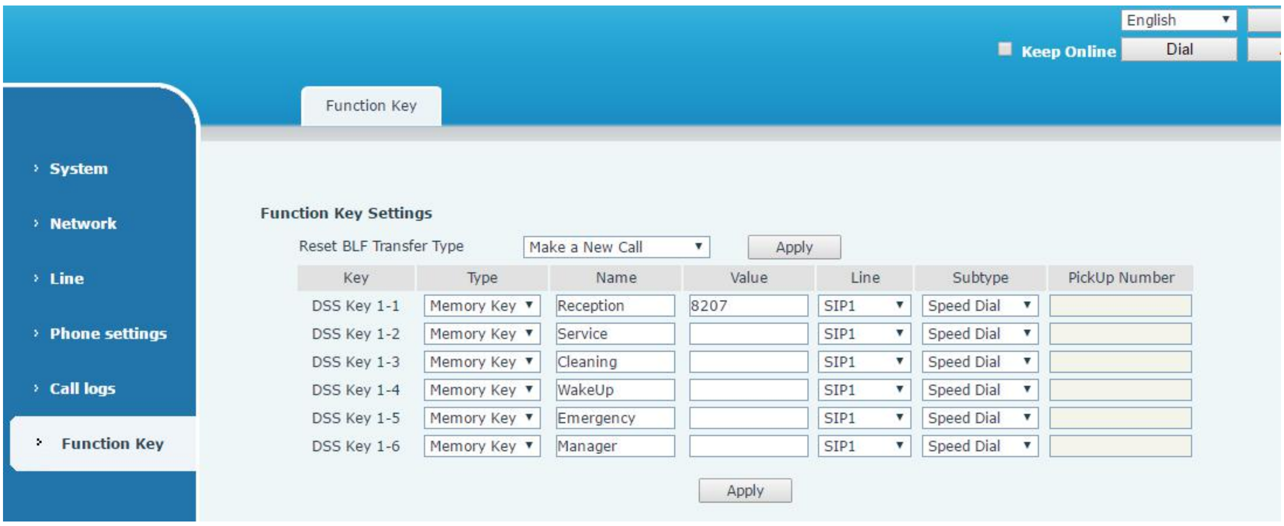

Figure 4 - Memory Key Setting

6 Basic Operation

6.1 Making call

There are two ways to make a call, using dial pad or memory button.

- Lift the handset or press **hands-free** key.
- Dial the number on the dial pad or press memory key, end with **#** as default.
- End a call, hang up handset.

6.2 Answering call

When your telephone rings and the light flashes.

- Lift the handset or **hands-free** key and start to talk.
- End a call, hang up handset.

6.3 Holding call

- While on a call, press the **Hold** key the call will be held.
- To retrieve a held call, press the **Hold** key again.

6.4 Redialing

Press redial to dial the last number you dialed.

- Lift handset or **hands-free** key.
- Press **Redial** key to dial the last number you dialed.

7 Advanced Operation

7.1 Call transfer

- Blind transfer

During a call, you want to transfer the call to another one without talking.

- Press Transfer key, get the second dial tone , and the first call is held automatically。
- Dial the number which you want to transfer to, and then press # or Transfer button。
- You will hear the busy tone, the call have been transferred successfully

- Attended transfer

During a call, you want to transfer the call to another one after talking.

- Press **Transfer** key, get the second dial tone, and the first call is held automatically.
- Dial the number you want to transfer to, press **Redial** key, the second call connected
- Press **Transfer** key again, you will hear the busy tone, the call have been transferred successfully.

7.2 Messages waiting

When the LED indicator flashes and there is no incoming call, you need to dial the feature access code for message retrieving. Once the messages have been retrieved, the lights up will stop. You can save your messages waiting feature access code on a memory button, when you listen voice messages usually.

8 Web Portal

8.1 Web Portal Authentication

User can log in onto the device web portal to manage the device or user's profile. User must provide correct username and password to be able to log in.

8.2 SYSTEM / Information

User can get the system information of the device in this page including.

