ESI eSIP and eCloud

ePhone3 V2 User's Guide

This guide instructs the user on ePhone3 V2 features and how to use them.

 0455-0301 Rev C

Contents

Safety

Please read the following safety notices.

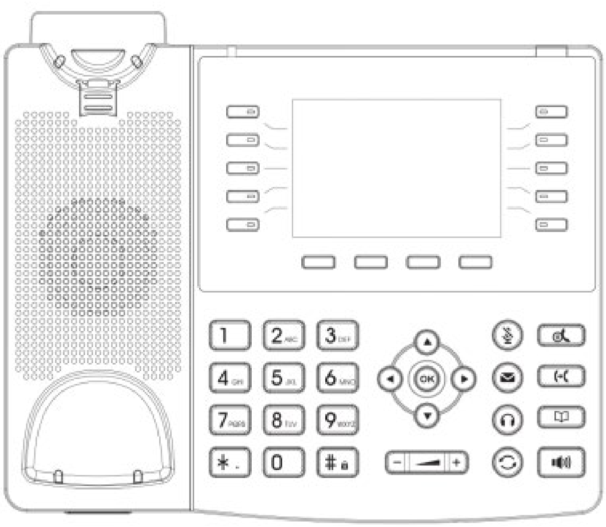
- If using a power supply, please use the power supply offered by ESI. Other power supply may cause damage to the phone, affect the behavior or induce noise.
- Verify that building power is good. Bad power may cause noise, fire or damage.
- Do not damage the power cord. If power cord or plug is damaged, it may cause fire or electric shock.
- Do not expose the phone to extreme forces such as dropping it. Rough handling can damage it.
- This phone is designed for indoor use. Do not install the phone in places where there is direct sunlight. Do not get the phone wet. Do not install the phone in a poorly ventilated area.
- Avoid exposing the phone to high temperature or below 32°F (0°C) or high humidity.
- Do not attempt to open the phone. It could be damaged or induce electric shot. It will void warranty.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean the phone. Clean the phone with a soft cloth that has been slightly dampened in a mild soap and water solution.
- Use caution when lightening is in the area.
- Before installing equipment and cabling, be familiar with the facility and any hazards such as electrical wires in the walls.

Overview

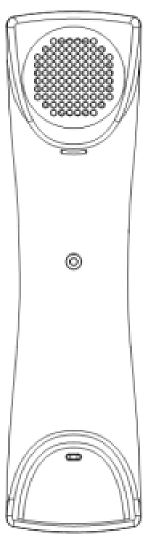
Introduction

The ESI ePhone3 V2 Phone is a cost effective feature rich phone that offers HD Audio, echo cancellation and more. It is an ideal choice for day to day office communications.

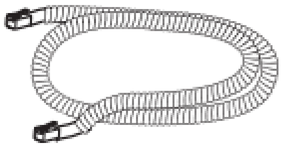
Packing Contents



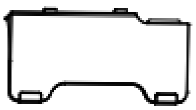
Phone



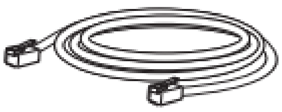
Handset



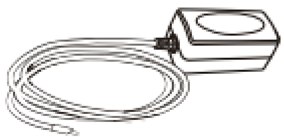
Handset cord



Stand



Network cable



Power adapter (optional)

Desktop Installation

PoE and external power adapters

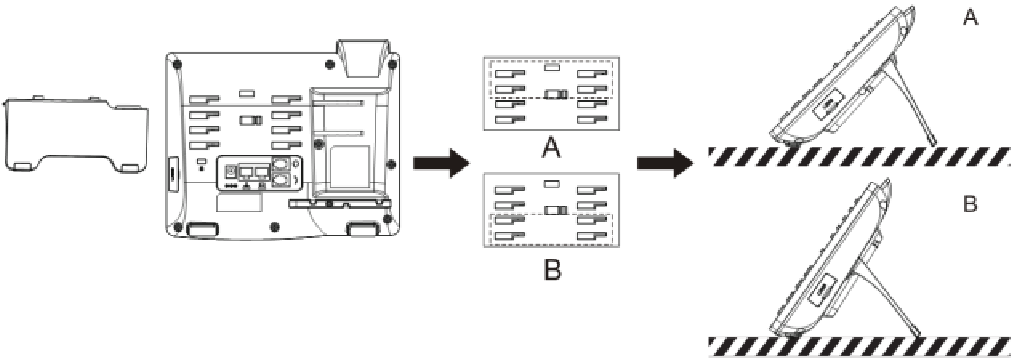
The phone can be powered up two ways, external power adapter or Power over Ethernet (PoE).

With a PoE switch, the phone is provided power through a single Ethernet cable which is also used for data.

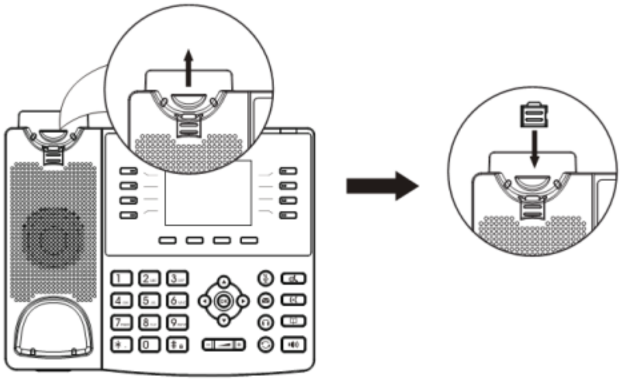
For users who do not have PoE equipment, a traditional power adaptor that plugs into a wall outlet should be used. If the phone is connected to a PoE switch and power adapter at the same time, the power adapter will be used in priority and will switch to PoE if the power adapter fails.

Desktop installation

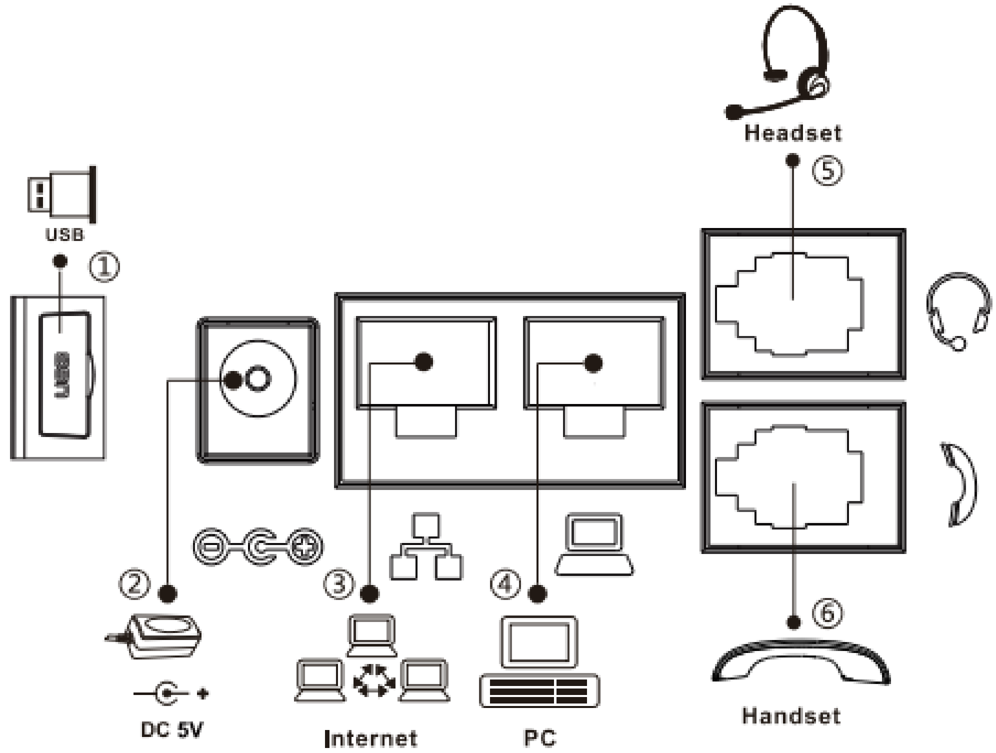
Follow the instructions in the image below to install the phone stand.



Orient handset hook to desired orientation as shown below.



Connect the optional power adapter, network, PC, handset and headset to the appropriate port as shown in the below.



Index	Interface	Description
①	USB Port (on side of phone)	Connect USB Disk for recordings
②	Power Port	Connect Power Adapter
③	Network Port	Connect to LAN or Internet
④	PC Port	Network pass through for Connecting Computer
⑤	Headset Port	Connecting Headset
⑥	Handset Port	Connect Handset

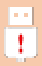
Icons

Keypad Icons

Icon	Description
	Redial
	Contacts (Phone Book)
	Hands-free speaker
	Mute Microphone (During Call). Mute Ring (while phone is idle)
	Volume up / down
	Hold
	Headset
	Voicemail (MWI)
	Transfer

Status Prompt and Notification Icons

Screen Icon	Description
	Call out
	Call in
	Call Hold
	Network Disconnected
	Open VLAN
	Open VPN
	Keypad Locked
	Call forward calls
	Outgoing calls
	Incoming calls
	Missed calls
	SMS
	New voice message waiting

	Do-Not-Disturb inactivated on Phone
	Call forward activated
	Auto-answering activated
	Hands-free (HF) Mode
	Headphone (HP) Mode
	Handset (HS) Mode
	Mute Microphone
	Voice call quality
	Voice encryption while on a call
	High Definition Audio
	Record
	SIP Hotspot
	NA
	NA
	USB Insert
	USB overload

Keypad 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
		*	*.+
		#	#
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	(space)
		*	.,*/+-:_=
		#	# ^!&$%
ABC	Upper Case Alphabets	1	@:;()<>
		2	A B C
		3	D E F
		4	G H I
		5	J K L
		6	M N O
		7	P Q R S
		8	T U V
		9	WZYX
		0	(space)
		*	.,*/+-:_=
		#	# ^!&$%
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	0
		*	.,*/+-:_=
		#	# ^!&$%

DSS key LED state definitions

Type	LED Light	State
Line Key	Off	Line inactive
	Green On	Line ready (Registered)
	Green Blinking	Ringing
	Red Blinking	Line is trying to register
	Red Blinking	Line error (Registration failure)
	Red On	Dialing/Line in use (Talking)
	Yellow Blinking	Call holding
BLF	Green On	Subscription number is idle.
	Red On	Subscription number is busy.
	Red On	Subscription number is dialing.
	Off	Subscription number is unavailable.
Presence	Green On	Subscription number is idle.
	Red On	Subscription number is busy.
	Red On	Subscription number is dialing.
	Off	Subscription number is unavailable.
DND	Red On	Enable DND
	Off	Disable DND
MWI	Green Blinking	New voice message waiting
	Off	No new voice message

User Introduction

Keypad

Number	The keypad names	Instruction
1	Side Keys	The keys along the side of the display can be programmed to perform various functions.
2	Soft-menu Keys	These four keys provide different functions which correspond to the soft-menu displayed on the screen.
3	Keypad	The 12 standard telephone keys provide the same functionality as standard telephones. Some keys also provide special a function by long-pressing the key. Example: Key # - Long-pressed to lock the phone.
4	Navigate/OK Keys	Navigate keys: Press the up/down navigation key to move the cursor in a list. On some Settings, the user can press the left/right navigation key to change options or move the cursor. OK key: Press OK to confirm an action.
5	Volume Down, Volume Up Key	While the phone is in the idle standby state, press this key to increase or decrease ringing volume. While on a call, press this key to increase or decrease the volume.

6	Mute Key	During a call, the user can press this key to mute the microphone.
7	Hold Key	Press the Hold key during a call to put the call on hold, and press the Hold key again to resume the call.
8	Voice Mail	Press the Voice Mail key to enter the voicemail list.
9	Transfer Key	Press the Transfer key to transfer a call to another extension.
10	Headset Key	Press Headset key to enable headset.
11	Contacts Key	Press the Contact key to enter contacts, which is also referred to as address book and phone book.
12	Redial	Press the Redial key to redial the last number dialed
13	Hands-free Key	Press the Hands-free key to enable speakerphone.

Using Handset / Hands-free Speaker / Headset

- Using Handset
  To talk over handset, lift the handset and dial the number, or dial the number first, then lift the handset and the number will be dialed.
- Using Hands-free Speaker
  To talk over hands-free speaker, user should press the hands-free key then dial the number, or just dial the number.
- Using Headset
  To use headset, press the headset key then dial the number, or dial the number then press the headset key.
- Using Line Keys (Defined by DSS Key)
  User can make or answer a call on a specific line using a DSS key that has been programmed as a handsfree key, answer key, etc.

Idle Screen

The image above shows the default standby screen, which is the user interface most of the time.

The upper half of the home screen shows the status and information such as voice messages, missed calls, auto answer, do not disturb, lock status, network connection status, etc.

The lower half of the screen is the function menu keys, through which users can operate the phone. Users can restore the phone to the default standby screen interface by picking up and dropping the handle.

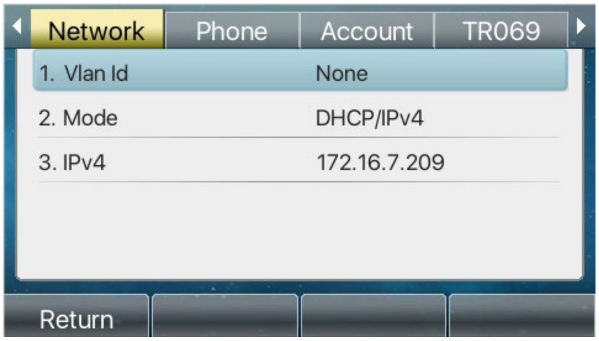
The left and right side of the screen show the side key configurations, which can be customized by the user.

Phone Status

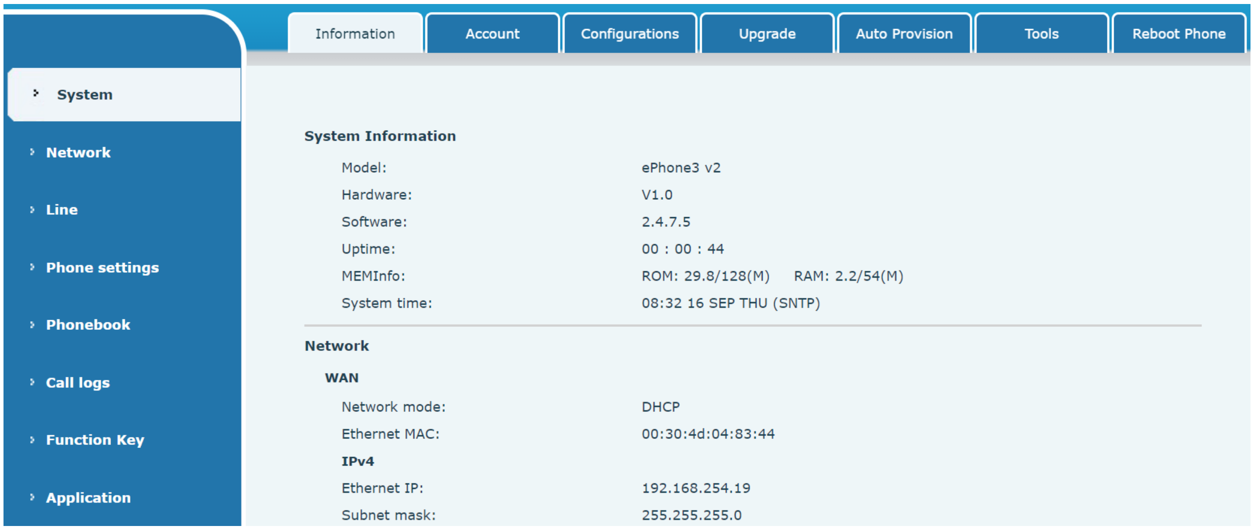
From the phone, press the Menu key then select Status. The phone status includes the following phone information:

- Network status:
  VLAN ID
  IPv4 or IPv6 status
  IP Address
  Network Mode
- Phone Information:
  Mac Address
  Phone Model
  Hardware Version number
  Software Version number
  Phone Storage (RAM and ROM)
  System Running Time
- SIP Account Information:
  SIP Account
  SIP Account Status (register / uncommitted / registering / time out)
- TR069 Connect Status (Displays only in the phone interface state)
  The user can view the phone status through the phone interface and the web interface.

To view phone status from the phone, press Menu then select Status.

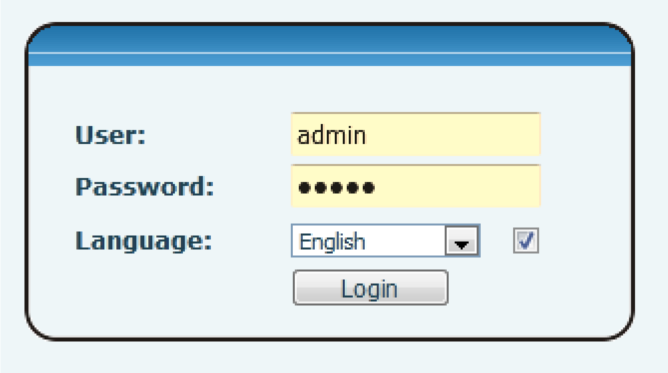


WEB interface: Refer to **Web Management**. Log in the phone page and navigate to [System] >> [Information] as shown below:



Web Management

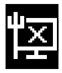
Phone can be configured and managed from its web interface. Enter the IP address of the phone and log in. The user can find the IP address of the phone by navigating to [Menu] >> [Status] at the phone.



User must correctly enter the user name and password to log in to the web page. The default user name and password are "admin". For specific details on the web interface, refer to **Web Configurations** section.

Network Configurations

The phone requires an IP network to operate. From the phone, go to [Menu] >> [Advanced Settings] >> [Network] >> [Network].
The default password for advanced settings is 123.

NOTICE! If user sees  'WAN Disconnected' icon flashing in the middle of screen, it means the network cable is not correctly connected to the device's network port. Check that the cable is connected correctly between the phone and to the network switch.

The phone supports three types of networks; IPv4, IPv6, and IPv4&IPv6
There are three common IP configuration modes for IPv4:

- DHCP (Dynamic Host Configuration Protocol) – The phone will automatically retrieve its network configuration from a DHCP server. Users do not need to configure any parameters manually. This is recommended for the most users. This is the phones default network setting.
- Static IP Configuration – This option allows user to configure each IP parameters manually, including IP Address, Subnet Mask, Default Gateway, and DNS servers. This method requires network knowledge to implement.
- PPPoE - This option is often used to connect the phone to a broadband modem or router. To establish a PPPoE connection, user should configure username and password provided by the service provider.

  The device is default configured in DHCP mode.

There are two common IP configuration modes for IPv6

- DHCP (Dynamic Host Configuration Protocol) – The phone will automatically retrieve its network configuration from a DHCP server. Users do not need to configure any parameters manually. This is recommended for the most users. This is the phones default network setting.
- Static IP Configuration – This option allows user to configure each IP parameters manually, including IP Address, Subnet Mask, Default Gateway, and DNS servers. This method requires network knowledge to implement. Refer to **Network Settings** for details.

SIP Configurations

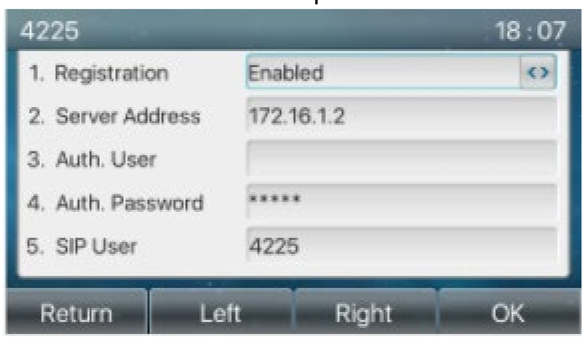
A line must be configured properly for telephony service. When the phone line is configured, the phone will register to the service provider with the server's address and user's authentication as stored in the configurations.

The user can configure the interface at the phone or the webpage, and input the corresponding information at the registered address, registered user name, registered password, SIP user and registered port respectively, which are provided by the SIP server administrator.

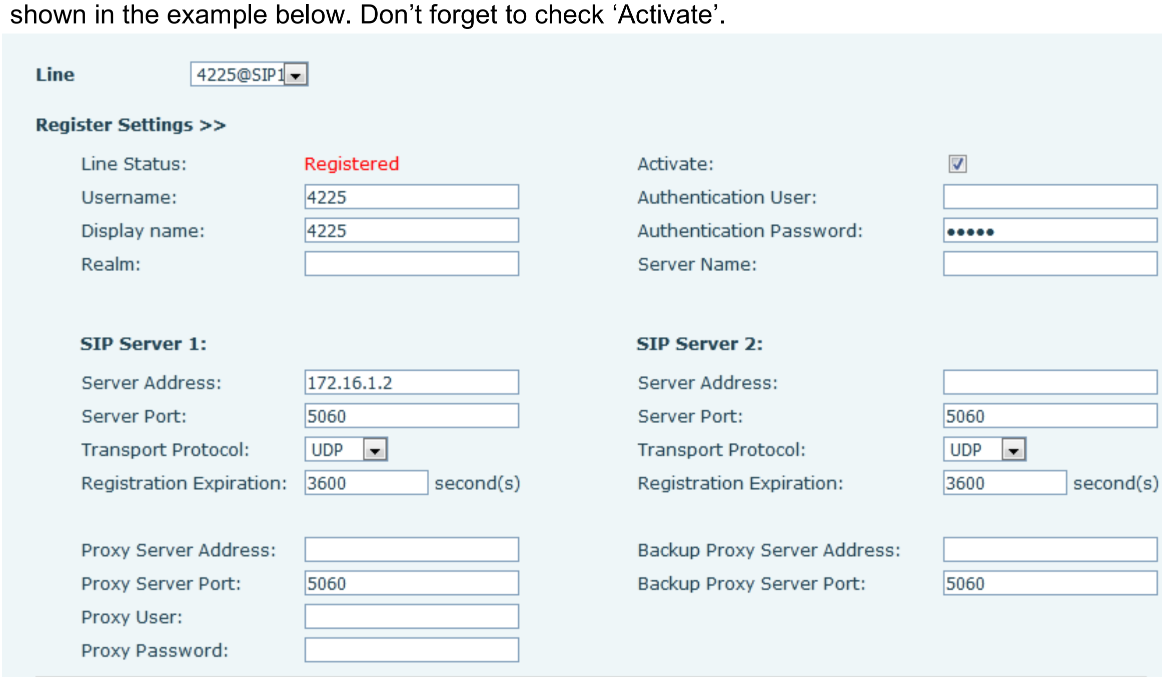
- Phone interface: To manually configure a line, the user can long press a line key, or configure a line. Go to [Menu] >> [Advanced] >> [Accounts] >> [SIP n]. When done configuring the line, select OK to save the configuration.

NOTICE! User must enter correct password to access Advanced settings. The default password is 123.

The Basic parameters of the SIP line are shown in the picture below.



- WEB interface: After logging into the phone page, enter [Line] >> [SIP] and select the line to configure from the drop down list. Enter SIP register information and click Apply to complete registration as shown in the example below. Don't forget to check 'Activate'.

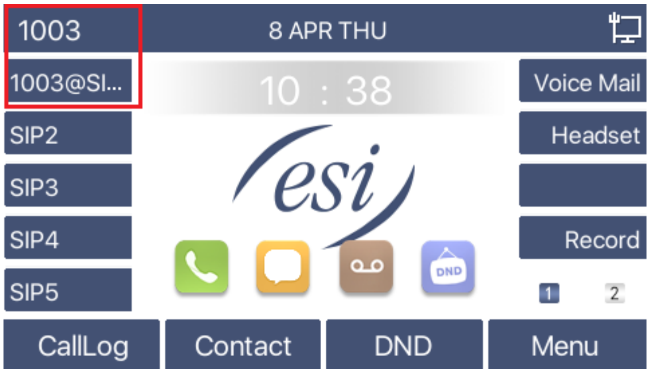


Basic Function

Making Phone Calls

- Default Line

The device provides 8 lines of service. If two lines are configured, user can make or receive phone calls on either line. If default line is configured by user, there will be a default line for making outgoing calls indicated in the top left corner of the screen. To change the default line, user can press left/right navigator keys to switch between two lines
Enable or disable default line: From the phone, press [Menu] >> [Features] >> [General] >> [Default Line] and toggle to Enabled or Disabled. Press OK to save.
From the Web Interface go to [Phone Settings] >> [Features], and check or uncheck [Enable Default Line].



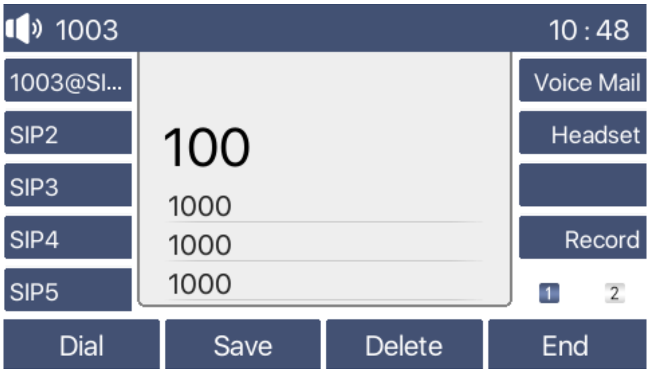
- Dialing Methods

User can dial a number by,

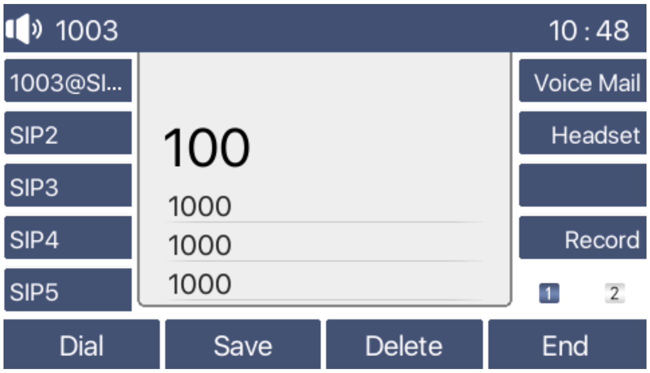
  - Entering the number directly.
  - Selecting a phone number from Local Contacts. Refer to **Local Contacts**.
  - Selecting a phone number from Cloud Contacts. Refer to **Cloud Phonebook**.
  - Selecting a phone number from Call Logs (Refer to **Call Log**.
  - Press Redial to dial the last dialed number.

- Dialing Number then Open Audio

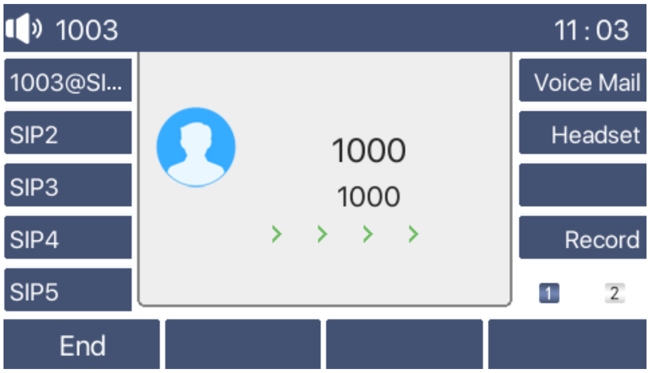
  To make a phone call, user can dial a number by one of the above methods. When the dialed number is completed, user can press [Dial] key on the soft-menu, or press hand-free key to turn on the speaker or headphone, or lift the handset to call out with the current line, or user can press line key (Configured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



- Go offhook then Dial the Number
  Go offhook by lifting the handset, or pressing the hands-free speaker key, or pressing the headset key, or pressing a line key, and then dial the number with one of the above methods. When number has been dialed, user can press [Dial] key or [OK] key to call out, or the number will be dialed automatically after timeout.

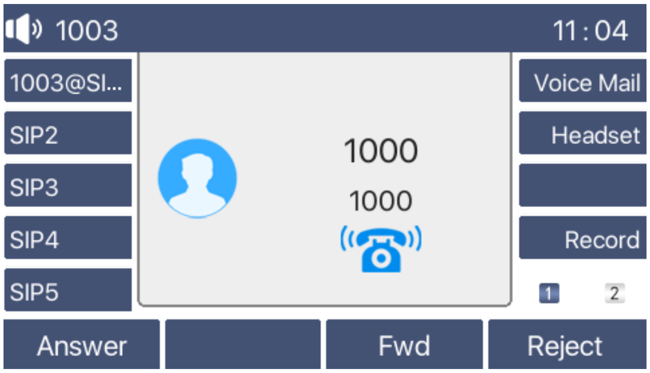


- Cancel Call
  While calling the number, user can disconnect the call by placing back the handset, pressing the hands-free key or by pressing the End key.



Answering Calls

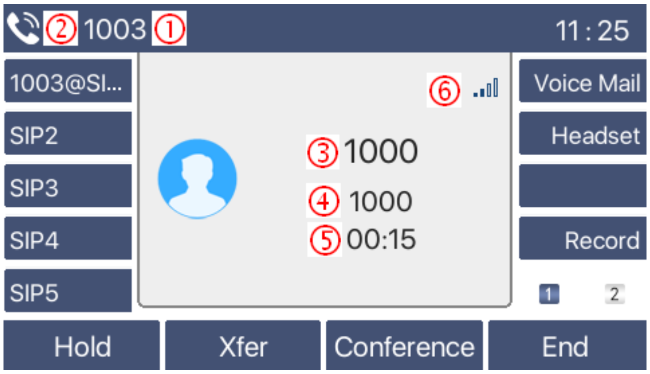
When there is an incoming call while the device is idle, the user will see the incoming call on the screen as shown below.



User can answer the call by lifting the handset, by pressing the headset key, by pressing the hands-free speaker key, or by pressing the Answer key. To reject the incoming call, press Reject key.

Talking

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

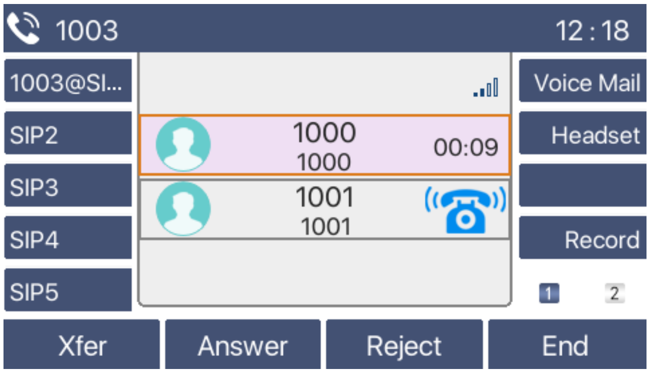


Number	Name	Description
①	Default line	The line that is currently in use by the phone.
②	Voice channel	The voice channel mode being used (handset, handsfree, headset).
③	Current call	The call that is currently in progress.
④	Call status	Shows the calls that are on a line.
⑤	Call Duration	The duration of the current call.
⑥	Speech quality	Displays the current voice quality of the call.

