Akuvox

IT82 Series

Indoor Monitor User Manual

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1 Production Overview

1.1 Production Description

IT82 series is an Android SIP-based with smooth touch-screen Indoor monitor. It can be connected with Akuvox door phone for unlock and monitor. Residents can communicate with visitors via audio and video call, and support remote unlock the door. It is more convenient and safe for residents to check the visitor identity through its video preview function. IT82 series is often applicable in villas , apartments, building and so on.

FCC Caution:

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions : (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Note : This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

—Reorient or relocate the receiving antenna.

—Increase the separation between the equipment and receiver.

—Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

—Consult the dealer or an experienced radio/TV technician for help.

FCC Radiation Exposure Statement:

This equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment .
This transmitter must not be co-located or operating in conjunction with any other antenna or transmitter.
This equipment should be installed and operated with minimum distance 20cm between the radiator& your body.

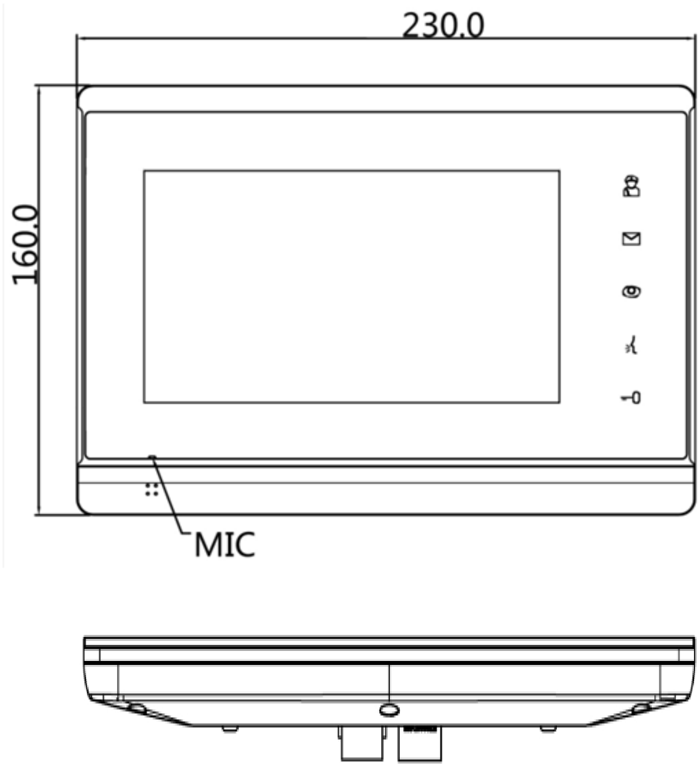
1.2 Technical Specification

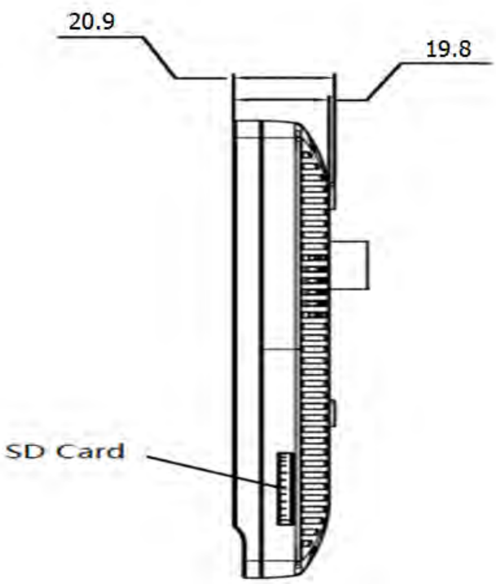
Model	IT82 series
Graphic Display	● 7 inch capacitive touch screen TFT LCD, 1024x600 resolution, 16:9 wide screen aspect ratio
Network Protocol	● SIP RFC3261,TCP/UDP/IP,RTP
Voice Codec	● G.711A/U, G.729, G.722, iLBC_13_3, iLBC_15_2,OPUS
Video Codec	● H.264, H.263,H.265,MJPEG
Network Interface	● Dual switched 10/100Mbps port
IP assignment	● Static IP, DHCP
Memory	● RAM:2GB, Flash:4GB
Management	● LCD Menu Configuration, WebUI
Dimension	● 230x160x30mm
Storage	● Album ● External SD device

1.3 Button Instruction

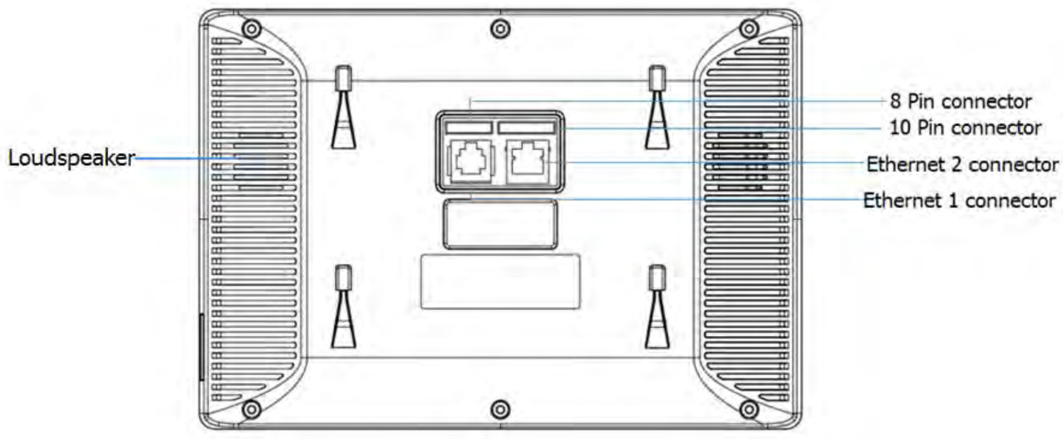
Interface	Description
	Manage center button. Click to make a call with manage center.
	Message button. Click to direct access to message interface.
	Monitor button. Click to view the monitoring from outdoor environment.
	Intercom button. Click to enter the dialing interface.
	Unlock sensor button. The physical Unlock key is only used for E10S now.

1.4 Dimension



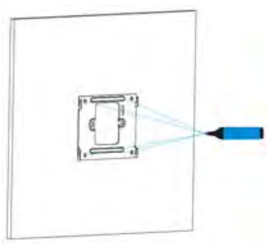


1.5 Equipment Appearance And Interface Description



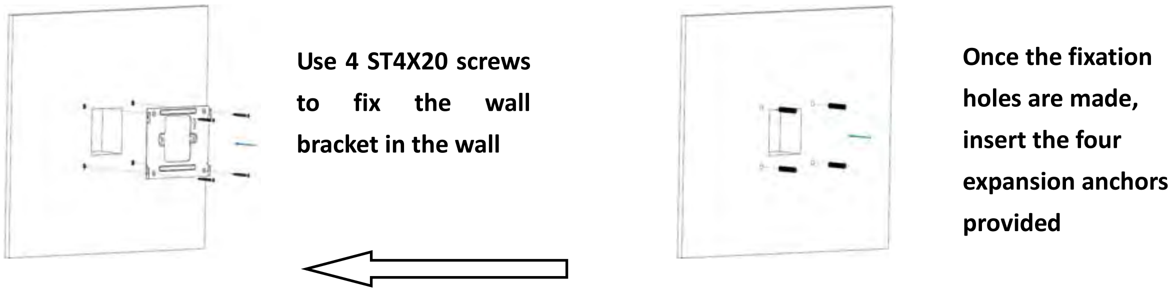
Interface	Description
8 PIN connector	+12V | GND | X | X | 485+ | 485- | NO | COM For power supply and 485 connector
10PINconnect	IO1 | IO2 | IO3 | IO4 | IO5 | IO6 | IO7 | IO8 | X | GND For 8 security connector
Ethernet 1	Network interface can be connected to a hub, switch or other network access devices.
Ethernet 2	Share the network access from ethernet 1 port, and for PC and other equipment connection.

1.6 Installation

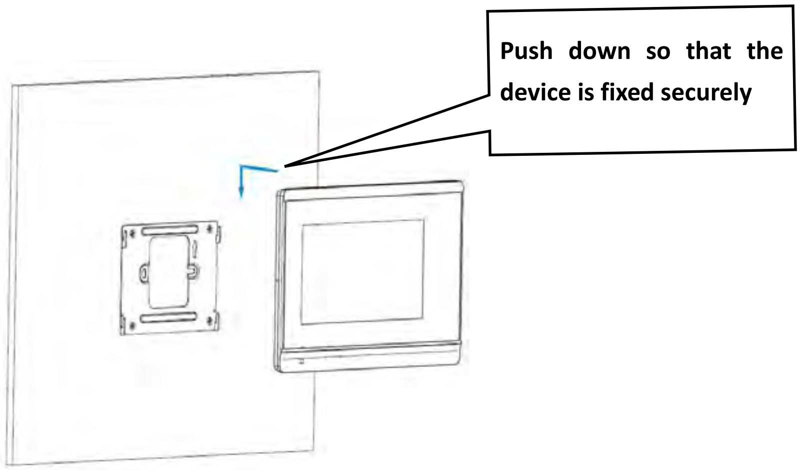


Use the pencil to mark 4 fixation holes through wall bracket on the wall

Use a 5mm hand drill to make the 4 fixation holes on the wall



Device Mounting



1.7 Installation Considerations

Here are some safety recommendations about the installation and the usage:

- Do not use this product near water, such as: bath, washbasin, kitchen sink and other damp places and so on.
- Place the device in a place away from heat.
- Place the device away from traffic areas to prevent collisions.
- Please use the equipment with the matching power adapter or POE.