- Model
- Hardware Version
- Software Version
- Uptime
- Last uptime
- MEMInfo

And also summarization of network status,

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

Besides, summarization of SIP account status,

- SIP User
- SIP account status (Registered / Unapplied / Trying / Timeout)

8.3 SYSTEM / Account

User may change his/her web authentication password in this page.
For users with Administrators privilege, the user can also manage user accounts by adding or deleting user account and assign privilege and password to new account.
There are two types of user privilege, Administrators and Users. If a user account is created as Users privilege, this account will have limited accessibility to the device and cannot change some device settings. The user account can be used to operate the device or access the device web portal by login to the device or its web. User should log in to device web portal with his/her username and web password.
NOTICE! The device is shipped with a default Administrators user account. The username and password for the default account is 'admin' which has been printed on the brand and model label at the bottom side of the device.

8.4 SYSTEM / Configurations

Users with administrators privilege can export or import the device configuration in this page and reset the device to factory default.

8.5 SYSTEM / Upgrade

The device supports online upgrade by periodically checking the software release version on the cloud server. Meanwhile, user can download the software and upgrade the device manually when there is trouble for the device to connect to the cloud server.

8.6 SYSTEM / Auto Provision

The Auto Provision settings help IT manager or service provider to easily deploy and manage the devices in mass volume.

8.7 SYSTEM / Tools

Tools provided in this page help users to identify issues at trouble shooting. Please refer to **10 Trouble Shooting** for more detail.

8.8 NETWORK / Basic

User can configure the network connection type and parameters in this page.

8.9 NETWORK / Advanced

The network advanced settings is often configured by IT manager to enhance the quality of service of the device.

8.10 NETWORK / VPN

User may configure a VPN connection in this page. Please refer to **9.1 VPN** for more detail.

8.11 LINES / SIP

The SIP service of the line is configured in this page.

Table 1 - SIP Settings for Lines on Web

Parameters	Description
Basic Settings	
Line Status	Display the current line status at page loading. To get the up to date line status, user has to refresh the page manually.
Username	Enter the username of the service account.
Display Name	Enter the display name to be sent in a call request.
Authentication Name	Enter the authentication name of the service account
Authentication Password	Enter the authentication password of the service account
SIP Proxy Server Address	Enter the IP or FQDN address of the SIP proxy server
SIP Proxy Server Port	Enter the SIP proxy server port, default is 5060
Outbound Proxy Address	Enter the IP or FQDN address of outbound proxy server provided by the service provider
Outbound Proxy Port	Enter the outbound proxy port, default is 5060
Realm	Enter the SIP domain if requested by the service provider
Activate	Whether the service of the line should be activated
Codec Settings	Set the priority and availability of the codecs by adding or remove them from the list.
Advanced Settings	
Call Forward Unconditional	Enable unconditional call forward, all incoming calls will be forwarded to the number specified in the next field
Call Forward Number for Unconditional	Set the number of unconditional call forward
Call Forward on Busy	Enable call forward on busy, when 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 of call forward on busy
Call Forward on No Answer	Enable call forward on no answer, when an incoming call is not answered within the configured delay time, the call will be forwarded to the number specified in the next field
Call Forward Number for No Answer	Set the number of call forward on no answer
Call Forward Delay for No Answer	Set the delay time of not answered call before being forwarded
Enable Hotline	Enable hotline configuration, the device will dial to the specific number immediately at audio channel opened by off-hook handset or turn on hands-free or headset
Hotline Number	Set the hotline dialing number

Hotline Delay	Set the delay for hotline before the system automatically dialed it
Enable Auto Answering	Enable auto-answering, the incoming calls will be answered automatically after the delay time
Auto Answering Delay	Set the delay for incoming call before the system automatically answered it
Subscribe For Voice Message	Enable the device to subscribe a voice message waiting notification, if enabled, the device will receive notification from the server if there is voice message waiting on the server
Voice Message Number	Set the number for retrieving voice message
Voice Message Subscribe Period	Set the interval of voice message notification subscription
Enable DND	Enable Do-not-disturb, any incoming call to this line will be rejected automatically
Blocking Anonymous Call	Reject any incoming call without presenting caller ID
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response
Anonymous Call Standard	Set the standard to be used for anonymous
Dial Without Registered	Set call out by proxy without registration
User Agent	Set the user agent, the default is Model with Software Version.
Use Quote in Display Name	Whether to add quote in display name
Ring Type	Set the ring tone type for the line
Conference Type	Set the type of call conference, Local=set up call conference by the device itself, maximum supports two remote parties, Server=set up call conference by dialing to a conference room on the server
Server Conference Number	Set the conference room number when conference type is set to be Server
Transfer Timeout	Set the timeout of call transfer process
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Enable Missed Call Log	If enabled, the phone will save missed calls into the call history record.
Response Single Codec	If setting enabled, the device will use single codec in response to an incoming call request
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 feature code to the server by dialing the number specified in each feature code field.
Enable DND	Set the feature code to dial to the server

