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CooFone-H83 VoIP Phone
User Manual

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Safety Instruction

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

- 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 AC power voltage. Inaccurate power voltage may cause electric shock or burn the power adaptor.
- 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 designed for indoor use only. Do not install the phone in places where there is direct sunlight. Also do not put the phone on carpets or cushions in case of overheat or fire.
- Avoid exposure the phone to high temperature or below 0℃ or high humidity.
- Avoid wetting the phone with any liquid.
- Do not attempt to open it. Non-expert handling of the phone could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock.
- 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.

1   Overview

The new H83 color screen IP Phone is a high-end enterprise desktop phone which comes with intelligent LCD the DSS Key-mapping to increase enterprise users' productivity at a cost-effective price.

The DSS key-mapping is designed with intelligence to support dynamic usage to substitute expansion modules. There are 7 DSS keys corresponded to the color LCD display to provide dynamic Line/DSS/BLF functions up to four pages at virtualized total 28 DSS keys. User may configure/customize each DSS key in each page. Every DSS key has a LED indication in green, red, and yellow colors to reflect the key state. A page shortcut button is also designed to allow user to quickly switch between pages. H83 is the most economic choice for SMB office and enterprise supervisors.

H83 inherits all features from enterprise phones, such as HD voice in handset, headset, and full-duplex speakerphone modes, PoE, Fast/Gigabit Ethernet, QoS, secure transmission, auto-provisioning, and more.

H83 is a great office productivity appliance for enterprise users. The old DSS key label is inconvenient and not environmental friendly. H83's intelligent LCD the DSS Key-mapping provides users the flexibility to change DSS key definition and display through easy configuration. Meanwhile, with its intelligent design of the DSS key, it can be multiplied as expansion modules to save space and cost. H83 will provide the best user experience to advance enterprise users."

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 H83.

2 Installation

2.1 Use PoE or external Power Adapter

H83 color screen IP phone, called as 'H83' hereafter, supports two power supply modes, power supply from external power adapter and H83 also supports 802.3af Class 1 Power over Ethernet (PoE) complied switch.

PoE power supply saves the space and cost of providing the phone additional power outlet. With a PoE switch, the phone can be powered through a single Ethernet cable which is also used for data transmission. By attaching UPS system to PoE switch, the phone 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 phone 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 phone work properly.

2.2 Desktop and Wall-mount Installation

H83 supports two installation mode, desktop and wall-mount. To set up the phone to be used on desktop, please follow the instructions in below picture to install the phone.
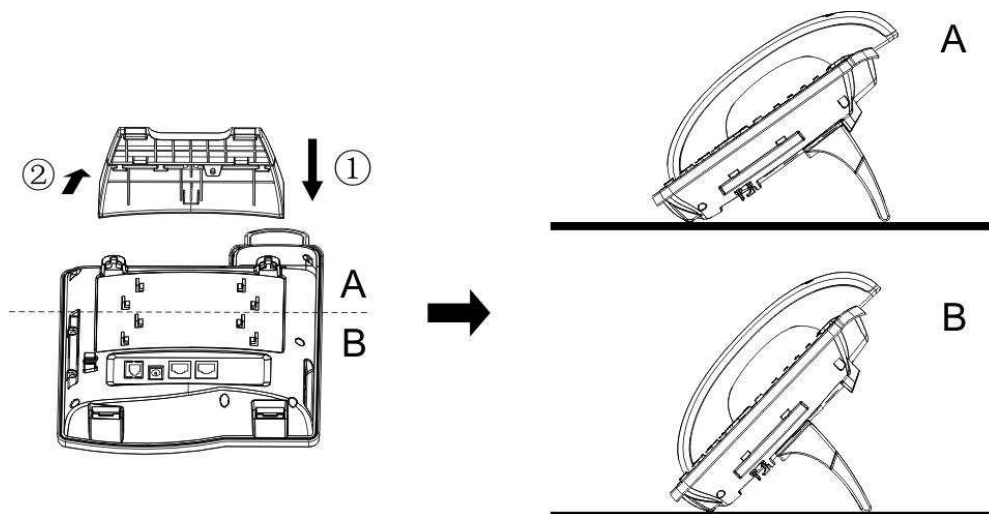
Desktop Installation

To mount the phone on the wall, please follow the instructions in below picture.
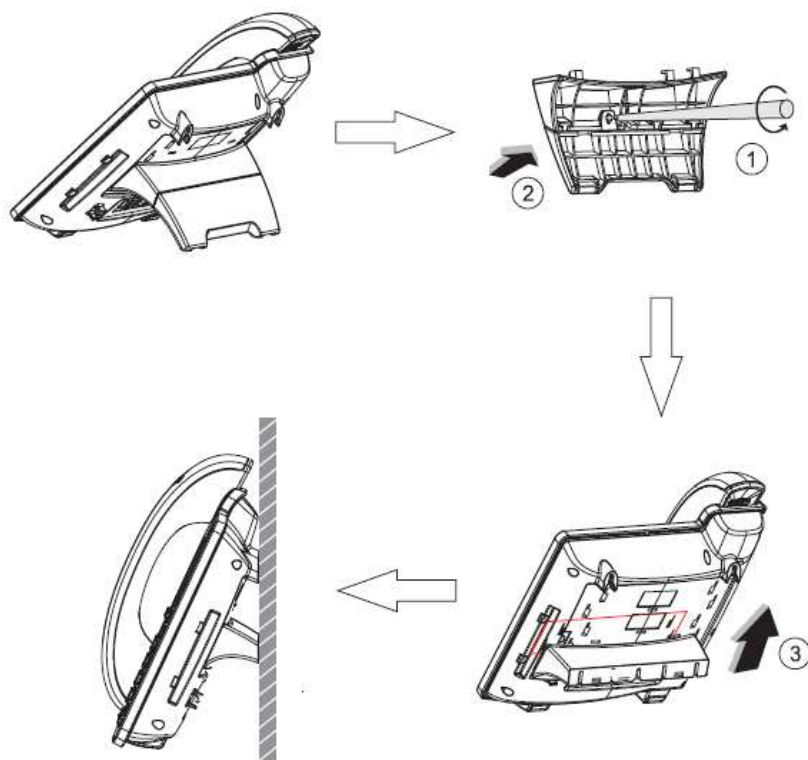
Wall-mount Installation

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

Connecting Peripherals and Network

3   Introduction to the Phone User Interface

3.1 Keypad

1. DSS keys
2. LCD page switch key
3. Soft keys
4. Power key
5. Headset key
6. MWI key
7. Mute key
8. Navigation+OK
9. Numeric key
10. Hold key
11. Transfer key
12. Conference key
13. Redial key
14. Volume -/+ key
15. Speaker key
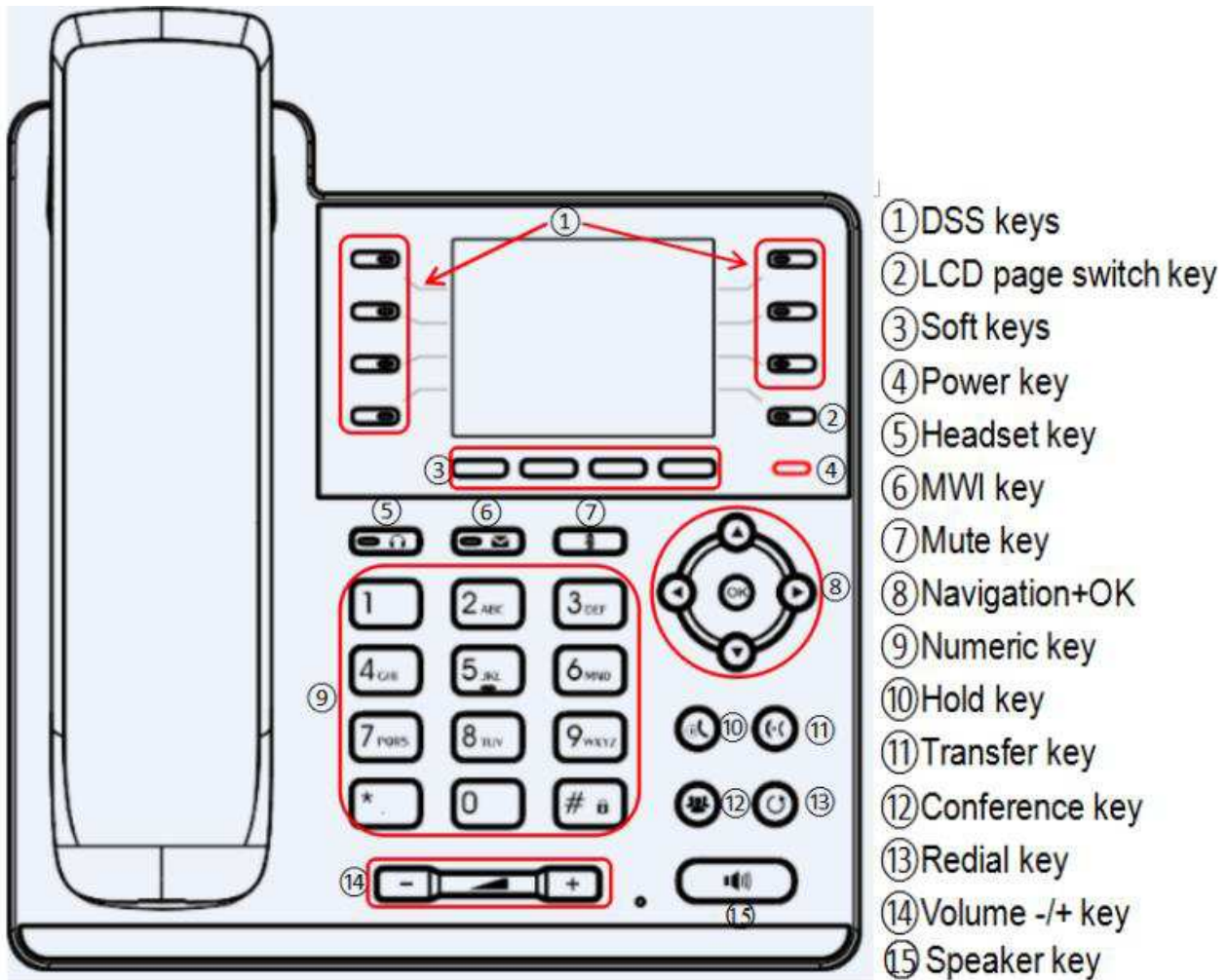
Keypad

The above picture shows the keypad layout of H83. 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.