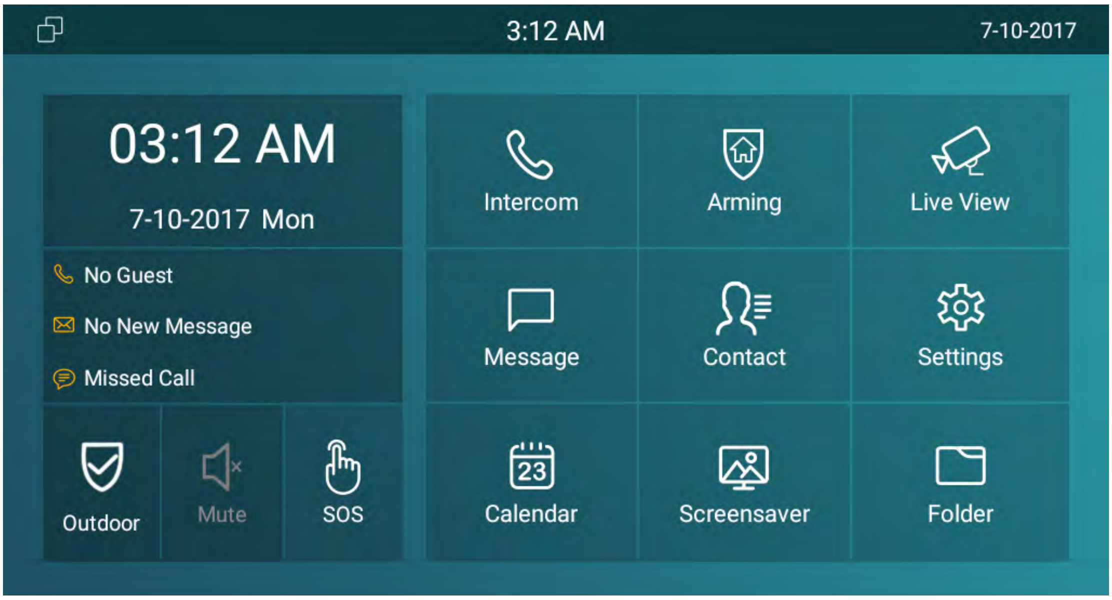
1.8 Equipment Packaging

Name	Quantity
IT82	1
Wall bracket	1
Quick Start Guide	1
10 Pin cable	1
8 Pin cable	1
Expansion anchor	4
ST4x20 screw	4

2 Setting

2.1 Main interface instruction

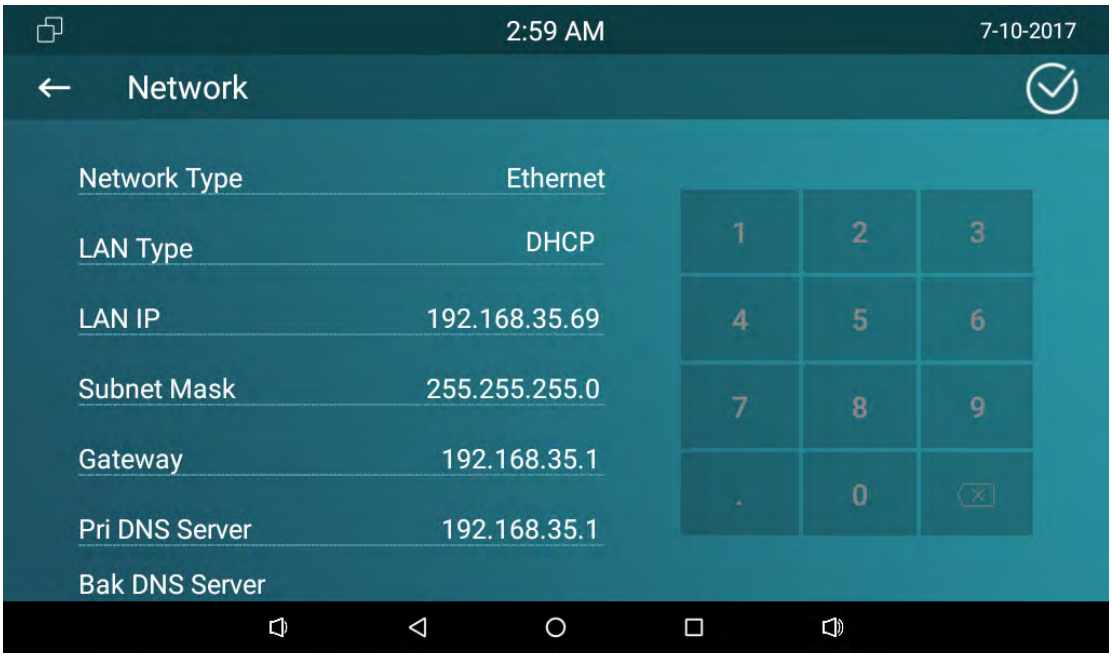
It82 supports two pages of main interface. Click the corresponding area to operate.



2.2 Network

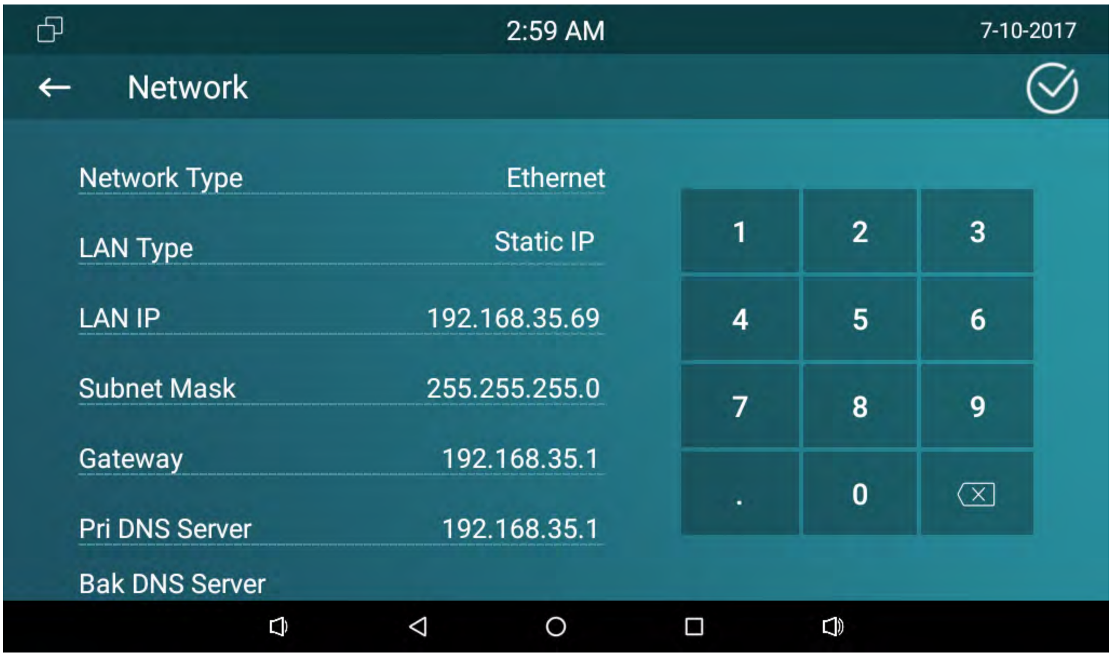
2.2.1 DHCP

IT82 series use HCP mode to get IP address by default. Please go to Setting-Advanced (password:123456)-Network
Choose DHCP,press CONFIRM, the phone will get IP address automatically.



2.2.2 Static IP

Select Static IP in LAN Type. Enter the IP address parameters in the corresponding area.