Make / Receive Second Call

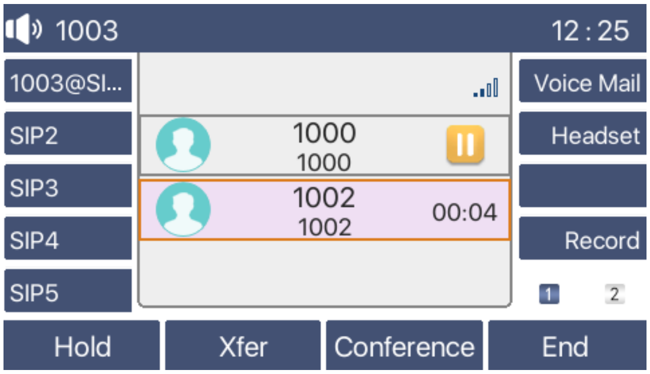
The device can support up to two concurrent calls. When there is already a call established, user can still answer another incoming call on either line or make a second call on either line.

- Second Incoming Call
  When there is another incoming call during a phone call, this call will be waiting for user to answer. User will see the call message in the middle of current screen. The device 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 the same way as a normal incoming call. When the waiting call is answered, the first call will be placed on hold automatically.



- Second Outgoing Call
  To make a second call, press Xfer or Conf key 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 or dial out from the configured Keys (BLF/Speed Dial).

- Switching between Two Calls
  When there are two calls established, user will see a dual calls screen as the following picture.



Press up/down navigator keys to switch between callers. Press the Resume key to pick up the caller.

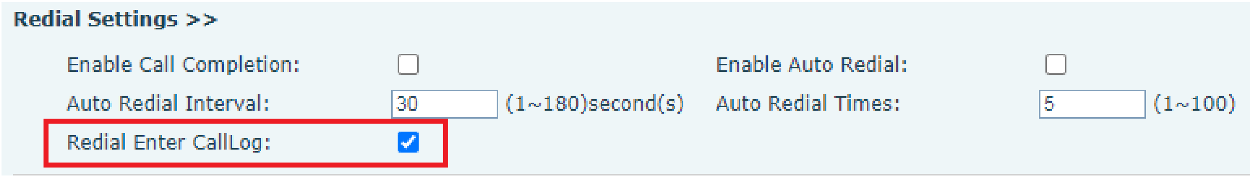
- Ending One Call
  Hang up the current call by pressing the End key. The phone will return to single call mode in holding state.

Ending the Call

End a call by hanging up the handset, pressing the hands-free key or by pressing the End key.

Redial

- Redial the last outgoing number:
  When the phone is in standby mode, press the redial key and the phone will call the last outgoing number.
- Call any number with the redial key:
  Enter the number, press the redial key, and the phone will call the number.
- Press the redial key to view call records:
  Log into the phone web page and go to [Phone Settings] >> [Features] >> [Redial Settings]. Check Redial Enter CallLog. Now user can press the Redial key to enter the call records page when the phone is idle.



Dial-up Query

Dial-up inquiry function is enabled by default. Go offhook and enter two or more numbers. The dial interface will automatically try to match the numbers dialed. Use the up/down navigation keys to select the number, press the Dial key to dial the number.

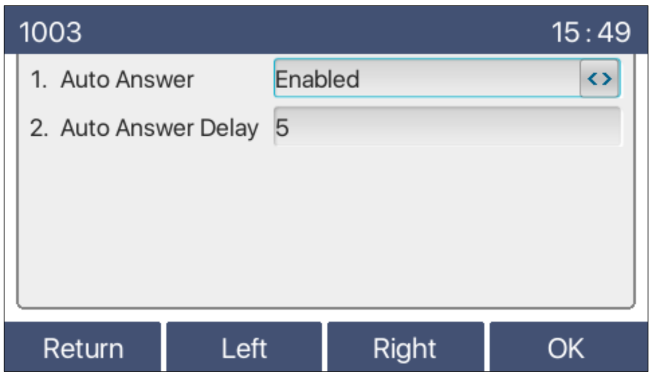
Auto-Answering

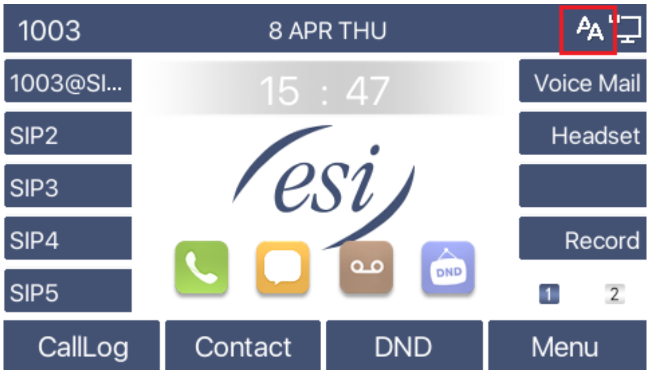
When enabled, any incoming call will be automatically answered (not including call waiting). Auto-answering can be enabled on line by line basis.

The user can enable the automatic answer function at the phone interface or the webpage interface.

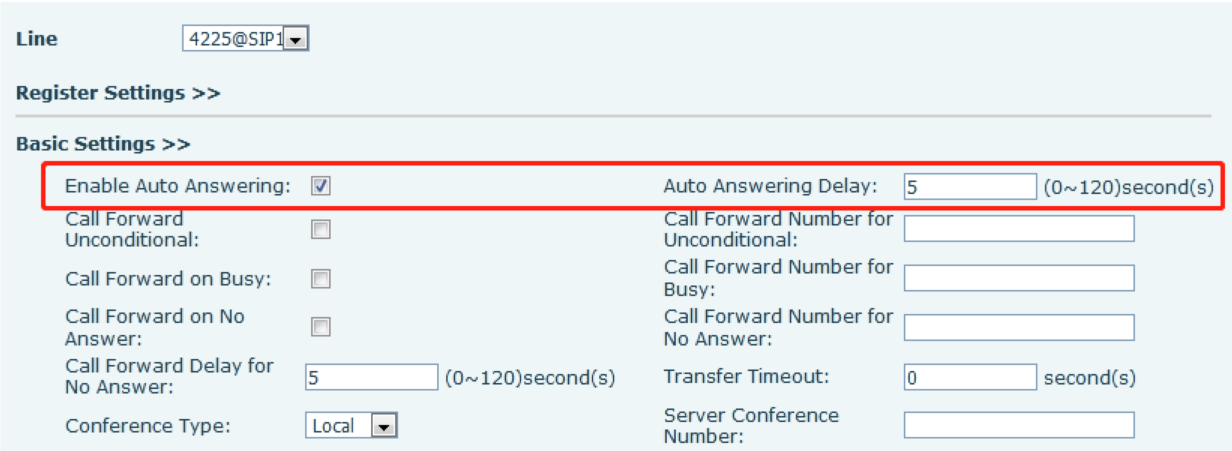
- Phone interface:
  While the phone is idle, press [Menu] >> [Features] >> [Auto Answer].
  Scroll to a line and press OK to select the line. Use the left/right navigation keys to enable/disable auto answer, then scroll down to the next line and set the Auto Answer Delay. Default is 5 seconds.
  When done, press [OK] key to save.

The icon in the upper right corner of the screen  indicates that auto answer is enabled.





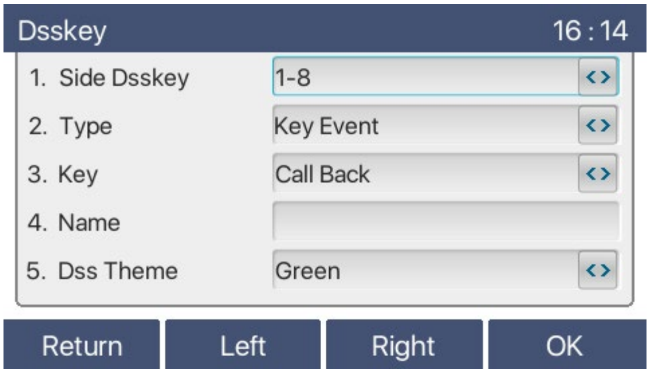
- WEB interface:
  Log in the phone web page, enter [Line] >> [SIP] >> select a SIP line from the drop down list >> [Basic settings], check Enable Auto Answering. Set Auto Answer Delay and click Apply.



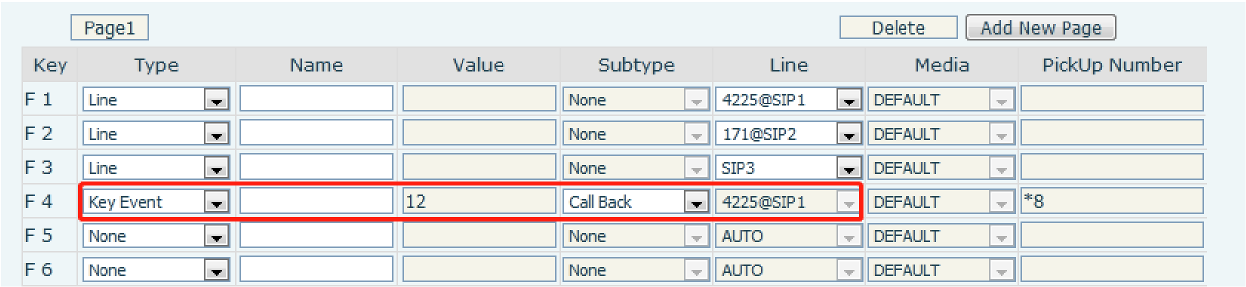
Callback

A Callback key, like the Redial key, will dial back the number of the last call. If there is no call history, pressing the Callback key will result with the message "can't process".

- Set the callback key through the phone interface:
  From the phone, press [Menu] >> [Basic] >> [Keyboard] >> [DSS Key Settings, or Soft DSS Key Settings]. Select the function key to program, set key Type to Key Event, set key to Call Back, enter the key name and press OK to save.



- Set the callback key through the web interface:
  Log in to the phone web page. Go to [Function Key] >> [Side Key] or [Softkey]. Select a function key to program, set the type as Key Event, and set the subtype as the Call Back, as shown below:



Page1 Delete Add New Page

Key	Type	Name	Value	Subtype	Line	Media	PickUp Number
F 1	Line			None	4225@SIP1	DEFAULT	
F 2	Line			None	171@SIP2	DEFAULT	
F 3	Line			None	SIP3	DEFAULT	
F 4	Key Event		12	Call Back	4225@SIP1	DEFAULT	*8
F 5	None			None	AUTO	DEFAULT	
F 6	None			None	AUTO	DEFAULT	

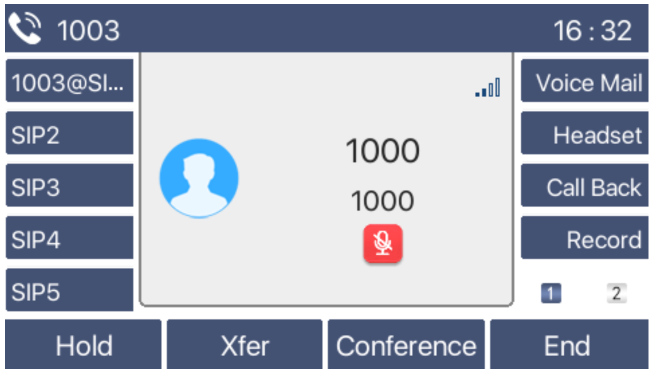
Mute

Press the mute key during a call to disable the microphone so that the local voice is not heard. Normally, mute mode is automatically turned off at the end of a call. User can also press mute to silence the ringtone when there is an incoming call.

Mute mode can be enabled in all call modes (handset, headset or hands-free).

Mute the Call

- During the conversation, press the mute key on the phone. A red mute icon is displayed on the phone screen, as shown in the figure below:

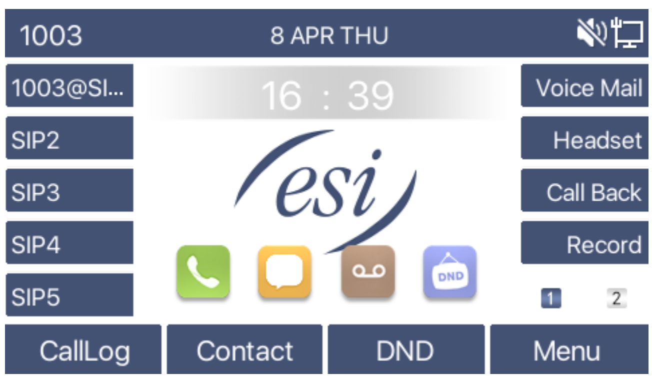


- Cancel mute: Press Mute key again to turn off mute. The red mute icon will no longer be displayed on the call screen.

Ringing Mute

- Mute: Press the mute key when the phone is in standby mode:

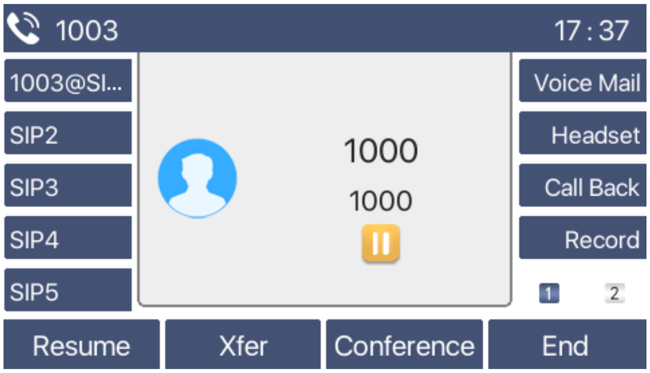
  The top right corner of the phone will show a mute speaker icon . The status lamp in upper right corner of phone will blink red when there is an incoming call. The phone will display the incoming call but will not ring.



- Cancel ring tone mute: On the standby or incoming call screen, press the mute key again to cancel ring tone mute. The mute speaker icon will no longer show in upper right corner and the phone will ring as normal when there is an incoming call.

Call Hold/Resume

Press the Hold key to put a call on hold. The Hold key will change to a Resume key. Press the Resume key to retrieve the call.



DND

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

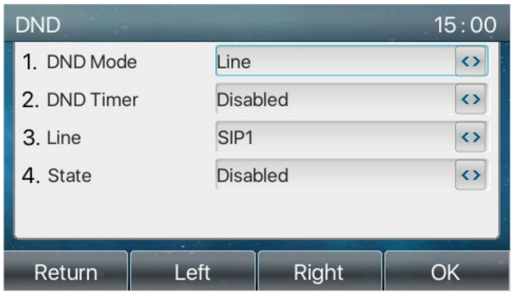
Enable/Disable DND for all lines and all calls：

- From the phone:
  - Press DND key to enable the DND. All calls will go to voicemail.
  - Press DND key again to disable DND. All calls will ring to the phone.

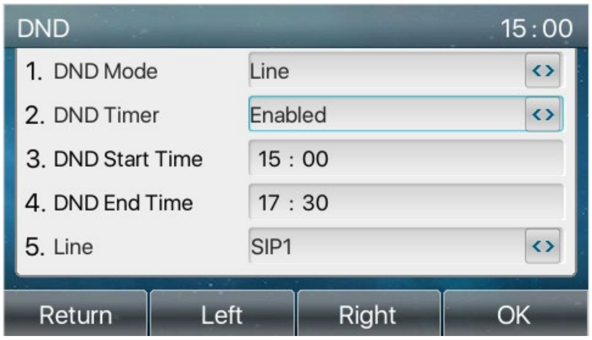
Enable/disable Do Not Disturb on a specific line:

- From the phone menu:
  - Press [Menu] >> [Features] >> [DND].
  - For DND Mode, use the left/right navigation keys to select Line. Use the up/down navigation keys to scroll down to DND Timer and enable or disable the timer. Select which line to place into DND, toggle State to Enabled and press [OK] to save1

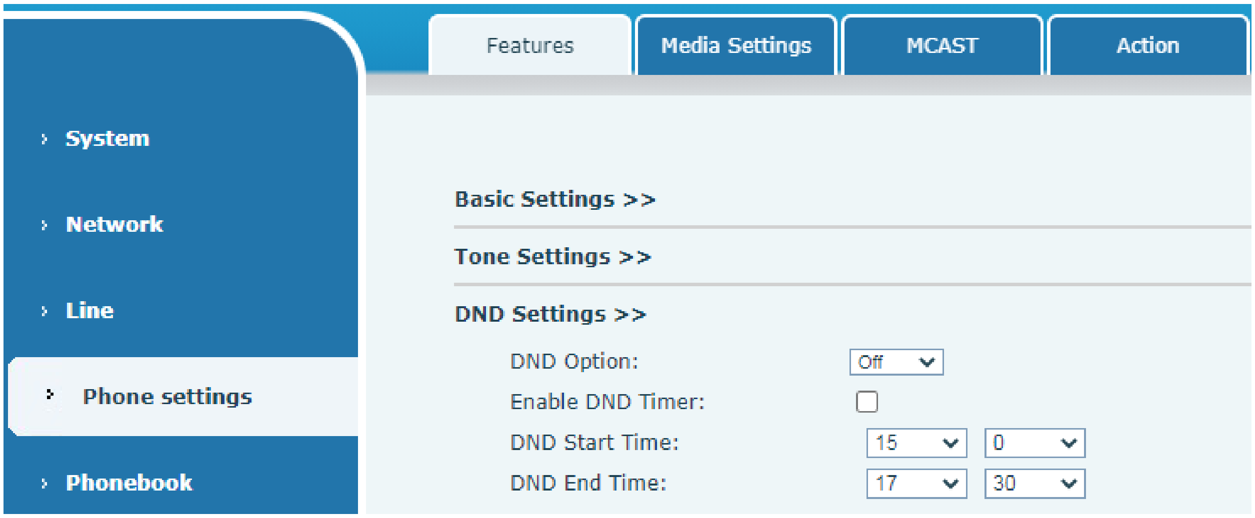
The user will see the DND icon turn red, and the sip-line will be in DND.



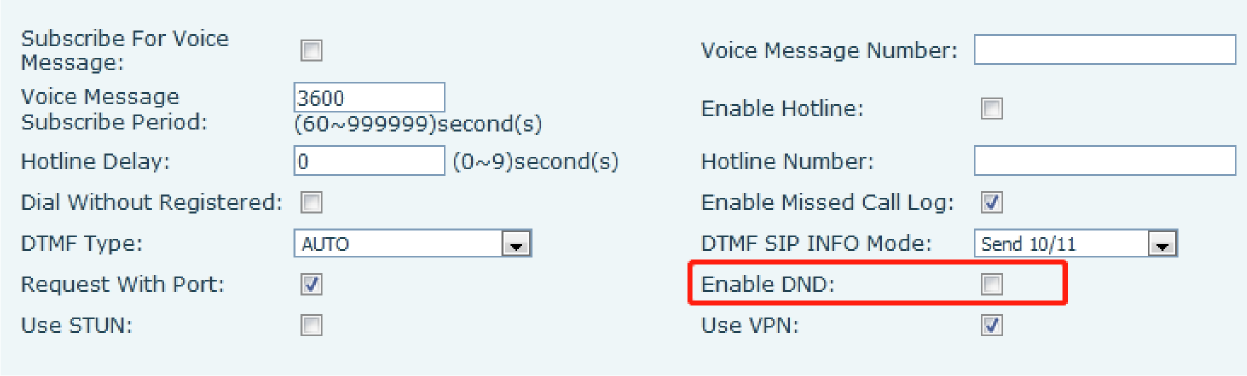
The user can also use the DND timer by using the left/right navigator keys to enable DND Timer. After enabling the DND timer, set the DND Start Time and DND End Time then press OK to save. DND will automatically activate at the designated time and the DND icon will turn red.



- WEB interface: Go to [Phone setting] >> [Features] >> [DND settings]. Set the DND Option (off, phone, line). Enable DND Timer, if needed. Set DND Start Time and DND End Time, if needed, and click Apply.



To enable DND for a specific Line route on the web page go to [Line] >> [SIP], select a [Line] from the drop down list >> [Basic settings], check Enable DND and click Apply.



Call Forward

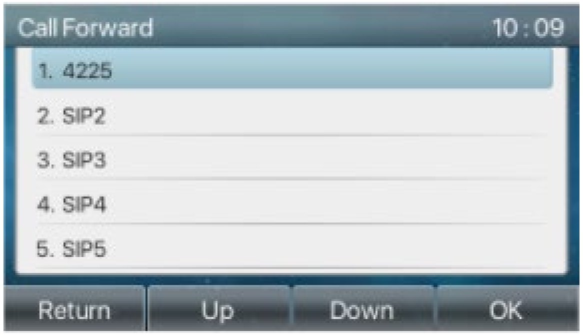
Call forward diverts an incoming call to a specific number based on conditions and configurations. User can configure the call forward settings of each line.

There are three types of Call Forwarding,

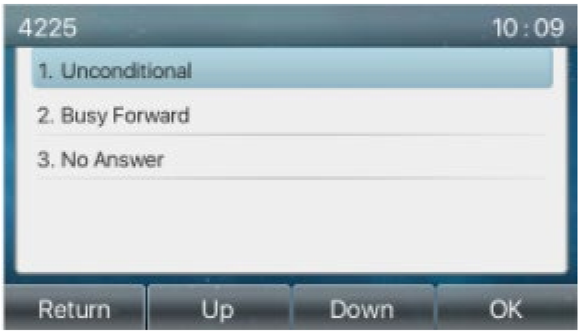
- Unconditional – Forward any incoming call to the configured number.
- Busy Forward – Forward incoming call when user is on phone.
- No Answer – Forward incoming call if user does not answer within a configured delay time.

Phone interface: Default standby mode:

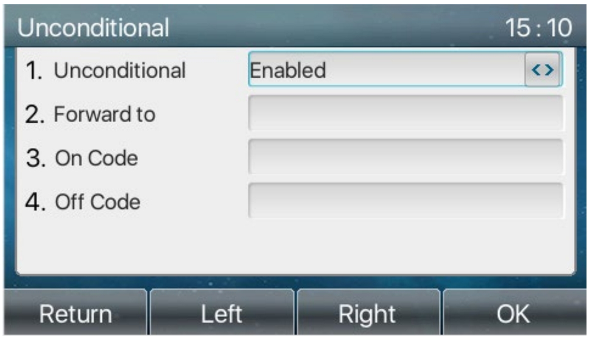
- Press [Menu] >> [Features] >> [Call Forward], select a line using up/down navigation keys and press OK.



- Select the call forward type by pressing the up/down navigation keys and press [OK].



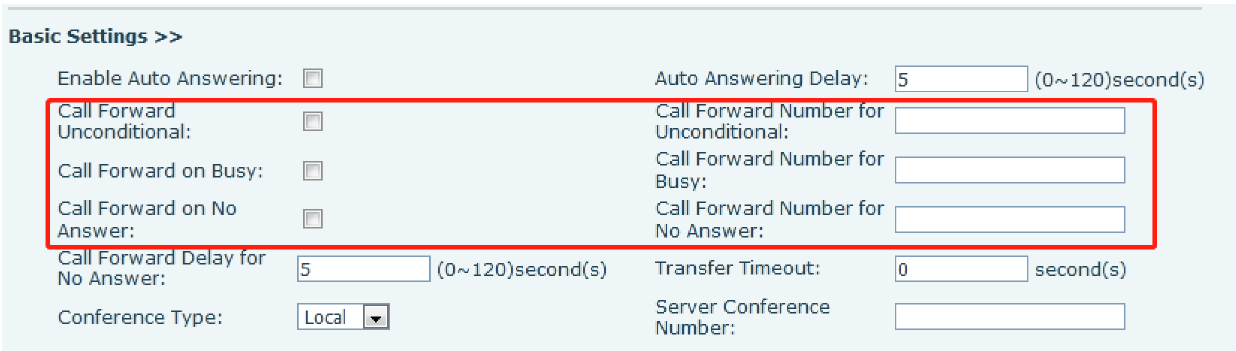
- Select enable/disable by pressing the left/right navigation keys.



- Use the up/down navigation keys to enter the required information in the remaining fields. When finished, press the [OK] key to save changes.

WEB interface:

- Enter [Line] >> [SIP], Select a [Line] >> [Basic settings].
- Check Call Forward Unconditional, Call Forward on Busy, or Call Forward on No Answer.
- Set Call Forward Delay for No Answer and Call Forward Number.
- Click Apply.



Call Transfer

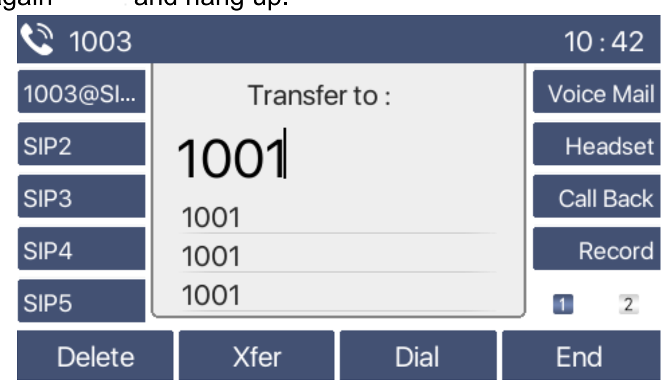
When the user is talking with a remote party and wishes to transfer the call to another remote party, there are three way to transfer the call, blind transfer, attended transfer and Semi-Attended transfer.

- Blind transfer: No need to negotiate with the other side, directly transfer the call to the other side.
- Semi-Attended transfer: When you hear the ring back, transfer the call to the other party.
- Attended transfer: When the caller answers the call, transfer the call to the other party.

For more transfer settings, refer to **Line >> Dial Plan**.

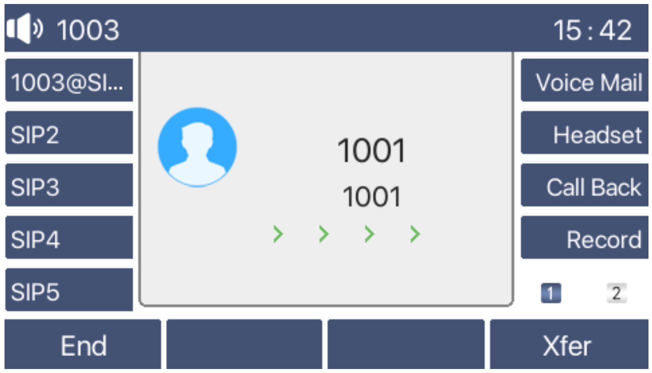
Blind transfer

During a call, press Transfer key or Xfer soft key on the phone, enter the number to transfer to, press the Transfer key again and hang up.



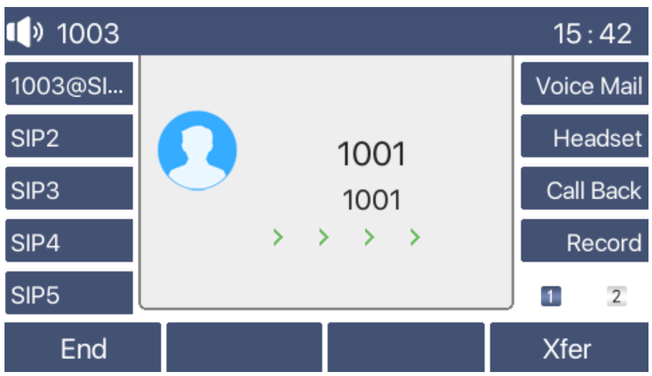
Semi-Attended transfer

During the call, press the Transfer key or Xfer soft key on the phone, enter the number to transfer to. When ringback is heard, press the Transfer key again and hang up.



Attended transfer

Attended transfer is also known as "courtesy mode", which is to transfer the call by calling the other party and waiting for the other party to answer the call then announcing that a call will be transferred to them. While on a call, press the Transfer key, wait for the other party to answer, announce the call then hang up.

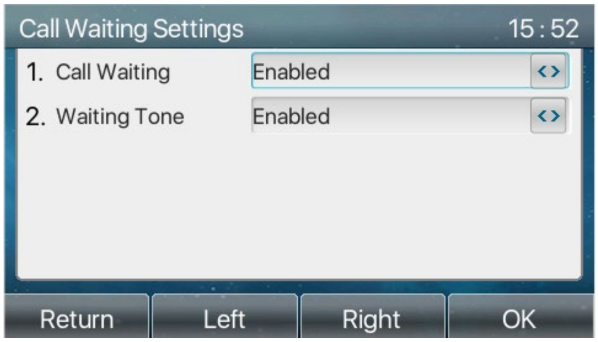


Call Waiting

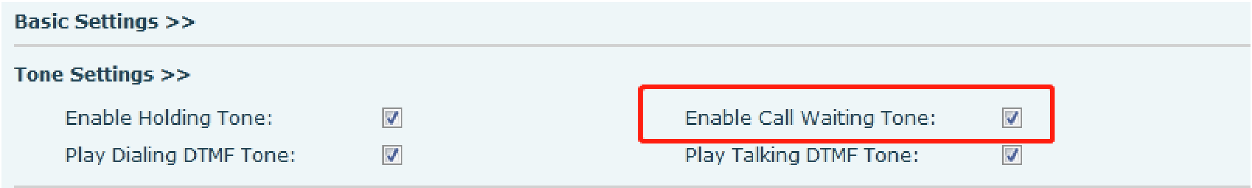
- Enable call waiting: new calls can be accepted during a call.
- Disable call waiting: new calls will be automatically rejected and a busy tone will play.
- Enable call waiting tone: when a new call is received while on a call, user will hear a beep, which indicates an incoming call.

The user can enable/disable the call waiting at the phone interface and the web interface.

- Phone interface: Press [Menu] >> [Features] >> [Call waiting] >> [Call Waiting Settings]. Use navigation keys to enable Call Waiting and Waiting Tone.