Disable DND	Set the feature code to dial to the server
Enable Call Forward Unconditional	Set the feature code to dial to the server
Disable Call Forward Unconditional	Set the feature code to dial to the server
Enable Call Forward on Busy	Set the feature code to dial to the server
Disable Call Forward on Busy	Set the feature code to dial to the server
Enable Call Forward on No Answer	Set the feature code to dial to the server
Disable Call Forward on No Answer	Set the feature code to dial to the server
Enable Blocking Anonymous Call	Set the feature code to dial to the server
Disable Blocking Anonymous Call	Set the feature code to dial to the server
Specific Server Type	Set the line to collaborate with specific server type.
Registration Expiration	Set the SIP expiration interval
Use VPN	Set the line to use VPN restrict route
Use STUN	Set the line to use STUN for NAT traversal
Convert URI	Convert digit and alphabet characters to %hh hex code
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'
Transport Protocol	Set the line to use TCP or UDP for SIP transmission
SIP Version	Set the SIP version
Caller ID Header	Set the Caller ID Header
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from the server， it will use the source IP address, not the address in via field.
Enable user=phone	Sets user=phone in SIP messages.
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
Enable BLF List	Enable/Disable BLF List
Enable DNS SRV	Set the line to use DNS SRV which will resolve the FQDN in proxy server into a service list
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT pinhole opened
Keep Alive Interval	Set the keep alive packet transmitting interval
Sync Clock Time	Time Sycn with server
Enable Session Timer	Set the line to enable call ending by session timer refreshment. The call session will be ended if there is not new session timer

	event update received after the timeout period
Session Timeout	Set the session timer timeout period
Enable Rport	Set the line to add rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
Keep Authentication	Keep the authentication parameters from previous authentication
Auto TCP	Using TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes
Enable Feature Sync	Feature Sycn with server
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
BLF Server	The registered server will receive the subscription package from ordinary application of BLF phone. Please enter the BLF server, if the sever does not support subscription package, the registered server and subscription server will be separated.
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
SIP Encryption	Enable SIP encryption such that SIP transmission will be encrypted
SIP Encryption Key	Set the pass phrase for SIP encryption
RTP Encryption	Enable RTP encryption such that RTP transmission will be encrypted
RTP Encryption Key	Set the pass phrase for RTP encryption

8.12 LINES / Dial Peer

This functionality offers you more flexible dial rule, you can refer to the following content to know how to use this dial rule.

Table 2 - Dial Peer Settings for Lines on Web

Parameters	Description
Phone number	There are two types of matching: Full Matching or Prefix Matching. In Full matching, the entire phone number is entered and then mapped per the Dial Peer rules. In prefix matching, only part of the number is entered followed by T. The mapping with then take place whenever these digits are dialed. Prefix mode supports a maximum of 30 digits.
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 Destination address. This is for IP direct.
Port	Set the Signal port, and 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 at the end of the phone number. It is an optional item.
Delete 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.

Examples of different alias application

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

Example 1: Global Substitution

It seems like a shortcut to dial out. When user dial “32”, the dialed number will be replaced of “833333”. But if user dials “322”, the device will still send “322” rather than “8333332”. The replacement rules should be matched globally.