Some keys support long-press function. User can press and hold the key for 1.5 seconds to trigger the long-press function.

- DSS keys - You could customize these DSS keys with some preset functions.
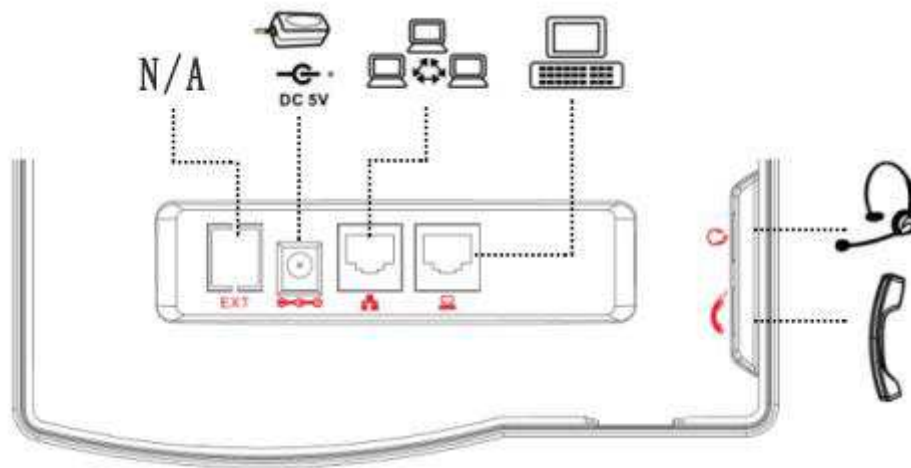- LCD page switch key - You could switch the LCD pages by pressing this key.
- Soft Keys – These four buttons provide different functions corresponding to the soft-menu displayed on the screen.
- Power key - The key will green flash when the phone reboot with power supply.
- Headset key - When you pressed the key, you could make a call using the headset.
- MWI key - You could view the voice messages by pressing this key.
- Mute key - Mute you phone. you can also mute the microphone with this button during talking mode.

- Navigation Keys – User can press up/down navigation keys to change line focus in talking screen or move cursor in a screen with list items; in some configuration or text editor screen, user can press left/right navigation keys to switch option or move cursor to the left / right.
- Numeric Keys – The 12 standard telephone keys provide the same function as standard telephones, but further to the standard function, some keys also provide special function by long-pressing the keys.
  - Key # - Long-press to lock the phone. (Default PIN is 123)
  - Key Page - Long-press to open LCD page configuration.
- Hold key - you could hold a call by pressing this key.
- Transfer key - You could transfer a call by pressing this key.
- Conference key - You could invite a third part to join your call by pressing this key.
- Redial – By pressing 'Redial' button, user can redial the last dialed number.
- Volume -/+key – In standby, ringing, ring configuration screen, user can press 2 buttons to lower/increase the ringtone volume, in talking and audio volume adjustment screen, user can press this button to lower/increase the audio volume.
- Speaker – By pressing this button once, user can turn on the audio channel of hands-free speaker

3.2 Using Handset / Hands-free Speaker / Headphone

■ **Using Handset**

To talk over handset, user should lift the handset off the phone and dial the number, or dial the number first, then lift the handset and the number will be dialed. User can switch audio channel to handset by lifting the handset when audio channel is opened in speaker or headphone.

■ **Using Hands-free Speaker**

To talk over hands-free speaker, user should press the hands-free button then dial the number, or dial the number first then press the hands-free button. User can switch audio channel to the speaker from handset by pressing the hands-free button when audio channel is opened in handset.

■ **Using Headphone**

To use headphone, by default, user should headset button which is defined by DSS key to turn on the headphone. Same as handset and hands-free speaker, user can dial the number before or after headphone turned on.

■ **Using Line Keys(Defined by DSS Key)**

User can use line key to make or answer a call on specific line. If handset has been lifted, the audio channel will be opened in handset, otherwise, the audio channel will be opened in hands-free speaker or headphone.

3.3 Screen User Interface

Screen Layout / Default Standby Screen

The screen user interface is mostly presented in the above layout except some prompt messages. The upper area is the main screen to display the phone status and information or data for viewing or editing. The lower area is the software menu (soft-menu) buttons which will change against user's action or phone status.

The above picture shows the default standby screen which is also the root of the soft-menu. The default standby screen shows the greeting words and effective feature indications like voice message, missed call, auto-answering, do-not-disturb, call forward, lock state, and the network connectivity. User can get back to the default standby screen mostly by lifting and putting by the handset.

The icon illustration is described in **Appendix I - Icon Illustration**.
In some screens, there are more items or long text to be displayed which could not fit into the screen. They will be arranged in a list or multiple lines with a scroll bar. If user sees a scroll bar, user can use up/down navigation buttons to scroll the list. By long-pressed the navigation keys, user can scroll the list or items in a faster speed.
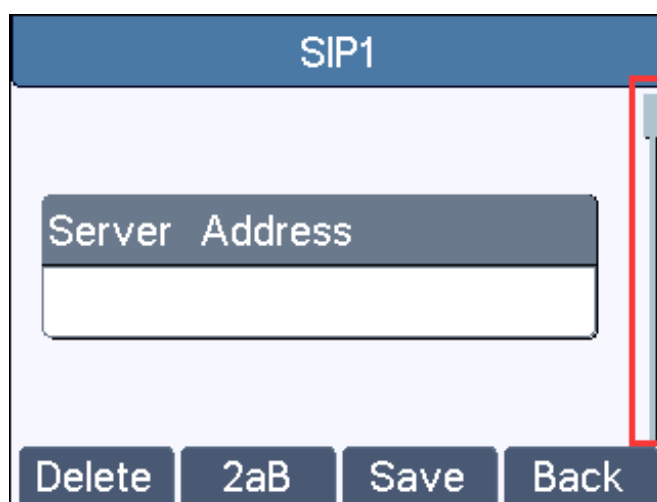
Vertical Scroll Bar

3.4 Web Portal

User can also use the phone's web portal to manage or operate the phone. User should open the phone's web portal page by entering the IP address in the Web browser. To get the phone's IP address, user could press the soft-menu button [Menu] -> [Status] or by pressing [Down] navigation key.
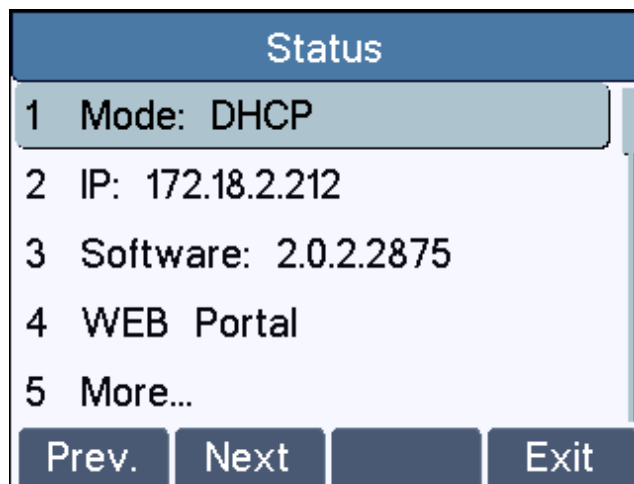
Check the phone's IP Address

The first screen of the phone's web portal is the login page.
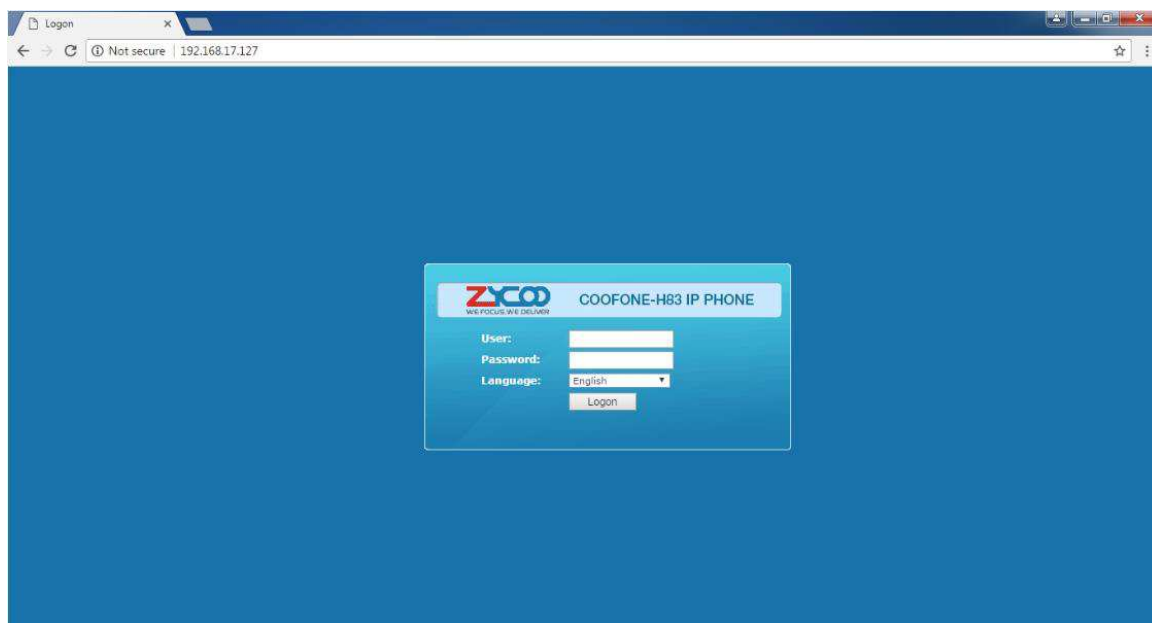
Web Portal Login Page

User must enter the username and password to log in to the web portal. **The default username and password are both 'admin'**. For the detail of web portal operations, please refer to **7 Web Portal**.