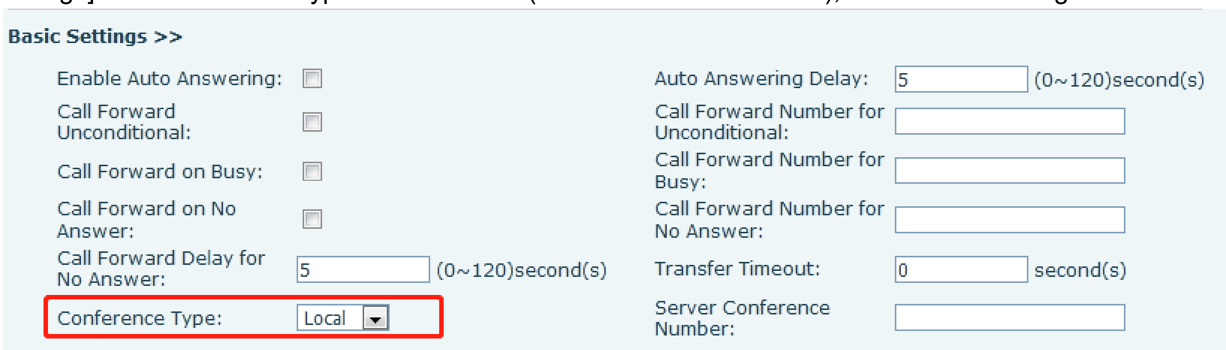
- WEB interface: Enter [Phone Settings] >> [Features] >> [Basic Settings], enable/disable call waiting by checking or unchecking Enable Call Waiting. Go to [Phone Settings] >> [Features] >> [Tone Settings], enable/disable call waiting tone by checking or unchecking Enable Call Waiting Tone.



Conference

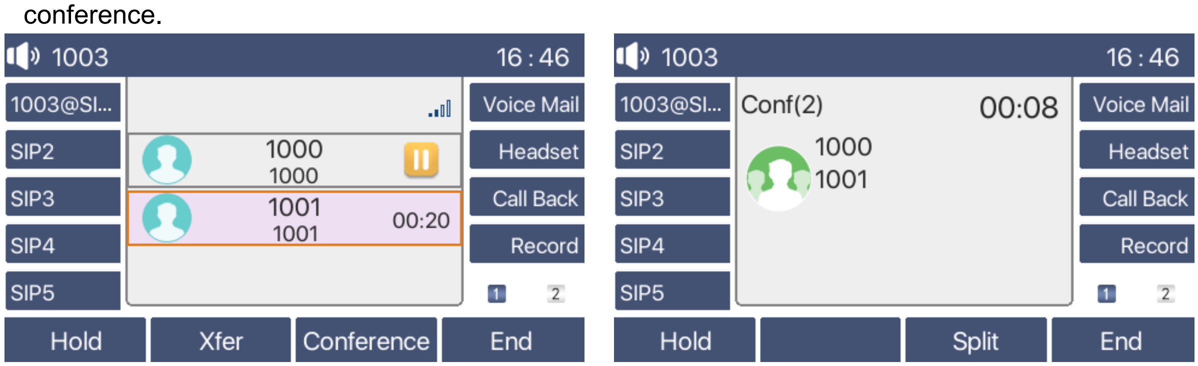
Local Conference

To conduct local conference, log in the phone webpage and enter [Line] >> [SIP], select a line, >> [Basic settings]. The conference type is set as local (the default is local mode), as shown in the figure.



Two ways to create a local conference:

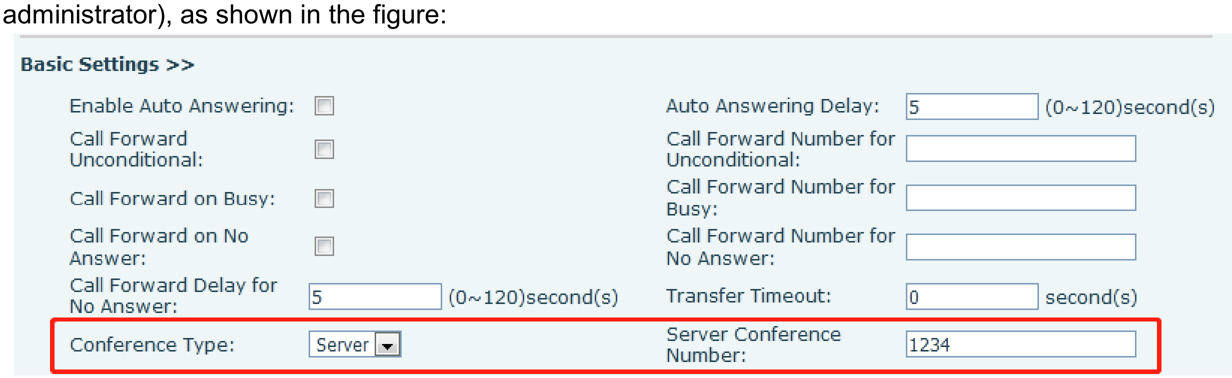
- Dial a number or select it from Contacts and wait for called party to pick up.
- When call is established, press the Conference key and dial the next number to add to the conference.
- When call is answered, press Conference key again and all three parties will be connected to the conference.



Note: During the conference, press the Split soft key to split the conference. Use the up/down Navigation keys and Resume soft key to toggle between the two conference members, User can also put users in the conference on Hold or press End to end the call with one of the conference members.

Network Conference

Users need server support for network conference.
Log in the phone web page, enter [Line] >> [SIP], select a line, >> [Basic settings], set the conference type as server mode (default is local mode), set the server conference room number (consult your system administrator), as shown in the figure:



Method to join a network conference:

- Call number of network conference room, enter the password then all enter the conference room.
- Two phones have established common calls. Press the conference key to invite new members to the conference. Follow the voice prompt to operate.

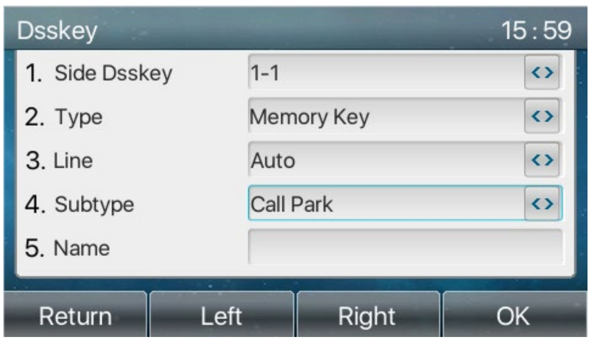
Note: The upper limit of the number of participants in the network conference varies according to the server.

Call Park

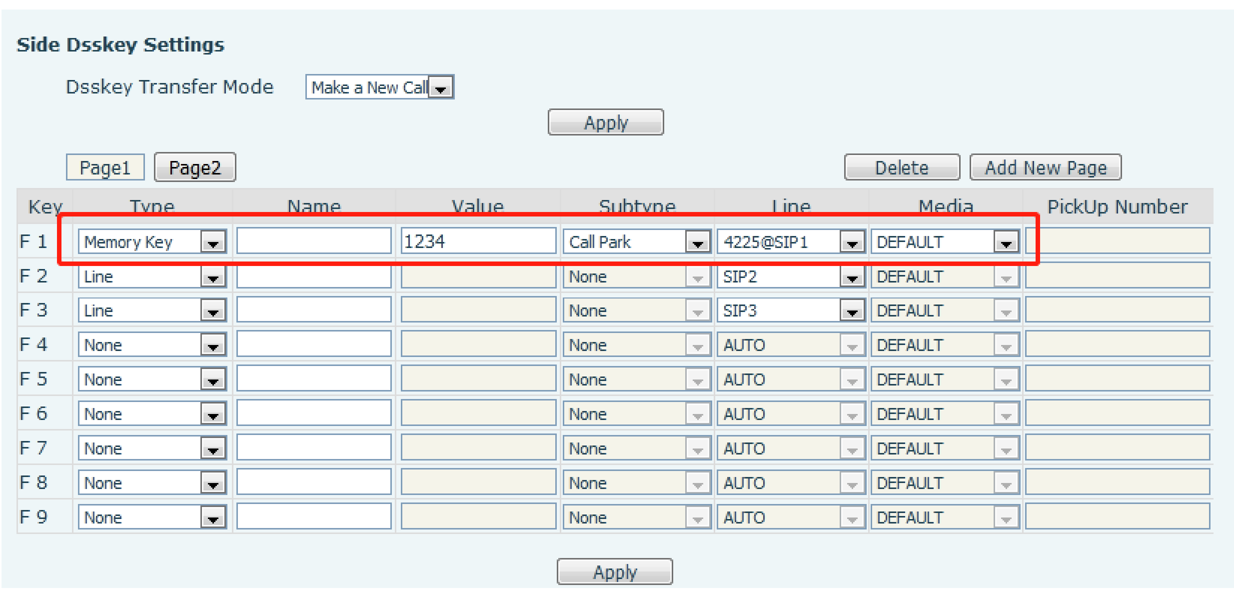
Call Park requires server support. Consult system administrator for support.
While on a call, if it is not convenient to answer an incoming call, press the configured park key to hold the call. After a successful park, user can resume the call by pressing the configured park key.

Set the call park button:

- Phone interface: long press a function key to enter the function key Settings interface, or go to [Menu] >> [Basic] >> [Keyboard], select and enter a key category (DSS key, Soft DSS key). Set the key function type as Memory Key and subtypes as Call Park, enter server call park number, set up corresponding SIP lines.



- WEB interface: log in the web page, enter the [Function Key] >> [Side Key], select a DSS key, set the key type as Memory Key, set the subtype as Call Park, set the Value as the call park number of the server, and set the corresponding SIP line.

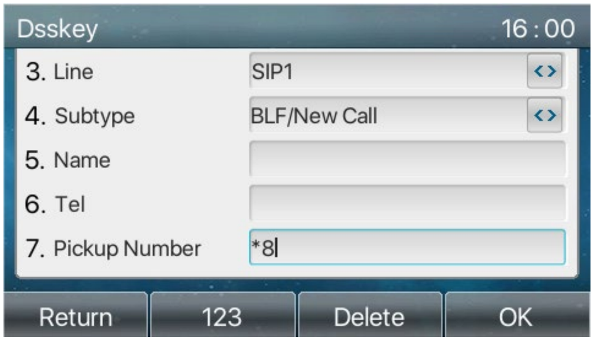


Pick Up

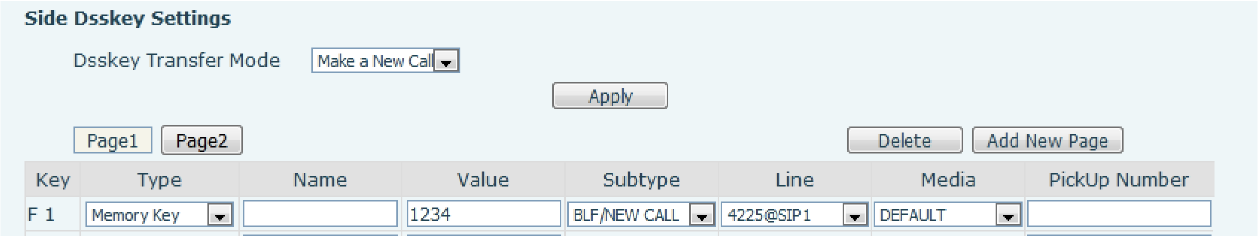
Pick Up requires server support. Consult system administrator for support.
A user can use the Pick Up function to answer incoming calls from other users. The phone can pick up incoming calls by configuring DSSkey for BLF and setting the Pick Up code.

Phone interface: press [Menu] >> [Basic] >> [Keyboard] >> [DSS Key Settings], select the function key to configure.

- Set the line, function key type as Memory Key, subtype as BLF/NEW CALL, set subscribed Telephone number, and pickup Number code. Press OK to save.
- A call comes in to the Telephone number that is in the subscribed to the pickup number and the phone rings.
- Press the DSS key to pick up the call.



WEB interface: Log in the phone webpage, enter the [Function Key] page, select a DSSkey, set the key type as Memory Key, the subtype as BLF/NEW CALL, and set the corresponding SIP line and pick up codes. Click Apply to save.

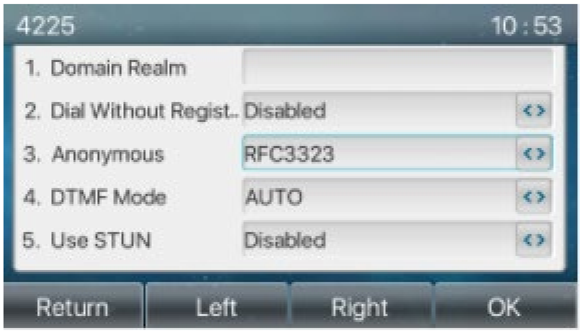


Anonymous Call

Anonymous Call

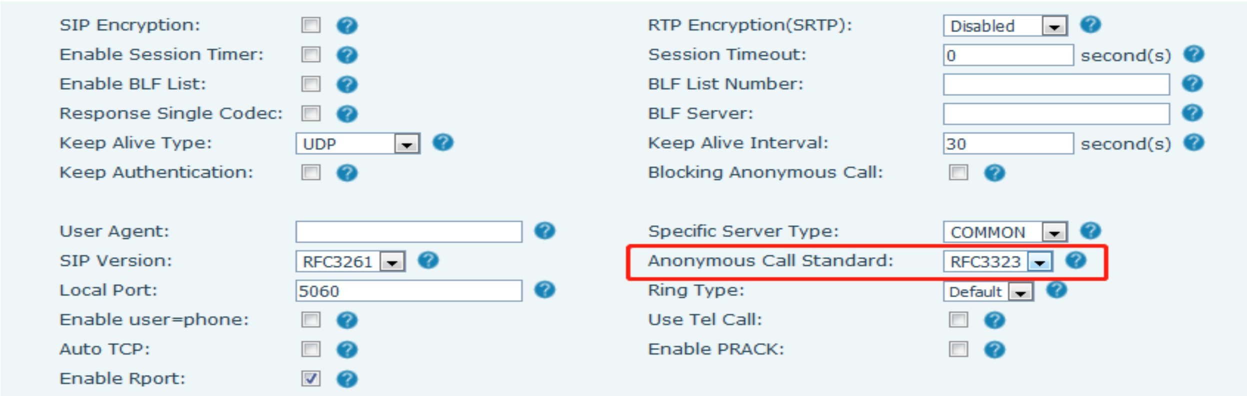
Phone interface: The phone can be set up to hide the calling number and the calling name.

- Go to [Menu] >> [Advanced Settings] >> [Accounts] >>select a SIP line to configure >> [Advanced].
- Enter relevant information. Select an anonymous call standard (RFC3323 and RFC3325).

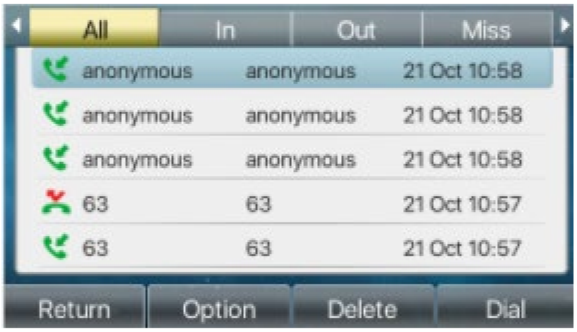


Web page:

- Go to [Line] >> [SIP], select a line to configure from the drop down list >> [Advanced Settings].
- Select an Anonymous Call Standard.



The following is a transcript of an anonymous call received by the phone.

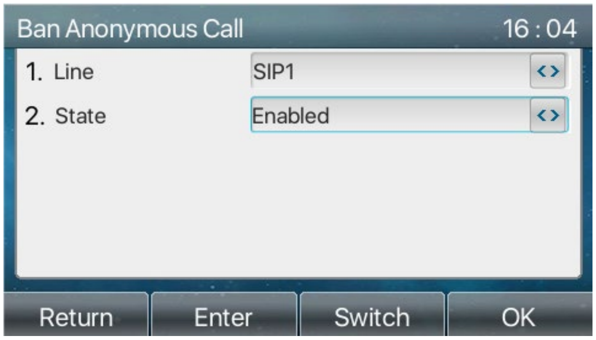


Ban Anonymous Call

The phone can be set to prohibit anonymous calls, so that anonymous calls to the number will be directly rejected.

Phone interface:

- Go to [Menu] >> [Features] >> [Ban anonymous call].
- Use left/right navigation keys to select a Line.
- Use left/right navigation keys to enable the State.



Web page:

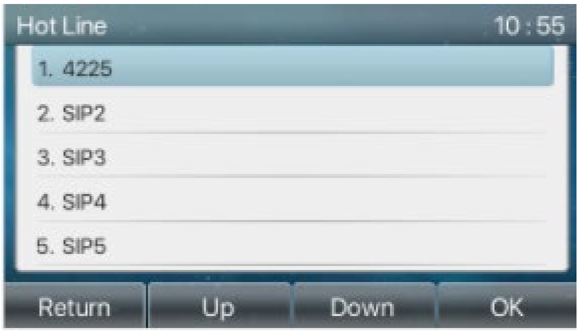
- Go to [Line] >> [SIP] >> [Advanced Settings], select a Line from the drop down list.
- Check Blocking Anonymous Call.

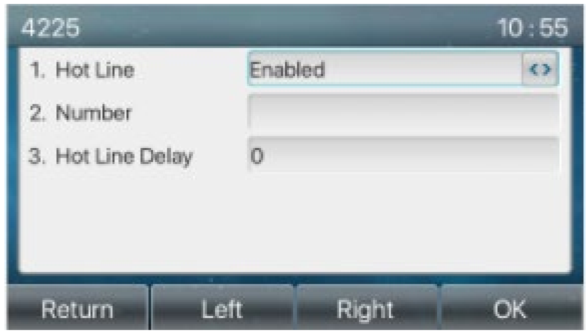
Hotline

The phone supports hotline dialing. After setting up the hotline dialing, go offhook and the phone will automatically call according to the hotline delay time.

Phone interface:

- Go to [Menu] >> [Features] >> [Advanced] >> [Hotline] and select which sip line to configure.
- Use the left/right navigation keys to enable Hot Line.
- Set the hotline Number.
- Set the Hot Line Delay.

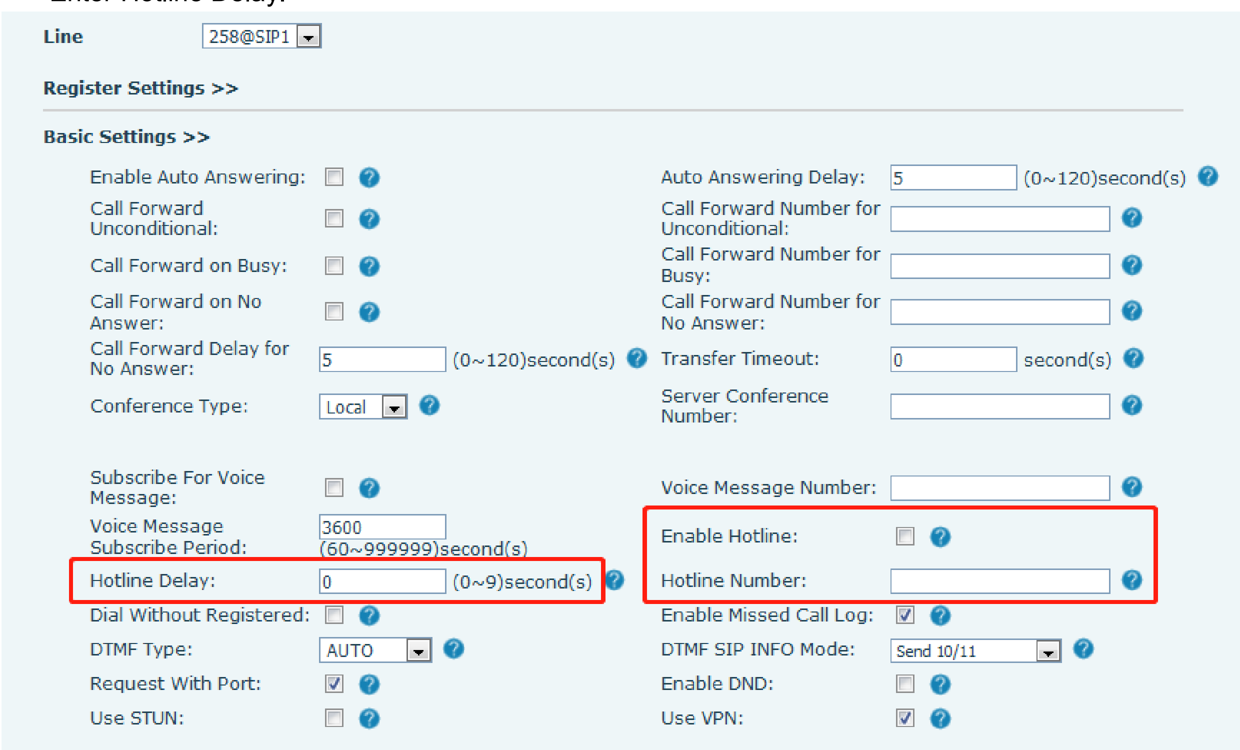




Web interface:

Go to [Line] >> [SIP], select a line to configure from the drop down list >> [Basic Settings].

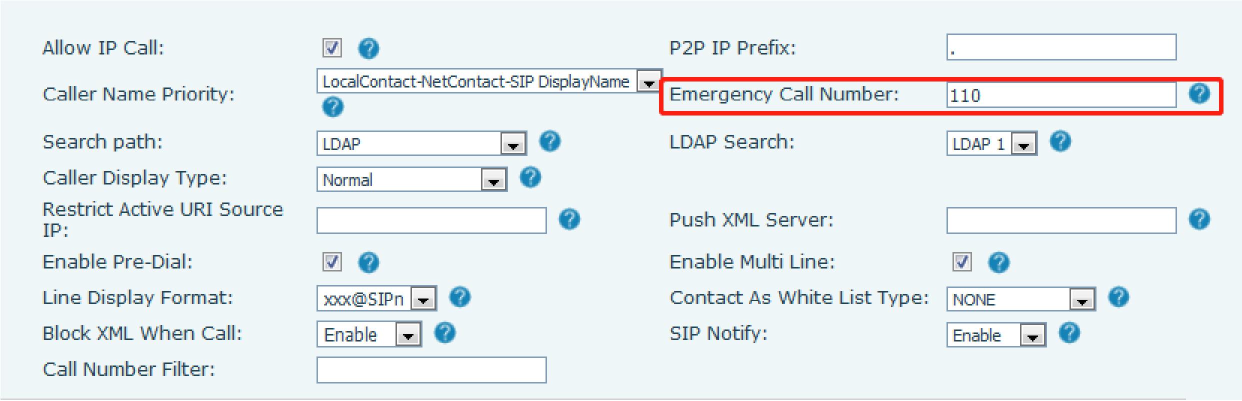
- Check Enable Hotline.
- Enter Hotline Number.
- Enter Hotline Delay.



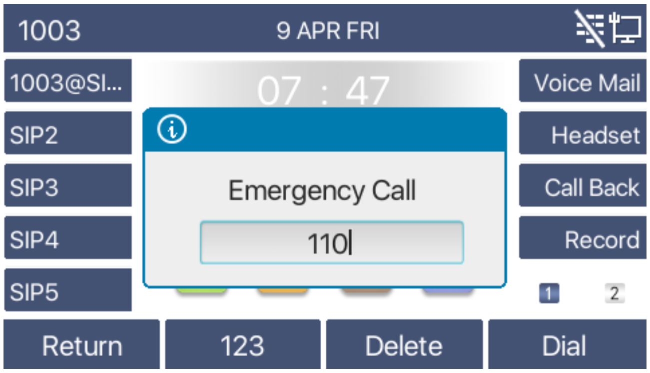
Emergency Call

The emergency call function is used to call an emergency call number when the keypad is locked.

- To configure the emergency call number, log in the phone web page, enter the [Phone Settings] >> [Features]>> [Basic Settings], and enter an Emergency Call Number. If more than one emergency call number needs to be set, use a comma ", " to separate the numbers.



- When the phone keypad is locked, user can call the emergency call number without unlocking the phone, as shown in the figure: Dial the emergency number then press Dial.

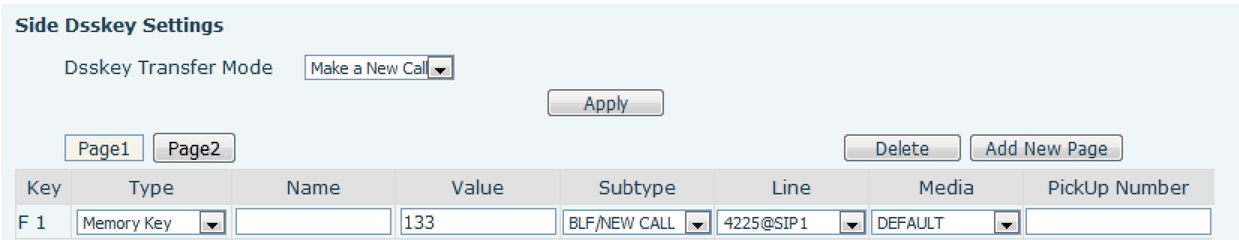


Advance Function

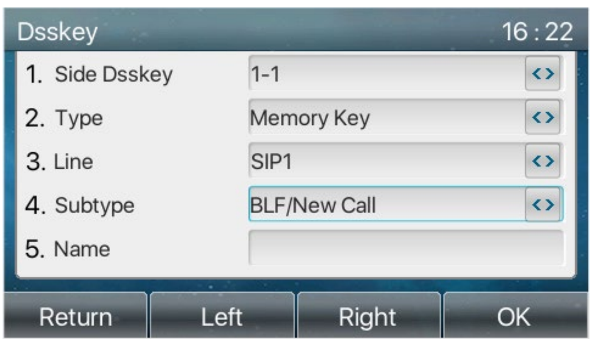
BLF (Busy Lamp Field)

Configure the BLF Functionality

- Web interface: log in the phone web page, enter the [Function key] >> [Side Key or Softkey], select a DSS key, set the function key type as Memory Key. Choose a subtype among BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF. For BLF/DTMF, set the value as the number to be subscribed. Set the corresponding SIP line. The pickup number is provided by the server. Click Apply to save. For usage, refer to **Pick up**.



- Phone interface: Long press a function key to enter the function key or go to [Menu] >> [Basic] >> [Keyboard] >> [DSS Key Settings or Soft DSS Key Settings].Set the Type as Memory Key and a subtype as BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONF, BLF/DTMF. Set the Tel value, which is the subscription number. Set the SIP line. Press OK to save.



Subtype	Standby is described	Calling is described
BLF/NEW CALL	Press the BLF key while on standby to dial the subscriber number.	When this BLF key is pressed while talking to another user, you create a new call along with the subscribed number.
BLF/BXFER	Press the BLF key while on standby to dial the subscriber number.	When this BLF key is pressed while talking to another user, you blind transfer the call to the subscribed number.
BLF/AXFER	Press the BLF key while on standby to dial the subscriber number.	When this BLF key is pressed while talking to another user, you attendance transfer the call to the subscribed number.
BLF/Conference	Press the BLF key while on standby to dial the subscriber number.	When this BLF key is pressed while talking to another user, you invite the subscriber number to join the meeting.
BLF/DTMF	Press the BLF key while on standby to dial the subscriber number.	When this BLF key is pressed while talking to another user, the phone automatically sends the DTMF that corresponds to the BLF key number.

Use the BLF Function

BLF, also known as a "busy lamp field," notifies the user of the status of the subscribed object and is used by the server to pick up the call. BLF helps you monitor other peoples status (idle, ringing, talking, off).

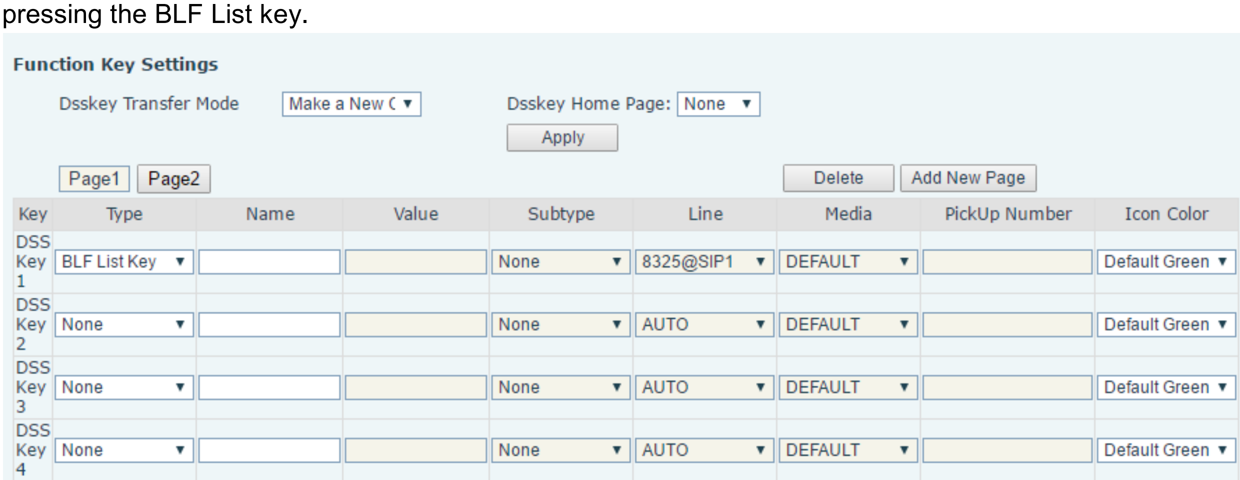
BLF function:

- Monitor the status of subscribed phones.
  When the subscription state of the number is changed (idle, ringing, talking), the function key LED status will have a corresponding change. Refer to **LED Definition** for LED states and meanings.
- Call the subscribed number.
  When the phone is in standby mode, press the configured BLF key to call out the subscribed number.
- Transfer calls to the subscribed number.
  The BLF key can be used for blind rotation, attention-rotation and semi-attention-rotation of the current call, and can also invite the subscribed number to join the call and send DTMF, etc. Refer to BLF function key table.
- Pick up incoming calls from subscribed number.
  When the subscription number telephone rings, the BLF LED.will turn red. At this point, press the BLF key to answer the incoming call from the subscribed number. Refer to **BLF LED** table.

BLF List

BLF List Key puts the number to be subscribed into a group on the server side, and the phone uses the URL of this group for unified subscription. The specific information, number, name and status of each number can be resolved based on the notify sent from the server. A Memory Key is then set as the BLF List Key. If the state of the subscription object changes later, the corresponding led light state will change.
Configure BLF List function: log into the phone web page, enter [Line] >> [SIP], select a line to configure, >> [Advanced settings], check Enable BLF List, and configure the BLF List number.