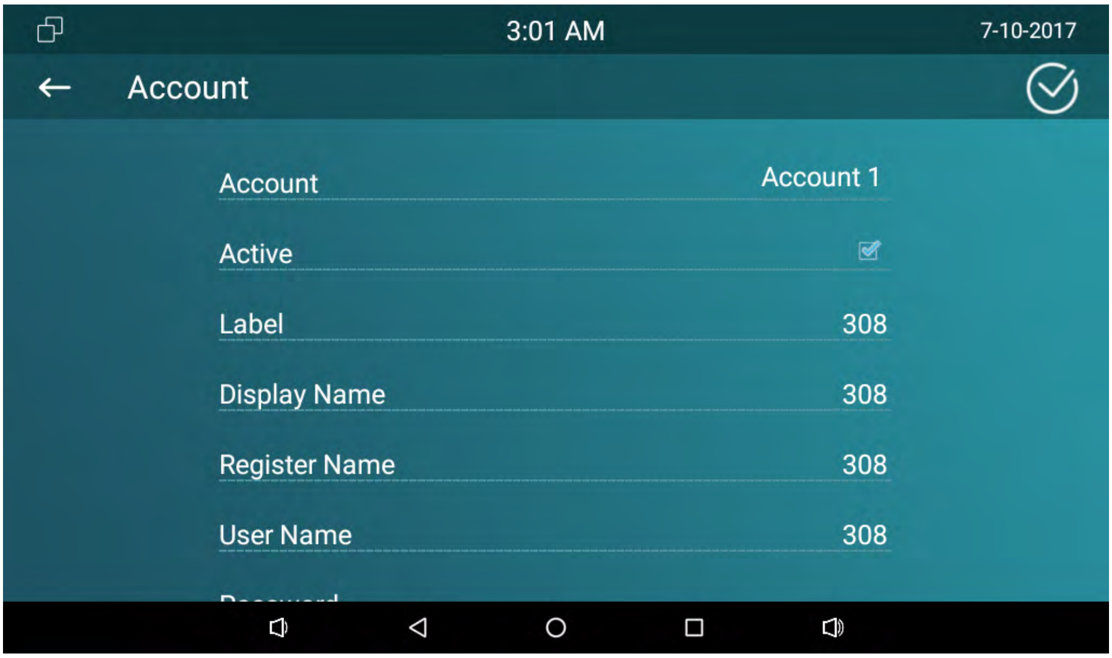


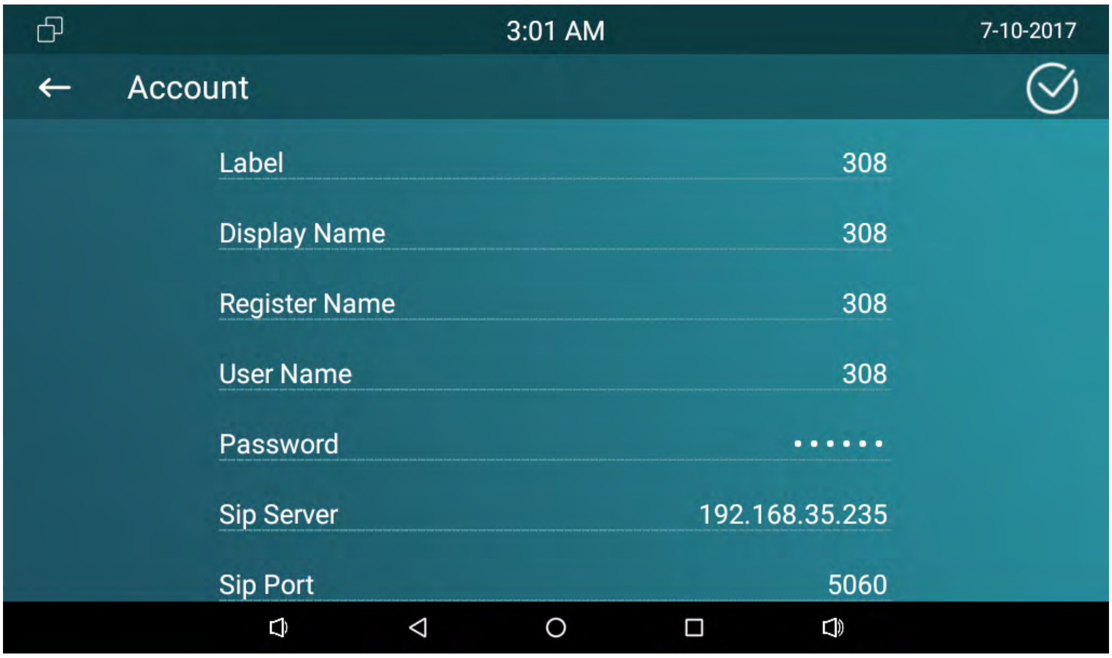
Parameter	Specification
IP address	set the IP address of the device

Subnet Mask	set the subnet mask of the device
Gateway	set the default gateway
DNS 1	set the DNS address
DNS 2	set the backup DNS address

2.3 Register Account

SIP account is provided by SIP server. Go to Setting-Advanced-Sip Account . Please consult administrator about sip server information.





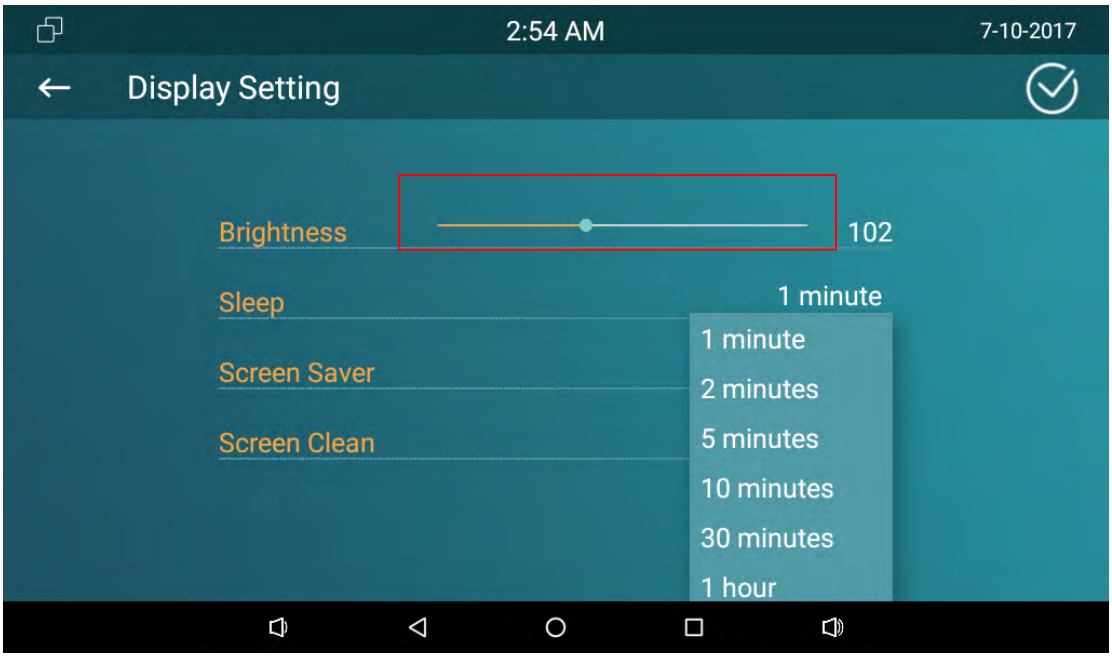
Fill the parameters in the corresponding area. Click CONFIRM to save.

Parameter	Specification
Register Name	SIP Account ID provided by ISP
User Name	SIP User Name provided by ISP
Password	SIP Password provided by ISP
Display Name	SIP Display name
Reg Server IP	SIP Register Server, format: domain/IP, for example: 194.168.1.2
Reg Server Port	The default port is 5060.

2.4 Display

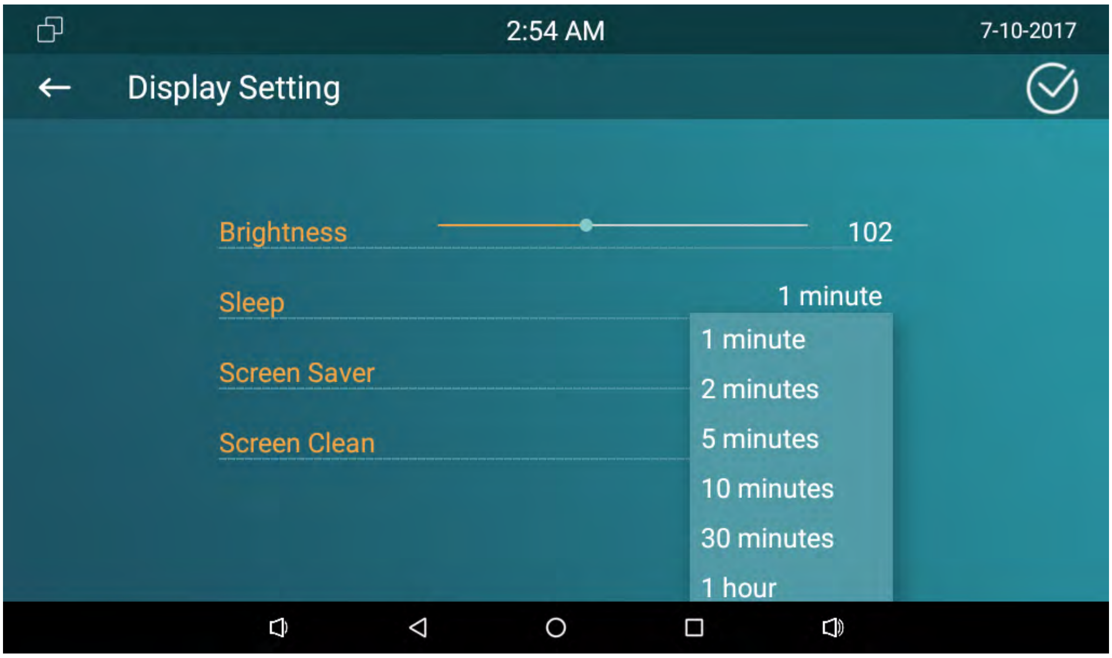
2.4.1 Brightness

Slide the point left or right to adjust the screen brightness, click CONFIRM to save.

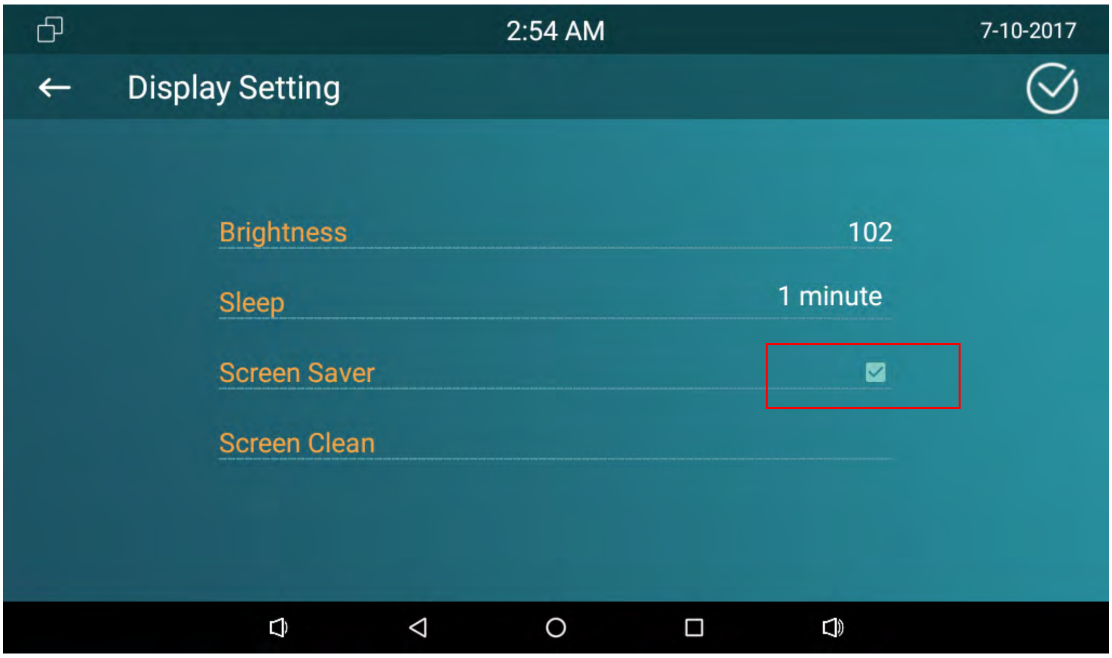


2.4.2 Sleep

Choose the sleep time from 15 seconds to Never. IT82 series setup 1minute by default. After the sleep time without any operation, the phone will black screen. Touch it to wake up.