4   Setting Up

In order to get the phone ready for making and receiving phone calls, the phone must be configured with correct network configurations and at least one of the lines must be configured with an SIP telephony service.

4.1 Network Configuration

The phone relies on IP network connection to provide service. Unlike traditional phone system based on a circuit switched wire technology, IP phones are connected to each other over the network and exchange data in packet basis based on the phones' IP address.

To enable the phone, the network parameters must be configured properly first. To configure network parameters, user should open the network configuration screen through soft-menu [Menu] -> [Settings] -> [Advanced Settings] -> [Network] -> [Network Settings] from standby screen.

NOTICE!

If user saw a **'Network Disconnected' icon flashing in the middle of screen, it means the network cable was not correctly connected to the phone's network port. Please check if the cable is properly connected to the phone and to the network switch, router, or modem.**

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 don't have to configure any parameters manually. All configuration parameters will be getting from DHCP server and applied to the phone. 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 by advanced users.
- PPPoE – This option is often used by users who connect the phone directly to a broadband modem or router. To establish a PPPoE connection, user should configure username and password provided by the service provider.

The phone is default configured in DHCP mode.
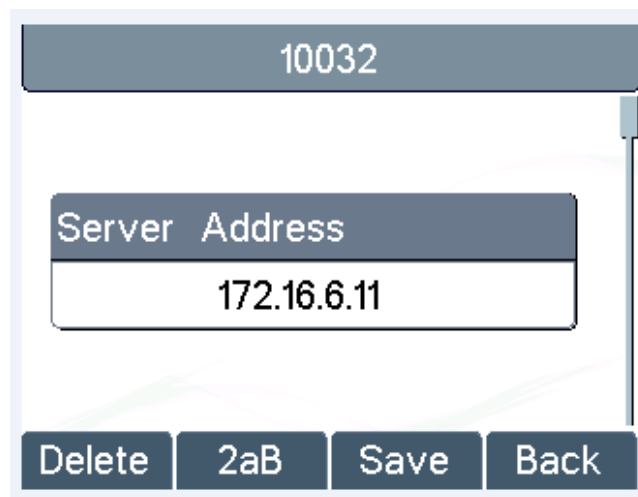
4.2 Line Configurations

A SIP line (SIP account) must be configured properly to be able to provide telephony service. The line configuration is like a virtualized SIM card. Just like a SIM card on a mobile phone, it stores the service provider and the account information used for registration and authentication. When the phone is applied with the configuration, it will register the phone to the service provider with the server's address and user's authentication as stored in the configurations.

To configure a line manually, user may open the line configuration screen through soft-menu button [Menu] -> [Settings] -> [Advanced Settings] -> [Accounts] -> [SIP1] / [SIP2] / [SIP3] / [SIP4] -> [Basic Settings] from the standby screen.

NOTICE!

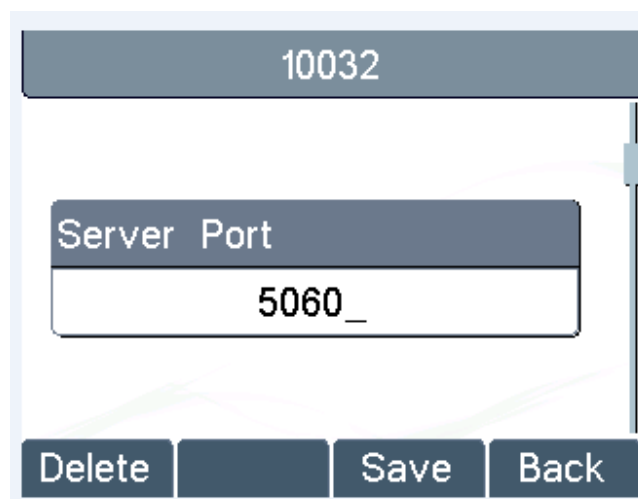**User must enter correct PIN code to be able to enter advanced settings to edit line configuration. (The default PIN is 123)**
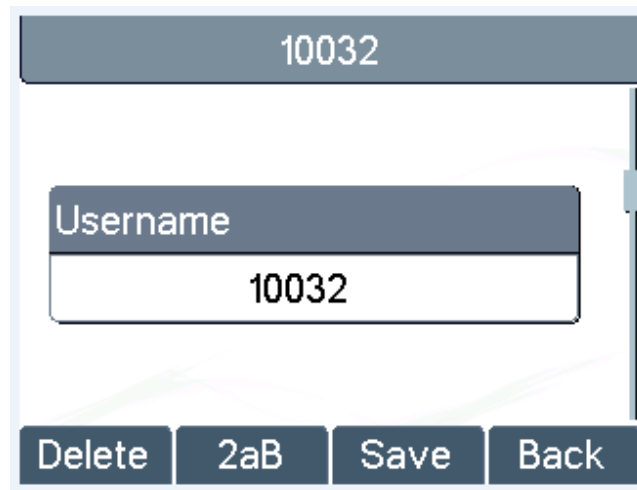
The parameters and screens are listed in below pictures.

Configure SIP Server (Proxy) Address

Configure SIP Server (Proxy) Port

Configure SIP Username

Configure Authentication Name

Configure SIP Authentication Password

Configure Display Name

Enable / Disable Outbound Proxy

Enable / Disable Sip Line

Save the configurations by pressing [OK] or [Save] button when done.

For users who want to configure more options, user should use web management portal to modify or

Advanced Settings in accounts on the individual line to configure those options.

Configure Advanced Line Options

Configure SIP Realm/Domain

Enable / Disable Anonymous

Configure Unregistered Dial

Configure DTMF Mode

10032	10032
Use STUN	**MWI Number**
Disabled ↔	123456_
Save Back	Delete 123 Save Back

Enable / Disable STUN

Configure MWI Number

10032	10032
Local Port	**Pickup Number**
5060	999_
Delete Save Back	Delete 123 Save Back

Configure Local Port

Configure Pickup Number

10032	10032
Ring Type	**Park Number**
Default ↔	888
Save Back	Delete 123 Save Back

Configure Ring Type

Configure Park Number

Configure Join Call Number

Enable / Disable Feature Sync

Enable / Disable Missed Call

Enable / Disable SCA

NOTICE!

On the phone screen, user can only configure whether STUN / Outbound Proxy is enabled, to change the STUN / Outbound Proxy server address, user must edit in the phone web portal. (Please refer to 7.12 LINES/SIP and 7.15 LINES/Basic Settings)

5   Using the Phone

5.1 Making Phone Calls

■   **Default Line**

You can configure up to 4 lines (accounts) on H83 IP phone. If all lines are configured, user can make or receive phone calls on any line. If default line is configured by user, there will be a default line to be used for making outgoing call which is indicated on the top left corner. To change the default line, user can press up/down navigation buttons to switch between all lines. Enable or disable default line, user can press [Menu] - > [Features] -> [Default Line] or configure from Web Interface (Web / Phone settings / Features).

Default Line

■   **Dialing Methods**

User can dial a number by,

- ● Entering the number directly
- ● Selecting a phone number from phonebook contacts (Refer to **5.2 Phonebook**)
- ● Selecting a phone number from cloud phonebook contacts (Refer to **5.3 Using Cloud Phonebook**)
- ● Selecting a phone number from call logs (Refer to **5.4 Call Logs**)
- ● Redialing the last dialed number

■   **Dial Number then Open Audio Channel**

To make a phone call, user can firstly dial a number by one of the above methods. When the dialed number is completed, user can press [Dial] button on the soft-menu, or press hand-free button to turn on the speaker or headphone, or lift the handset to call out with the current line, or user can press line key(Configured by DSS keys) to call out with specified line.

Dialing a Number before Audio Channel Opened

■ **Open Audio Channel then Dial the Number**

Another alternative is the traditional way to firstly open the audio channel by lifting the handset, turning on the hands-free speaker or headphone by pressing hands-free button, or line key, and then dial the number with one of the above methods. When number dialed completed, user can press [Dial] button or [OK] button to call out, or the number will be dialed out automatically after timeout.

Dial a Number after Audio Channel Opened

NOTICE!

For some users who get used to dial a number immediately by pressing # key, the user must login to the web to enable 'Press "#" to invoke dialing' option in page [Line] -> [Dial Plan] -> "Basic Settings" and apply it.

■ **Cancel Call**

While calling the number, user can press [End] button or close the audio channel by put back the handset or press the hands-free button to drop the call.

Calling Screen

■ **Answering Incoming Call**

When there is an incoming call while the phone is idle, user will see the following incoming call alerting screen

Incoming Call Screen

User can answer the call by lifting the handset, open headphone or speaker phone by pressing the hands-free button, or the [OK] / [Answer] button. To divert the incoming call, user should press [Divert] button. To reject the incoming call, user should press [Reject] button.

■ **Talking**

When the call is connected, user will see a talking mode screen as the following figure.

In Call Screen

1.Audio Channel – The icon reflects the current audio channel being used.

2.Current Line – The line is being used on the call.

3.Remote Party – The name or number of the remote party.

4.Talking time – The time passed since the call established.

5.Simutaneous Calls – The number of calls. User can press up/down navigation buttons to switch between calls.

■ **Call Holding /Resuming**

User can hold the remote party by pressing [Hold] button and the button will be changed to [Resume] icon. User can press the [Resume] button to resume the call.

Call Holding Screen

■ **Call Ended**

When user finished the call, user can put the handset back to the phone to hang up the call or press the hands-free button to close the audio channel to hang up.

NOTICE!

When the call is held or in holding state, the user must press [Resume] button to back to call mode, or putting the handset back to the phone or pressing Speaker-free button to hang up is not available.

5.1.1   Make / Receive Second Call

H83 IP phone can support multichannel concurrent calls. When there is already a call established, user can still answer another incoming call on either lines or make a second call on either lines.

■   **Second Incoming Call**

When there is another incoming call during talking a phone call, this call will be waiting for user to answer it. User will see the call message in the middle of current screen. The phone will not be ringing but playing call waiting tone in the audio channel of the current call and the LED will be flashing in green. User can accept or reject the call same as normal incoming call. When the waiting call is answered, the first call will be put on hold automatically.