Using the BLF List function: when the configuration is completed, the phone will automatically subscribe to the contents of the BLF List group. Users can monitor, call and transfer the corresponding number by pressing the BLF List key.



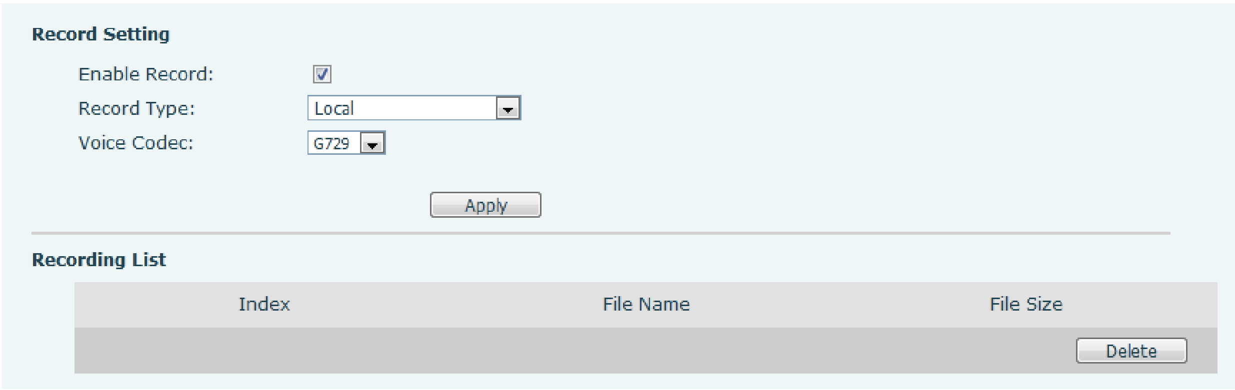
Record

Record a conversation while on a call.

Local Recording (USB flash drive required)

When using local recording, it is necessary configure recording on the phone web page.

- Begin by inserting a USB drive into the USB port on the phone.
- Web interface: Go to [Application] >> [Manage recording].
- Select Recording Type as Local.
- Set which Voice Codec to use.
- Click Apply.
- Go to [Function Key] >> [Side Key or Softkey] and configure a key Type as Key Event and Subtype as Record.



Local recording steps:

- Make a call and press the key that was configured as a record key.
- End the recording by pressing the record key again or by hanging up.

View local recording:

From the Phone:

- Select [Menu] >> [Application] >> [USB] >> [Browse Audio].
- You will see a list of recordings. From here you can select a recording using the Navigation up/down keys and press Play to listen to the recording.

From the phone web page:

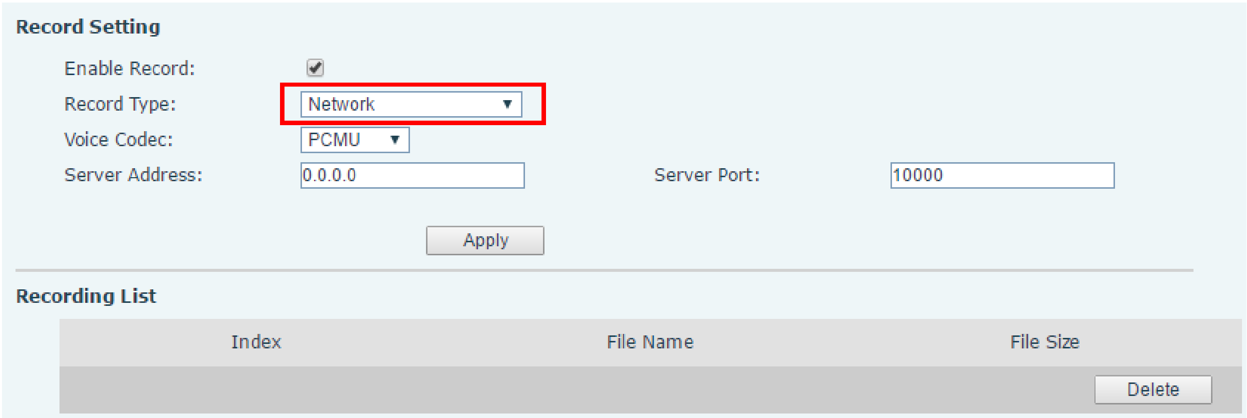
- Enter [Application] >> [Manage Recording].
- View the recording file. From here you can delete the recording file by selecting it and clicking Delete.

Listen to the recording

- Select [Menu] >> [Application] >> [USB] >> [Browse Audio].
- You will see a list of recordings. Select a recording using the Navigation up/down keys and press Play.

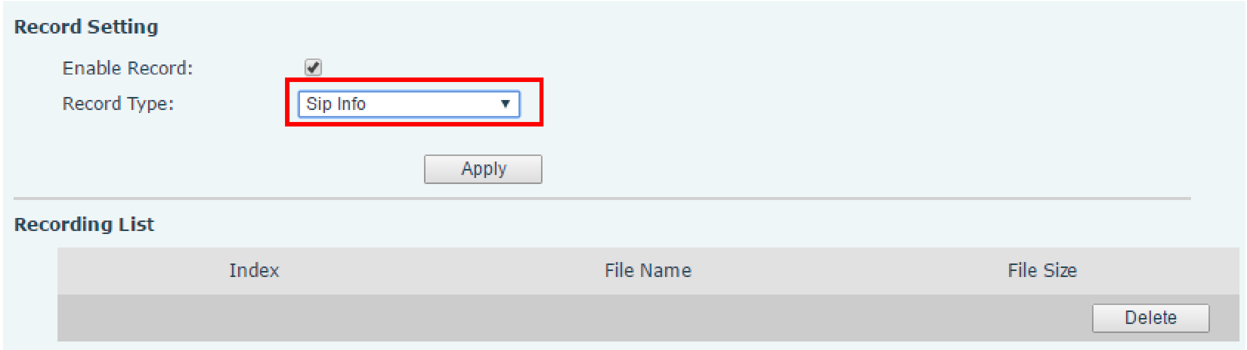
Server Recording

When using the network server to record, go to phone web page and navigate to [Application] >> [Manage recording]. Select Type as Network, select a Voice Codec, enter Server Address and Server port as shown in the image below:



SIP INFO Record

Register the phone with a server that supports SIP INFO recording. After registering the account, go to [Application] >> [Manage recording], check Enable Record, set Record Type to SIP INFO and click Apply.



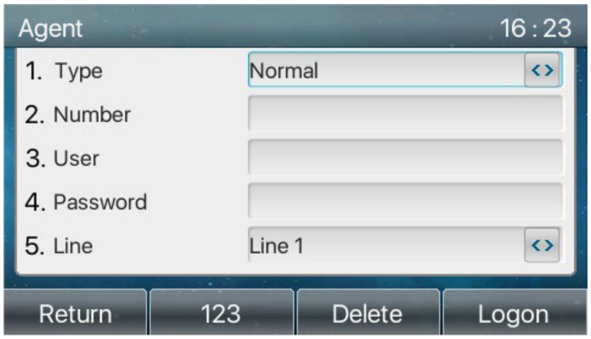
Agent

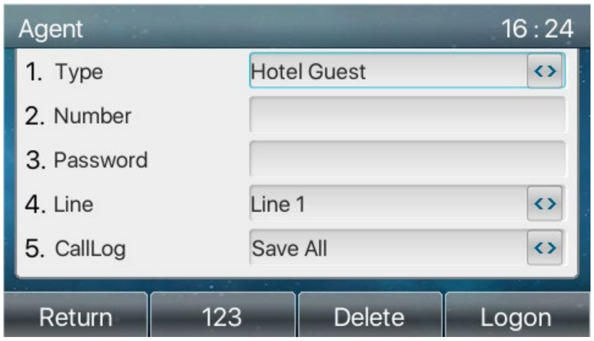
When multiple people use a phone for Agent services at different times, they can quickly register their SIP account on the same server. The Agent functions of the phone can be divided into Normal and Hotel Guest. The Hotel Guest mode requires server support.

Normal Mode:

To configure agent function from the phone, go to [Menu] >> [Features] >> [Agent]. The password to enter the agent page is default 123.

The SIP server needs to be configured before the account can be configured.

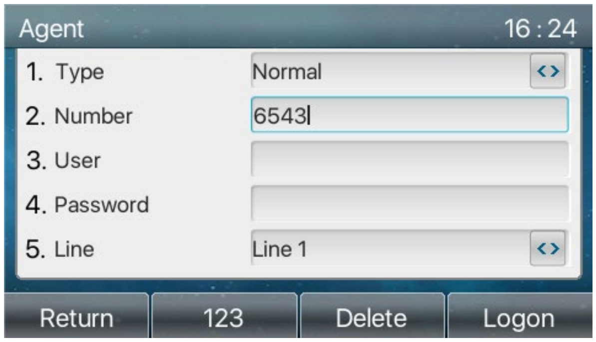




Parameter	Description
Normal	
Number	Set the proxy account number.
User	Set the proxy account number to verify the user name.
Password	Set the proxy account number to verify the password.
Line	Select the SIP line.
CallLog	Users can choose to save all or delete all.
Hotel Guest	
Number	Set the proxy account number.
Password	Set the proxy account number to verify the password.
Line	Select the SIP line.
CallLog	Users can choose to save all or delete all.
Status	Consists of Logon, Logoff, Unavailable, Available, Wrap-up.

Using agent functions:

- When the phone has been configured on SIP server, enter the correct number, user name and password, press Logon and the phone will register to the SIP server.
- After registration, click Logout and the phone will delete the user name and password and log out of the SIP account.
- Click Unregister, the phone will retain the user name and password and log out of the SIP account.



Intercom

When Intercom is enabled, phone will automatically connect to incoming calls through the hands free speaker.

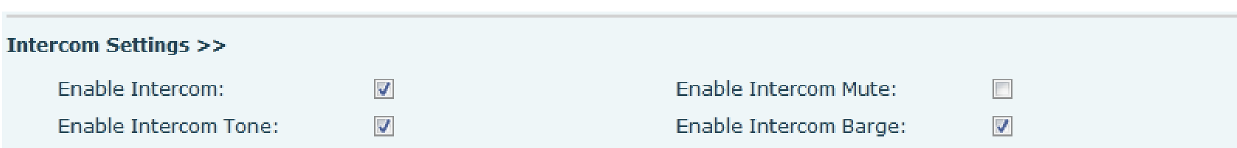
Phone interface:

- Press [Menu] >> [Features] >> [Intercom].
- Enable all Intercom settings that apply by using the left/righ navigation keys to toggle each selection from Disabled to Enabled.

Web interface:

- Set up Intercom on all applicable phones by going to [Phone Settings] >> [Intercom Settings] and check all applicable intercom settings. Click Apply to save.
- Go to [Function Key] and configure a key. Set type as Key Event and subtype as Intercom.

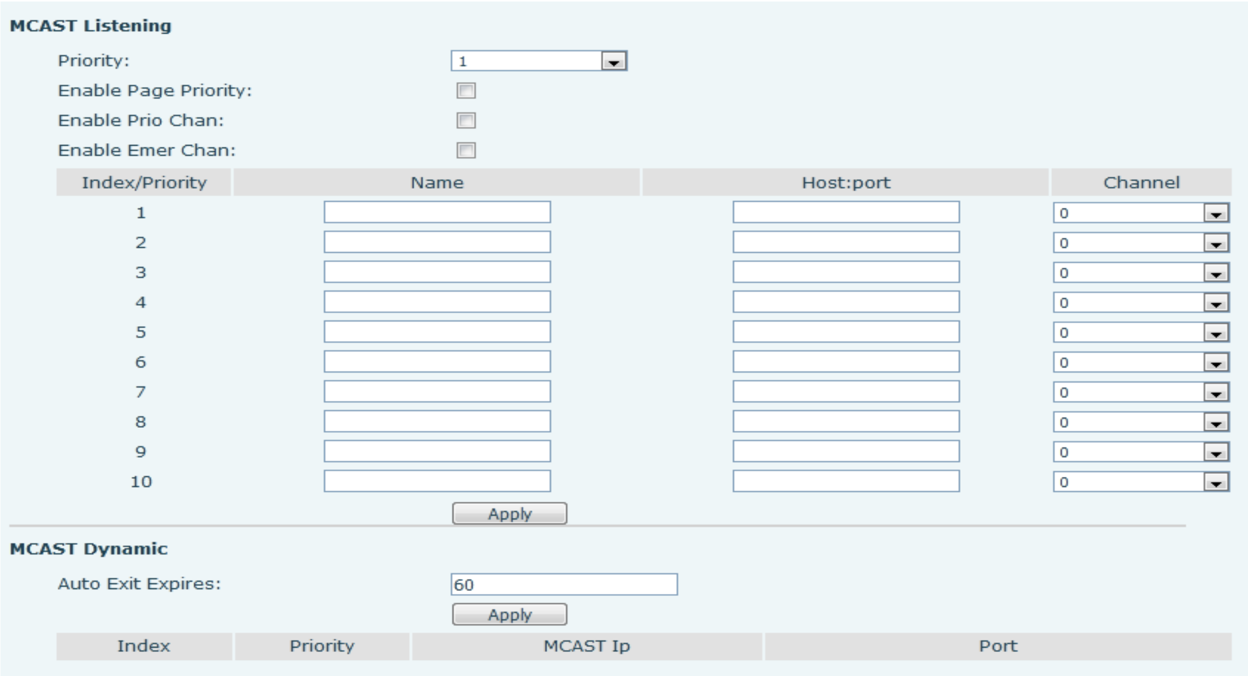
Press the intercom key and dial another extension that has intercom enabled. The other extension will play a barge in tone and connect automatically.
Note that the extension that you are barging into must have intercom enabled.



Parameter	Description
Enable Intercom	When intercom is enabled, the device will accept the incoming call request with a SIP header of Alert-Info instruction to automatically answer the call after 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	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

MCAST

The multicast feature allows user to make a broadcast call to people who are in a 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 without involving SIP signaling. You can also configure the phone to receive an RTP stream from a pre-configured multicast listening address without involving SIP signaling. Up to 10 multicast listening addresses can be entered.



Parameters	Description
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.

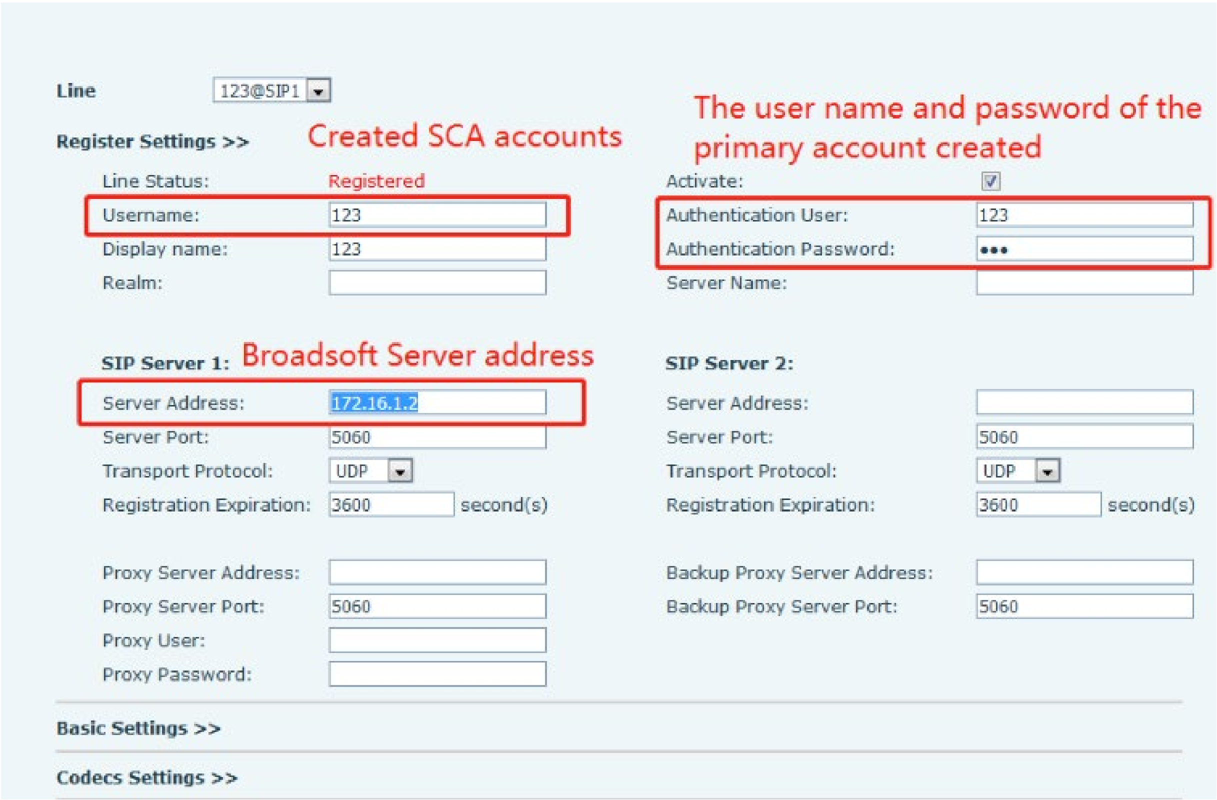
To set up a function key for multicast:

- Log into phone web interface.
- Go to [Function Key] >> [Side Key or Softkey], select MCAST Paging or MCAST Listening, set the multicast address Value, and select the codec.
- Click Apply.
- Go to [Phone Settings] >> [MCAST] and set the Name and Host:port of the receiving multicast.
- From the phone, press the DSS key that was set as a Multicast key.
- Receive end will receive multicast call and play multicast automatically.

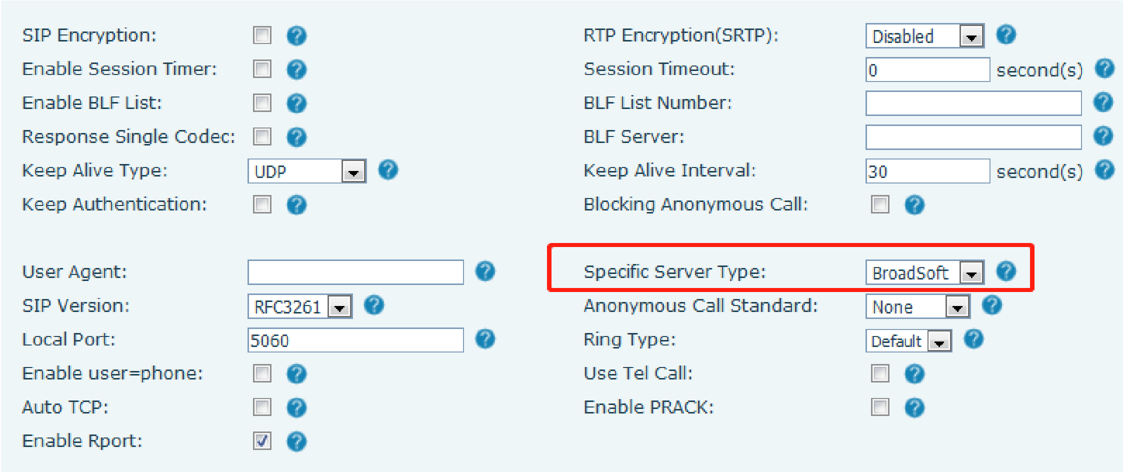
SCA (Shared Call Appearance)

SCA function requires server support.

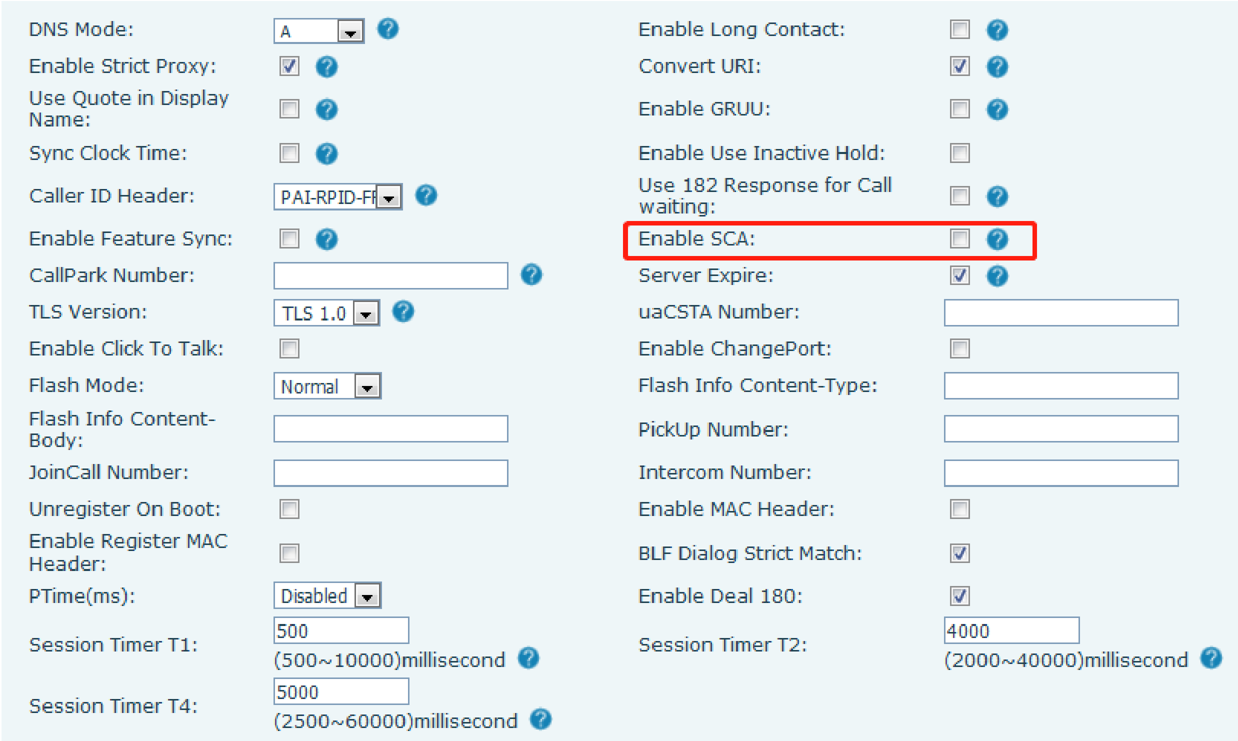
- Configuring the phone
  - When registering with the BroadSoft server, the phone can register the account created previously on multiple terminals. To configure the phone from the web page, go to [Line] >> [SIP], select a line and enter Register Settings information and SIP Server 1 information. Select Activate and click Apply to save.



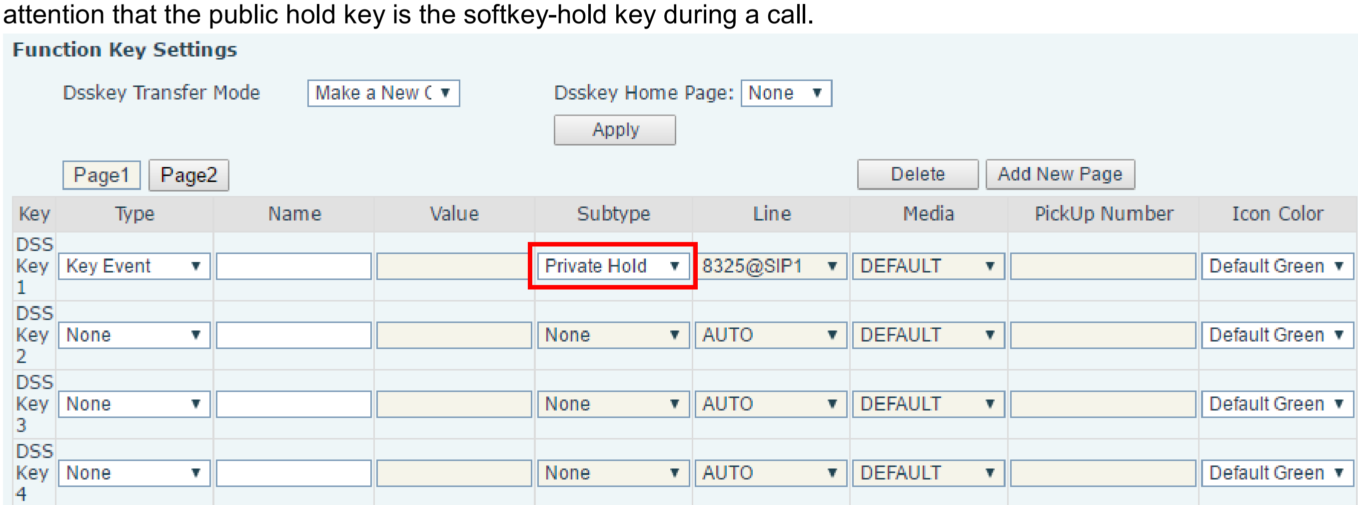
  - After the phone registers with the BroadSoft server, a server type needs to be set. Specifically, log in to the webpage of the phone, choose [Line] >> [SIP] >> [Advanced Settings] and set Specific Server Type to BroadSoft, as shown in the following figure.



- If a phone needs to enable the SCA function, log in to the phone webpage, go to [Line] >> [SIP] >> [Advanced Settings], and select Enable SCA. If SCA is not enabled, the registered line is the private line.



After an account is configured and successfully registered, you can configure lines whose DSS Key is Shared Call Appearance on the Function Key page to facilitate viewing the call status of the group. Each line key represents a call appearance. Understand the call status by referring to **LED DEFINITION** table. To facilitate private hold, configure keys whose DSS Key is Private Hold on the Function Key page. Pay attention that the public hold key is the softkey-hold key during a call.



- Each phone registered with the BroadSoft server should be configured as above, then the SCA function can be used.

- LED Status
  To facilitate viewing the call status of a group, configure the DSS Key as SCA. The following table describes the line LEDs in different states.

State&Direction	Local	Remote
Idle	Off	Off
Seized	Steady green	Steady red
Progressing (outgoing call)	Steady green	Steady red
Alerting (incoming call)	Fast blinking green	Fast blinking green
Active	Steady green	Steady red
Public Held (hold)	Slow blinking green	Slow blinking red
Held-private (private hold)	Slow blinking yellow	Steady red
Bridge-active (Barge-in)	Steady green	Steady red
Bridge-held	Steady green	Steady red

- Shared Call Appearance(SCA)
  See the following scenarios to help facilitate understanding of the feature.
  In the following scenarios, the manager and secretary are registered to the same SCA account.

  - Scenario 1: When this account receives an incoming call, the phones of both the manager and the secretary will receive the call and ring. If the manager is busy, the manager can reject the call and the manager's phone stops ringing but the secretary's phone keeps ringing until the secretary rejects/answers the call or the call times out.

  - Scenario 2: When this account receives an incoming call, if the secretary answers the call first and the manager is required to answer the call, the secretary can press the Public Hold key to hold this call and notify the manager. The manager can press the line key corresponding to the Shared Call Appearance to answer the call.

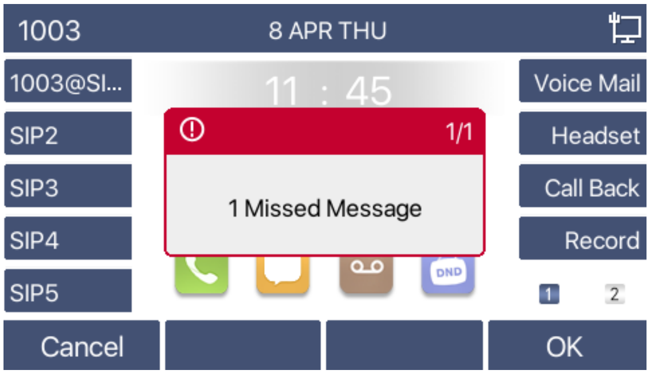
  - Scenario 3: The manager is on an important call with a customer and needs to leave their office for a moment. If the manager does not want others to retrieve this call, the manager can press the Private Hold key.

  - Scenario 4: The manager is in a call with a customer and requires the secretary to join the call to take notes. The secretary can press the corresponding SCA line key to barge into this call.

Message

SMS (Short Message Service)

If the service of the line supports SMS, when the other end sends a text message to the number, the user will receive the SMS message on the standby screen.



Send messages:

- From the phone, go to [Menu] >> [Message] >> [SMS].
- Select New Message
- Type a message using the keypad, press Send, then enter a “To” number and press OK to send.

View SMS:

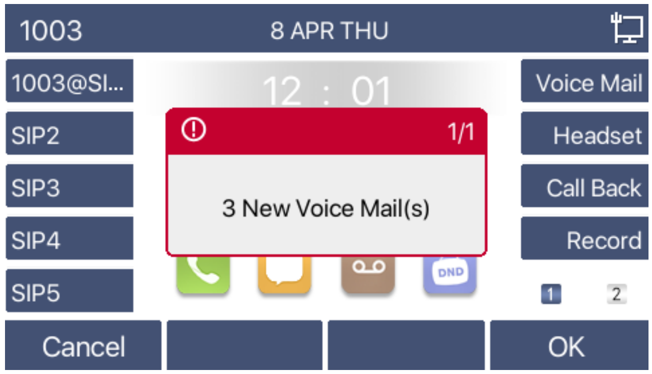
- From the phone, go to [Menu] >> [Message] >> [SMS].
- Use the up/down navigation keys to select Inbox.
- Press [OK] to enter the SMS inbox interface.
- Select the unread message and press [OK] or [View] to read the unread message.

Reply to SMS:

- Use the navigation keys to select the message.
- Press [OK] or [View] to view the message.
- Press the [Reply] key, type your reply and press Send.

MWI (Message Waiting Indicator)

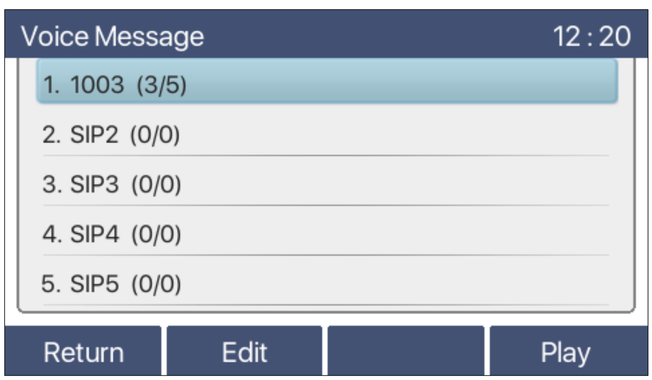
If the line service supports voice message feature, when the user is not available to answer the call, the caller can leave a voice message on the server to the user. User will receive voice message notification from the server and device will prompt a voice message waiting icon on the standby screen.