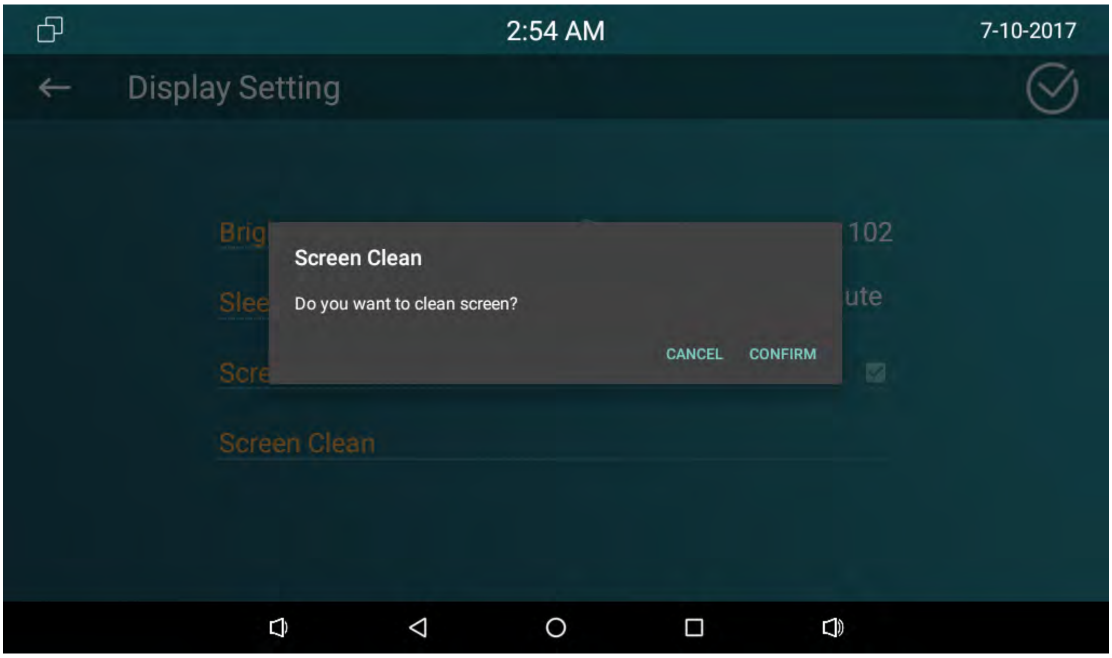


2.4.3 Screen Saver

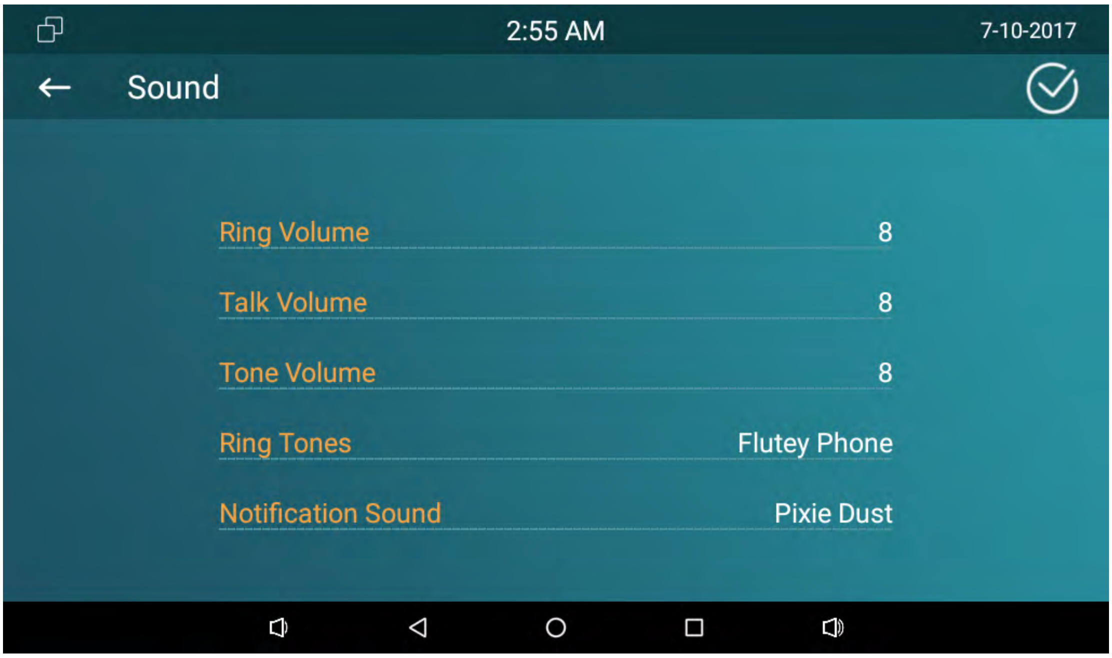


2.4.4 Screen Clean

This function is used to protect the device from being pressed any keys when users clean the screen.



2.5 Sound



2.5.1 Volume

Slide the point right or left to adjust the Ring / Talk/Tone Volume.

2.5.2 Ring tones

To setup the Phone Ringtone, click OK to confirm

2.5.3 Notification Sound

Select the suitable notification sound, click OK to confirm.

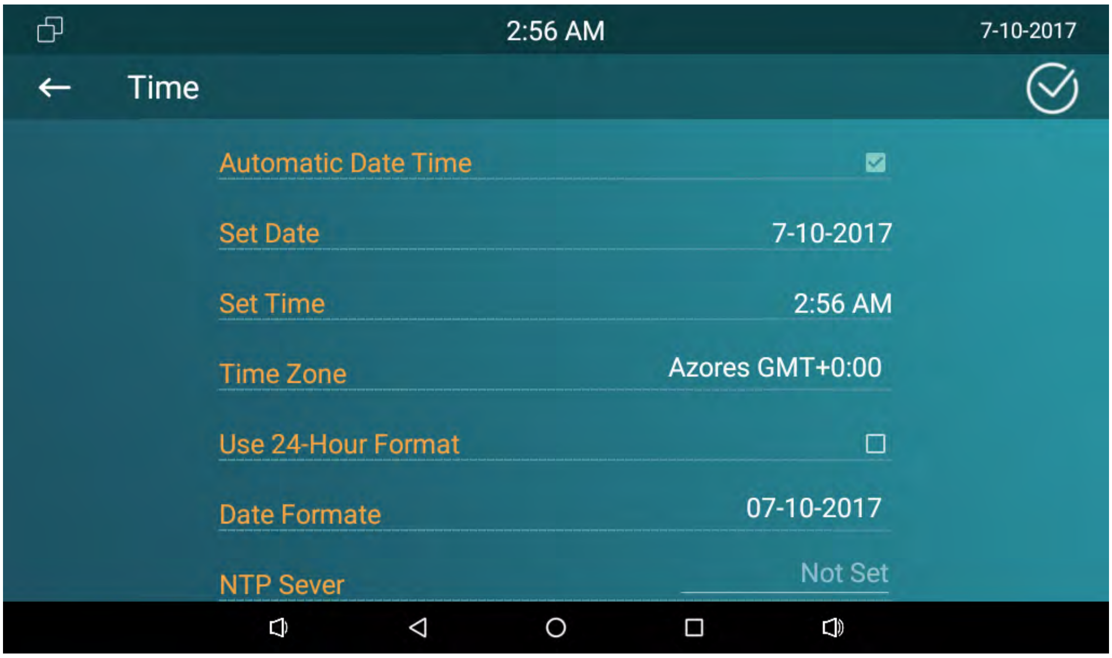
2.6 Time

Setup the date and time in the corresponding area.

Enable Automatic Date Time , the phone will get the Date and Time automatically.

If you uncheck Automatic Date Time, users need to setup the Date and Time manually.

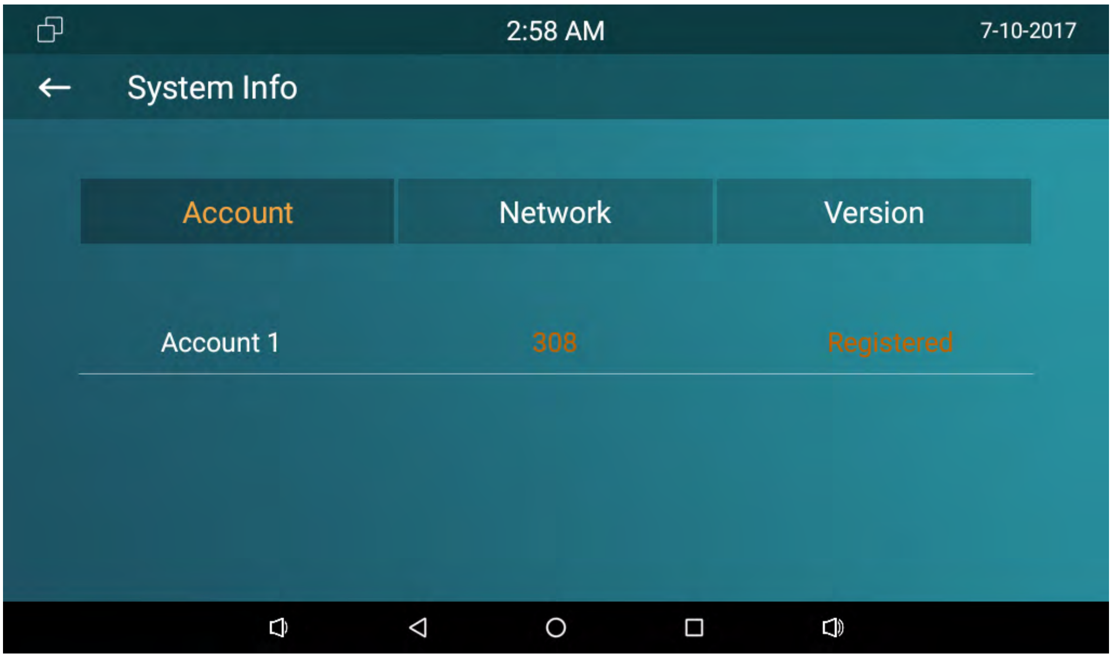
Users can also select the suitable Time Zone and use 24-Hour format.