Second Incoming Call Screen

■   **Second Outgoing Call**

To make a second call, user may press [XFER] / [Conf] button to make a new call on the default line or press the line key to make new call on specific line. Then dial the number the same way as making a phone call. Another alternative for making second call is to press DSS Keys dial out from the configured keys (BLF / Speed Dial). When user is making a second call with the above methods, the first call could be place on hold manually first or will be put on hold automatically at second dial.

■   **Switching between Two Calls**

When there are two calls established, user will see a dual calls screen as the following pictures

Dual Calls

User can press up/down navigation buttons to switch between call screens, and switch calls to talk by pressing [Hold] / [Resume] button.

■ **Ending One Call**

User may hang up the current talking call by closing the audio channel or press [End] button. The phone will return to single call mode in holding state.

5.1.2   Merge / Split Two Calls (3-way Local Conference)

In the dual call mode, user can merge two calls into a conference call by pressing [Conf] button. When two calls are merged, user can split them by pressing [Split] button.

3-way Conference Call

5.1.3   Call Transfer

When the user is talking with a remote party and wish to transfer the call to another remote party, there are two ways to transfer the call, attended transfer and blind transfer.

■ **Attended Transfer**

The attended call transfer which is to dial through the other remote party, wait for the remote party to answer the call and then transfer the call.

This is the same procedure as making two concurrent calls. In the dual-call mode, press the [XFER] button to transfer the first party to the second one.

■ **Blind Transfer**

Instead of connecting to the second party first and confirming the second call is established, user press [XFER] button first then dial the second party number. When the number is dialed, user can press[XFER] button again and the first party will be transferred to the second.

This is like helping the first party to dial to the second one. However, the transfer could be successful if the second party answered it, or could be failed if the second party is busy or rejected.
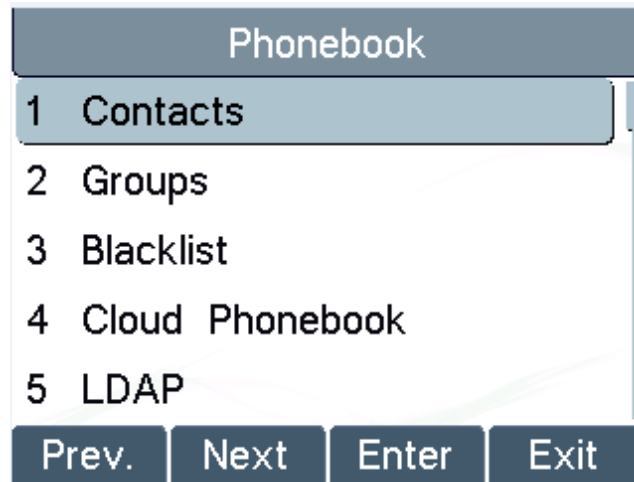
NOTICE!

Except transfer feature of the phone, you could also use the transfer feature of the IPPBX. For example, on ZYCOO IPPBX system you can initiate attended transfer by feature code *2, or blind transfer by feature code #.

5.2 Phonebook

Users can save contacts' information in the phonebook and dial the contact's phone number(s) from the phonebook. To open the phonebook, user can press soft-menu button [Dir] in the default standby screen or keypad.

By default the phonebook is empty, user may add contact(s) into the phonebook manually or from call logs.

Phonebook screen

NOTICE!
The H83 IP phone can save up to total 500 contact records.
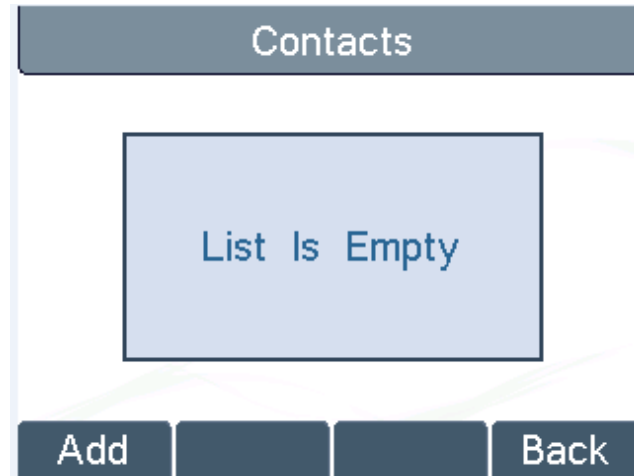
Phonebook is Empty

When there are contact records in the phonebook, the contact records will be arranged in the alphabet order. User may browse the contacts with up/down navigation keys. User may fast scroll up/down the contact list by holding the up/down navigation keys. The record indicator tells user which contact is currently focused. User may check the contact's information by pressing [OK] button.
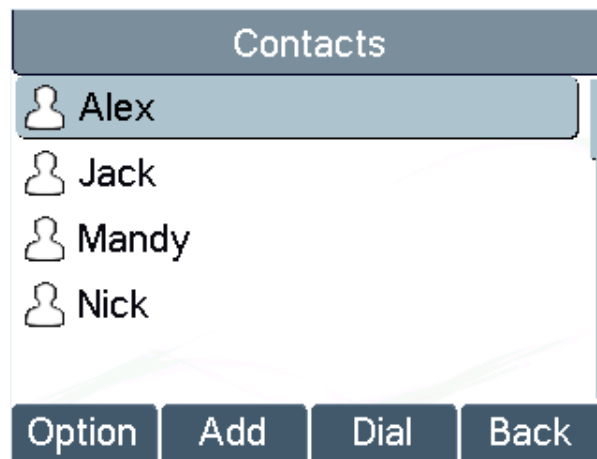
Browsing Phonebook

5.2.1   Add / Edit / Delete Contact

To add a new contact, user should press [Add] button to open Add Contact screen and enter the contact information of the followings.

● Contact Name
● Tel. Number
● Mobile Number
● Other Number
● Ring Tone
● Contact Group

Add New Contact

User can edit a contact by pressing [Option] -> [Edit] button.

To delete a contact, user should move the record indicator to the position of the contact to be deleted, press [Option] -> [Delete] button and confirm with [Yes].

5.2.2 Add / Edit / Delete Group

User can open the group list by pressing [Groups] button in phonebook screen.

By default, the group list is blank. User can create his/her own groups, edit the group name, add or remove contacts in the group, and delete a group.

To add a group, press [Add] button.

To delete a group, press [Option] -> [Delete] button.

To edit a group, press [Edit] button.
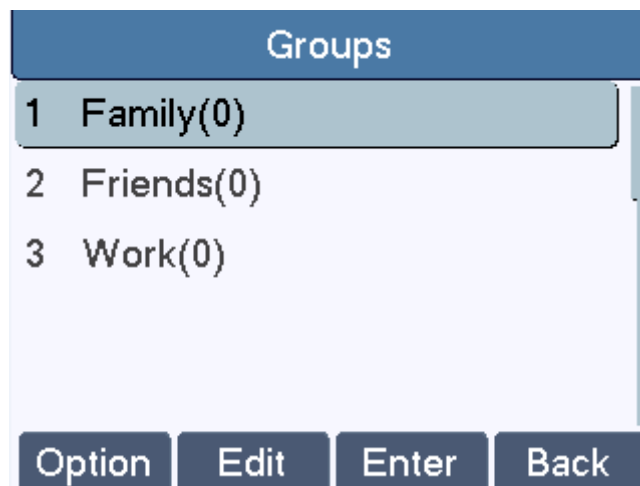
Group List

5.2.3 Browse and Add / Remove Contacts in Group

User can browse contacts in a group by opening the group in group list with [OK] button.
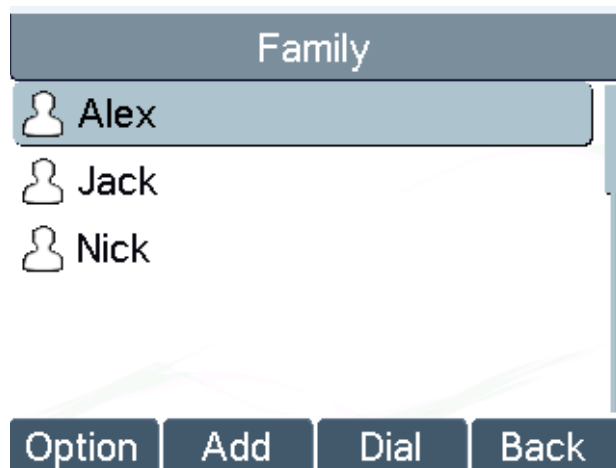
Browsing Contacts in a Group

When user is browsing contacts of a group, user can also add or remove contacts in that group by pressing [Add] button to enter the group contacts management screen. Press [Add] button to create a new contact for the group; press [Option] button to remove the contact from the group.
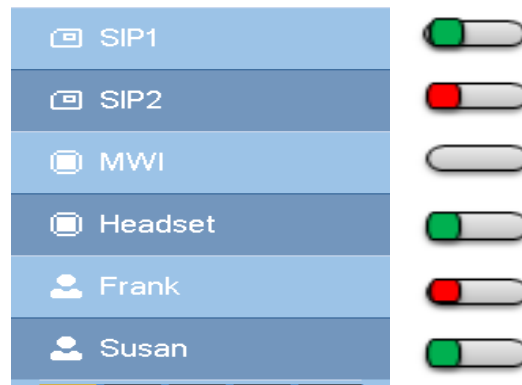
Add/Remove Contacts in a Group

5.3　Using Cloud Phonebook

Cloud phonebook allows user to configure the phone to download a phonebook from a cloud server. This is very useful for office users to use the phonebook from a single source and save the effort to create and maintain the contact list individually. It is also a useful tool for user to synchronize his/her phonebook from a personal mobile phone to the phone with Cloud Phonebook Service and App which is to be provided publicly soon.

NOTICE!

The cloud phonebook is ONLY temporarily downloaded to the phone each time it is opened on the phone to ensure the user gets the most up to date phonebook. However, the downloading may take a couple seconds depending on the network condition. Therefore, it is highly recommended to the users to save important contacts from cloud to local phonebook to save the time of waiting for downloading.