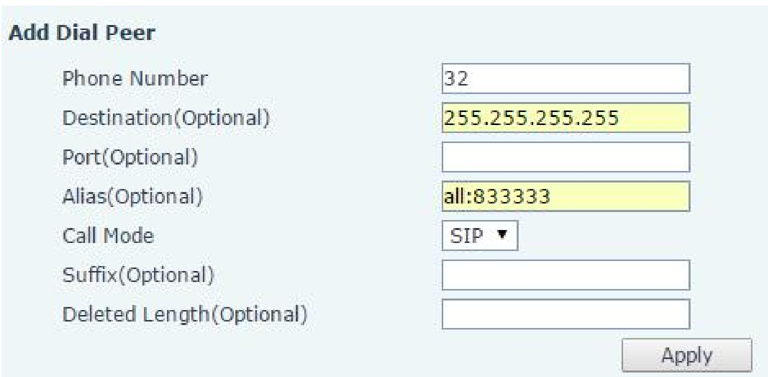

Figure 5 - Global Substitution Configuration

Example 2: Local Substitution

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

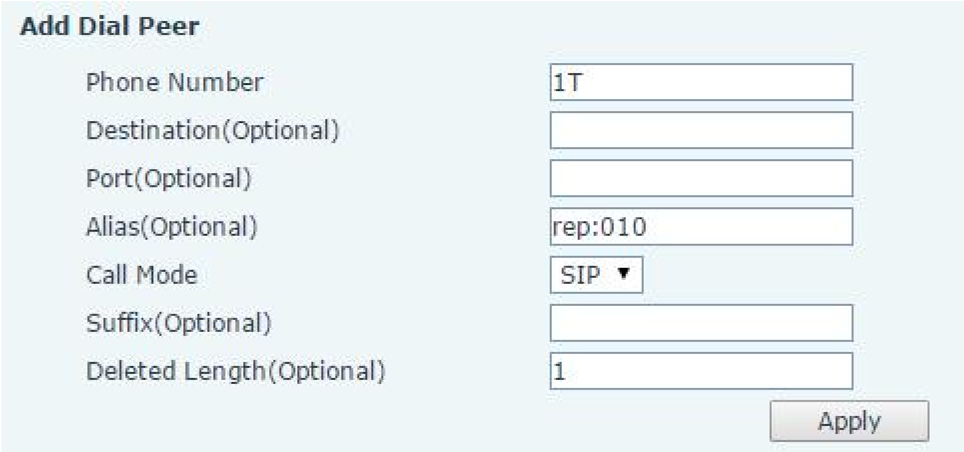

Figure 6 - Local Substitution Configuration

Example 3: Add Prefixes

If the dialed number starts with the fixed prefix number, the phone will send out your dialed phone number adding prefix number automatically.

For example, when users dial “9312”, the device will send out “0079312”.

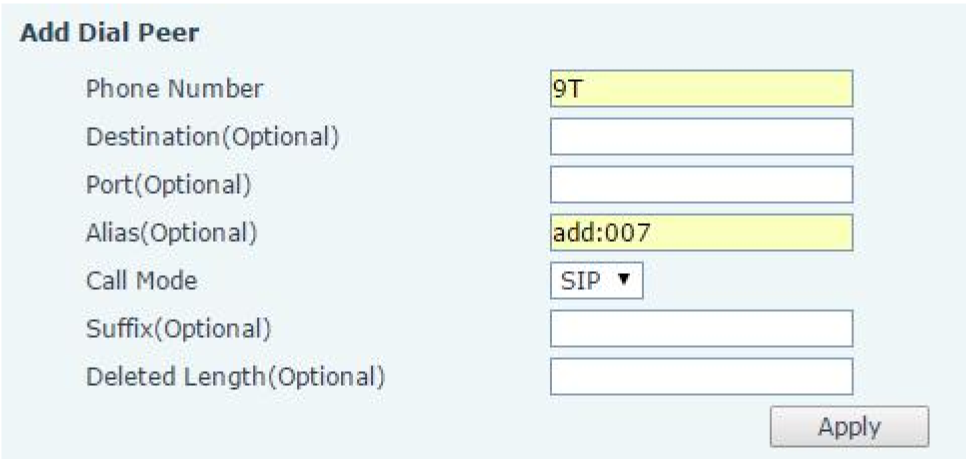

Figure 7 - Add Prefixes Configuration

Example 4: Add Suffixes

If the dialed number ends with the fixed suffix number, the phone will send out your dialed phone number adding suffix number automatically.

For example, when users dial “1383322”, the device will send out “13833220088” .