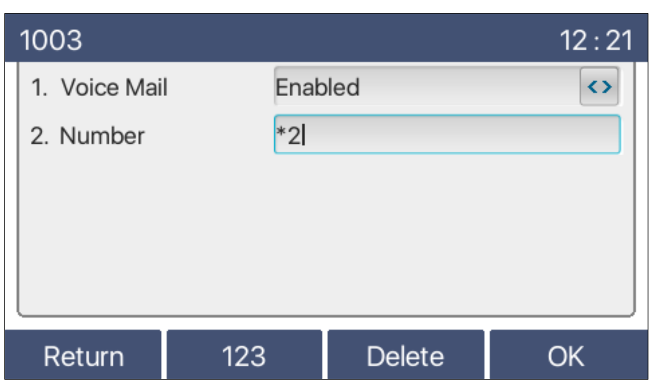


 The Voice message icon  on the home screen will also show the number of messages missed.

To enable Voice Mail:

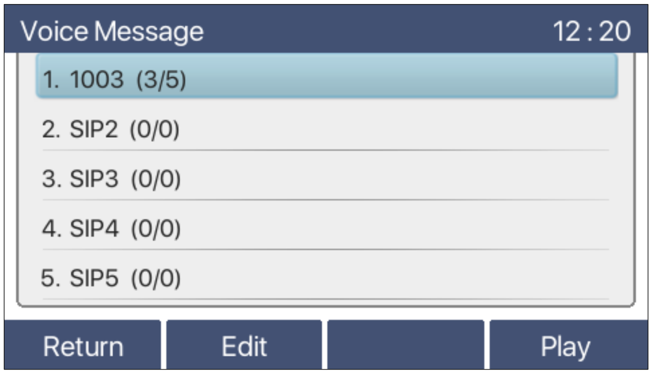
- Press the Voice Mail key.
- Use the up/down navigation keys to select the appropriate line and press OK.
- Use the left/right navigation keys to toggle Voice Mail from Disabled to Enabled.
- Enter the code that will allow the phone to access the server voicemail.
- Press OK to save.





To listen to a voice message:

- Press the Voice Mail key.
- Use the up/down navigation keys to select the line and press Play.
- In the following picture, (3/5) represents 3 new voice messages, and 5 total voice messages.



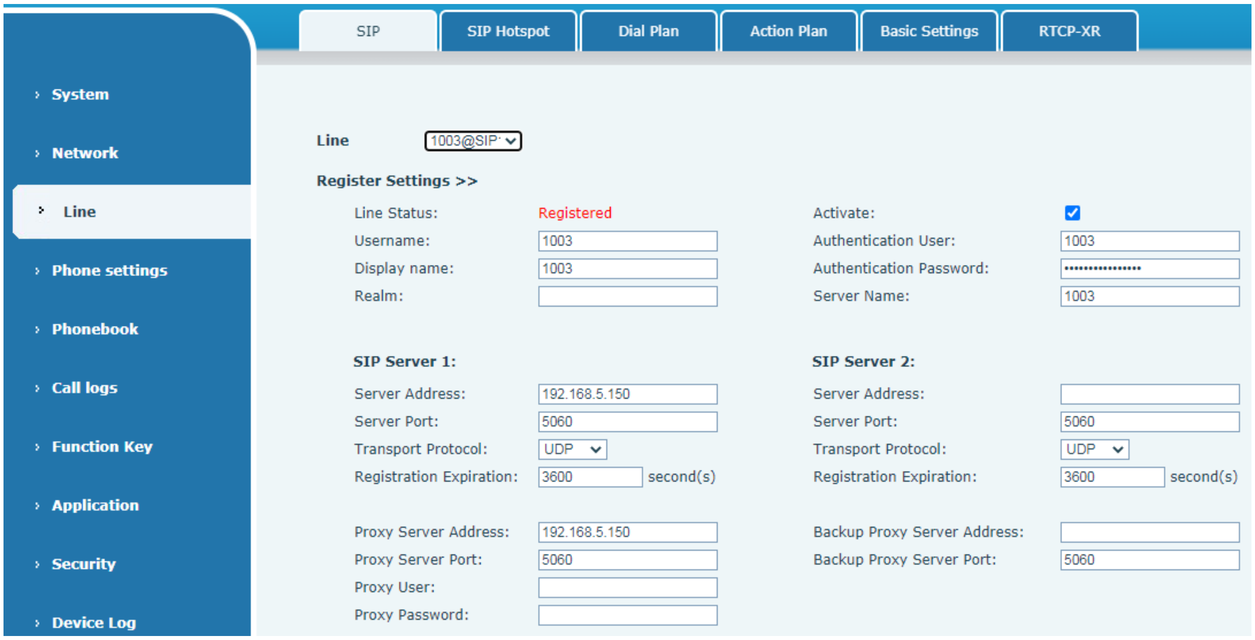
SIP Hotspot

SIP accounts can be expanded using SIP hotspot, a simple but practical function that can implement group ringing.

Call example: Phone (A) is set up as a Hotspot and phones (B) and (C) are set up as clients. When someone calls phone (A), phones (A), (B), and (C) all ring. When any phone answers the call, other phones stop ringing. The call can be answered by only one phone. When phone (B) or (C) initiates a call, the SIP number registered to phone (A) is the calling number.

To create a SIP hotspot, at least one SIP account must be configured and registered before proceeding.

To create a SIP account from the web interface, go to [Line] >> [SIP] and fill in the appropriate information.

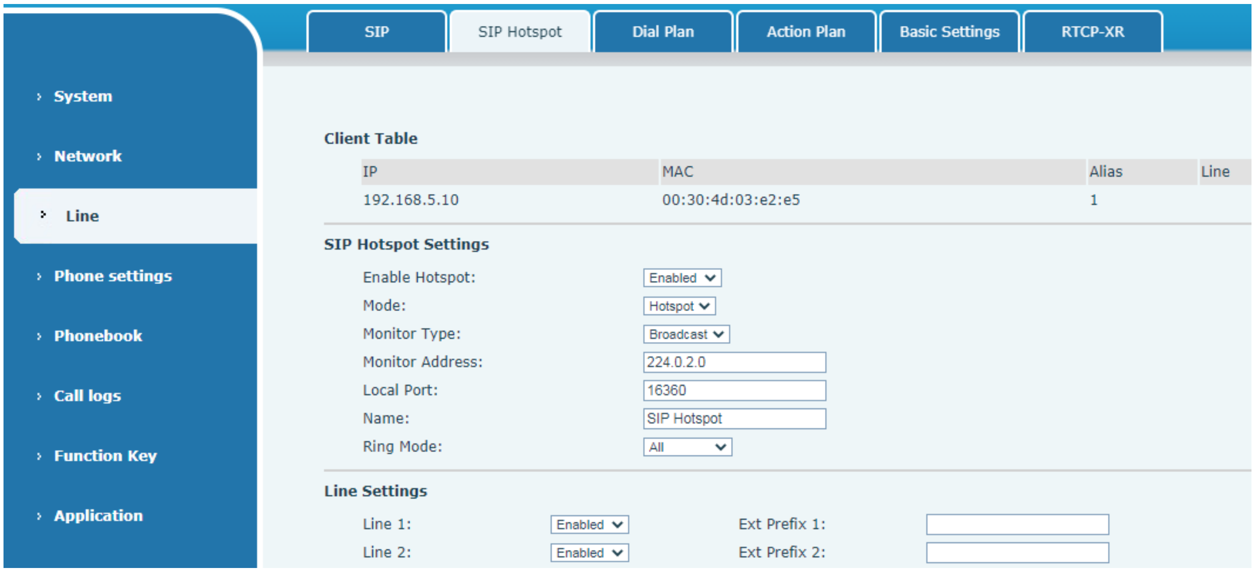


There are two hotspot modes, Client and Hotspot.

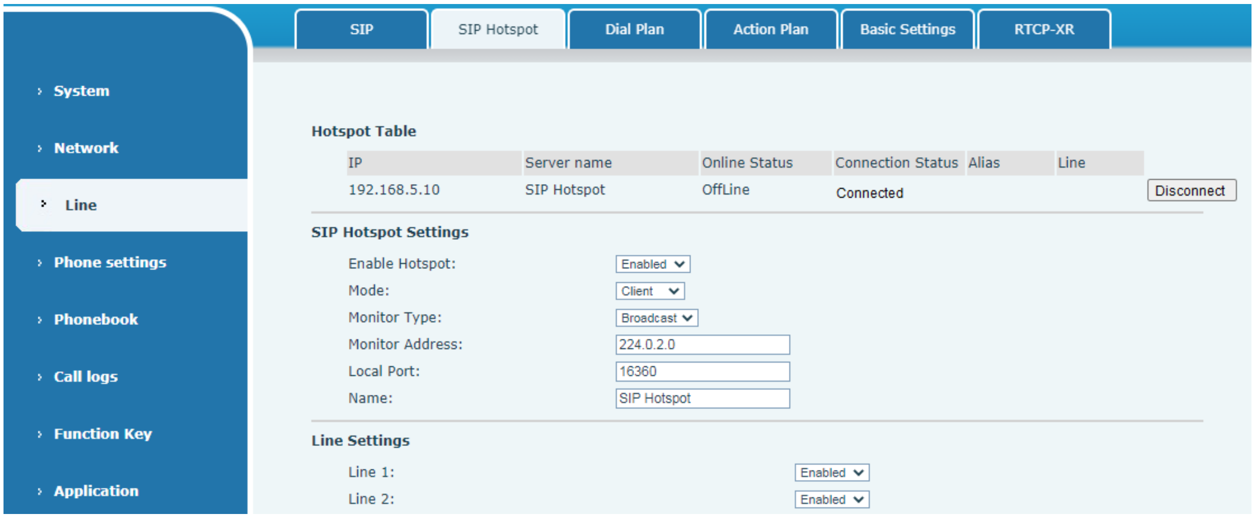
- If your phone is set up as a “Hotspot”, the device table will display as Client Table, devices that can connect to your phone.
- If your phone is set up as a “Client”, the device table will display as Hotspot Table, devices to which your phone can connect.

Parameters	Description
Enable hotspot	Set it to Enabled to enable the feature.
Mode	Choose whether the phone will be a hotspot client or server.”
Monitor Type	Choose Broadcast or Multicast. If you want to limit the broadcast packets, use Broadcast. If client chooses Broadcast, the SIP hotspot phone must be broadcast.
Monitor Address	This is the address of broadcast. Hotspot server and hotspot client must be the same.
Local Port	This is the port number used for access.

SIP hotspot server:
As the hotspot server, the default extension number is 0. When the phone is used as the client, the extension number is increased from 1 (example:1003-1). You can view the extension number Alias through the [SIP Hotspot] page.



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



Call extension number:

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

Phone Settings

Basic Settings

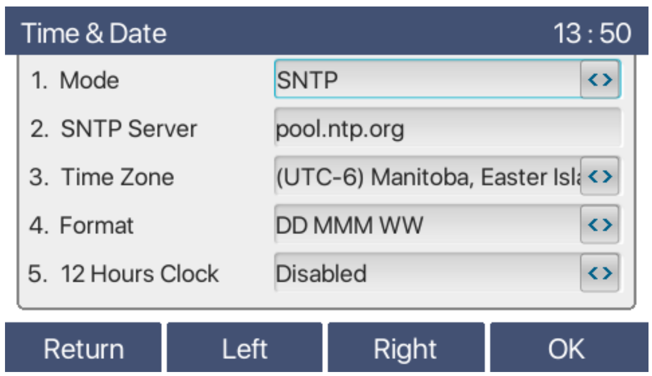
Language

The phone only supports English.

Time & Date

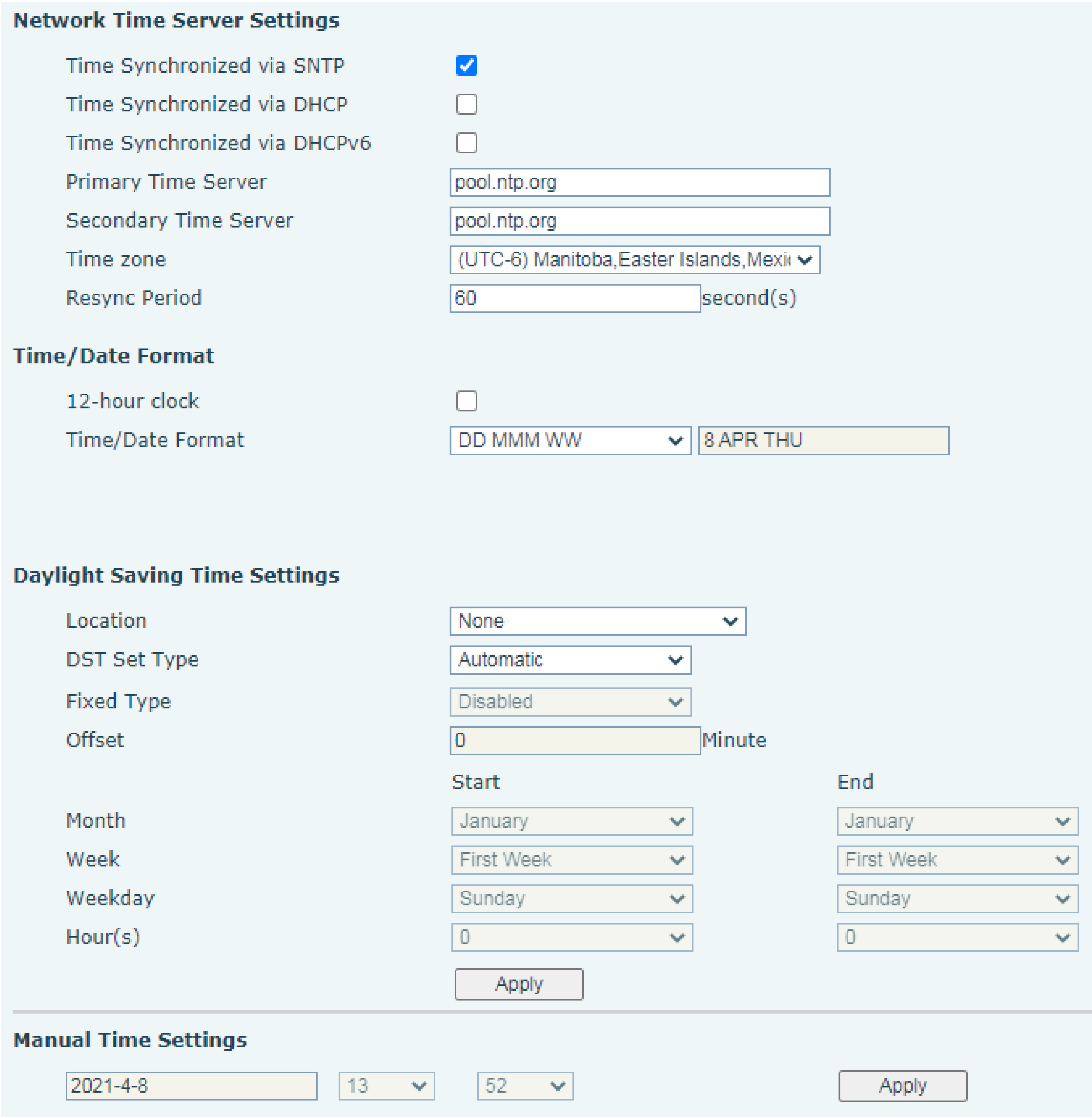
Users can set the phone time through the phone interface and web interface.

- Phone interface: When the phone is in the default state, press [Menu] >> [Basic] >> [Time & Date]. Use the up/down navigation keys to edit parameters. Press [OK] to save.



Parameters	Description
Mode	SNTP/Manual SNTP: Enable network time synchronization via SNTP protocol, default enabled. Manual: User can modify time and date manually.
SNTP Server	SNTP server address
Time zone	Select the time zone
Time format	Select time format from one of the following: 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
12-Hour Clock	Display the clock in 12-hour format
Daylight Saving Time	Enable or Disable the Daylight Saving Time

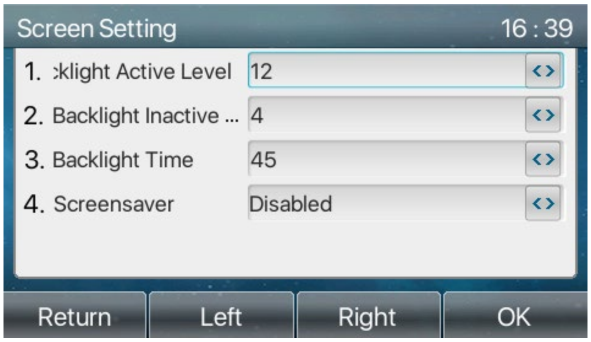
- Web interface: Go to [Phone Settings] >> [Time/Date] , as shown below:



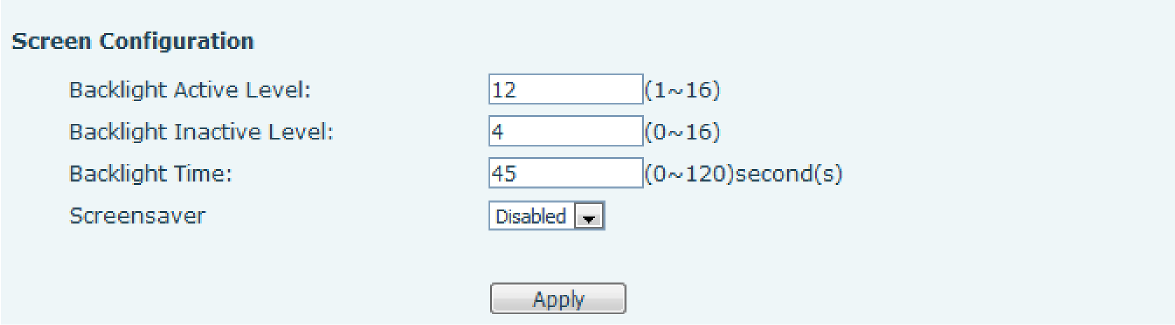
Screen Settings

The user can set the phone screen parameters through both of the phone interface and web interface.

- Phone interface: When the phone is in the default standby state, go to [Menu] >> [Basic] >> [Screen] to edit the screen parameters. After editing, click [OK] to save as shown in the figure:



- Web interface: Go to [Phone Settings] >> [Advanced]. Edit the screen parameters, and click Apply to save.



- Backlight Active Level: Set the Backlight Active Level (brightness) from 1 to 16. Use navigation keys [<] or [>] to adjust brightness level.
- Backlight Inactive Level: Set the Backlight Inactive Level (idle brightness) from 0 to 16. Use Navigation keys [<] or [>] adjust brightness level.
- Backlight Timer: Set the Backlight Timer. Timer range is Always On, Custom, 15 seconds, 30 seconds, 1 minute, 2 minutes, 5 minutes, 10 minutes, 30 minutes, 1 hour, 2 hours, 3 hours, 6 hours, 15 hours.
- Screensaver: The screen saver can be Enable and Disabled: Enabled by default.
- Click Apply to save.

Ring Settings

When the phone is in the default standby mode,

- Press [Menu] >> [Basic] >> [Ring].
- Adjust Headset or Handsfree ring volume using the left / right navigation keys and press OK to save.
- Scroll to Ring Type, press left / right navigation keys to change the ring type.
- Press OK to save.

Voice Volume

When the phone is in the default standby mode,

- Press [Menu] >> [Basic] >> [Voice Volume].
- Adjust Handset, Headset and Handsfree volume with the left / right navigator keys.
- Press OK to save.

Greeting Words

When the phone is in the default standby mode,

- Press [Menu] >> [Basic] >> [Greeting Words].
- Edit the Greetings Words and press OK to save.

NOTICE! The welcome message can only be displayed in the upper left corner of standby screen when this option is disabled.

Reboot

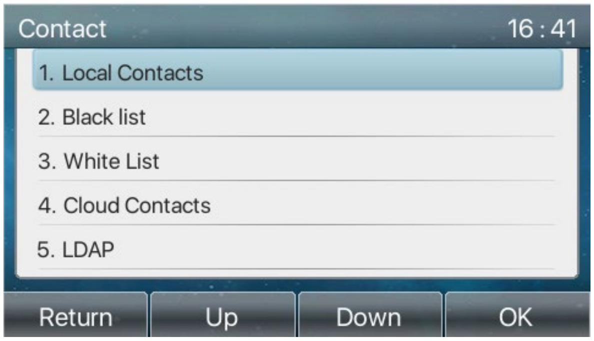
When the device is in the default standby mode,

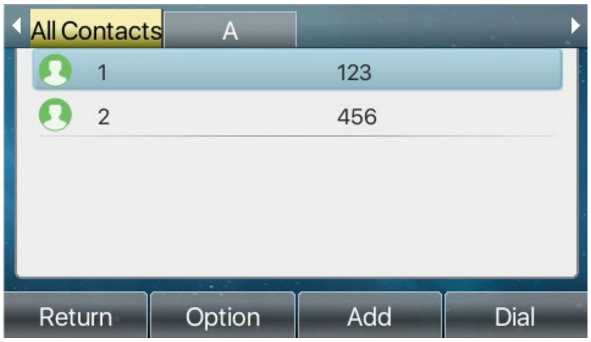
- Press [MENU] >> [BASIC] >> [Reboot System].
- Press OK. A prompt message will read, “Reboot Now?”
- Press OK to reboot the phone or Cancel.

Phone Book

Local Contact

User can save contact information in the phonebook and dial contact phone numbers from the phonebook. To open the phone book, press Contacts key or Contact soft key while phone is idle.
By default the phonebook is empty. User can add contacts to the phone book manually or from call logs.



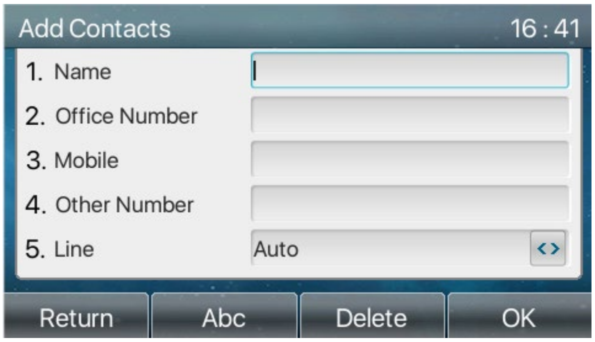


When there are contact records in the phone book, the contact records will be arranged in alphabetical order. User may browse the contacts with up/down navigator keys. User may check the contact information by pressing [OK] key.

Add / Edit / Delete Contact

To add a new contact, press contacts key then press [Add] key to enter the Add Contacts screen and enter the contact information as shown below,

- Contact Name
- Tel. Number
- Mobile Number
- Other Number
- Line
- Ring Tone
- Contact Group
- Photo

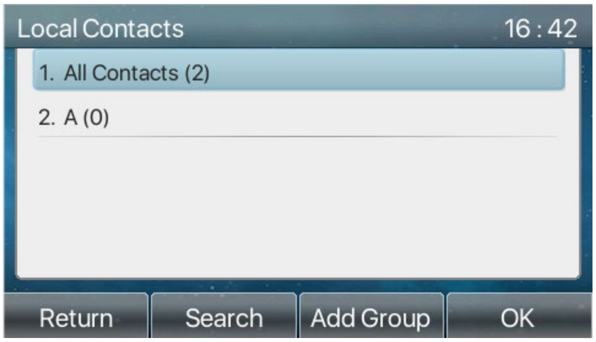


Edit a contact by selecting the contact then pressing [Option] >> [Edit].
To delete a contact, select the contact to be deleted, press [Option] >> [Delete] and confirm with [OK].

Add / Edit / Delete Group

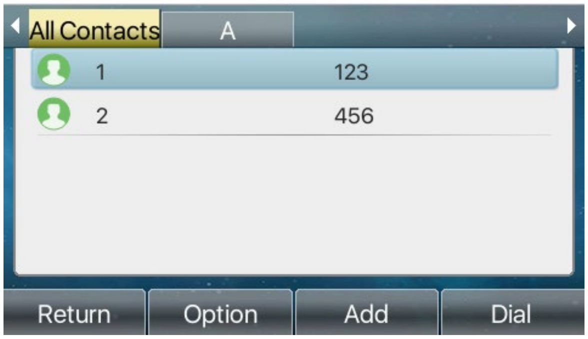
User can create groups, edit the group name, add or remove contacts in the group, and delete a group.

- To add a group, press [Contact] >> [Add Group].
- To delete a group, press [Option] >> [Delete].
- To edit a group, press [Edit].
- The Number behind the group name means the total number of contacts within a group.

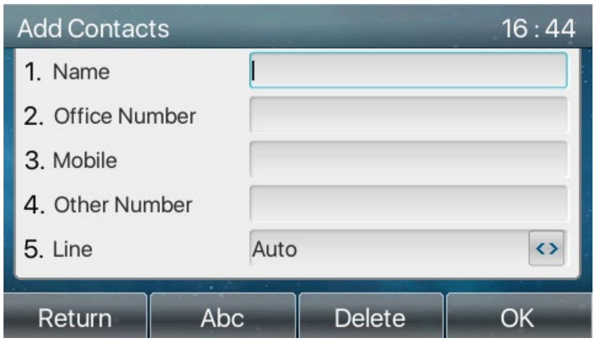


Browse and Add / Remove Contacts in Group

User can browse contacts in a group pressing the Contacts key or Contact soft key.



When user is browsing contacts of a group, user can also add contacts to that group by pressing [Add] then pressing [OK] to save the contact. The contact will also be added to the local phonebook. User can delete contact from group by selecting [Option] >> [Delete].

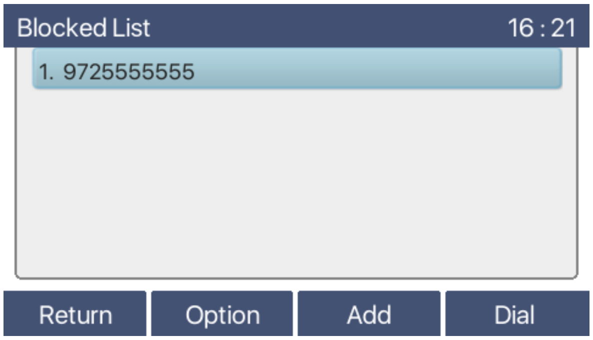


Blocked List

A number added to the Blacklist will be automatically rejected. User can still call a block listed Number.

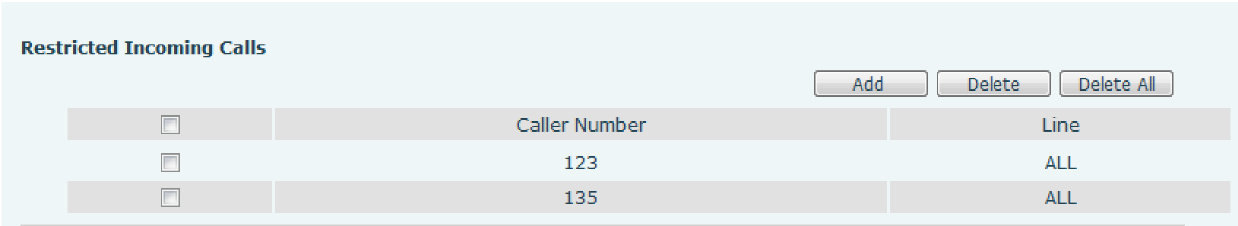
Phone interface: There are multiple ways to add a number to Block List.

- Option 1: A number can be added by going to [Menu] >> [PhoneBook] >> [Bocked List] >> [Add].
- Option 2: Select a number from the phone book (both local and network), press Options then select Add to Blocked List.
- Option 3::Select any number in the call log then press Options, then select Add to Blocked List.



Web interface: There are various ways to add number to the blacklist on web page.

- Option 1: Go to [Phonebook] >> [Call list] >> [Restricted Incoming Calls] and Click Add.
- Option 2: Go to [Phonebook] >> [Contacts]. Select any number in the phone book (both local and network) then click Add to Blacklist.
- Option 3: Go to [Call logs]. Select any number in the call log and Click Add to Blacklist.

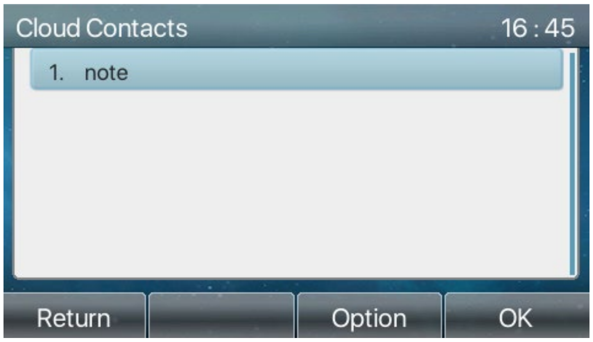


Cloud Phone Book

Configure Cloud Phone book

Cloud phonebook allows user to download a phonebook from a cloud server. This method ensures that all office personnel are using a phonebook from a single source and saves the effort of creating and maintaining individual contact lists.

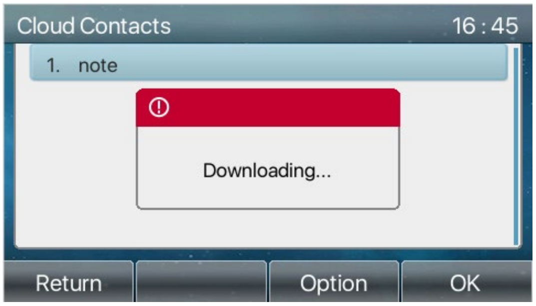
NOTICE! The cloud phonebook is ONLY temporarily downloaded to the phone each time it is opened to ensure the user gets the latest phonebook. However, downloading may take a couple seconds depending on network conditions. Therefore, it is highly recommended that important contacts are stored locally to the phone to save download time.
Phone interface: Press [Menu] >> [PhoneBook] >> [Cloud Contacts].

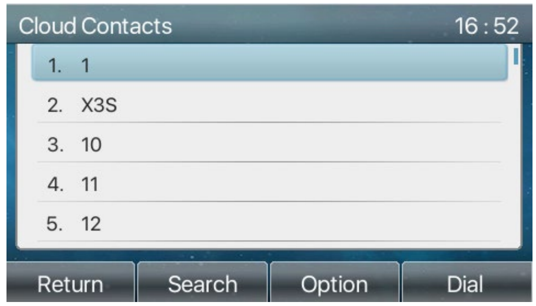


Downloading Cloud Phone book

In Cloud Contacts screen, open a cloud phone book by pressing [OK]. The phone will start downloading the phone book. The user will be prompted with a warning message if the download fails.