To open cloud phonebook list, press [Cloud Phonebook] in phonebook screen.
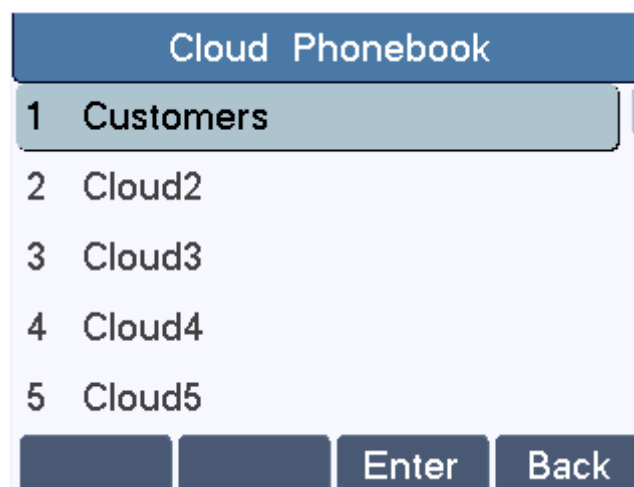
Cloud Phonebook List

5.3.1 Open Cloud Phonebook

In cloud phonebook screen, user can open a cloud phonebook by pressing [OK] / [Enter] button. The phone will start downloading the phonebook. If downloading failed, user will be prompted with a warning message. Once the cloud phonebook is downloaded completely, user may browse the contact list and dial the contact number same as in local phonebook.
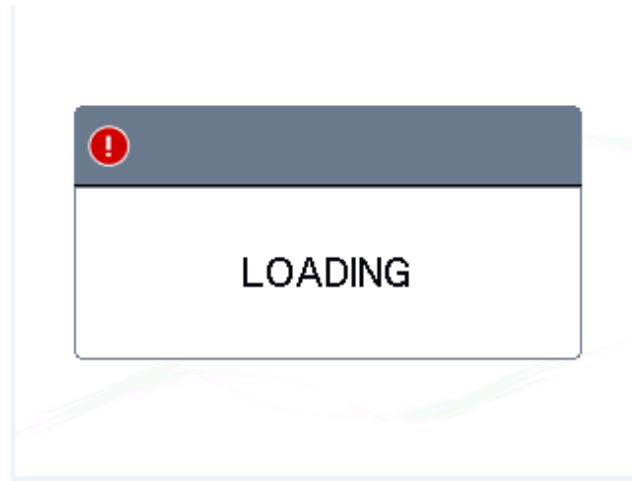
Downloading Cloud Phonebook
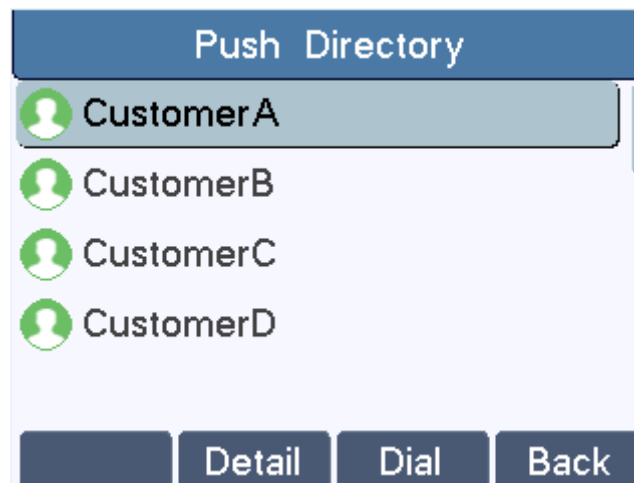
Browsing Contacts in Cloud Phonebook

5.4 Call Logs

The phone can store up to 600 call log records and user can open the call logs to check all incoming, outgoing, and missed call records by pressing soft-menu button [History]
In the call logs screen, user may browse the call logs with up/down navigation keys.
Each call log record is presented with 'call type' and 'call party number / name'. User can check further call log detail by pressing [OK] button and dial the number with [Dial] button, or add the call log number to phonebook with pressing [Option] -> [Add to Contact] .

User can delete a call log by pressing [Delete] button and can clear all call logs by pressing [Clear] button.

Call Logs

User can also filter call logs with specific call log type to narrow down the call log records by pressing the left/right navigation button and select one of the call log types in the soft-menu buttons.

 - Missed Calls

 - Received Calls / Incoming Calls

 - Dialel Calls / Outgoing Calls

Filter Call Log Type

5.5   Voice Message

When the user is not available to answer the call, the caller can leave a voice message on the server to the user. User will receive voice message notification from the server and phone will prompt a voice message waiting icon on the standby screen.

New Voice Message Notification

To retrieve the voice messages, user must configure the voice message number first. Once the voice message number is configured, user can retrieve the voice message of a line by pressing the [Dial] button in the voice message screen.

When the phone is in standby state.
● You can press [Menu] / [Applications] / [Voice Massage] to enter MWI interface
● When you opened voice message configuration interface, select the line to be configured with up/down navigation keys.
● Press [Edit] button to enable and edit the voice message number, when done, press [OK] / [Save] button to save the configuration.

Voice Message Screen

Configure the Voice Message Number

5.6   Do-Not-Disturb

User may enable Do-Not-Disturb (DND) feature on the phone to reject incoming calls (including call waiting). DND can be enabled on line basis.

To quickly enable or disable the DND on all lines. When the phone is in standby state.

- Press [DND] button to enable DND on all line and icon ![DND] will be displayed on the screen.

- Press [DND] button again to disable DND on all lines and icon ![DND] will disappear from the screen.

DND Enabled on All Lines

33

If user wishes to enable or disable DND on a specific line, user could change the DND mode in DND configurations.

- Press soft-button [Menu] till you find the [Features] item.
- Enter [Features] item till you find the [DND] item.
- Enter [DND] item till you find [DND item.
- Press left/right navigation key to change the DND mode or DND state on specific line. When done, press [Save] button to save the changes.
- User will see icon 'Do Not Disturb' against the enabled line.

Configure DND on Line

5.7   Auto-Answering

User may enable auto-answering feature on the phone and any incoming call will be automatically answered (not including call waiting). Auto-answering can be enabled on line basis.

When the phone is in standby state, if user wishes to enable or disable auto-answering on a specific line or change the auto-answering delay time, user could change the auto-answering configuration by following steps.

- Press soft-button [Menu] till you find the [Features] item.
- Enter [Features] item till you find the [Auto Answer] item.
- Enter [Auto Answer] item to change the auto-answering configuration on a specific line
- Press left/right navigation button to select the auto-answering option. When done, press [Save] button to save the changes.
- The default auto-answering delay is 5 seconds.

Configure Auto-answering on Line1

Auto-answering Enabled on Line1

5.8   Call Forward
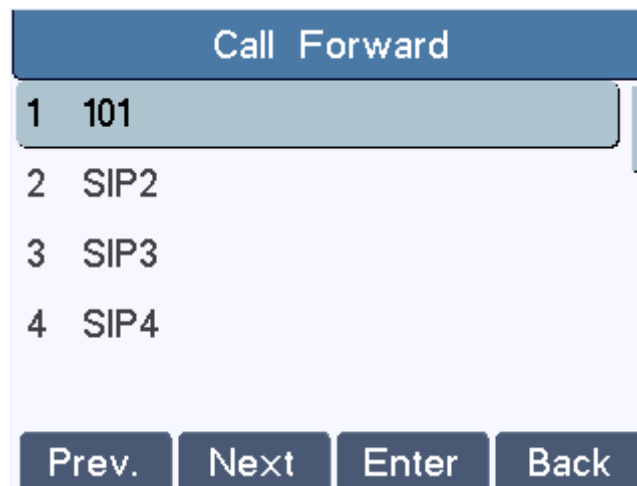
Call forward is also known as 'Call Divert' which is to divert the incoming call to a specific number based on the conditions and configurations. User can configure the call forward settings of each line.

There are three types of call forwarding.

● **Unconditional Call Forward –** Forward any incoming call to the configured number.
● **Call Forward on Busy –** When user is busy, the incoming call will be forwarded to the configured number.
● **Call Forward on No Answer –** When user does not answer the incoming call after the configured delay time, the incoming call will be forwarded to the configured number.
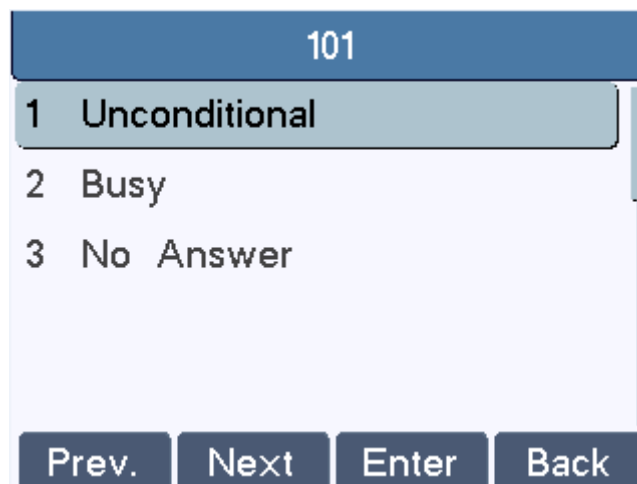
To configure call forward, when the phone is in the default standby mode,

● Press soft-button [Menu] till you find the [Features] item.

● Enter [Features] item till you find the [Call Forward] item.

● Press [Call Forward] button to open call forward configuration screen, select the line to be configured with up/down navigation keys.

● Press [Enter] button to edit the call forward settings.

● Select the call forward type with up/down navigation keys. Click [Enter] button to configure the call forward number and delay time, if applicable.

● Enable or Disable call forward with left/right navigation buttons against specific lines and types.

● If select 'Enable', browse the setting parameters with up/down navigation keys and enter required information. When done, press [Save] / [OK] button to save the changes.