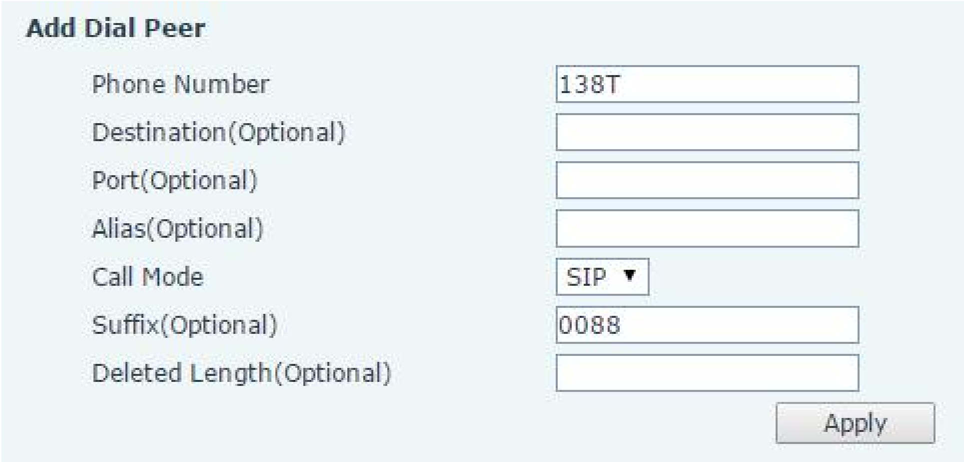

Figure 8 - Add Suffixes Configuration

Example 5: Deletion

If the dialed number ends with the fixed prefix number, the phone will send out your dialed phone number deleting prefix number automatically.

For example, when users dial “98322”, the device will send out “8322” .

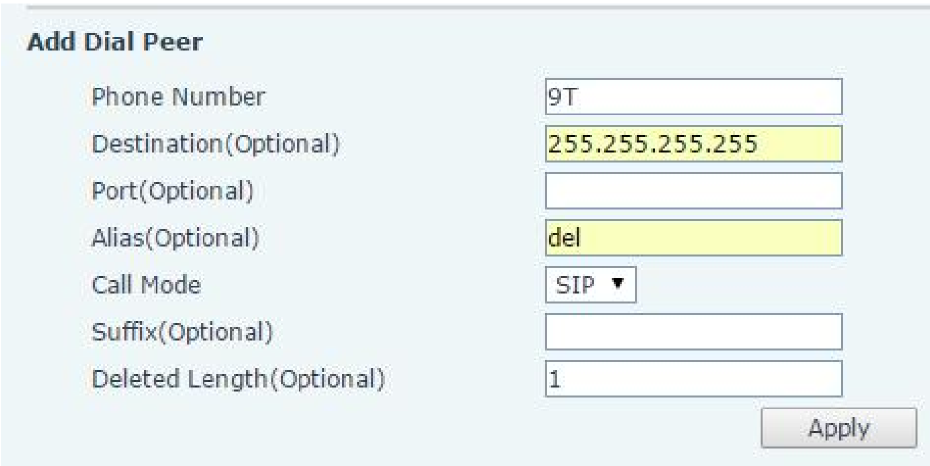

Figure 9 - Deletion Configuration

8.13 LINES / Dial Plan

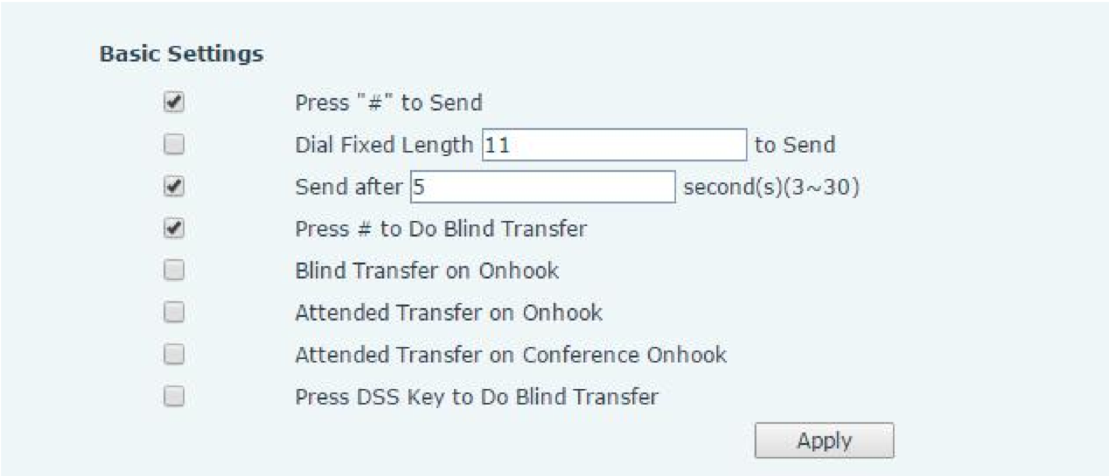

Figure 10 - Dial Plan Configuration

The device supports 8 dialing modes:

- Press # to Send - Dial the desired number, and press # to send it to the server.
- Dial Fixed Length – Configure the fixed length to dial out
- Send after seconds – Number will be sent to the server after the specified time.
- Press # to Do Blind Transfer - Press # after entering the target number for the transfer. The phone will transfer the current call to the third party.
- Blind Transfer on Onhook - Hang up after entering the target number for the transfer. The phone will transfer the current call to the third party.
- Attended Transfer on Onhook - Hang up after the third party answers. The phone will transfer the current call to the third party.
- Attended Transfer on Conference Onhook - Hang up during a 3-way conference call, the other two ways will make a call.
- Press DSS Key to Do Blind Transfer – When user is in the 'XFER' screen, user can fulfill Blind Transfer by pressing DSS Key.

8.14 LINES / Basic Settings

Configure basic settings for lines.

Table 3 - Basic Settings for Lines on Web

Parameters	**Description**
SIP Settings	
Local SIP Port	Set the local SIP port used to send/receive SIP messages.
Registration Failure Retry Interval	Set the retry interval of SIP REGISTRATION when registration failed.

STUN Settings	
Server Address	Set the STUN server address
Server Port	Set the STUN server port, default is 3478
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 sending SIP messages
TLS Certification File	Upload or delete the TLS certification file used for encrypted SIP transmission.

8.15 PHONE / Features

Configure the phone features

Table 4 - Common Phone Feature Settings on Web

Parameters	**Description**
DND Mode	Configure the Phone DND If enable Phone DND, the phone rejects any incoming call, the caller will automatically prompt hang up.
Ban Outgoing	If you select Ban Outgoing to enable it, and you cannot dial out any number.
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. Default enabled.
Enable Call Waiting Tone	Turn off this feature, and you will not hear a ‘beep’ sound in talking mode when there is another incoming call
Hand down Time	Specify Auto hand down time, the phone will hang up and return to the idle automatically after Auto Hand down time at hands-free mode, and play dial tone Auto hand down time at handset mode
Enable Call Completion	Enable Call Completion by selecting it, If the dialed line is busy, the sip server will inspect the dialed line status at intervals. If the dialed line is idle, the server will send notify message to inform the caller whether redial.
Hide DTMF	Configure the hide DTMF mode
Enable Pre-Dial	Disable this feature, user entering number will open audio channel automatically. Enable the feature, user enter the number without opening audio channel.
Enable Silent Mode	Enable silent mode by selecting it, the phone incoming call indicator will blink to remind that there is a incoming call instead of playing ring tone.

Disable Mute for Ring	Disable mute for ring.
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 specific delay.
Enable Intercom Mute	Enable mute mode during the intercom call
Enable Intercom Tone	If the incoming call is 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 intercom call, the phone will reject the second intercom call
Auto Answer By Headset	When this item is checked, the device will auto-answer phone calls by headset if the auto-answer or intercom is enabled.
Ring From Headset	Enable Ring From Handset by selecting it, the phone plays ring tone from handset.
Emergency Call Number	Configure the emergency call number. Despite the keyboard is locked, you can dial the emergency call number.
Enable Password Dial	Enable Password Dial by selecting it, When number entered is beginning with the password prefix, the following N numbers after the password prefix will be hidden as *, N stand for the value which you enter in the Password Length field. For example: you set the password prefix is 3, enter the Password Length is 2, then you enter the number 34567, it will display 3**67 on the phone.
Password Dial Prefix	Configure the prefix of the password call number
Enable Phone DND	Enable phone DND feature.
DND Response Code	Set the SIP response code on call rejection on DND
Busy Response Code	Set the SIP response code on line busy
Reject Response Code	Set the SIP response code on call rejection
Restrict Active URI Source IP	Set the device to accept Active URI command from specific IP address.
Push XML Server	Configure the Push XML Server, when phone receives request, it will determine whether to display corresponding content on the phone which sent by the specified server or not.
Allow IP Call	If enabled, user can dial out with IP address
Enable Multi Line	Enable phone to make calls for 10 lines max, or disable for 2 lines max.
Enable Default Line	If enabled, user can assign default SIP line for dialing out rather than SIP1.
Enable Auto Switch Line	Enable phone to select an available SIP line as default automatically
Play Talking DTMF Tone	Play DTMF tone on the device when user pressed a phone digits during taking, default enabled.