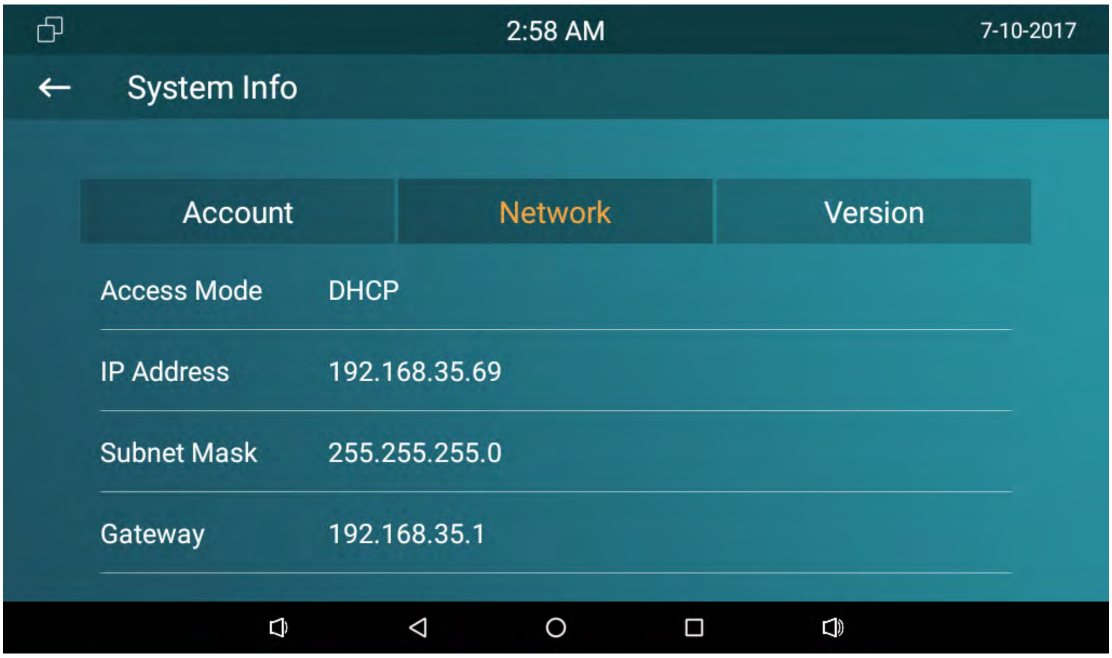
2.7 System Info

Enter System Info interface to check Account, Network and Version information.

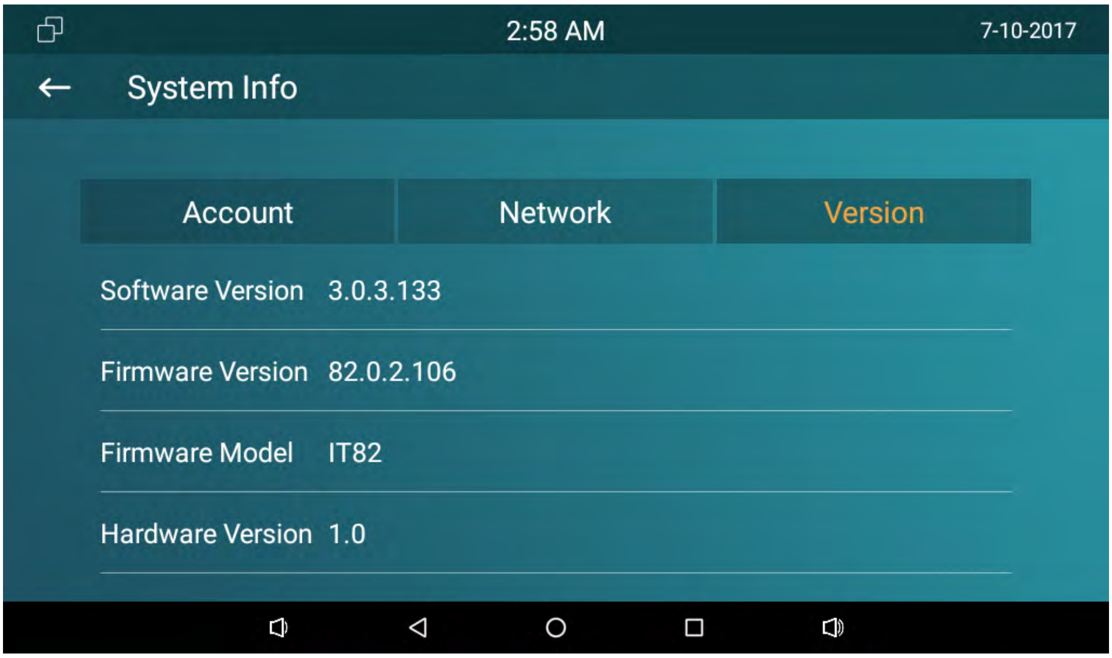
To check the SIP account status



To check the Network Access Mode And parameters.

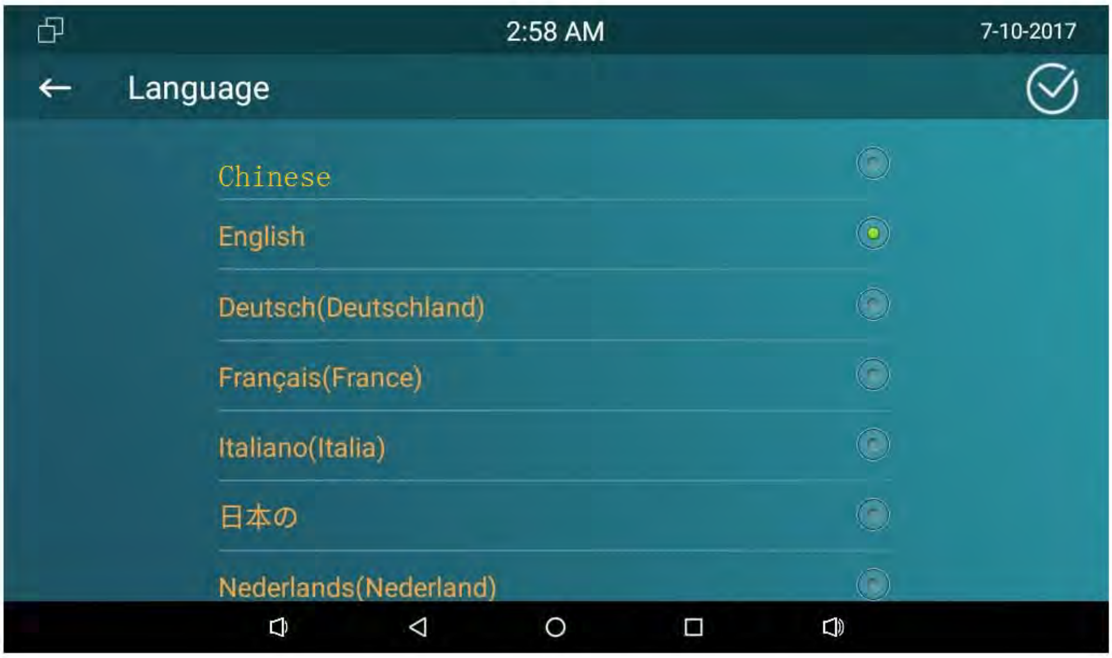


To check the Software version, Firmware version, Firmware Model and Hardware version.



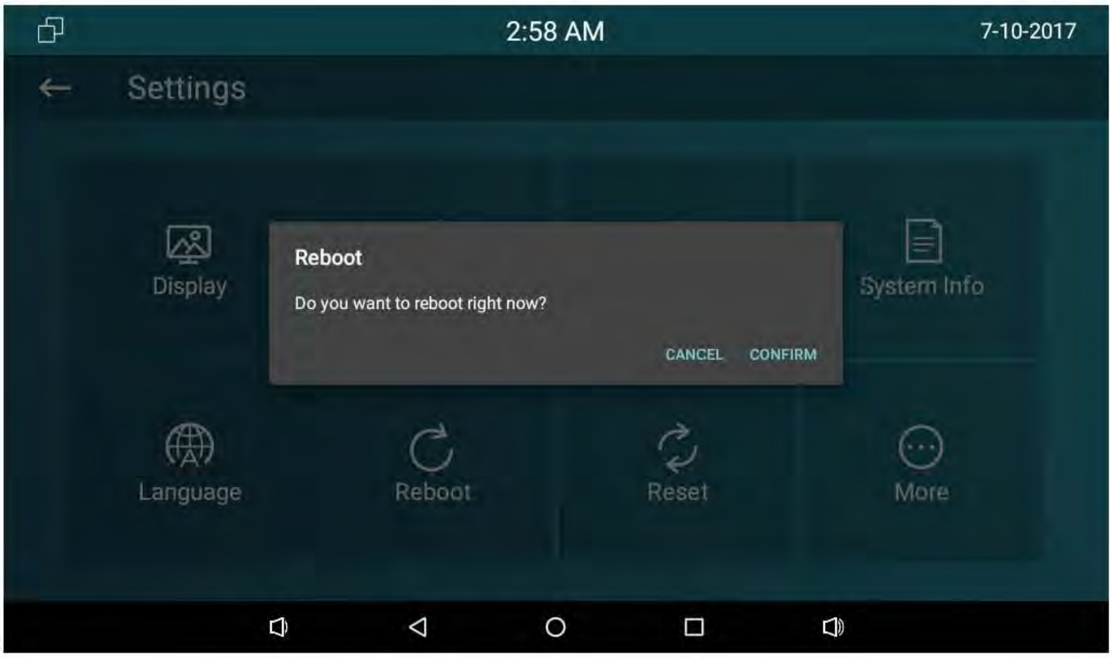
2.8 Language

Choose the suitable phone language. IT82 series use English by default.