Select Line for Call Forward Configuration

Select Call Forward Type

Activate Call Forward and configure Call Forward Number

Set Delay Time for Call Forward on No Answer

5.9   DSS LCD Configuration

There are 7 DSS keys corresponded to the LCD display to provide dynamic Line/DSS/BLF functions up to four pages at virtualized total 28 DSS keys. User may configure/customize each DSS key in each page. User can enter the DSS Key page configuration by long-press DSS page key, where user can Delete/Add page(s) for the DSS LCD. Besides, user can modify DDS key configuration by long-press each one.
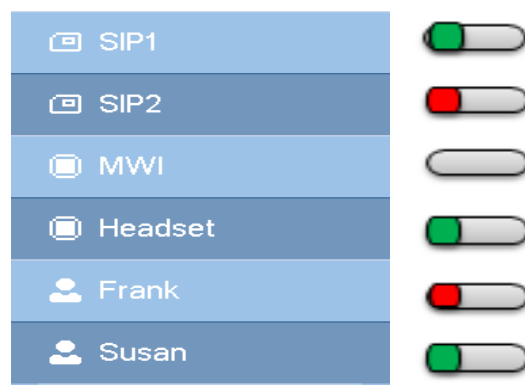
DSS Page Configuration Screen

The phone provide DSS Key configuration, such as.

Memory Key

- ■ Speed Dial/Intercom/BLF/Presence/Call Park/Call Forward(For someone)
- ◆ Line
- ◆ Key Event
  - ■ MWI/DND/Hold/Transfer/Phonebook/Redial/Pickup/Call Forward(for entitle line)
- ◆ DTMF
- ◆ URL
- ◆ BLF List Key
- ◆ Multicast

BLF/Intercom/Speed Dial number can match contact intelligently as the label, which is stored in phonebook, and DSS LCD will display the contact name rather than extension number. Moreover, user also can add the user-defined title for the DSS Keys, which is configured as Memory Key / Line / URL / Multicast.
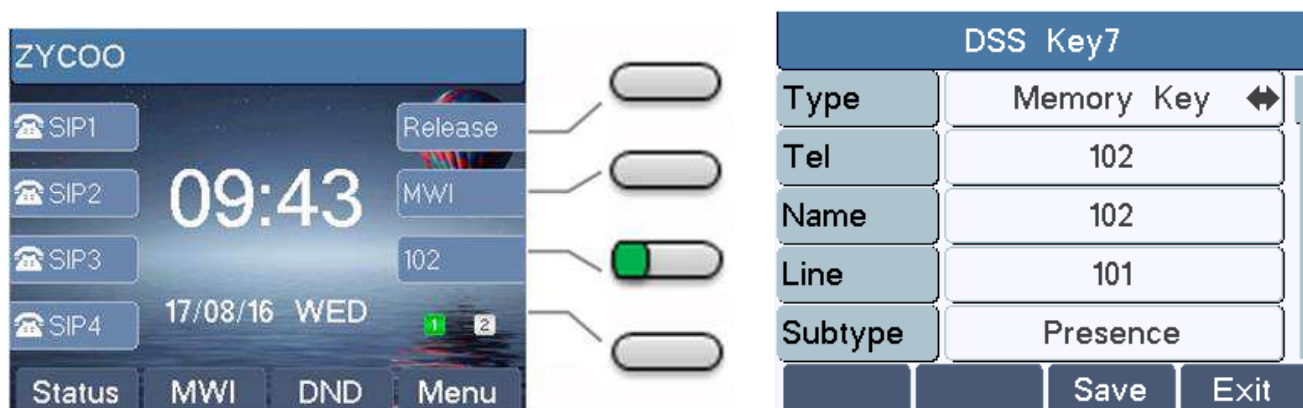
DSS Key Configuration

NOTICE!

User-defined title is limited to 15 characters at most.

6   Phone Settings

6.1 Adjust Audio Volume

When the phone is in standby state,

- Press soft-button [Menu] till you find the [Settings] item.
- Enter [Settings] item till you find the [Basic Settings] item.
- Enter [Basic Settings] item till you find [Voice Volume] item.
- Enter [Voice Volume] item and you will find [Handset Volume],[Handsfree Volume] and [Headset Volume] item.
- Enter [Handset Volume] or [Handsfree Volume] or [Headset Volume] item, press Left / Right navigation keys to adjust the audio volume for different mode.
- Save the adjustment by pressing [OK] or [Save] button when done.

6.2 Set Ring Tone Volume and Type

When the phone is in standby state,

- Press soft-button [Menu] till you find the [Settings] item.
- Enter [Settings] item till you find the [Basic Settings] item.
- Enter [Basic Settings] item till you find [Ring Settings] item.
- Enter [Ring Settings] item and you will find [Ring Volume] and [Ring Type] item.
  - Enter [Ring Volume] item and you find [Headset Volume] and [Handsfree Volume] item.
    - Enter [Headset Volume] or [Handsfree Volume] item, press left / right navigation keys to adjust the ring volume.
    - Save the adjustment by pressing [OK] button when done.
  - Enter [Ring Type] item, press left / right navigation keys to change the ring type.
    - Save the adjustment by pressing [OK] button when done.

6.3 Adjust LCD Energy Saving

When the phone is in standby state,

- Press soft-button [Menu] till you find the [Settings] item.
- Enter [Settings] item till you find the [Basic Settings] item.
- Enter [Basic Settings] item till you find [Screen Settings] item.
- Enter [Screen Settings] item till you find [Energysaving] item.
- Enter [Energysaving] item, press Left / Right navigation keys to adjust the LCD Energysaving mode.
- If select 'Enabled', browse the setting parameters with up/down navigation keys and enter required information. When done, press [Save] / [OK] button to save the changes.
- The default Energysaving time is 30 seconds.

6.4 Time/Date Settings

When the phone is in standby state,

● Press soft-button [Menu] till you find the [Settings] item.
● Enter [Settings] item till you find the [Basic Settings] item.
● Enter [Basic Settings] item till you find [Time & Date] item.
● Enter [Time & Date] item, use up/down navigation keys to edit the time/date parameters and save the
   settings by pressing [OK] or [Save] button when done.

Time Settings Parameters

Parameters	Description
Mode	Auto/Manual Auto: Enable network time synchronization via SNTP protocol, default enabled. Manual: User can modify data manually.
SNTP Server	SNTP server address
Time zone	Select the time zone
Daylight Saving Time	Enable or Disable the Daylight Saving Time
12-Hour Clock	Display the clock in 12-hour format
Time format	Select time format from one of the followings: ■ 1 JAN, MON ■ 1 January, Monday ■ JAN 1, MON ■ January 1, Monday ■ MON, 1 JAN ■ Monday, 1 January ■ MON, JAN 1 ■ Monday, January 1 ■ DD-MM-YY ■ DD-MM-YYYY ■ MM-DD-YY ■ MM-DD-YYYY ■ YY-MM-DD ■ YYYY-MM-DD
Date(Only Mode=Manual)	Edit the date of the phone in the format of yyyy/mm/dd.
Time(Only Mode=Manual)	Edit the time of the phone in the format of hh:mm:ss. Hh=00~23, mm=00~59, ss=00~59

6.5 Set Language

When the phone is in standby state,

- Press soft-button [Menu] till you find the [Settings] item.
- Enter [Settings] item till you find the [Basic Settings] item.
- Enter [Basic Settings] item till you find [Language] item.
- User can change the language by using the navigation keys.

6.6 Reboot the Phone

When the phone is in standby state,

- Press soft-button [Menu] till you find the [Reboot System] item.
- Press [OK] or [Enter] button, a warning message "Reboot Now?" will be prompt to user.
- Press [Yes] or [OK] button to execute the reset command, or [No] to exit.

6.7 Reset to Factory Default

When the phone is in standby state,

- Press soft-button [Menu] till you find the [Settings] item, and press [OK] or [Enter] button.
- Select [Advanced Settings] item, then input PIN (Default PIN is 123) to enter the menu.
- Select [Reset to Default] item, then a warning message "Factory Reset, Continue?" will be prompt to user.
- Press [Yes] or [OK] button to execute the reset command, or [No] to exit.

7 Web Portal

7.1 Web Portal Authentication

User can login in to the phone web portal to manage the phone or user profiles. User must provide correct username and password to be able to login in.

7.2 Web Dial

Besides making and receiving phone calls from the phone directly, user could also make and receive phone calls from the web portal.

Web dial is particularly suitable for users who use headphone. User can dial out a number by clicking the [Dial] button on top of the web page, or click a contact in the phonebook, or click a number in the call logs, to make phone call and press [Answer] button to answer incoming call. When call ended, user can click [Hang up] button to end the call. All these operation can be done from the web portal without touching the phone.

NOTICE!

The web call is limited to make a single call. To make or answer a second call is not allowed.
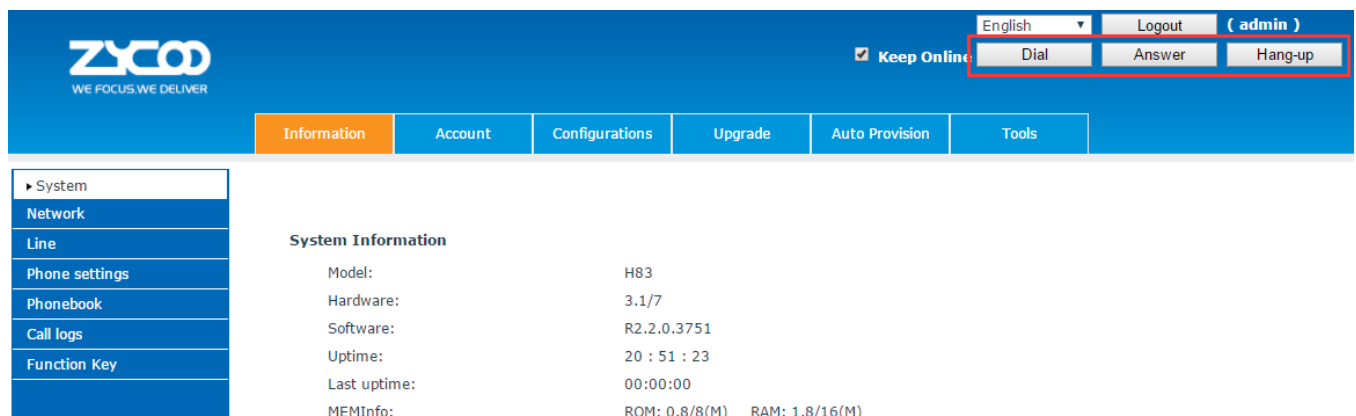
Web Call Controls

Web Dial, Entering Number and Select Line

7.3 SYSTEM / Information

User can get the system information of the phone on this page including,

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

And network status,

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

And also SIP account status,

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

7.4 SYSTEM / Account

User may change his/her web authentication password on 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 phone and cannot change phone settings.