Play Dialing DTMF Tone	Play DTMF tone on the device when user pressed a phone digits at dialing, default enabled.
Caller ID Display Priority	Change caller ID display priority. The default priority is “Phonebook” > “SIP Display Name” > “SIP URI”. User may select one of the options to change the desired caller ID display priority.
Hotline Number	Set the hot line number
Hotline Delay	Set the hot line delay time.

Action URL

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

8.16 PHONE / Audio

Table 5 - Audio Settings on Web

Parameters	Description
First Codec	The first preferential DSP codec:G.711A/U,G.722,G.723,G.729,G.726-32, ILBC,AMR,AMR-WB
Second Codec	The second preferential DSP codec: G.711A/U,G.722,G.723,G.729,G.726-32, ILBC,AMR,AMR-WB,NONE
Third Codec	The third preferential DSP codec: G.711A/U,G.722,G.723,G.729,G.726-32 , ILBC,AMR,AMR-WB,NONE
Fourth Codec	The forth preferential DSP codec: G.711A/U,G.722,G.723,G.729,G.726-32 , ILBC,AMR,AMR-WB,NONE
Fifth Codec	The fifth preferential DSP codec: G.711A/U,G.722,G.723,G.729,G.726-32, ILBC,AMR,AMR-WB,NONE
Sixth Codec	The sixth preferential DSP codec: G.711A/U,G.722,G.723,G.729,G.726-32, ILBC,AMR,AMR-WB,NONE
Onhook Time	Configure the least reflection time of Hand down, the default is 200ms.
Tone Standard	Set the country standard of call progress tones, including dial tone, busy tone, ring-back tone, etc.
Handset Volume	Set the Handset volume, the value must be 1~9

Default Ring Type	Set the default ring type. If the caller ID of an incoming call was not configured with specific ring type, the default ring will be used.
Speakerphone Volume	Set the speakerphone volume, the value must be 1~9
Headset Ring Volume	Set the ring volume in the headset, the value must be 0~9
Headset Volume	Set the Headset volume, the value must be 1~9
Speakerphone Ring Volume	Set the ring volume in the speakerphone, the value must be 0~9
Headset Volume Offset	This is to adjust the base volume of the headset. Please note when set the volume at the maximum level it may create noise and decrease the echo canceller.
Headset Mic Offset	This is to adjust the base volume of the headset Mic.
G.729AB Payload Length	Set G729 Payload Length.
G.723.1 Bit Rate	5.3kb/s or 6.3kb/s is available
G.722 Timestamps	160/20ms or 320/20ms is available
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
Enable VAD	Enable Voice Activity Detection. When enabled, the device will suppress the audio transmission with artificial comfort noise signal to save the bandwidth.
Enable MWI Tone	The phone will play MWI tone when a new MWI Comes

8.17 PHONE / MCAST

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

Table 6 - MCAST Parameters on Web

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.

8.18 PHONE / Time/Date

User can configure the device time settings in this page.

Table 7 - Time/Date Setting Parameters on Web

Parameters	Description
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol
Time Synchronized via DHCP	Enable time-sync through DHCP protocol
Primary Time Server	Set primary time server address
Secondary Time Server	Set secondary time server address, when primary server is not reachable, the device will try to connect to secondary time server to get time synchronization.
Timezone	Select the time zone
Resync Period	Time of re-synchronization with time server
12-Hour Clock	Set the time display in 12-hour mode
Date Format	Select the time/date display format
Daylight Saving Time Settings	
Location	Select the user's time zone specific area
DST Set Type	Select automatic DST according to the preset rules of DST, or the manually input rules
Offset	The DST offset time
Month Start	The DST start month
Week Start	The DST start week
Weekday Start	The DST start weekday
Hour Start	The DST start hour
Minute Start	The DST start minute
Month End	The DST end month
Week End	The DST end week
Weekday End	The DST end weekday
Hour End	The DST end hour
Manual Time Settings	Set time manually

8.19 CALL LOGS

User can browse complete call logs in this page, order the call logs by time, caller ID, contact name, duration, or line, and can also filter the call logs by the call log types, in, out, missed, or all.

User can save a call log into his/her phonebook or add it to the blacklist.

User can also make web call by click on the number of a call log.