Once the cloud phone book is downloaded completely, the user can browse the contact list and dial the contact number the same as in the local phonebook.

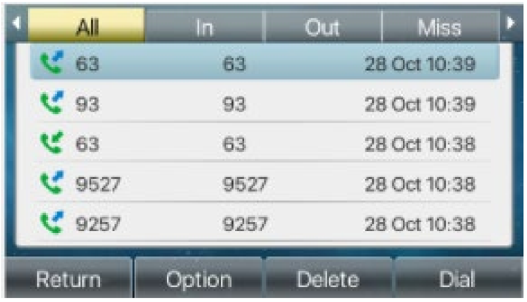




Call Log

The phone stores call records.

- Press [CallLog] to open the call record and view all incoming calls, outgoing calls and missed calls.
- Browse the call logs with up/down navigation keys.
  - Each call log record shows the call type, call party number and name'. User can view call log details by pressing [OK], dial the number by pressing [Dial], or add the call log number to phonebook by pressing [Option] >> [Add to Contacts].
  - Delete a call log by pressing [Delete] and clear all call logs by pressing [Options], [Delete All].



Users can filter the call records of specific call types to narrow down the scope of search records, and select a call record type by using the left / right navigation keys.

 - Missed Call Log

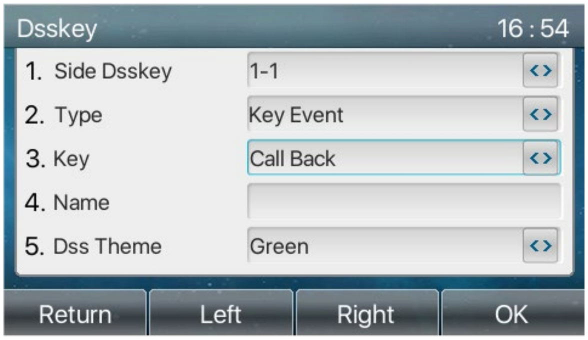
 - Incoming Call Log

 - Outgoing Call Log

 - Forward Call Log

Function Key

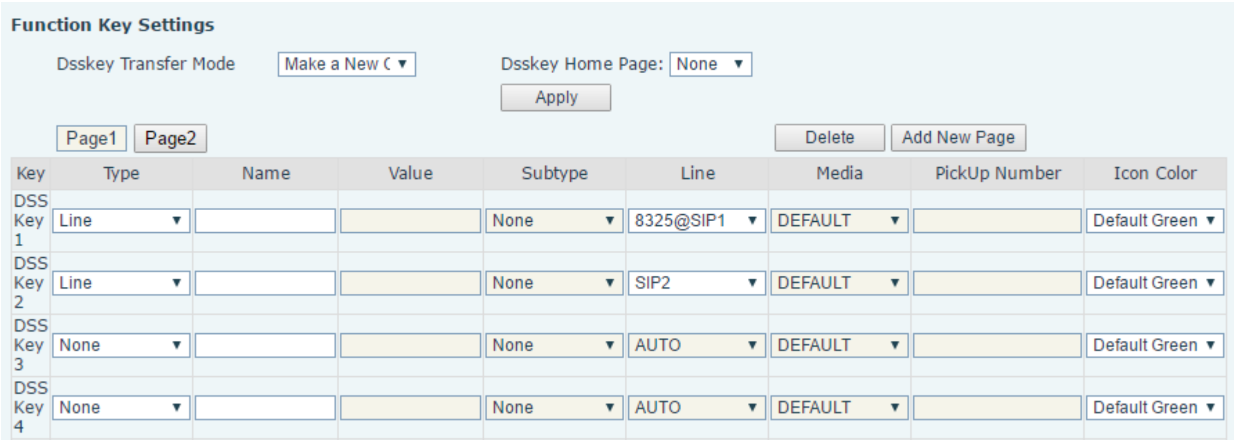
Users can use the page switch key (side key 10) to switch DSS display pages quickly. In addition, the user can also long press each DSS key to modify the corresponding key Settings.



The DSS Key can be configured as follows,

- Memory Key
  - Voice mail/Speed Dial/Intercom/BLF/Presence/Call Park/Call Forward.
- Line
- Key Event
  - /Lock/VoiceMail/Directory/Join/CallLog/Flash/Memo/MWI/DND/Hold/Transfer/Phonebook/Redial/Pickup/Call Forward (to specified line)/Headset/ SMS/Release
- DTMF
- BLF List Key
- Multicast Paging and Listening
- Action URL
- XML Browser

Webpage interface: [Function key] >> [Side Key or Softkey].

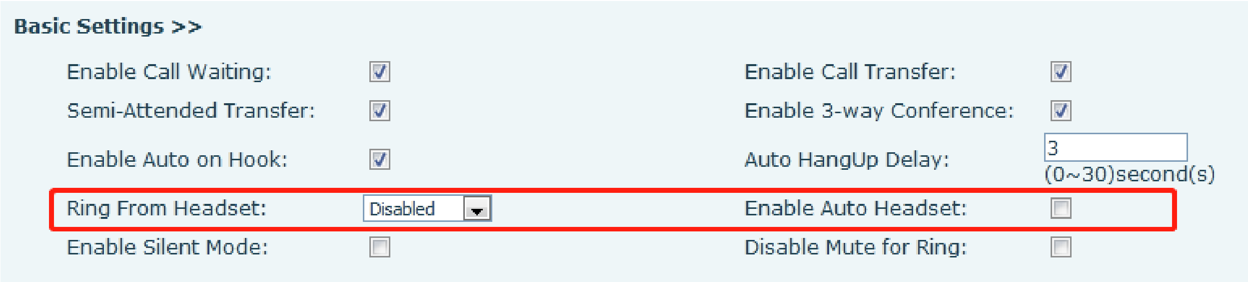


For more information refer to **Function Key** and **LED Definition**.

Headset

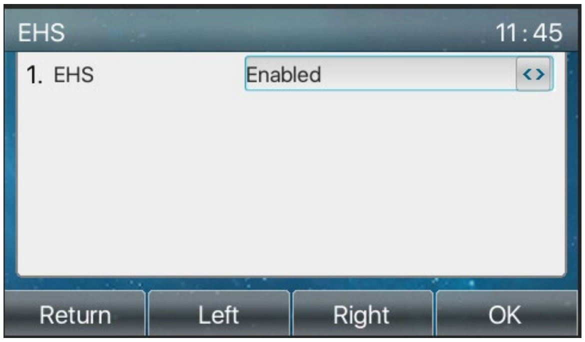
Wired Headset

- The phone supports a wired headset with an RJ9 connector.
- After connecting the headset to the phone, the Headset DSS key will light green.
- The headset answering mode and ring tone can be set through the phone interface or web interface.
    - Web interface: Go to [Phone settings] >> [Features] >> [Basic Settings] and enable Ring From Headset.
    - Phone interface: Go to [Menu] >> [[Features] >> [General] >> and enabler [Ring From Headset].



EHS Headset

To enable EHS from the phone, go to [Menu] >> [Features] >> [Advanced] >> [EHS]. Use the left/right navigation keys to toggle EHS from Disabled to Enabled and press OK.



Bluetooth Headset

The ePhone3 supports a Bluetooth headset via the ESI Bluetooth adapter.

Bluetooth device compatibility

Bluetooth for ESI ePhones - Supported BT Versions

	ePhone3 v2	ePhone4x v2	ePhoneX Built-in BT	ePhone8 (Built-in BT)	ePhone3	ePhone4x
BT 1.0 & 1.0B	Yes*	Yes	Yes	Yes	N/A	N/A
BT 1.1	Yes*	Yes	Yes	Yes	N/A	N/A
BT 1.2	Yes*	Yes	Yes	Yes	N/A	N/A
BT 2.0	Yes*	Yes	Yes	Yes	N/A	N/A
BT 2.1	Yes*	Yes	Yes	Yes	N/A	N/A
BT 3.0	Yes*	Yes	No	Yes	N/A	N/A
BT 4.0	No	Yes	No	Yes	N/A	N/A
BT 4.1	No	No	No	Yes	N/A	N/A
BT 4.2	No	No	No	Yes	N/A	N/A

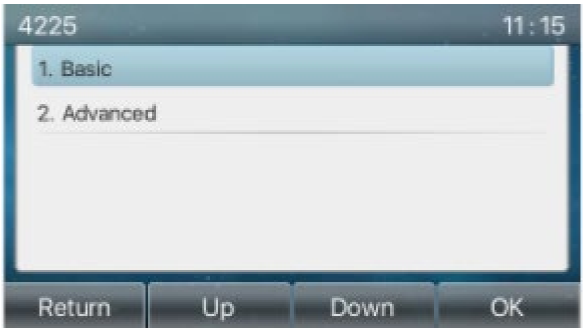
*There is no menu option in the e3v2 for Bluetooth but inserting the BT dongle that comes with ESI's BT headset will function and support audio.

Note: Bluetooth is backwards compatible. Therefore devices that support BT v4.2 will actually support headsets that user an older BT version. The difference between Bluetooth version comes down to speed. Therefore the closer the phone and headset BT versions are to v4.2 the higher bandwidth. This allows for faster data sharing with less lag and quicker response times between devices.

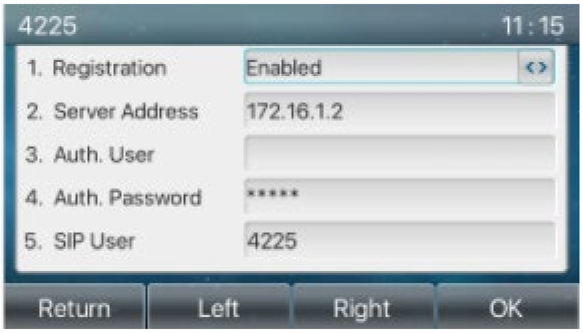
Advanced

Line Configurations

- From the phone, press [Menu] >> [Advanced] >> [Accounts], select a line account and press OK.



- Select [Basic] and press OK.
- Toggle Registration to Enabled by using the left/right navigation keys.
- Enter account information and press OK to save.



The same can be done from the web portal by going to [Line] >> [SIP].

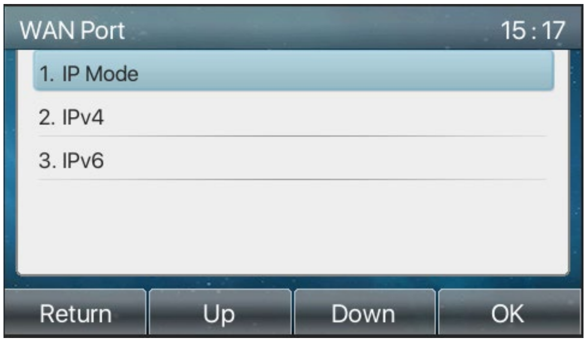
Network Settings

From the phone, go to [Menu] >> [Advanced] >> [Network] >> [Network].

Network

- IP Mode

There are 3 IP Mode options, IPv4, IPv6 and IPv4 & IPv6.
Use the left/right navigation keys to select a mode and press OK to save.

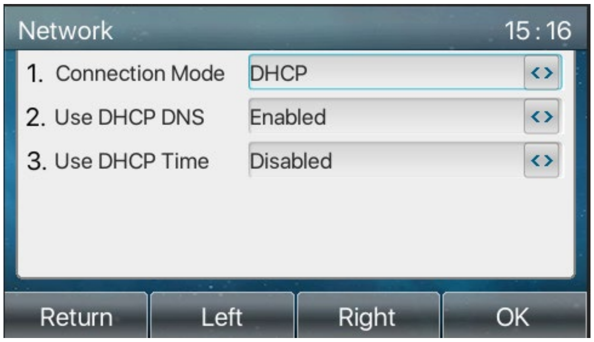


- IPv4

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

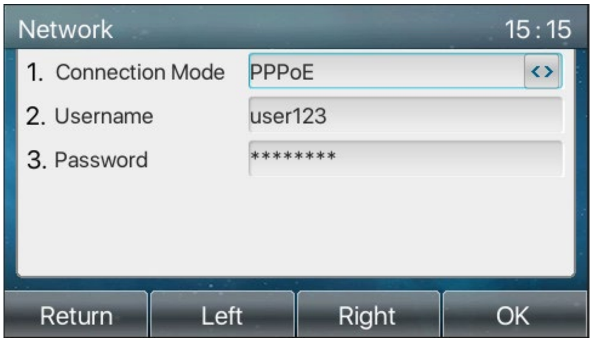
When using DHCP mode, phone will get the IP address from DHCP server (router).

- Use DHCP DNS: It is enabled by default. When enabled, phone will get DNS address from DHCP server.
- Use DHCP time: It is disabled by default. When enabled, phone will get DNS address from DHCP server.



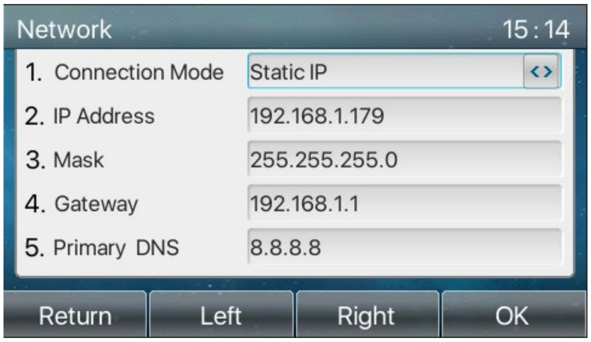
When using PPPoE, phone will get the IP address from PPPoE server.

- Username: PPPoE user name.
- Password: PPPoE password.



When using Static IP mode, user must configure the IP address manually.

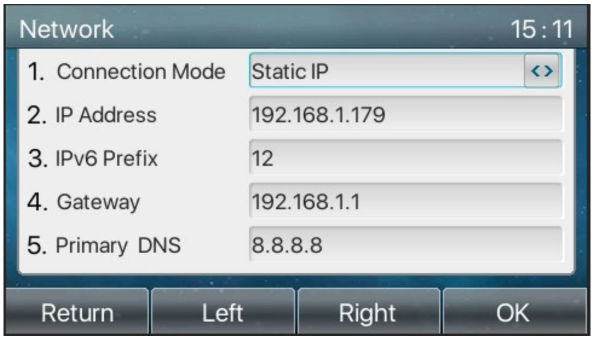
- IP Address: Phone IP address.
- Mask: subnet mask of your LAN.
- Gateway: The gateway IP address.
- Primary DNS: Primary DNS address. The default is 8.8.8.8, Google DNS server address.
- Secondary DNS: Phone will use secondary DNS when primary DNS is not available.



- IPv6

In IPv6, there are 2 connection modes, DHCP and Static.

  - DHCP: Phone will retrieve IP information from a DHCP server automatically.
  - Static IP: Phone IP information must be configured manually.



QoS & VLAN

- LLDP

Link Layer Discovery Protocol (LLDP) is a vendor independent link layer protocol used by network devices for advertising their identity, capabilities to neighbors on a LAN segment.
Phone can use LLDP to find the VLAN switch or other VLAN devices.

- CDP

CDP (Cisco Discovery Protocol) is a not-for-profit charity that runs the global disclosure system for investors, companies, cities, states and regions to manage their environmental impacts. According to the CDP, Cisco devices can share the OS version, IP address, hardware version and so on.

- WAN VLAN

Use navigation keys to enable/disable.

- LAN VLAN

Use navigation keys to enable/disable.

Parameters	Description
LLDP setting	
Report	Enable or Disable LLDP
Interval	LLDP requests interval time
QoS	
QoS Mode	Enable or Disable QoS, configure SIP DSCP and Audio DSCP
WAN VLAN	
WAN VLAN	Enable or Disable WAN VLAN and configure WAN VLAN
LAN VLAN	
LAN VLAN	Enable or Disable LAN VLAN and configure LAN VLAN
CDP	
CDP	Enable or Disable CDP. Set CDP interval time

VPN

Virtual Private Network (VPN) allows the 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 may need to be established before activating a line of service. 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. The phone only supports non-encrypted basic authentication and non-encrypted data tunneling. For users who need data encryption, use OpenVPN.

- L2TP

To establish an L2TP connection, log into the phone web portal, and go to [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. Click Apply. The phone will try to connect to the L2TP server. This can also be done from the phone by going to Menu >> Advanced >> Network >> VPN.

When the VPN connection is established, the VPN IP Address should be displayed in the VPN status. There may be a delay in the connection being established. User may need to refresh the page to update the status.

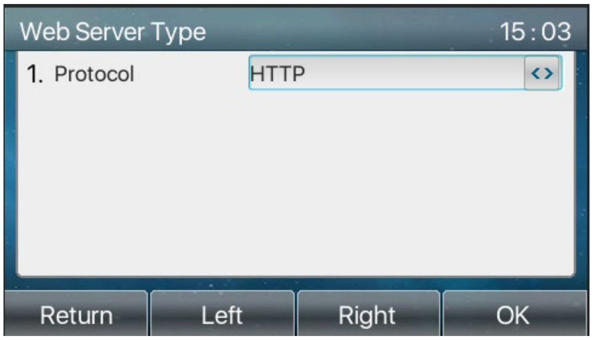
Once the VPN is configured, the phone will connect with the VPN automatically until user disables it. Sometimes, if the VPN connection does not establish immediately, user can reboot the phone and check if VPN connected after reboot.

- OpenVPN
  - To establish an OpenVPN connection, get the following authentication and configuration files from the OpenVPN hosting provider and name them as follows,
    OpenVPN Configuration file: client.ovpn
    CA Root Certification: ca.crt
    Client Certification: client.crt
    Client Key: client.key
  - Log in to the phone web page.
  - Go to [Network] >> [VPN] >>[OpenVPN Files], and select OpenVPN Configuration file.
  - Upload the files to the phone.
  - Check "Enable VPN"
  - Select "OpenVPN" in VPN Mode.
  - Click "Apply" to enable OpenVPN connection.

Once configured, the phone will connect with the VPN automatically until user disables it.

Web Server Type

Configure the Web Server mode to be HTTP or HTTPS and will be activated after the reboot. Then user could use http/https protocol to access pone web page.

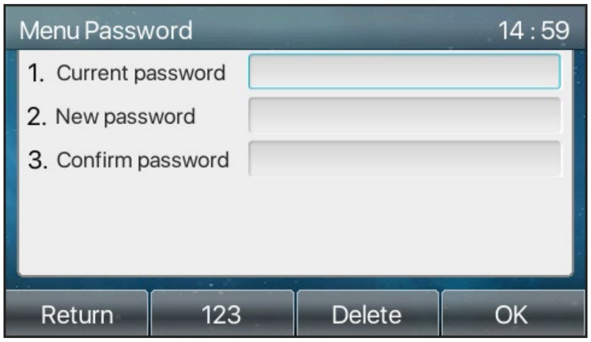


Set Passwords

Menu Password:

When the phone is in the default standby mode,

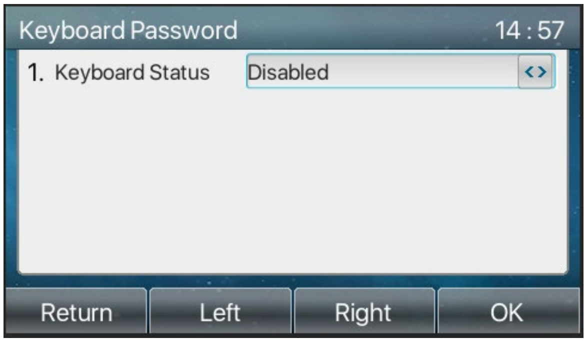
- Select [Menu] >> [Advanced setting] >> [Security] >> [Menu Password], enter Current password, New password, Confirm password and press [OK].
  As default, the Advance setting password is 123.
  Menu password grants permission to access the advanced setting.



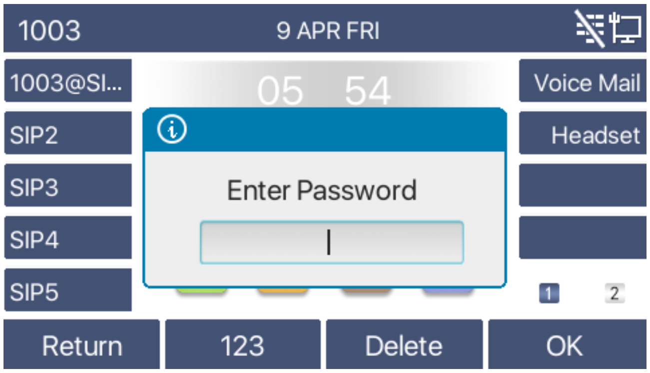
Keyboard Password:

Keyboard password is used to unlock the phone once it's locked.

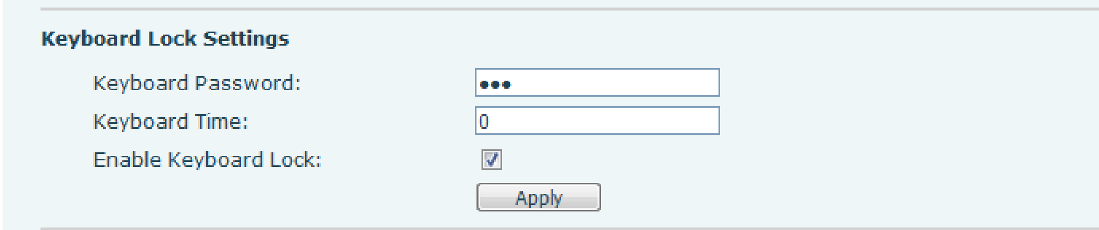
- Select [Menu] >> [Advanced setting] >> [Security] >> [Keyboard Password].
- Use the left/right navigation keys to toggle Keyboard Status lock to All, Menu or DSSKey.
- Set KeyLock Timout
- Press OK to save.



- Once keyboard lock is set, long press # to lock the phone. A lock icon will appear in the top right corner of the screen. Phone will prompt "Enter Password" when any key is pressed.

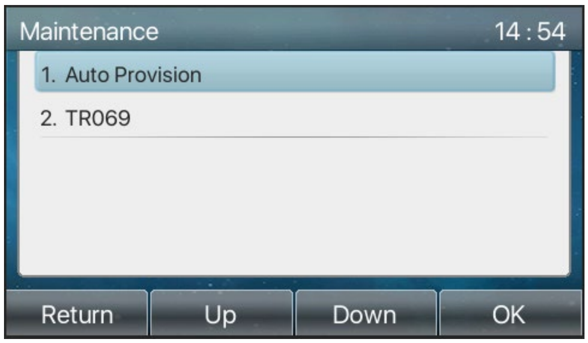


Keyboard Lock Settings can also be set from the phone web page by going to [Phone Settings] >> {Advanced] >> [Keyboard Lock Settings].

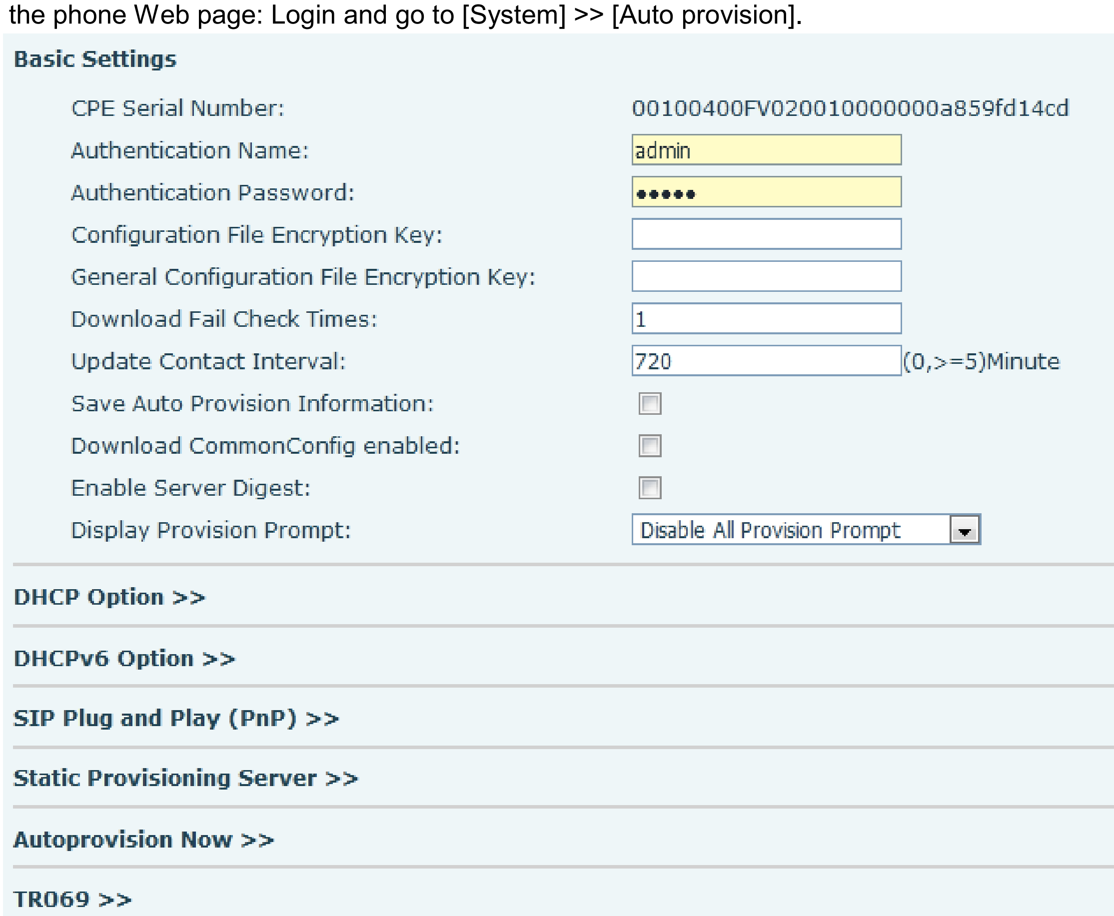


Maintenance

Configure phone web interface access:
From the phone, go to [Menu] >> [Advanced] >> [Maintenance].



From the phone Web page: Login and go to [System] >> [Auto provision].



The options are SIP PnP, DHCP options, Static Provisioning Server, TR069. If all of the 4 methods are enabled, the priority from highest to lowest will be **PNP>DHCP>TR069>Static Provisioning**.

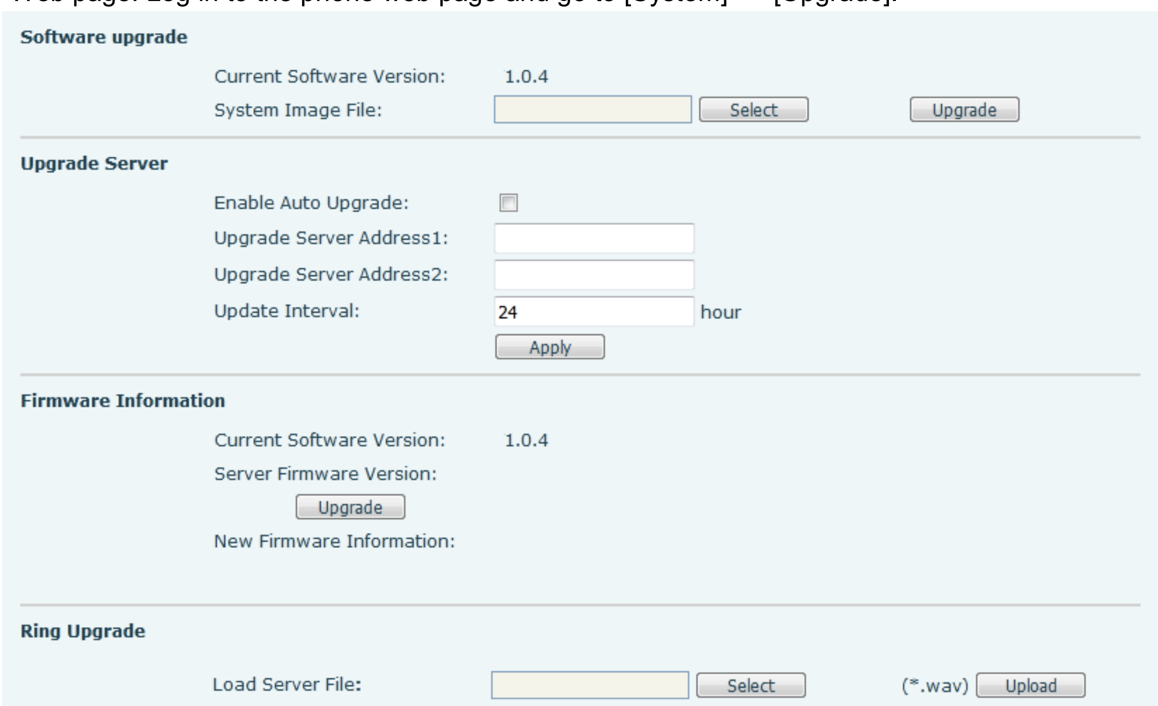
The transfer protocols available are FTP, TFTP, HTTP, HTTPS.

Parameters	Description
Basic settings	
CPE Serial Number	Phone serial number
Authentication Name	The user name of provision server
Authentication Password	The password of provision server
Configuration File Encryption Key	If the phone configuration file is encrypted, user will add the encryption key here.
General Configuration File Encryption Key	If the common configuration file is encrypted, user will add the encryption key here.
Download Fail Check Times	If download fails, phone will retry with the configured times.
Update Contact Interval	Phone will update the phonebook at the configured interval time. If it is 0, the feature is disabled.
Save Auto Provision Information	When checked, the phone will save the HTTP/HTTPS/FTP user name and password. If the provision URL is kept, the information will be kept.
Download Common Config enabled	When checked, the phone will download the common configuration file.
Enable Server Digest	When this feature is enabled, if the configuration of server is changed, phone will download and update.
DHCP Option and DHCPv6 Option	
Option Value	Configure DHCP option. DHCP option supports three methods to get the provision URL, DHCP custom option / DHCP option 66 / DHCP option 43. The default is Option 66.
Custom Option Value	Custom Option value is allowed from 128 to 254. The option value must be same as server define.
Enable DHCP Option 120	Use Option120 to get the SIP server address from DHCP server.
SIP Plug and Play (PnP)	
Enable SIP PnP	Check this option to enable PNP. If PnP is enabled, phone will send a SIP SUBSCRIBE message with broadcast method. Any server that supports this feature will respond and send a Notify with URL to phone. Phone can get the configuration file with the URL.
Server Address	Broadcast address. Default is 224.0.0.0.
Server Port	PnP access port
Transport Protocol	PnP protocols are TCP or UDP.
Update Interval	PnP message interval.
Static Provisioning Server	
Server Address	Provisioning server address. Support both IP address and domain address.
Configuration File Name	The configuration file name. If it is left empty, phone will request the common file and device file which is named as its MAC address. The file name can be a common name, $mac.cfg, $input.cfg. The file format supports CFG/TXT/XML.
Protocol Type	Transferring protocol type. Supports FTP, TFTP, HTTP and HTTPS
Update Interval	Configuration file update interval time. Default is 1, means phone will check the update every 1 hour.