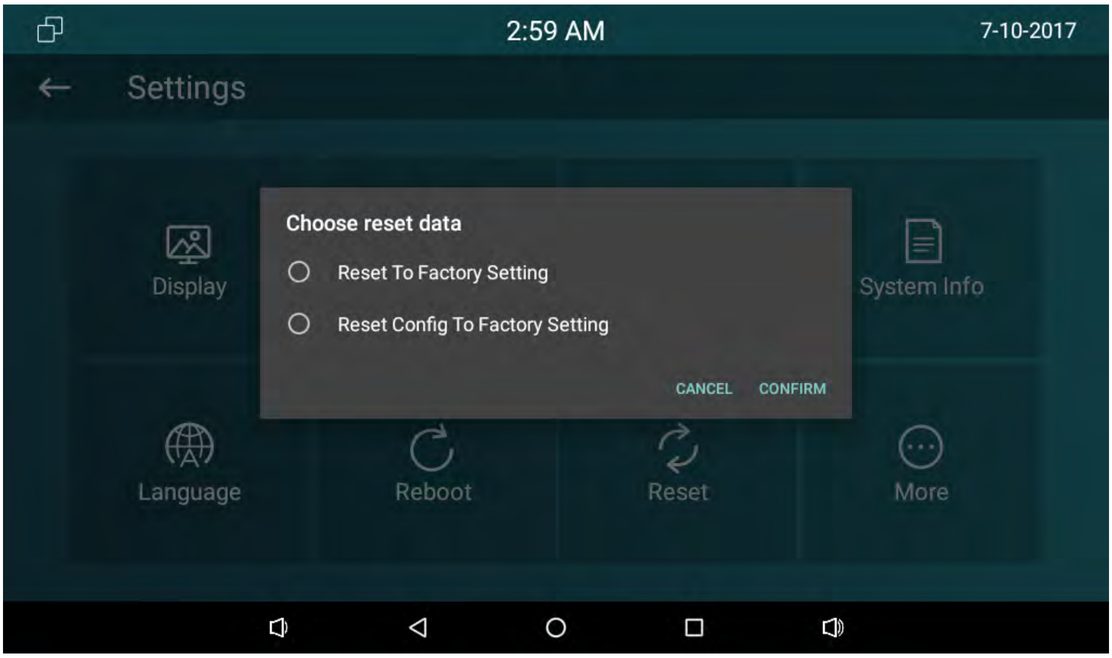
2.9 Reboot

To reboot the device, click CONFIRM when you see the prompt. The phone will reboot.



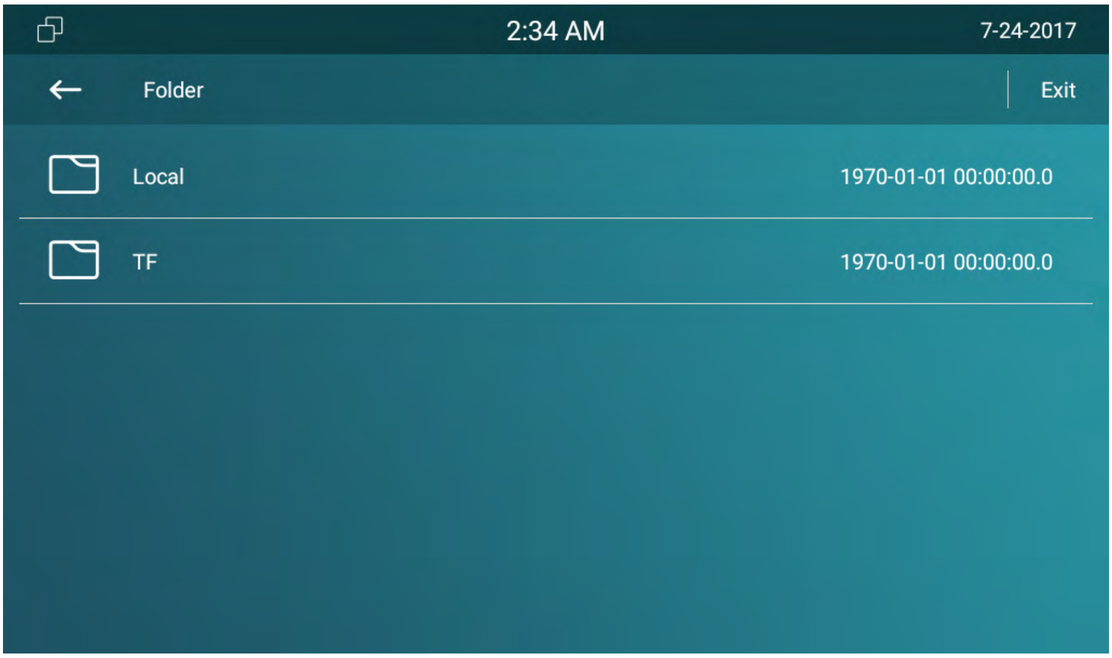
2.10 Reset

Go to Advance-Reset. Choose the Reset mode,click CONFIRM when you see the prompt. The phone will reset to factory setting automatically.



2.11 Folder

Click Folder to check the local and SD card files. IT82 series supports external SD card.


2:34 AM
7-24-2017
Folder
Exit
Local
1970-01-01 00:00:00.0
TF
1970-01-01 00:00:00.0

3 Function

3.1 Make a call

User can directly dial from the soft keypad, select from the contact list or from call log to call out the number.

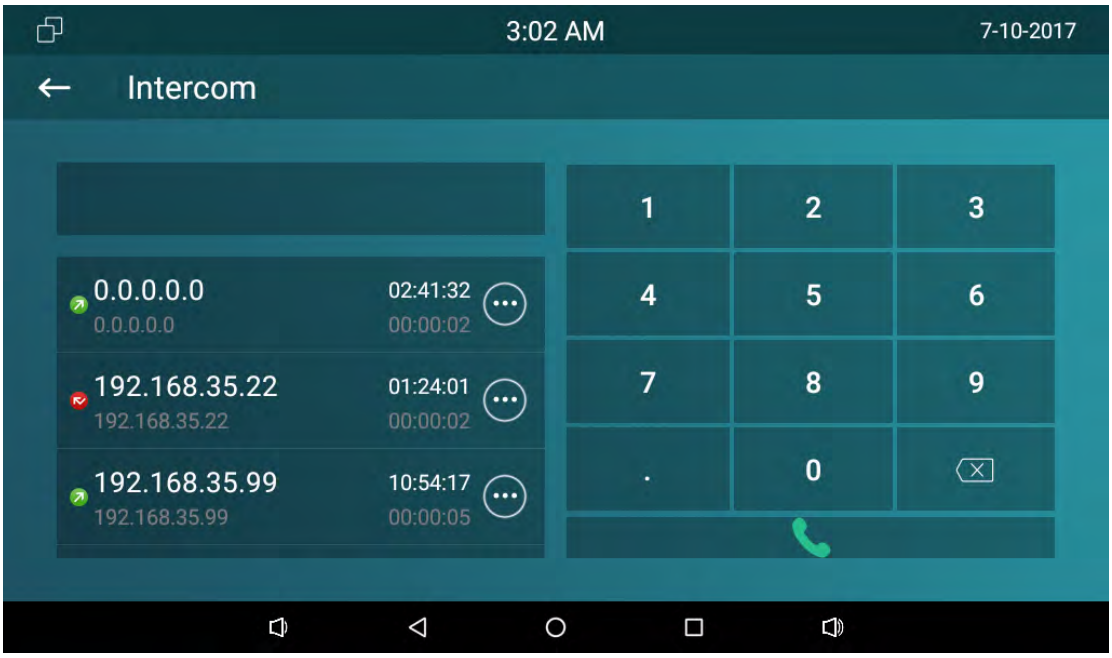
User can choose one of the following steps to enter the dialing interface.

Mode 1: Click Call icon  on the main screen.

Mode 2: Click Intercom button .

3.1.1 Call SIP

If you want to call sip number, you can select Call SIP label. Click the sip number to make a call. The more information about sip account please consult administrator.