8.20 FUNCTION KEY / Function Key

The device provides 6 user-define DSS Keys at most. User may configure/customize each DSS key in this webpage.

Table 8 - DSS Key Setting Parameters on Web

Parameters	Description
Memory Key	**BLF(NEW CALL/BXFE /AXFER)**: It is used to prompt user the state of the subscribe extension, and it can also pick up the subscribed number, which help user monitor the state of subscribe extension (idle, ringing, a call). There are 3 types for one-touch BLF transfer method. p.s. User should enter the pick-up number for specific BLF key to fulfill the pick-up operation. **Presence**: Compared to BLF, the Presence is also able to view whether the user is online. Note: You cannot subscribe the same number for BLF and Presence at the same time **Speed Dial**: You can call the number directly which you set. This feature is convenient for you to dial the number which you frequently dialed. **Intercom**: This feature allows the operator or the secretary to connect the phone quickly; it is widely used in office environments. **MWI**：You can set the speed dial key for the voice messages. **Call Park**： You can retrieve the held call by using the call park code. **Call forward**：You can transfer the call to the set number.

9 Advanced Features

9.1 VPN

Virtual Private Network (VPN) is a technology to allow device to create a tunneling connection to a server and becomes 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 to be established before activate a line registration. 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.

9.1.1 L2TP

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

To establish a L2TP connection, users should log in to the device web portal, open page [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" then the device will try to connect to the L2TP server.

When the VPN connection established, the VPN IP Address should be displayed in the VPN status. There may be some delay of the connection establishment. User may need to refresh the page to update the status.

Once the VPN is configured, the device will try to connect to the VPN automatically when the device boots up every time until user disable it. Sometimes, if the VPN connection does not established immediately, user may try to reboot the device and check if VPN connection established after reboot.

9.1.2 OpenVPN

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

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

User then upload these files to the device in the web page [Network] -> [VPN], Section OpenVPN Files.

Then user should check “Enable VPN” and select “OpenVPN” in VPN Mode and click “Apply” to enable OpenVPN connection.

Same as L2TP connection, the connection will be established every time when system rebooted until user disable it manually.

10 Trouble Shooting

When the device does not work properly, users may try the following methods to recover the device or gather relative information and send an issue report to support.

10.1 Upgrade to the latest software

Manufacturer will keep publishing software update to fix bugs and improve device features. The device will check for new software release on manufacturer cloud server automatically and periodically.

10.2 Reset Device to Factory Default

Reset Device to Factory Default will erase all user’s configuration, preference, database and profiles on the device and restore the device back to the state as factory default.

To perform a factory default reset, user should [system] -> [configurations]. Then choose [Reset to factory Default] and click [Reset], and confirm the action by [OK]. The device will be rebooted into a clean factory default state.

10.3 Network Packets Capture

Sometimes it is helpful to dump the network packets of the device for issue identification. To get the packets dump of the device, user needs to log in the device web portal, open page [System] -> [Tools] and click [Start] in “Network Packets Capture” section. A pop-up message will be prompt to ask user to save the capture file. User then should perform relevant operations such as activate/deactivate line or making phone calls and click [Stop] button in the web page when operation finished. The network packets of the device during the period have been dumped to the saved file. User may examine the packets with a packet analyzer or send it to support.

10.4 Common Trouble Cases

Table 9 - Trouble Cases

Trouble Case	Solution
Device could not boot up	1. The device is powered by external power supply via power adapter or PoE switch. Please use standard power adapter provided or PoE switch met with the specification requirements and check if device is well connected to power source
Device could not register to a service provider	1. Please check if device is well connected to the network. The network Ethernet cable should be connected to the [Network] port NOT the [PC] port. 2. Pick up the handset or press hands-free key, and input "# * 111" botton, then Checking the IP address information. If the device does not have an IP address, Please check if the network configurations is correct. 3. If network connection is fine, please check again your line configurations. If all configurations are correct, please kindly contact your service provider to get support, or follow the instructions in "**10.3 Network Packets Capture**" to get the network packet capture of registration process and send it to support to analyze the issue.
No Audio or Poor Audio in Handset	1. Please check if Handset correct is connected. 2. The network bandwidth and delay may be not suitable for audio call at the moment.
Audio is chopping at far-end in Hands-free mode	This is usually due to loud volume feedback from speaker to microphone. Please lower down the speaker volume a little bit, the chopping will be gone.