Update Mode	Provision Mode. 1. Disabled. 2. Update after reboot. 3. Update after time interval.
TR069	
Enable TR069	Select to enable TR069
ACS Server Type	There are 2 Server Type options, Common and CTC.
ACS Server URL	ACS server address
ACS User	ACS server username (can be up to is 59 character)
ACS Password	ACS server password (can be up to is 59 character)
Enable TR069 Warning Tone	If TR069 is enabled, there will be a prompt tone when connecting.
TLS Version	TLS version (TLS 1.0, TLS 1.1, TLS 1.2)
INFORM Sending Period	INFORM signal interval time. It ranges from 1s to 999999s
STUN Server Address	Configure STUN server address
STUN Enable	Enable STUN server for TR069

Firmware Upgrade

- Web page: Log in to the phone web page and go to [System] >> [Upgrade].



- Phone interface: Go to [Menu] >> [Advanced setting] >> [Firmware Upgrade].

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

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

  Xxxxx
  Xxxxx
  Xxxxx
  Xxxxx
- After the update interval arrives, if the server has available files and versions, the phone will prompt that a firmware upgrade is available. Click OK to view the version information and upgrade.

Factory Reset

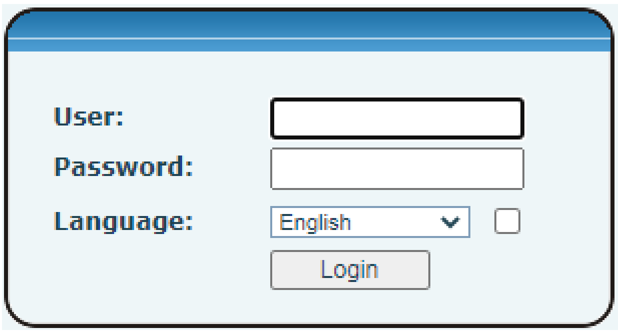
While the phone is in default standby mode:

- Press [Menu] >> [Advanced Settings], enter password and press [OK]. Default password is 123.
- Select [Factory reset].
- Use the left/right navigation keys to select all configurations that are to be factory reset.
- Press [OK] to factory reset phone. The phone will restart automatically after clearing the selected configurations.

Web Configurations

Web Page Authentication

The user can manage the phone from the phone web page. User must have user name and password to log into the web page.



System >> Information

User can view the system information of the device including,

- Model
- Hardware Version
- Software Version
- Uptime
- Network Mode
- MAC Address
- IP address
- Subnet Mask
- Default Gateway
- SIP User information
- SIP account status (Registered / Unapplied / Trying / Timeout)

System >> Account

On this page the user can change the password for the web login page.
Users with administrator rights can also add or delete users, manage users, and set permissions and passwords for new users.

System >> Configurations

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

- Clear Configurations
  Select the module in the configuration file to clear.
  SIP: account configuration.
  AUTOPROVISION: automatically upgrades the configuration
  TR069:TR069 related configuration
  MMI: MMI module, including authentication user information, web access protocol, etc.
  DSS Key: DSS Key configuration
- Clear Tables
  Select the local data table to be cleared, all tables are selected by default.
- Reset Phone
  The phone data will be cleared, including configuration and database tables.

System >> Upgrade

Upgrade the phone software version, customized ringtone, background, DSS Key icon, etc. Ring tone support ".wav" format.

System >> Auto Provision

The Auto Provision settings help IT manager or service provider to easily deploy and manage the phones in mass volume. For Auto Provision details, refer to **Auto Provision**.

System >> Tools

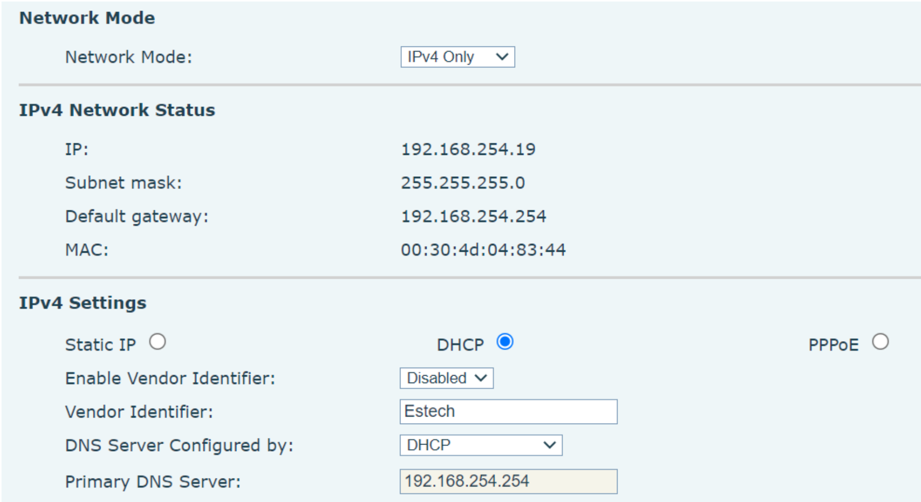
Tools provided in this page help users to identify and troubleshoot issues. Refer to **Trouble Shooting** for more detail.

System >> Reboot Phone

Restart the phone without factory resetting.

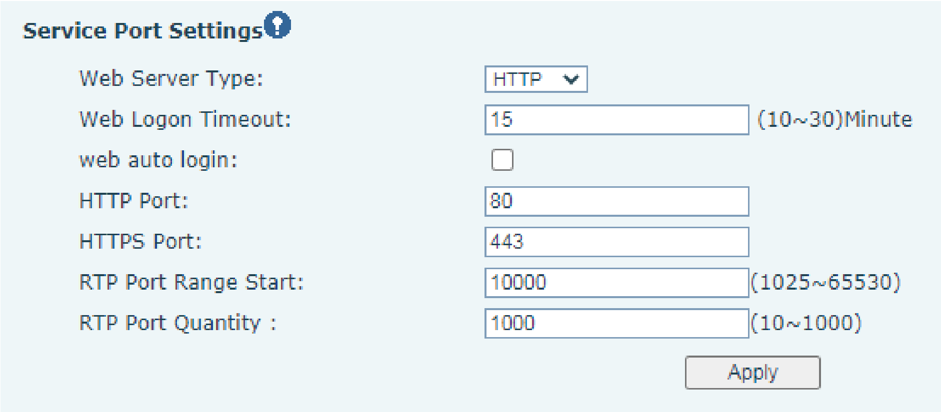
Network >> Basic

This page allows users to configure network connection types and parameters.
The default network priority is IPv4 Only.



Network >> Service Port

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



Parameter	Description
Web Server Type	Reboot is required for this option to take effect after settings. Optionally, the web page login is HTTP/HTTPS.
Web Logon Timeout	Default is 15 minutes. The timeout will automatically log out the user.
Web auto login	After the timeout, user does not need to enter a user name password.
HTTP Port	Default is 80. User can set ports other than 80 for better security. Such as :8080, webpage login: HTTP://ip:8080
HTTPS Port	Default is 443.
RTP Port Range Start	The value range is 1025 to 65530. The value of the RTP port starts from the initial value set. For each call, the value of voice port adds 2.
RTP Port Quantity	Number of calls.

Network >> VPN

Users can configure a VPN connection on this page. Refer to **VPN** section for more details.

Network >> Advanced

Advanced network Settings are typically configured by the IT administrator to improve the quality of the phone service.

Line >> SIP

Configure Line service on this page.
Select a line then configure its parameters listed below.

Parameters	Description
Register Settings	
Line Status	Displays the current line status at page loading. To get the up to date line status, user must refresh the page manually.
Activate	Select to activate the line
Username	Enter the username of the service account.
Authentication User	Enter the authentication user of the service account
Display Name	Enter the display name to be sent in a call request.
Authentication Password	Enter the authentication password of the service account
Realm	Enter the SIP domain if requested by the service provider
Server Name	Input server name.
SIP Server 1	
Server Address	Enter the IP or FQDN address of the SIP server
Server Port	Enter the SIP server port, default is 5060
Transport Protocol	Set up the SIP transport line using TCP, UDP or TLS.
Registration Expiration	Set SIP expiration date.
SIP Server 2	
Server Address	Enter the IP or FQDN address of the SIP server
Server Port	Enter the SIP server port, default is 5060
Transport Protocol	Set up the SIP transport line using TCP, UDP or TLS.
Registration Expiration	Set SIP expiration time in seconds.
Proxy Server Address	Enter the IP or FQDN address of the SIP proxy server.
Proxy Server Port	Enter the SIP proxy server port, default is 5060.
Proxy User	Enter the SIP proxy user.
Proxy Password	Enter the SIP proxy password.
Backup Proxy Server Address	Enter the IP or FQDN address of the backup proxy server.
Backup Proxy Server Port	Enter the backup proxy server port, default is 5060.
Basic Settings	
Enable Auto Answering	When enabled, the incoming calls will be answered automatically after the delay time
Auto Answering Delay	Set the delay for incoming call before the call is automatically answered
Call Forward Unconditional	When enabled, 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 before an un-answered gets forwarded.
Transfer Timeout	Set the timeout of call transfer process.
Conference Type	Set the type of call conference. Local=set up call conference by the device itself. Supports two remote parties maximum, 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 Server
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 Hotline	Enabling hotline configuration, the device will dial to the specific number immediately at audio channel opened by off-hook handset or hands-free speaker or headphone
Hotline Delay	Set the delay for hotline before the system automatically dials
Hotline Number	Set the hotline dialing number
Dial Without Registered	Set call out by proxy without registration
Enable Missed Call Log	If enabled, the phone will save missed calls to the call history record.
DTMF Type	Set the DTMF type to be used for the line
DTMF SIP INFO Mode	Set the SIP INFO mode to send '*' and '#' or '10' and '11'
Enable DND	Enable Do-not-disturb. Any incoming call to this line will be rejected
Use VPN	Enable to allow the line to use VPN route
Use STUN	Set the line to use STUN for NAT traversal
Enable Failback	Enable this option to switch back to primary server when it is available.
Failback Interval	A Register message is used to periodically detect the time interval for the availability of the main Proxy.
Signal Failback	Multiple proxy cases, whether to allow the invite/register request to also execute failback.
Signal Retry Counts	The number of attempts that the SIP Request considers proxy unavailable under multiple proxy scenarios.
Codecs Settings	Set the priority and availability of the codecs by adding or removing them from the Enabled Codecs list.
Advanced Settings	
Use Feature Code	When this setting is enabled, the features in this section will not be handled by the device itself but by the server instead. In order to control the enabling of the features, the device will send 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
DND Disabled	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	Set the feature code to dial to the server

Unconditional	
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
Call Waiting On Code	Set the feature code to dial to the server
Call Waiting Off Code	Set the feature code to dial to the server
Send Anonymous On Code	Set the feature code to dial to the server
Send Anonymous Off Code	Set the feature code to dial to the server
RTP Encryption	Enable RTP encryption such that RTP transmission will be encrypted
Enable Session Timer	Enable the call session timer. The call session will end if there is no new session timer event update received after the timeout period
Session Timeout	Set the session timer timeout period
Enable BLF List	Enable/Disable BLF List
BLF List Number	BLF List allows one BLF key to monitor the status of a group. Multiple BLF lists are supported.
Response Single Codec	If enabled, the phone will use a single codec in response to an incoming call request
BLF Server	The registered server will receive the subscription package from ordinary application of BLF phone. Enter the BLF server, if the sever does not support subscription package, the registered server and subscription server will be separated.
Keep Alive Type	Set the line to use dummy UDP or SIP OPTION packet to keep NAT pinhole opened
Keep Alive Interval	Set the keep alive packet transmission interval
Keep Authentication	Keep the authentication parameters from previous authentication
Blocking Anonymous Call	Reject any incoming call that does not present caller ID
User Agent	Set the user agent, the default is Model number and Software Version.
Specific Server Type	Set the line to collaborate with specific server type
SIP Version	Set the SIP version
Anonymous Call Standard	Set the standard to be used for anonymous calls
Local Port	Set the local port
Ring Type	Set the ring tone type for the line
Enable user=phone	Sets user=phone in SIP messages.
Use Tel Call	Set use tel call
Auto TCP	Use TCP protocol to guarantee usability of transport for SIP messages above 1500 bytes
Enable Rport	Set the line to add Rport in SIP headers
Enable PRACK	Set the line to support PRACK SIP message
DNS Mode	Select DNS mode, A, SRV, NAPTR
Enable Long Contact	Allow more parameters in contact field per RFC 3840
Enable Strict Proxy	Enables the use of strict routing. When the phone receives packets from

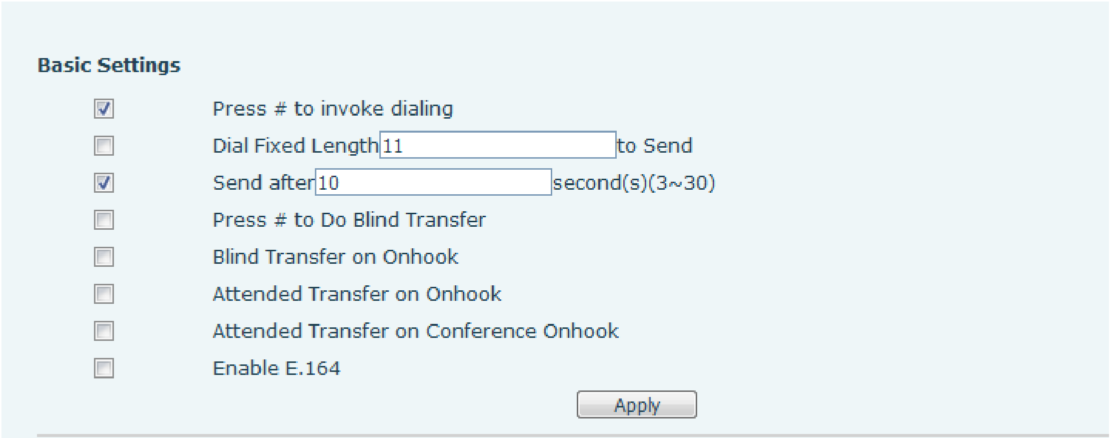
	the server, it will use the source IP address, not the address in via field.
Convert URI	Convert not digit and alphabet characters to %hh hex code
Use Quote in Display Name	Adds quotes in display name, i.e. “123” vs 123
Enable GRUU	Support Globally Routable User-Agent URI (GRUU)
Sync Clock Time	Time Sync with server
Enable Inactive Hold	With the post-call hold capture package enabled, you can see in the INVITE package taht SDP is inactive.
Caller ID Header	Set the Caller ID Header
Use 182 Response for Call waiting	Set the device to use 182 response code at call waiting response
Enable Feature Sync	Feature Sync with server
Enable SCA	Enable/Disable SCA (Shared Call Appearance)
CallPark Number	Set the CallPark number.
Server Expire	Set the timeout to use the server.
TLS Version	Choose TLS Version.
uaCSTA Number	Set uaCSTA Number.
Enable Click To Talk	With the use of a special server, click to call out directly after enabling.
Flash mode	Chose Flash mode, normal or SIP info.
Flash Info Content-Type	Set the SIP info content type.
Flash Info Content-Body	Set the SIP info content body.
PickUp Number	Set the scramble number when Pickup is enabled.
JoinCall Number	Set JoinCall Number.
Intercom Number	Set Intercom Number.
Unregister On Boot	Select to enable logout function.
Enable MAC Header	Open user registration with IP package and user agent MAC.
Enable Register MAC Header	Open user agent registration with MAC.
BLF Dialog Strict Match	Whether to enable accurate matching of BLF sessions.
PTime(ms)	Set Ptime field interval
SIP Global Settings	
Strict Branch	Set to strictly match the Branch field.
Enable Group	Set to enable group.
Enable RFC4475	Set to enable RFC4475.
Enable Strict UA Match	Enable strict UA matching.
Registration Failure Retry Time	Set the registration failure retry time.
Local SIP Port	Set the local SIP port.

Line >> SIP Hotspot

Refer to **SIP Hotspot** section.

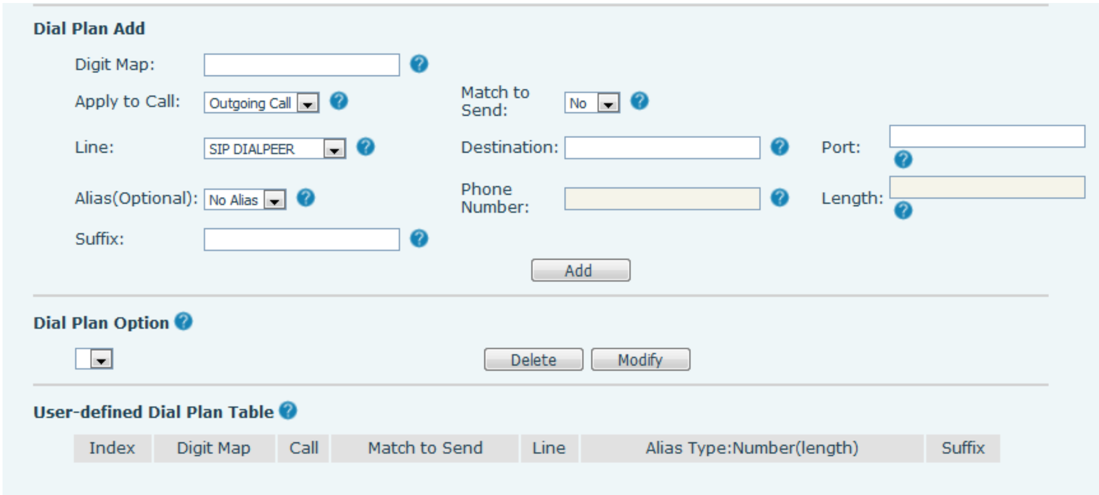
Line >> Dial Plan

Basic Settings



Parameters	Description
Press # to invoke dialing	The user dials the other party's number and then adds the # number to dial out
Dial Fixed Length	The number entered by the user is automatically dialed out when it reaches a fixed length
Timeout dial (send after x seconds)	The system dials automatically after timeout
Press # to Do Blind Transfer	The user enters the number to be transferred and then presses the "#" key to transfer the current call to a third party
Blind Transfer on Onhook	After the user enters the number, hang up the phone to transfer the current call to a third party.
Attended Transfer on Onhook	Hang up the phone when other end answers to transfer the current call to a third party.
Attended Transfer on Conference Onhook	During a three-way call, hang up the phone and the remaining two parties remain on the call.
Enable E.164	Refer to E.164 standard specification

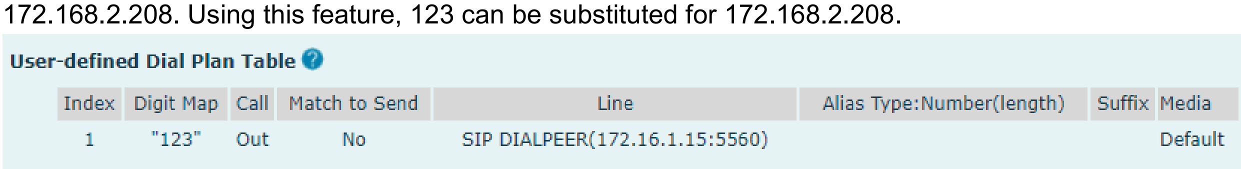
Dial Plan Add:



Parameters	Description
Dial rule	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. Default is 5060 for SIP.
Alias	Set the Alias. This is the text to be added, replaced or deleted.
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.
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 optional.

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

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


User-defined Dial Plan Table

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

Example 2: Partial Substitution -- To dial a long distance call 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.

User-defined Dial Plan Table

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

Example 3: Addition -- Two examples are shown. In the first case, it is assumed that 0 must be dialed before any 11 digit number beginning with 13. In the second case, it is assumed that 0 must be dialed before any 11 digit number beginning with 135, 136, 137, 138, or 139. Two different special characters are used.
x -- Matches any single digit that is dialed.
[ ] – Brackets specify a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.

Line >> Action Plan

1. When an IP phone calls a phone, an action plan can be put in place to handle the call.
2. When SIP calls, multicast calls or intercom calls are made, the device converts calls that conform to the number rules into group calls.

Parameter	Description
Number	Auxiliary phone number
Type	When connected
Direction	For call mode, incoming/outgoing/both
Line	Select outgoing lines.
Username	User name
Password	Password
URL	Mcast Address (mcast://IP:port)
User Agent	Set user agent information

Line >> Basic Settings

Set up the register global configuration.

Parameters	Description
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
SIP P2P Settings	
Enable Auto Answering	Set phone to auto answer an incoming call after auto answer delay time
Auto Answering Delay	Set the delay time
DTMF Type	Set DTMF type
DTMF SIP INFO Mode	Set to send 10/11 or */#
Enable Preview	Enable preview
Use VPN	Select to use VPN

Line >> RTCP-XR

RTCP-XR mode is based on RFC3611 (RTP Control Extended Report), which can measure and evaluate network packet loss, delay and voice quality by sending RTCP-XR packets.

Parameters	Description
VQ RTCP-XR Settings	
VQ RTCP-XR Session Report	VQ report on whether session mode is enabled or not.
VQ RTCP-XR Interval Report	Whether to turn on Interval mode for VQ report sending.
Period for Interval Report(5~99)	The time interval at which VQ reports are sent periodically.
Warning threshold for Moslq(15~40)	When the phone calculated the Moslq value x10 below the set threshold, a warning is issued.
Critical threshold for Moslq(15~40)	When the phone calculates the Moslq value x10 below the set threshold, the critical report is issued.
Warning Threshold for Delay(10~2000)	When the one-way delay of the phone is greater than the set threshold, warning is issued.
Critical Threshold for Delay(10~2000)	When the phone computes that the one-way delay is greater than the set threshold, the critical report is issued.
Display Report options on phone	Whether to display VQ report data for the last call throufh the phone.
Display Report Options on web	Whether to display the VQ report data for the last call through the web page.

Phone settings >> Features

Configuring phone features.

Parameters	Description
Basic Settings	
Enable Call Waiting	Enable this setting to allow user to take second incoming call during an established call. Default enabled.
Enable Call Transfer	Enable Call Transfer.
Semi-Attended Transfer	Enable Semi-Attended Transfer
Enable 3-Way Conference	Enable 3-way conference
Enable Auto On hook	The phone will hang up and return to idle automatically after call
Auto HangUp Delay	Specify Auto Onhook time delay, the phone will hang up and return to idle automatically after call
Ring from Headset	Enable Ring from Headset by selecting it, the phone plays ring tone from headset.
Enable Auto Headset	Enable this feature to enable headset auto answer.
Enable Silent Mode	When enabled, the ring volume is muted. User can use the volume keys and mute key to unmute.
Disable Mute for Ring	When enabled, user cannot mute the phone
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
Default Ext Line	Select the default line to use for outgoing calls
Ban Outgoing	When enabled, user cannot dial out to any number.
Hide DTMF	Configure the hide DTMF mode.
Enable CallLog	Select whether to save the call log.
Enable Restricted Incoming List	Enable restricted call list.
Enable Allowed Incoming List	Enable the allowed call list.
Enable Restricted Outgoing List	Enable the restricted allocation list.
Enable Country Code	When selected, the country code is enabled.
Country Code	Enter the country code.
Area Code	Enter the area code.
Enable Number Privacy	Enable number privacy.
Match Direction	Matching direction. There are two rules, from right to left and from left to right.
Start Position	Enable number privacy after the start of the defined position.
Hide Digits	Turn on number privacy to hide the number of digits.
Allow IP Call	If enabled, user can dial out with IP address
P2P IP Prefix	Prefix a point-to-point IP call.
Caller Name Priority	Change caller ID display priority.
Emergency Call Number	Set Emergency Call Number. This number can be dialed even when the keyboard is locked. Dial the number and press the dial key.
Search path	Select the search path for contacts.
LDAP Search	Select an LDAP for searching contacts
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 is sent by the specified server. .
Enable Pre-Dial	Disabled. User enters number and audio channel will open automatically. Enabled. User enters number without opening audio channel automatically.
Enable Multi Line	If enabled, up to 10 simultaneous calls can exist on the phone. If disabled, up to 2 simultaneous calls can exist on the phone.
Line Display Format	Custom line format: Options are SIPn, SIPn: xxx, xxx@SIPn
Contact As White List Type	Options are NONE, BOTH, DND White List, FWD White List
Block XML When Call	Disable XML push on call.
SIP notify	When enabled, the phone displays the information when it receives the relevant notify content.
Tone Settings	
Enable Holding Tone	When enabled, a tone plays when call is on hold
Enable Call Waiting Tone	When enabled, a tone plays when a call is waiting to be answered
Play Dialing DTMF Tone	Play DTMF tones on the phone when dialing digits. Default enabled.
Play Talking DTMF Tone	Play DTMF tone on the phone when user presses digits during talking. Default enabled.
DND Settings	
DND Option	Select an option to take effect on the line, on the phone, or off.
Enable DND Timer	Enable DND Timer, If enabled, DND is automatically turned on from the start time to the off time.
DND Start Time	Set DND Start Time
DND End Time	Set DND End Time
Intercom Settings	
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 a delay.
Enable Intercom Mute	Enable mute mode during the intercom call
Enable Intercom Tone	If the incoming call is an intercom call, the phone will play an intercom tone
Enable Intercom Barge	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.
Redial settings	
Enable Call Completion	Enable call completion
Enable Auto Redial	Redial the last number dialed automatically
Auto Redial Interval	Set a time interval to redial
Auto Redial Times	Set a number of times to attempt redial
Redial enter CallLog	Enter redial attempts into the call log
Response Code Settings	
DND Response Code	Set the SIP response code on call rejection while DND is enabled
Busy Response Code	Set the SIP response code on line busy
Reject Response Code	Set the SIP response code on call rejection
Password Dial Settings	
Enable Password Dial	Enable Password Dial by selecting it. When the number that is entered begins with the password prefix, the following N numbers after the password prefix will be hidden as *. N stand for the value which was enter in the Password Length field. Example: Set the password prefix to 3, enter the Password Length to 2, then enter the number 34567, it will display 3**67 on the phone.

Encryption Number Length	Configure the Encryption Number length
Password Dial Prefix	Configure the prefix of the password call number
Power LED	
Common	Standby power lamp state. Off when set to off. On is always bright red. Off by default.
SMS / Voice mail	This sets the status of power lamp when there is unread short message/voice message. Options are off/on/slow blink/fast blink. Default is slow blink.
Missed call	This sets the state of the power lamp when there is a missed call. Options are off/on/slow blink/fast blink. Default is slow blink.
Talk/Dial	The power lamp options are off and on. Off is off. On is always bright red. Default is off.
Ringing	Power lamp status when there is an incoming call, including off/on/slow blink/fast blink. Default is fast blink.
Mute	Power lamp status in mute mode, including off/on/slow flash/quick flash, off by default.
Hold/Held	The power lamp states are off/on/slow blink/fast blink. Default is off.
DssKey Settings	Set the BLF LED status and Line LED status. Options are on/off/slow blink/fast blink.
Notification Popups	
Display Missed Call Popup	Display a popup when a call is missed. Popup shows the number of missed calls.
Display Voice Mail Popup	Display a popup when there is a message waiting. Popup shows the number of messages waiting.
Display Device Connect Popup	Display a popup when a Wi-Fi dongle is plugged into phone USB port.
Display SMS Popup	Display a popup when an SMS message is waiting to be read.
Display Other Popup	Display a popup when the handset is not fully set on cradle, when registration fails, when IP acquisition fails, when TR069 connection fails, and other abnormalities.

Phone settings >> Media Settings

Change voice Settings.

Parameters	Description
Codecs Settings	Select voice codec: G.711A/U,G.722,G.729, G.726-16,G726-24,G726-32,G.726-40, G.723,ILBC,Opus
Enable / Disable Codecs	Select a codec
Media Settings	
Handset Volume	Set the Handset volume to 1~9
Default Ring Type	Configure default ringtones. If no special ringtone is set for the phone number, the default ringtone will be used.
Speakerphone Volume	Set the hands-free volume to 1-9.
Headset Ring Volume	Set the volume of the headset ringtone to 1~9.
Headset Volume	Set the volume of the headset to 1~9.
Speakerphone Ring Volume	Set the volume of hands-free ringtone to 1~9.
G.723.1 Bit Rate	5.3kb/s or 6.3kb/s is available.
DTMF Payload Type	Enter the DTMF payload type, the value must be 96~127.
Headset Mic Gain	Set the MIC gain to fit different models of headsets.
Handset Mic Gain	Set the handset MIC gain.
Handsfree Mic Gain	Set the handsfree MIC gain.
Opus Payload Type	Set Opus load type, range 96~127.
OPUS Sample Rate	Set Opus sampling rate, including opus-nb (8KHz) and opus-wb (16KHz).
ILBC Payload Type	Set the ILBC Payload Type, the value must be 96~127.
ILBC Payload Length	Set the ILBC Payload Length
Enable Voice Mail Tone	When there is a new voice message, the phone will start a special dial tone.
Enable VAD	Enable or disable voice activity detection.
Onhook Time	Configure a minimum response time
EHS Type	EHS headset is available after enabling.
Enable Hookflash	Enable or disable hookflash by checking the box.
RTP Control Protocol(RTCP) Settings	
CNAME user	Set CNAME user
CNAME host	Set CNAME host. Local IP address is set as the default
RTP Settings	
RTP keep alive	Hold the call and send the packet after 30s
RTP Relay	Forward an RTP stream from a source to either a multicast or multiple unicast destinations
Alert Info Ring Settings	
Value	Set the value to specify the ring type.
Line	Set which line to assign the value
Ring Type	Select a ring type value to None, Default, or 1 through 7

Phone settings >> MCAST

This feature allows user to a 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 without involving SIP signaling. User can also configure the phone to receive an RTP stream from pre-configured multicast listening address without involving SIP signaling. User can specify up to 10 multicast listening addresses.

Parameters	Description
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
Enable Prio Chan	Enable priority channel
Enable Emer Chan	Enable emergency channel
Host: port	Listened multicast server's multicast IP address and port.
Channel	Select a channel that m multicast group will join
MCAST Dynamic	Set an expire time

Phone settings >> Action

Action URL Event Settings

Action URL's are used for IPPBX systems to submit phone events.