3.1.2 Call resident

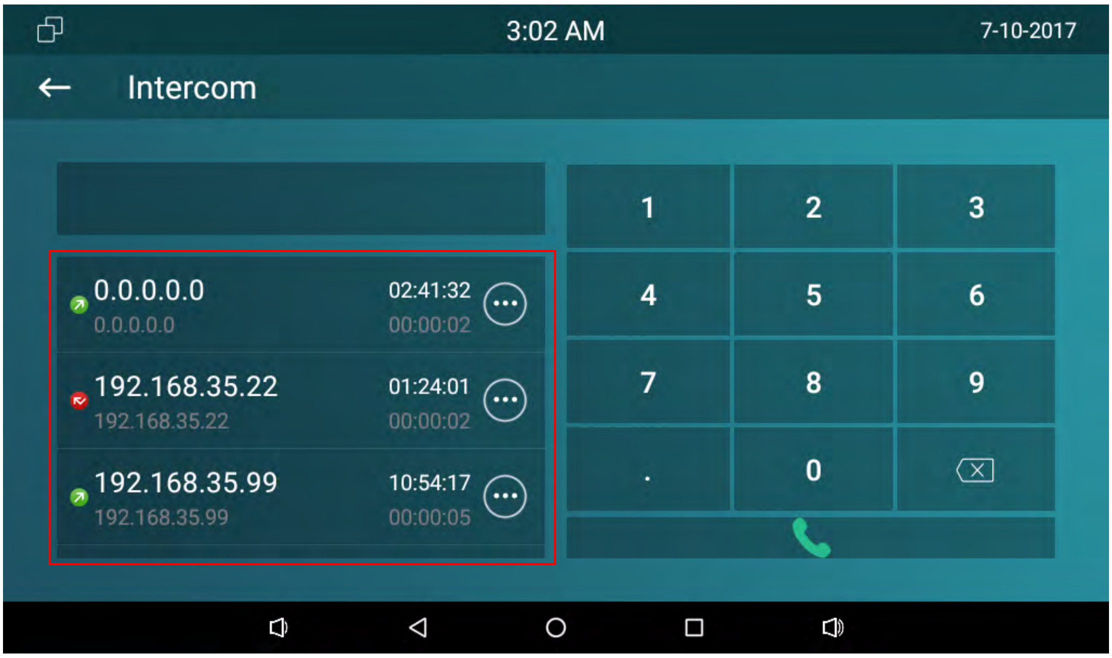
If you want to call room number, you can select Call Resident label. For example, if you want to call community 1, Building 1, Unit 1, Floor 1, Room1, you can input digits “001001010101”.

3.1.3 Call from Call Log

User can enter the call log page to make a call.

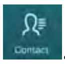
1. In the dialing interface->click Intercom icon 
2. Click Intercom button .

Directly slide up and down to choose the number from the call history. Click Dail key  to call out.



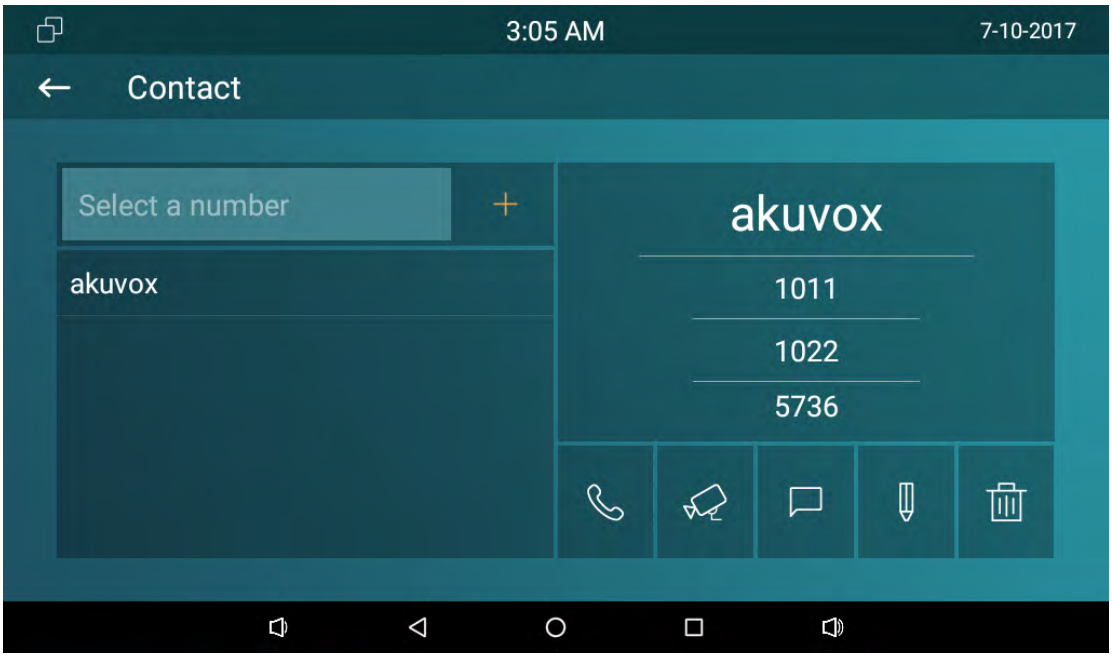
3.1.4 Call from Contacts

User can enter phonebook interface to make a call.

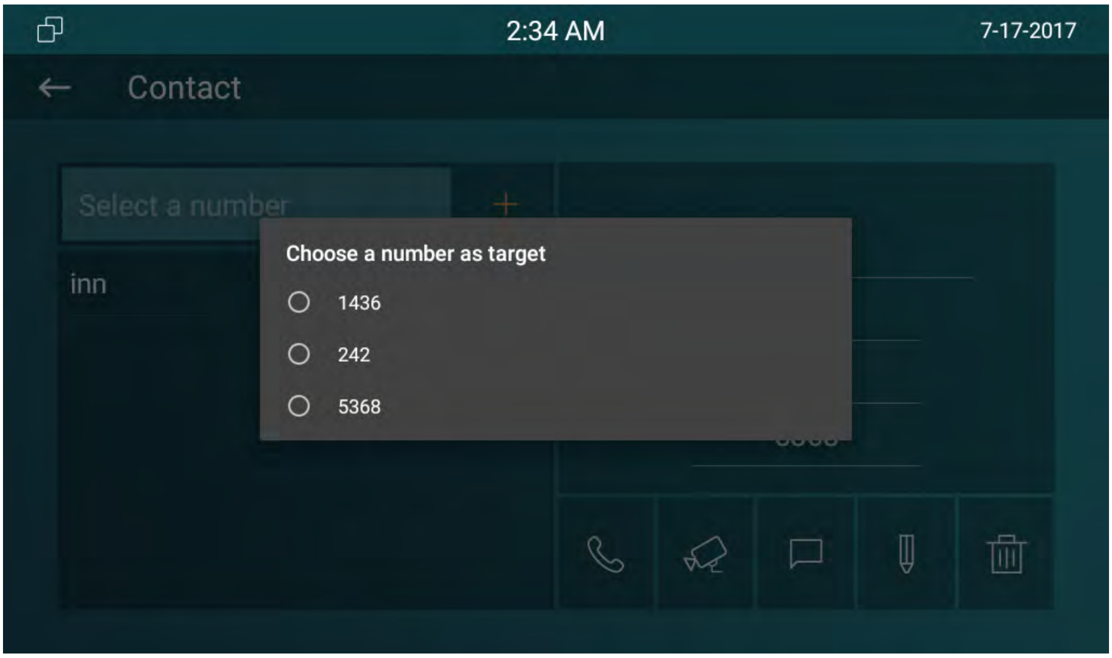
Directly click Contact icon .

If you want to make an audio call, click Audio Call icon . If you want to make a video call, click Video Call .

Contact is shown as below：



If the contact has multiple numbers, after clicking the calling mode, user need to choose the number you want to call.



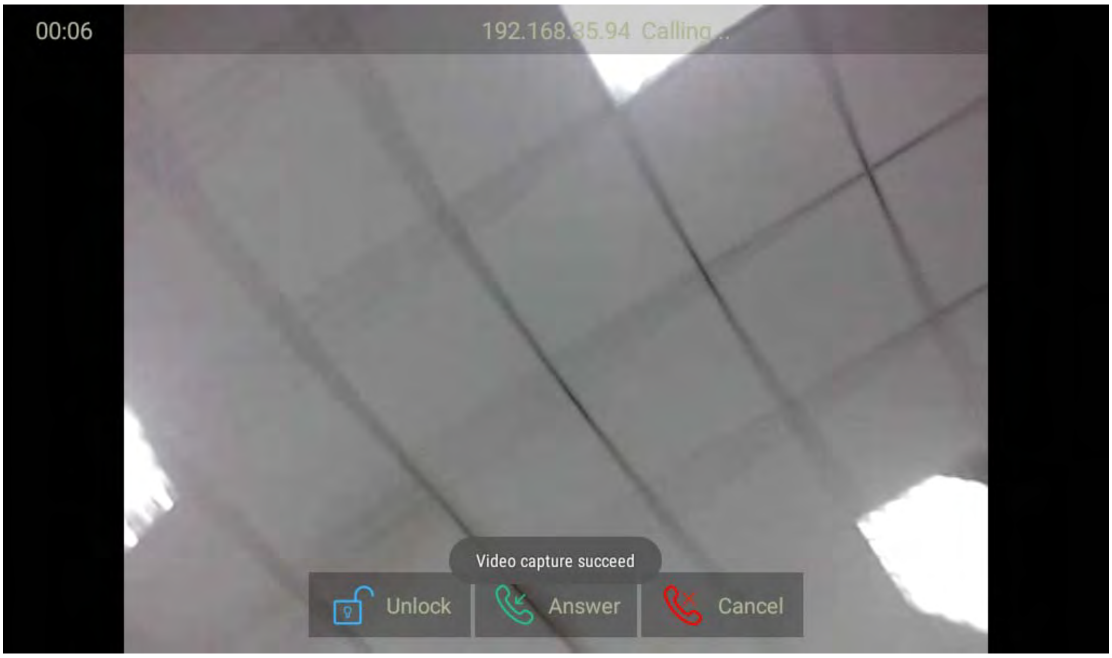
3.1.5 Call Center

If you want to call management center (SDMC system), you can press to call out. This function only can be used when IT82 series has registered from SDMC.