The user account can be used to operate the phone or access the phone web portal by login to the phone or its web. User should login in to phone web portal with his/her username and web password.
NOTICE!
The phone is shipped with a default Administrators user account. The username and password for the default *account* are both 'admin' which has been printed on the phone label, at the bottom side of the phone.

7.5 SYSTEM / Configurations

Users with administrator's privilege can export or import the phone configuration in this page and reset the phone to factory defaults.

7.6 SYSTEM / Upgrade

Users can download the new firmware from ZYCOO website and upgrade the phone firmware for new features or bug fixes from this page.

7.7 SYSTEM / Auto Provision

The Auto Provision settings can help IT manager or service provider to easily deploy and manage massive phones. In most cases the IPPBX system supports auto provision phones via DHCP option 66 and SIP Plug and Play (PNP), so you don't have to modify any settings here.
If you have your own FTP/TFTP/HTTP/HTTPS server for auto provisioning, please fill in the server details here on this page.

7.8 SYSTEM / Tools

Tools provided in this page help users to identify issues at troubleshooting. Please refer to **9 Troubleshooting** for more detail.
Besides, the phone also provided the screenshot feature to user.(LCD Screen)

7.9 NETWORK / Basic

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

7.10 NETWORK / Advanced

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

7.11 NETWORK / VPN

User may configure a VPN connection in this page. Please refer to **8.1 VPN** for more details.

7.12 LINES / SIP

The service of the SIP lines is configured in this page, choose the sip line (from SIP 1 – SIP 4) to configured. Click the dropdown arrow to adjust configuration according to each line.

SIP Settings for SIP 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 SIP account.
Display Name	Enter the display name to be sent in a call request.
Authentication Name	Enter the authentication name of the SIP account
Authentication Password	Enter the authentication password of the SIP 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 specified ring 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 ring time of not answered call before being forwarded
Enable Hotline	Enable hotline configuration, the phone will dial to the specific number immediately at audio channel opened by off-hook

	handset or turn on hands-free speaker or headphone
Hotline Number	Set the hotline dialing number
Hotline Delay	Set the delay for hotline before the system automatically dial it
Enable Auto Answering	Enable auto-answering, the incoming calls will be answered automatically after the time of delay.
Auto Answering Delay	Set the delay for incoming call before the system automatically answered it
Subscribe For Voice Message	Enable the phone to subscribe a voice message waiting notification, if enabled, the phone 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 phone to use 182 response code at call waiting response
Anonymous Call Standard	Set the standard to be used for anonymous calling
Dial Without Registered	Set call out by proxy without registration
User Agent	Set the user agent, the default value is phone Model.
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 phone itself, maximum supports two remote parties, Server=set up call conference by dialing to a conference room on the server (IPPBX)
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 enabled, the phone 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 phone itself but by the server instead. In order to control the enabling of the features, the phone 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

7.13 LINES / Dial Peer

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

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 phone will still send "322" rather than "833333". The replacement rules should be matched globally.

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.

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 phone will send out "0079312".

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 phone will send out "13833220088" .
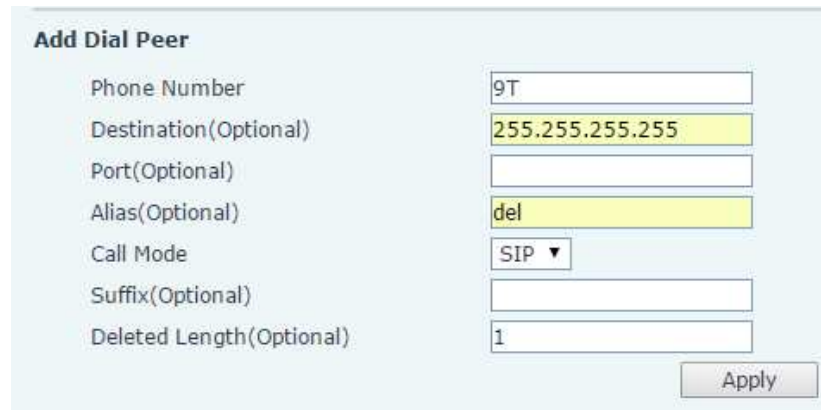

Add Suffixes Configuration

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.

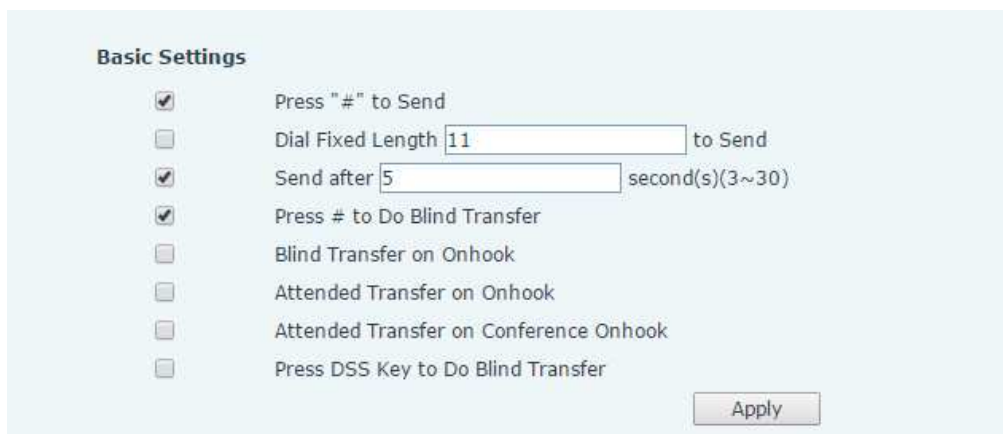For example, when users dial "98322", the phone will send out "8322" .

Deletion Configuration

7.14 LINES / Dial Plan

Dial Plan Configuration

The H83 IP phone 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.

7.15 LINES / Basic Settings

Configure basic settings for lines.

Basic Settings for Lines on Web

Parameters	Description
SIP Settings	
Local SIP Port	Set the local SIP port used to send/receive SIP signaling.
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.

7.16 PHONE / Features

Configure the phone features

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 and enable it, then you will not be able to 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
Auto Hand up Delay	Specify Auto handdown 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 handdown time at handset mode

	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.
Enable Call Completion	
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 light will red blink to remind that there is a missed call instead of playing ring tone.
Disable Mute for Ring	Disable Mute for Ring
Enable Intercom	When intercom is enabled, the phone 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 phone 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
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
Encryption Number Length	Configure the Encryption Number length
Restrict Active URI Source IP	Set the phone 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 phone when user pressed a phone digits during taking, default enabled.
Play Dialing DTMF Tone	Play DTMF tone on the phone 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.
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.

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

Restrict Outgoing Call

The user with Administrators privilege can set the rule to restrict some numbers from dialing out. This setting is phone-wide and applies to every user of the phone.

User can add restricted number by entering the number in the edit box and click "Add" button. To remove a number from the restriction list, select the number from the dropdown box and click "Delete" button to remove it.

7.17 PHONE / Audio

Change the audio settings.

<center>Audio Settings on Web</center>

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 phone 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

7.18 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.

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.

7.19 PHONE / Time/Date

User can configure the phone's time settings on this page.

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 phone 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

7.20   PHONE / Advanced

User can configure the advanced configuration settings on this page.

● Screen Configuration.

  ■ Enable Energysaving

  ■ Backlight Time

● LCD Menu Password Settings.

  The password is 123 by default.

● Keyboard Lock Settings.

● Configure Greeting Words

The greeting message will display on the top left corner of the LCD when the phone is idle, which is limited to 12 characters. The default string is 'ZYCOO'.

7.21   PHONEBOOK / Contacts

User can add, delete, or edit contacts in the phonebook in this page. User can browse the phonebook and sorting it by name, phones, or filter them out by group.

To add a new contact, user should enter contact's information and press "Add" button to add it.

To edit a contact, click on the checkbox in front of the contact, the contact information will be copied to the contact edit boxes, press "Modify" button after finished editing.

To delete one or multiple contacts, check on the checkbox in front of the contacts wished to be deleted and click the "Delete" button, or click the "Clear" button with selecting any contacts to clear the phonebook.

User can also add multiple contacts into a group by selecting the group in the dropdown options in front of "Add to Group" button at the bottom of the contact list, selecting contacts with checkbox and click "Add to Group" to add selected contacts into the group.

7.22   PHONEBOOK / Cloud Phonebook

Cloud Phonebook

User may configure up to 8 cloud phonebooks. Each cloud phonebook must be configured with an URL where an XML phonebook is stored. The URL may be based on HTTP/HTTPs or FTP protocol with or without authentication. If authentication is required, user must configure the username and password.

To configure a cloud phonebook, the following information should be entered,

  Phonebook name (must)

Phonebook URL (must)

Access username (optional)

Access password (optional)

LDAP Settings

The cloud phonebook allows user to retrieve contact list from a LDAP Server through LDAP protocols.

User must configure the LDAP Server information and Search Base to be able to use it on the phone. If the LDAP server requires authentication, user should also provide username and password.

To configure a LDAP phonebook, the following information should be entered,

Display Title (must)

LDAP Server Address (must)

LDAP Server Port (must)

Search Base (must)

Access username (optional)

Access password (optional)

7.23   PHONEBOOK / Blacklist

By adding a number into the blacklist, user will no longer receive phone call from that number and it will be rejected automatically by the phone until user delete it from the blacklist.

User can add specific number to be blocked, or a prefix where any numbers matched the prefix will all be blocked.

7.24   PHONEBOOK / Advanced

User may export current phonebook in xml, csv, or vcf format file and save it locally on a computer.

User can also import contacts into phonebook from an xml, csv, or vcf file.

NOTICE!

If user repeatedly imports a same phonebook, it will *not* create duplicated contacts in user's phonebook.

User can add new group in this page or delete an existing one. Deleting a contact group will not delete the contacts in that group.

7.25   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.