Phone settings >> Time/Date

The user can configure the time Settings of the phone on this page.

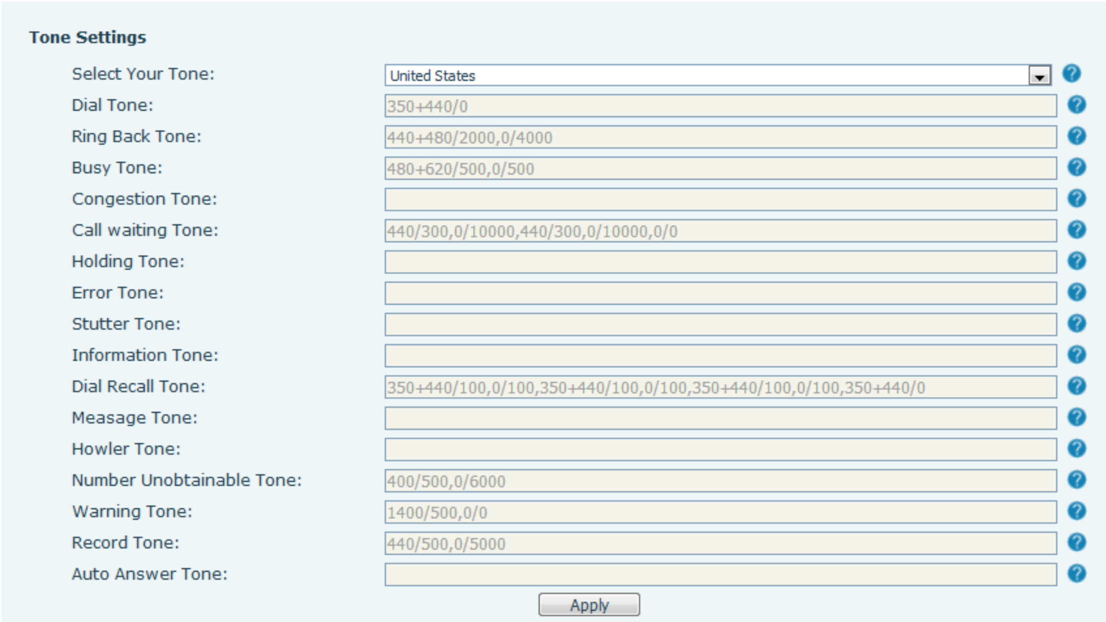
Parameters	Description
Network Time Server Settings	
Time Synchronized via SNTP	Enable time-sync through SNTP protocol
Time Synchronized via DHCP	Enable time-sync through DHCP protocol
Primary Time Server	Set primary time server address
Secondary Time Server	Set secondary time server address, when primary server is not reachable. The phone will try to connect to secondary time server to get time synchronization.
Time Zone	Select the time zone
Resync Period	Time of re-synchronization with time server
Time/Date Format	Select the time/date display format
12-Hour Clock	Select check box to set the time display in 12-hour mode
Daylight Saving Time Settings	
Location	Choose your location. Phone will set daylight saving time automatically based on the local
DST Set Type	Choose DST Set Type. If Manual, user will need to set the daylight savings start time and end time.
Fixed Type	Daylight saving time rules are based on specific dates or relative rule dates for conversion. Display in read-only mode in automatic mode.
Offset	Set the offset minutes when DST starts
Month start	The DST start month
Week start	The DST start week
Weekday start	The DST start weekday
Hour(s) start	The DST start hour
Month end	The DST end month
Week end	The DST end week
Weekday end	The DST end weekday
Hour(s) end	The DST end hour
Manual Time Settings	Set time manually

Phone settings >> Time plan

This page allows user to set a timed reboot, timed upgrade or timed forward for the phone.

Phone settings >> Tone

This page allows users to configure a phone prompt.
User can either select the country area or customize the area. If the area is selected, it will display the following information directly. If Custom is selected, user can modify the button tone, call back tone and other information.



Phone settings >> Advanced

User can configure the advanced configuration settings on this page.

- Screen Configuration.
  - Backlight Active Level
  - Backlight Inactive Level
  - Backlight Time
  - Screensaver
  - Timeout to Screensaver
- Power Saving
  Enable or disable power saving mode. Disabled by default.
- LCD Menu Password Settings.
  Set a Menu password. The password is 123 by default.
- UI Preference
  Customize the colors of the user interface.
- Keyboard Lock Settings.
  These settings will allow user to lock the phone manually by long pressing the lock key (# key) or to have the phone lock automatically after a set time.
- Greeting Words
  The greeting message will display on the top left corner of the LCD screen when the phone is idle, which is limited to 16 characters.

Phonebook >> Contact

User can add, delete, or edit contacts in the phonebook in this page. User can browse the phonebook and sort by name, phone, or filter contacts by group.
To add a new contact, click Add new contact, enter contact information and click “Add”.
To edit a contact, click Edit to the right of the contact. Edit the contact information and click OK to save.
To delete one or more contacts, check the box in front of the contacts to be deleted and click “Delete”.

Add multiple contacts into a group by selecting the group in the dropdown options in front of “Add to Group” at the bottom of the contact list. Note that a group must be added from the phone interface first. Select contacts with checkbox and click “Add to Group” to add selected contacts into the group. Similarly, user can select multiple users and add them into blacklist by clicking “Add to Blacklist”.

Phonebook >> Cloud phonebook

Cloud Phonebook

User can 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, enter the following information,

- Phonebook name (required)
- Phonebook URL (required)
- Authentication name (optional)
- Authentication 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 requests an authentication, user should also provide username and password.

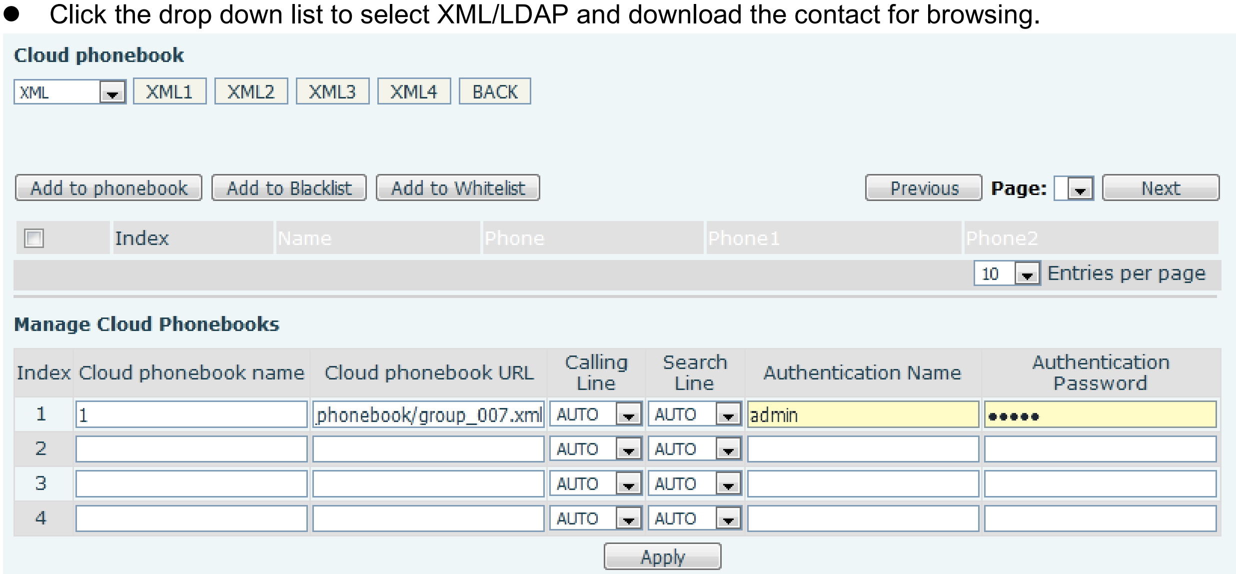
To configure a LDAP phonebook, the following information should be entered,

- Display Title (required)
- Server Address (required)
- Server Port (required)
- Search Base (required)
- Username (optional)
- Password (optional)

Note! Refer to the LDAP technical documentation before creating the LDAP phonebook and phonebook server.

After setting up the XML Voip directory or LDAPfrom the phone Web page,

- Select [Phone book] >> [Cloud phone book] >> [Cloud phone book] to select the type.
- Click the drop down list to select XML/LDAP and download the contact for browsing.



Phonebook >> Call List

- Restricted Incoming Calls:
  This is similar to a blacklist. Add the number to the blacklist, and the user will no longer receive calls from the stored number until the user removes it from the list.
  Users can add specific Numbers to the blacklist or add specific prefixes to the blacklist to block calls with all Numbers with this prefix.
- Allowed Incoming Calls:
  When DND is enabled, the incoming call number can still be called.
- Restricted Outgoing Calls:
  Adds a number that restricts outgoing calls and cannot be called until the number is removed from the table.

Phonebook >> Web Dial

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

Phonebook >> Advanced

Users can export the local phone book in XML, CSV, and VCF format and save it on the local computer.
Users can also import contacts into the phone book in XML, CSV, and VCF formats.

Attention! If the user imports the same phone book repeatedly, the same contact will be ignored. If the name is the same but the number is different, the contact is created again.
Users can delete groups or add new groups on this page. Deleting a contact group does not delete contacts in that group.

Call Logs

The user can browse the complete call record in this page. The call record can be sorted by time, call number, contact name or line, and the call record can be screened by call record type (incoming call, outgoing call, missed call, forward call).
The user can also save the number in the call record to his/her phone book or add it to the blacklist/whitelist.
Users can also dial the web page by clicking on the number in the call log.
Users can also download call records conditionally and save them locally.

Function Key >> Side Key

Side Key is a Key on both sides of the screen that functions as a shortcut Key. Each key can be customized in the phone webpage.

DSSkey Transfer Mode: Make a New Call, Blind Transfer, Attended Transfer, Conference Call, Play DTMF.
The side keys can be set as on the web page as shown below.

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.
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.
Action URL	The user can use a specific URL to make basic calls to the phone.
XML browser	Users can set the DSS Key for specific URL download and other operations.

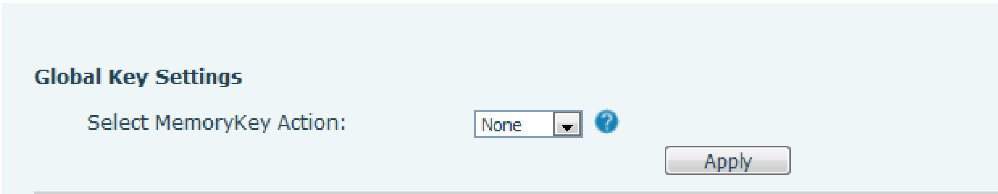
Function Key >> Softkey

The User Settings mode and display style, display page.

Parameter	Description
Softkey Mode	Selections are Disabled and More. Default is Disabled
Softkey Exit Style	Softkey Exit on Left or Right
Screen	
Call Dialer	2aB/Delete/Exit/Call Back/Dial/Join/Voice Mail (MWI)/Local Contacts/Pickup/Call Log/Missed/Clear/In/Dialed/Pause/ Next line/Prev line/Headset/Audio/Remote XML/DSS Key
Conference	Hold/Split/End/Release/Mute/DSS Key/Headset
Desktop	Call Log/Menu/Local Contacts/DND/Prev Account/Next Account/Blacklist/Call Back/Call Forward/ In/Lock/Memo/ Missed/Voice Mail (MWI)/Dialed/Reboot/Redial/Remote XML/SMS/ Headset/Status/DSS Key
Divert Dialer	2aB/Delete/Exit/Forward/Local Contacts/Call Log /Clear/Missed/Dialed/Headset/Audio/Remote XML /DSS Key/ In/Send
Ending	Redial/End/Headset/Release/DSS Key
Predictive Dialer	Dial/2aB/Delete/Exit/Call Back/Local Contacts/Redial/Clear /Pickup/Voice Mail (MWI)/Join/Call Log/Release/Missed/Pause/Dialed/ Headset/Audio/Remote XML/DSS Key/In/Next line /Prev line/Save
Ringing	Answer/Forward/Reject/Mute/Release/Headset/DSS key
Talking	Hold/Transfer/Conference/End/Mute/Release/New Call/ Local Contacts/Listen/Call Log/Next call/Prev call/RTP Private/Headset/DSS Key
Transfer Alerting	End/Transfer/Headset/Release/DSS Key
Transfer Dialer	Delete/Exit/2aB/Dial/Local Contacts/Transfer/ In Call Log/Clear/Missed/Dialed/Pause/Headset/Remote XML/DSS Key
Trying	End/Release/Headset/DSS Key
Waiting	Hold/Transfer/Conference/End/Answer/Forward/Mute/Next call/New call/Prev call/Reject/Release/Headset/Listen/DSS Key

Function Key >> Advanced

- Global Key Settings
  Select the function of the memory key while on a call. Selections are None, Call Hold, and Hangup. The configured memory key has a call path. If the global configuration is maintained, pressing the memory key again will maintain the call path. If configured to hang up, pressing the memory key again will hang up the call.



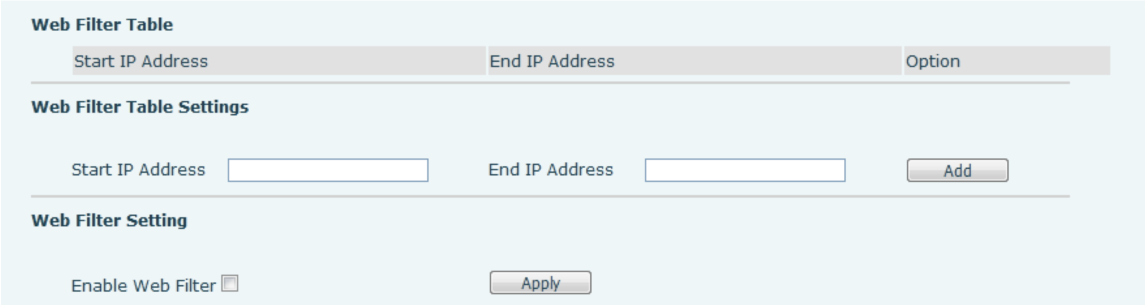
- Programmable key Settings
  The navigation keys can be programmed to perform other functions. The keys can be set to perform a different function from the Desktop, Dialer, while in call, or with a long press. Select a function from the drop down list and click Apply.

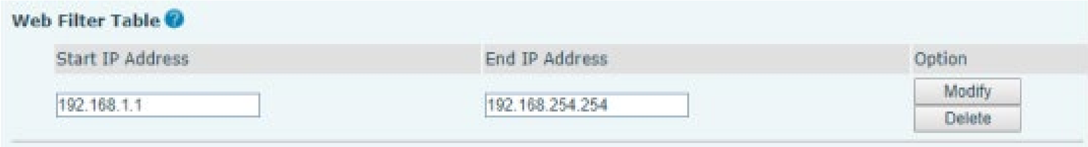
Application >> Manage Recording

Manage audio recordings. Refer to **Record** for details of recording.

Security >> Web Filter

The user can set up a configuration management phone that only allows devices with a specific network segment IP access.

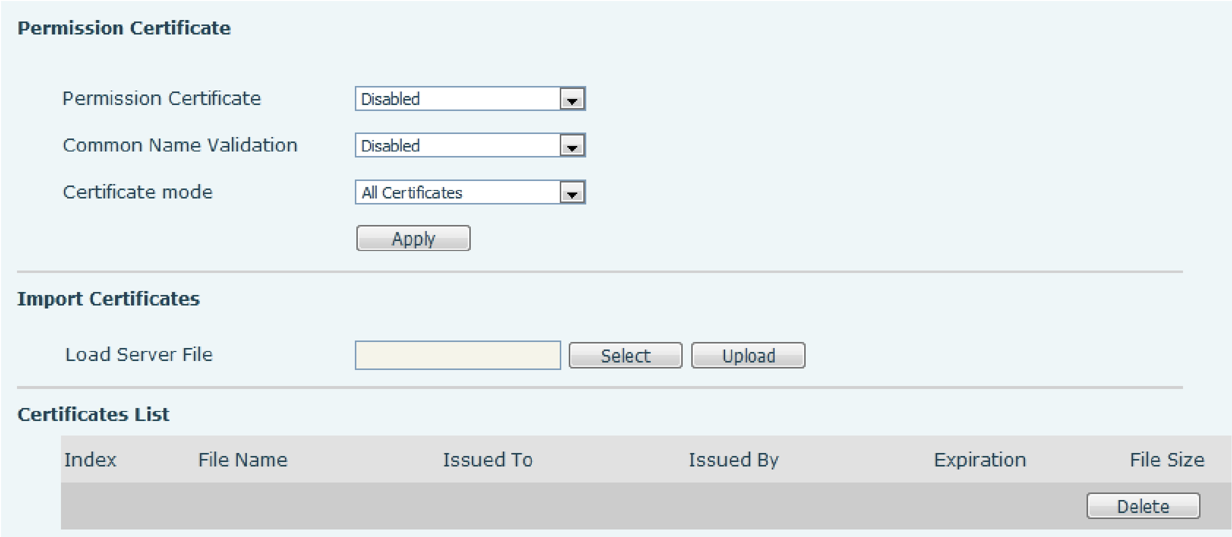




Configure the starting IP address within the start IP, end the IP address within the end IP, and click [Add] to take effect. A large network segment can be set, or it can be divided into several network segments to add. If the user wants to delete segment, select the initial IP of the network segment to be deleted and click [Delete] to take effect.
Enable web page filtering: select Enable Web Filter and click the Apply to take effect.
Note: If the device you are accessing is in the same network segment as the phone, do not configure the filter segment of the web page to be outside your own network segment. Otherwise you will not be able to log in the web page.

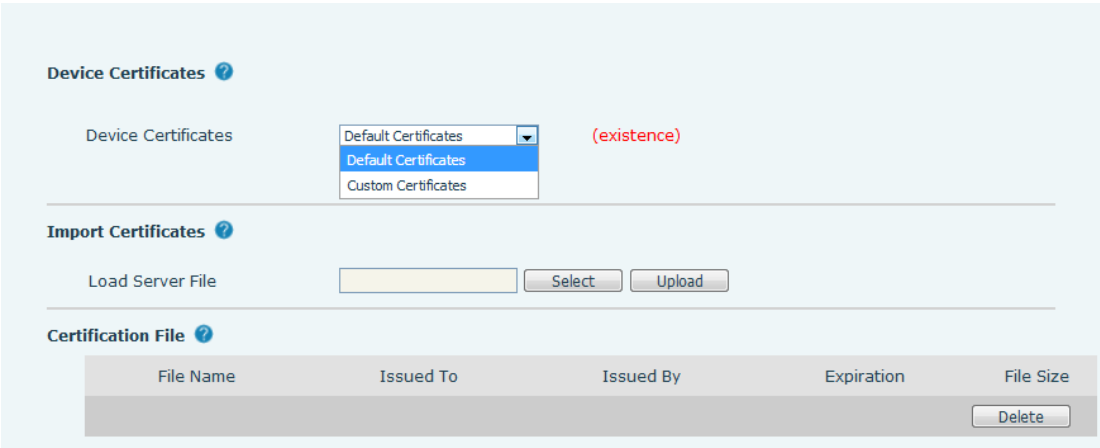
Security >> Trust Certificates

Enable/Disable permission certificate, common name validation and certificate mode.
Upload and delete certificates.

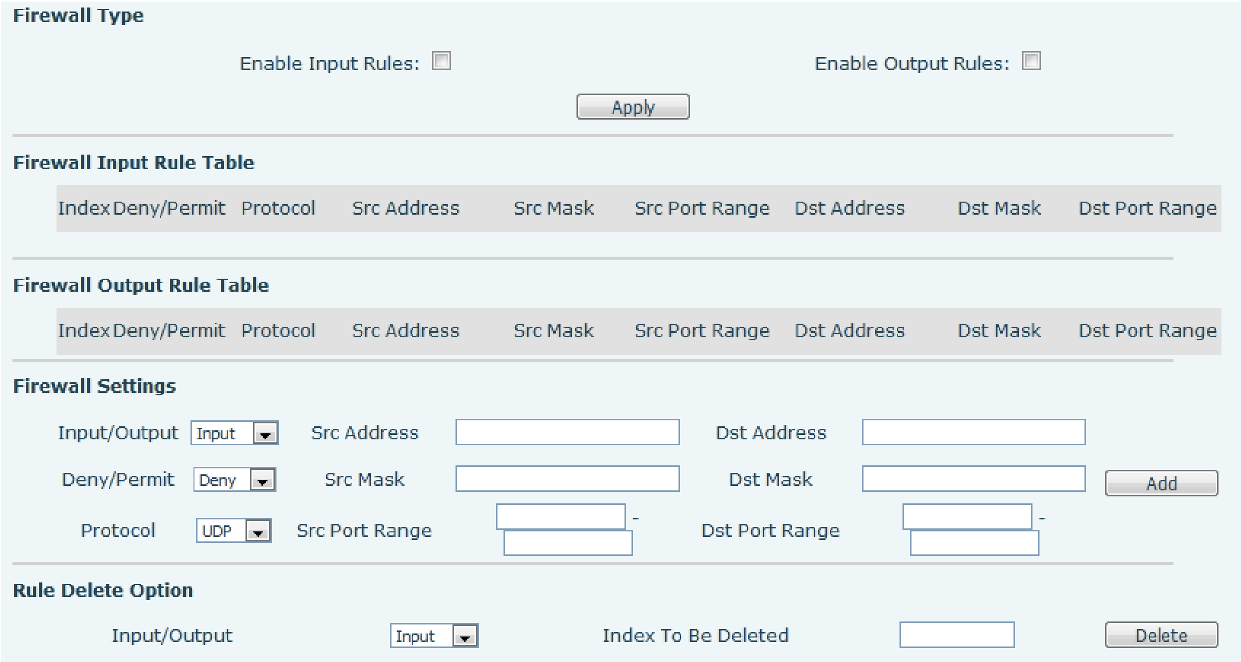


Security >> Device Certificates

Select the device certificates as default or custom certificate.
Upload and delete certificates.



Security >> Firewall



The user can set firewall rules from the phone web GUI. Using these Settings can prevent some malicious network access, or restrict internal user access to some resources of the external network, which can improve security.
This phone firewall supports two types of rules, input rules and output rules. Each rule is assigned an ordinal number, allowing up to 10 for each rule.
Considering the complexity of firewall settings, the following is an example to illustrate:

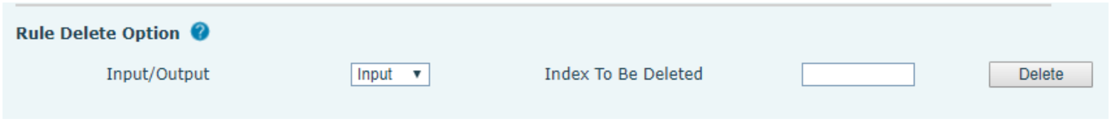
Parameter	Description
Enable Input Rules	Indicates that the input rule application is enabled.
Enable Output Rules	Indicates that the output rule application is enabled.
Input/Output	To select whether the currently added rule is an input or output rule.
Deny/Permit	To select whether the current rule configuration is disabled or allowed;
Protocol	There are three types of filtering protocols: TCP | UDP | ICMP.
Src Port Range	Filter port range
Src Address	Source address can be host address, network address, or all addresses 0.0.0.0; It can also be a network address similar to *.*.*.0, such as: 192.168.1.0.
Dst Address	The destination address can be either the specific IP address or the full address 0.0.0.0; It can also be a network address similar to *.*.*.0, such as: 192.168.1.0.
Src Mask	This is the source address mask. When configured as 255.255.255.255, it means that the host is specific. When set as 255.255.255.0, it means that a network segment is filtered.
Dst Mask	This is the destination address mask. When configured as 255.255.255.255, it means that the host is specific. When set as 255.255.255.0, it means that a network segment is filtered.
Dst Port Range	Filter port range.

After setting, click [Add] and a new item will be added in the firewall input rule, as shown in the figure below:

Firewall Input Rule Table

Index	Deny/Permit	Protocol	Src Address	Src Mask	Src Port Range	Dst Address	Dst Mask	Dst Port Range
1	deny	udp	192.168.1.0	192.168.1.154	0-9	255.255.255.0	255.255.255.0	0-9

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



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

Device Log >> Device Log

When a problem is encountered, user can collect the device log and use the log to diagnose a problem. Refer to **Get Log Information** section.

Trouble Shooting

The user can try the following methods to restore normal operation to the phone or collect relevant information.

Get Phone System Information

Users can view phone information from the phone by pressing the [Menu] >> [Status]. The following information will be provided:
Network information.
Phone information: Model, software and hardware version, etc.
Account: Is the line registered, attempting to register, etc.

Reboot Phone

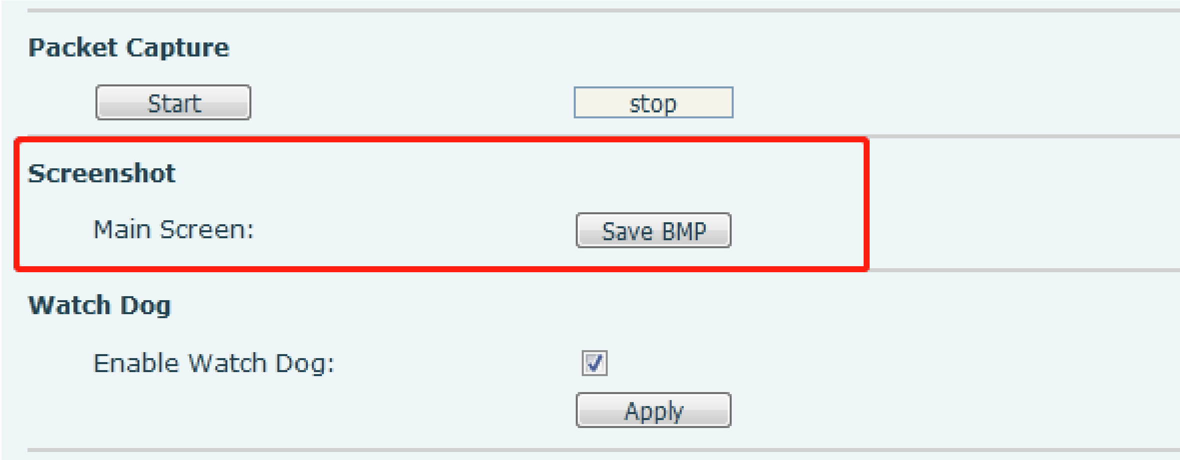
Reboot the phone from the phone by going to [Menu] >> [Basic] >> [Reboot System]. Confirm the action by pressing [OK]. An alternative method is to unplug phone power, wait 30 seconds then plug it back in.

Reset Phone to Factory Default

Resetting phone to Factory Default will erase all of the user's configuration, preferences, database and profiles and restore the phone back to factory default.
To perform a factory reset, press [Menu] >> [Advanced]. Input the password to enter the Advanced options, then choose [Factory Reset] and press [OK]. The phone will reboot to a clean factory default state.

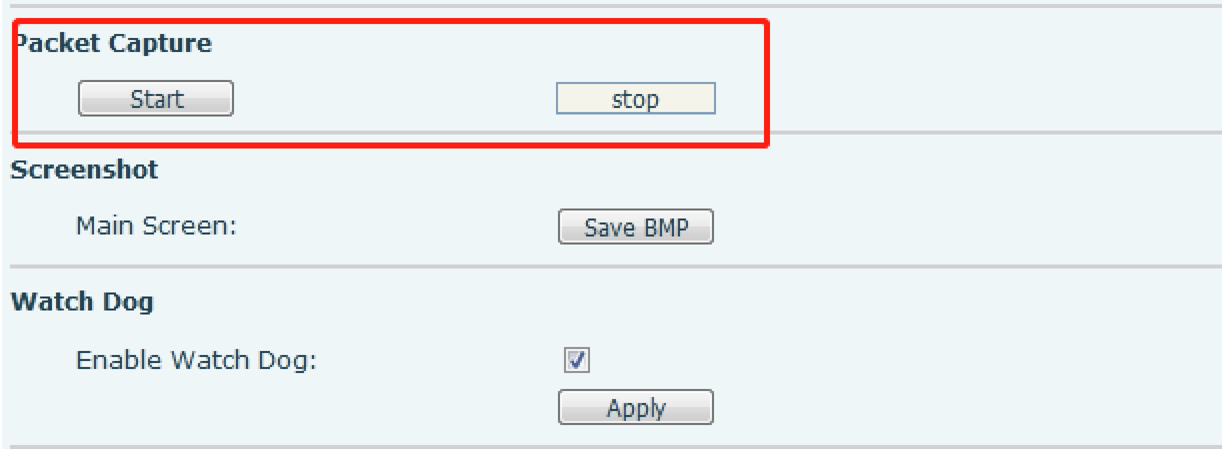
Screenshot

If there is a problem with the phone, a screenshot may help identify the problem. To obtain screen shots, log into the phone webpage and go to [System] >> [Tools], and click Save BMP to capture an image of the main screen.



Network Packets Capture

Sometimes it is helpful to capture the network packets of the phone for troubleshooting a problem. To capture network packets of the phone, log into the phone web portal, go to [System] >> [Tools] >> [Packet Capture] and click [Start]. A pop-up message will ask user to save the capture file. User will then duplicate the issue then click [Stop] when done. The network packets will be saved to a file for analysis.



Get Log Information

Log information is helpful when encountering an exception problem. In order to get the log information of the phone, log in the phone web page and go to [Device Log]. Click the [Start], duplicate the issue, click [End], and [Save] the file for analysis of the problem.

Common Trouble Cases

Trouble Case	Solution
Phone will not boot up	The phone is powered by an external power supply via power adapter or by a PoE switch. Use the power adapter provided by ESI or a PoE switch that meets power requirements and check that the device is well connected to power source. If "POST MODE" appears on phone screen, the phone system image is damaged.
Phone will not register to a service provider	Check that 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, the icon , [WAN disconnected] will be flashing in the middle of the screen. Check that the device has an IP address. Check system information, if the phone displays "Negotiating…", the phone does not have an IP address. Check that the network configurations are correct. If network connection is good, check line configurations again. If all configurations are correct, contact support for assistance.
No Audio or Poor Audio in Handset	Check that handset is connected to handset () port and NOT headset () port. Network bandwidth and delay may not be suitable for audio call at the moment.
Poor Audio or Low Volume in Headset	There are two Headset wiring types on the market. Verify that the headset you are using is compatible. 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. Lower the speaker volume until the chopping goes away.