7.26 FUNCTION KEY / Function Key
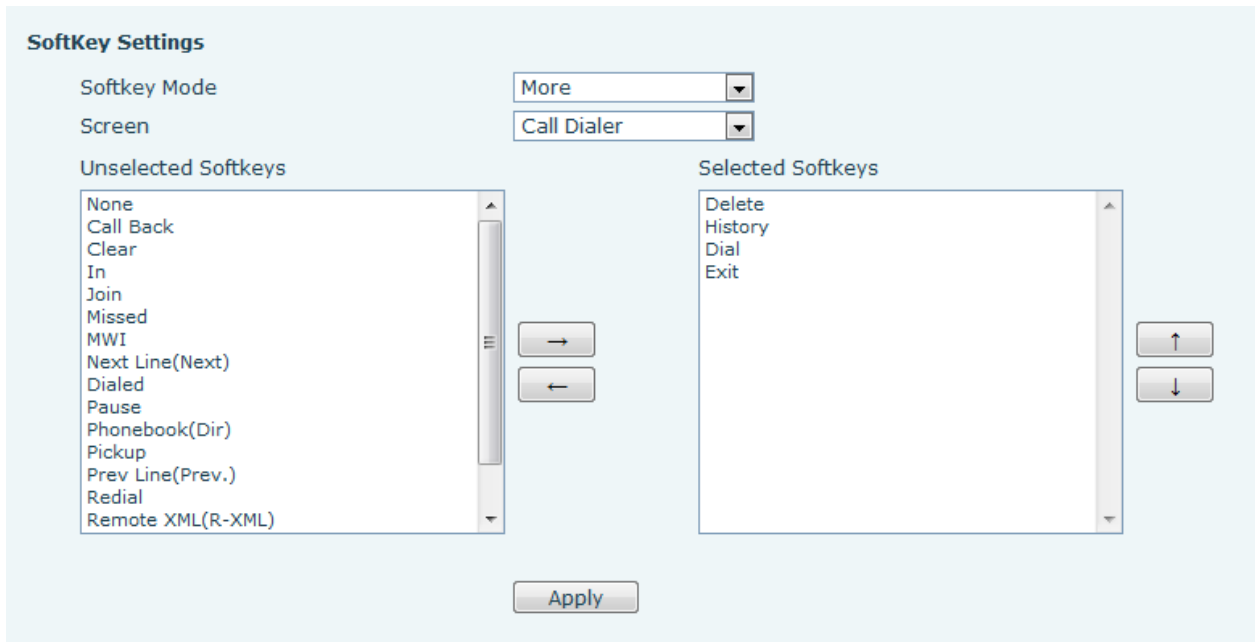
H83 IP phone provides 28 user-define DSS Keys at most. User may configure/customize each DSS key in this webpage.

DSS Key Setting Parameters on Web

Parameters	Description
Memory Key	**BLF(NEW CALL/BXFE /AXFER)**: It is used to indicate 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, in 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.
Line	It can be configured as a Line Key. User is able to make a call by pressing Line Key.
Key Event	User can select a key event as a shortcut to trigger. For example: MWI / DND / Release / Headset / Hold / etc.
DTMF	It allows user to dial or edit dial number easily.
URL	Open the specific URL directly.
Multicast	Configure the multicast address and audio codec. User presses the key to initiate the multicast.
BLF List Key	You can view the subscription status of the subscribed numbers.
LDAP	You can store the phonebook via the LDAP function

7.27 FUNCTION KEY / Softkey

User can configure different functions in different screens for each softkey.

Softkey Configuration

8   Advanced Features

8.1 VPN

Virtual Private Network (VPN) is a technology to allow H83 IP phone to create a tunneling connection to a server and becomes part of the server's network. The network transmission of the phone 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 phone supports two VPN modes, Layer 2 Transportation Protocol (L2TP) and OpenVPN.

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

8.2 L2TP

NOTICE!

The H83 IP phone 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 phone 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 phone 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 phone will try to connect to the VPN automatically when the phone boots up every time until user disable it. Sometimes, if the VPN connection does not established immediately, user may try to reboot the phone and check if VPN connection established after reboot.

8.3 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 phone 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.

9   Trouble Shooting

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

9.1 Get Phone System Information

Users may get the phone system information by pressing [Settings] -> [Status]. The following basic information will be provided:

Mode

IP

Software Version

More…

User can select [More] item to get more information in details.

9.2 Upgrade to the latest software

ZYCOO will keep publishing software update to fix bugs and improve phone features. Please download firmware package from ZYCOO official website to upgrade your phone.

9.3 Reboot Phone

Users may reboot the phone from soft-menu, [Menu] -> [Reboot System], and confirm the action by [Yes]. Or, simply remove the power supply and restore it again.

9.4 Reset Phone to Factory Default

Reset phone to Factory Default will erase all user configurations, preferences, database and profiles on the phone and restore the phone back to the state as factory default.

To perform a factory default reset, user should do it by [Menu] -> [Settings] -> [Advanced settings], and then input the password. Then choose [Reset to Default] and press [Enter], and confirm the action by [Yes]. The phone will then reboot and reset to factory default state.

9.5 Network Packets Capture

Sometimes it is helpful to sniff the network packets of the phone for issue identification. To get the packets of the phone captured, user needs to log in the phone 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 capture finished. The network packets of the phone during the period have been captured and saved as a file. User may examine the packets with a packet analyzer or send it to support.

9.6 Entering Post Mode and Recover System Image

WARNING!
PLEASE DO NOT UPDATE SYSTEM IMAGE FROM POST MODE UNLESS THE SYSTEM IMAGE IS DAMAGED.

If the phone could not boot up and entered into POST MODE, it is usually because of system image damaged due to power failure during software upgrading. When this occurs, users may follow below steps to restore the system image.

Step 1. Download the latest phone software.

Step 2. Setup a FTP server on a PC and put the downloaded software in the FTP repository. (Windows users can download free FTP server software such as WFTPD32 or 3CDaemon)

Step 3. Configure the PC with fixed IP address as 192.168.10.10

Step 4. Connect phone PC port with the PC.

Step 5. Telnet from command line to phone: >telnet 192.168.10.1

Step 6. When telnet connection established, it will run POST maintenance tools, please select '2' – FTP update image, and fill in required information to start the image update.

Step 7. Once update finished, please select '4' to reboot the phone.

9.7 Common Trouble Cases

Trouble Cases
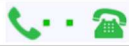
| Trouble Case	Solution
Phone could not boot up	1. The phone is powered by external power supply via power adapter or PoE switch. Please use standard power adapter provided or PoE switch met with the specification requirements and check if phone is well connected to power source. 2. If you saw "POST MODE" on the phone screen, the phone system image has been damaged. Please refer to the instructions in "**9.6 Entering Post Mode and Recover System Image**" to restore the system image.
Phone could not register to a service provider	1. Please check if phone is well connected to the network. The network Ethernet cable should be connected to the ▦ [Network] port NOT the ▯ [PC] port. If the cable is not well connected to the network icon ▭ [WAN disconnected] will be flashing in the middle of the screen. 2. Please check if the phone has an IP address. Check the system information, if the IP displays "Negotiating…", the phone 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 "9.5 Network Packet 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 is connected to the correct Handset (📞) port NOT Headphone (🎧) port. 2. The network bandwidth and delay may be not suitable for audio call at the moment.
Poor Audio or Low Volume in Headphone	1. There are two Headphone wire sequence in the market. Please use the Headphone provided by us, or consult the wire sequence if you wish to use a third party headphone. 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 speaker 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.

Appendix I - Icon Illustration

Keypad Icons

↻	Redial
🔊	Handsfree (HF) speaker
🔒	Phone lock = long-press key(#)
🎤̸	Mute microphone (during Call)
🔊+	Volume up
🔊−	Volume down

Status Prompt and Notification Icons

📞··☎	Call out
((☎))	Call in
⏸	Call hold
🖥	Network disconnected
🔒	Keypad locked
📞¹	Missed calls
✉¹	SMS
📧²	New voice message waiting
DND	Do-not-disturb activated on phone
📞	Auto-answering activated
📞	Call forward activated
🔊	Handsfree (HF) mode
🎧	Headphone (HP) mode
📞	Handset (HS) mode
🎤̸	Microphone muted
2aB	Mixed type input
abc	Lower case input
ABC	Upper case input
123	Numeric input

Appendix II – Text Input from Keypad

Look-up Table of Characters

Mode Icon	Text Mode	Key Button	Characters Of Each Press
123	Numeric	1	1
		2	2
		3	3
		4	4
		5	5
		6	6
		7	7
		8	8
		9	9
		0	0
		∗	∗#(space)@,.:/?<>[]%! &$~+-
		#	#
abc	Lower Case Alphabets	1	+ - * % / | @ ! :
		2	a b c
		3	d e f
		4	g h i
		5	j k l
		6	m n o
		7	p q r s
		8	t u v
		9	w x y z
		0	0
		∗	∗#(space)@,.:/?<>[]%! &$~+-
		#	#

ABC	Upper Case Alphabets	1	+ - * % / | @ ! :
		2	A B C
		3	D E F
		4	G H I
		5	J K L
		6	M N O
		7	P Q R S
		8	T U V
		9	W Z Y X
		0	(space)
		✳	✳#(space)@,.:/?<>[]%! &$~+-
		#	#
2aB	Mixed type input	1	1 + - * % / | @ ! :
		2	2 a b c A B C
		3	3 d e f D E F
		4	4 g h I G H I
		5	5 j k l J K L
		6	6 m n o M N O
		7	7 p q r s P Q R S
		8	8 t u v T U V
		9	9 w z y x W Z Y X
		0	(space)
		✳	✳#(space)@,.:/?<>[]%! &$~+-
		#	#

Appendix III – LED Definition

DSS KEY LED State

Type	LED Light	State
Line Key	Off	Line inactive
	Green On	Line ready (registered)
	Green Blinking	Dialing / ringing / call holding
BLF	Green On	Subscribed number is idle.
	Red On	Subscribed number is busy.
	Red Blinking	Subscribed number is dialing.
	Off	Subscribed number is unavailable
Headset Key	Green On	Enable headset
	Off	Disable headset
MWI	Green On	New voice message waiting
	Off	No new voice message