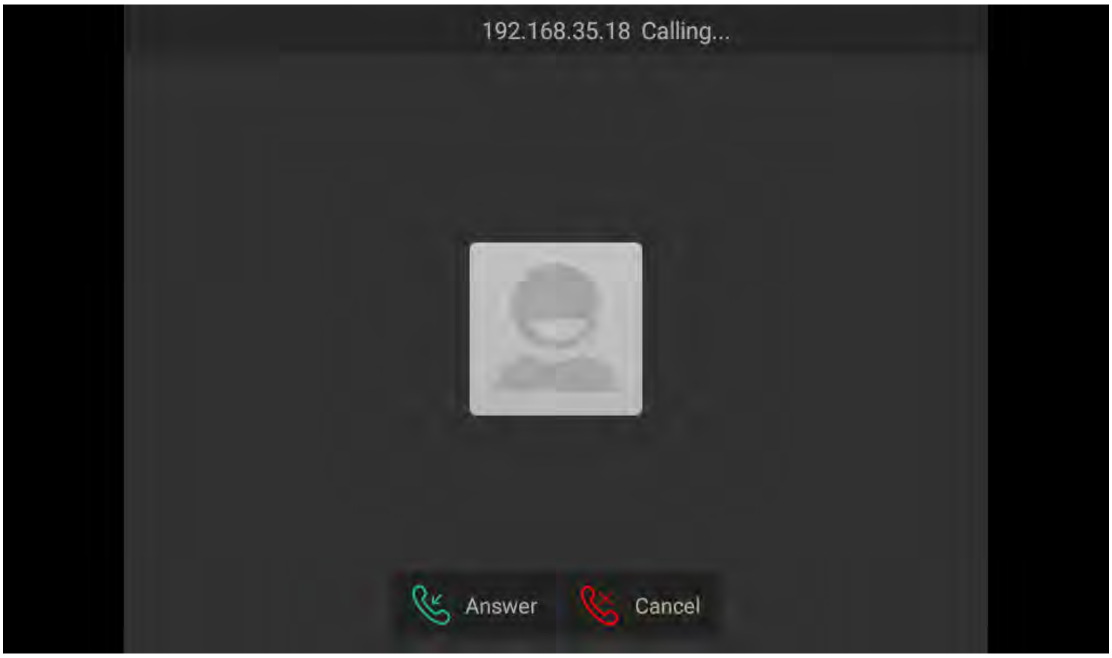
3.2 Receive a call

Incoming calls include audio and video calls. If the caller has been stored in the contacts, it will show the contact name, otherwise the caller number will be displayed. When IT82 series received the video call, it can automatically get a screenshot and save it in the Album.

Video call :



Audio Call :

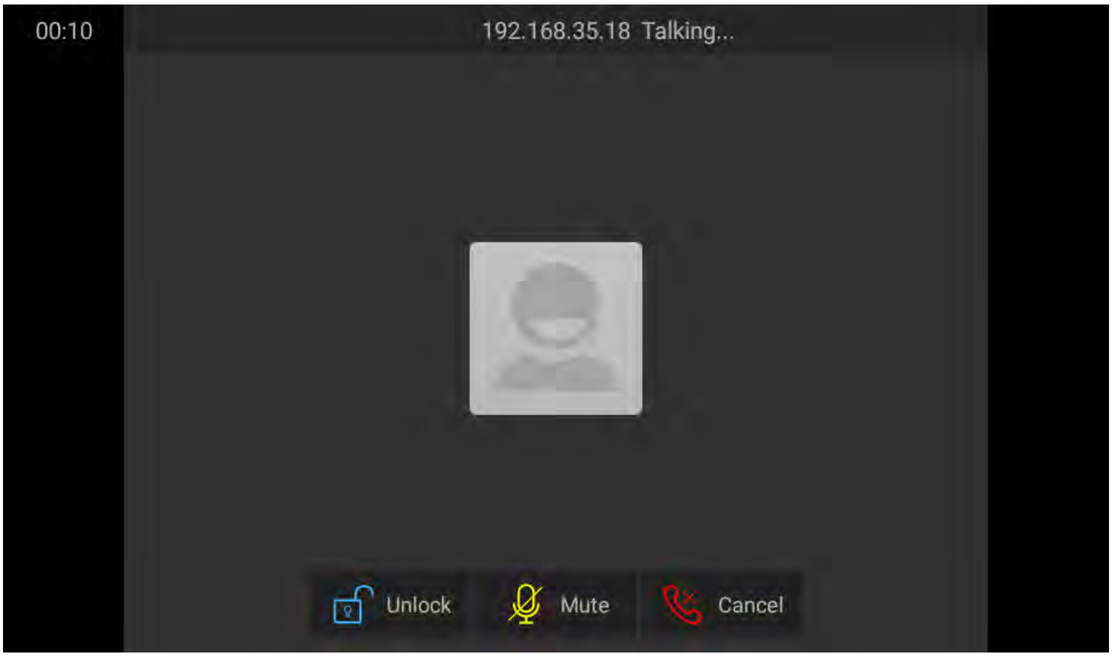


3.3 Answer a call

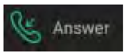
1) Answer a audio

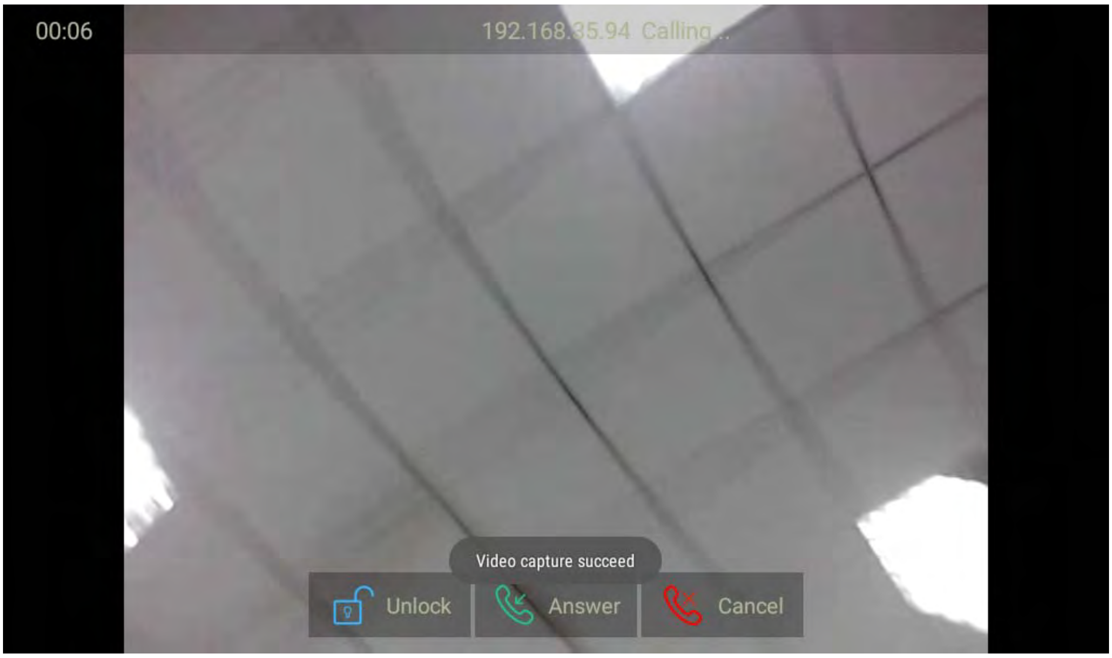
Click Answer Key 

 to answer the call, As shown below：



2) Answer a video call

When receiving an video call, the phone will automatically open the video preview so that resident can view the visitors identity. Press Answer key  to answer the call. As shown below：



3.4 Reject a call

If user want to reject a call, click Cancel label  in incoming interface.

3.5 Contacts

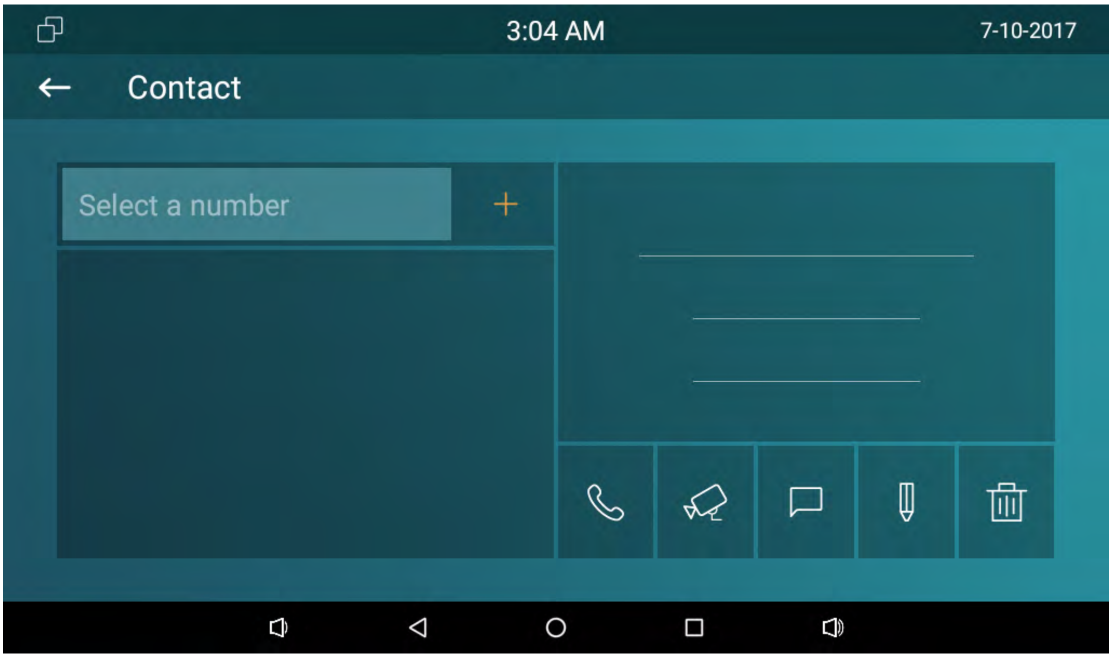
3.5.1 New contacts

1.Click Add button.

2.Enter the name.

3.Select the number type, and enter the parameters in the corresponding area.

4.Click CONFIRM to save.

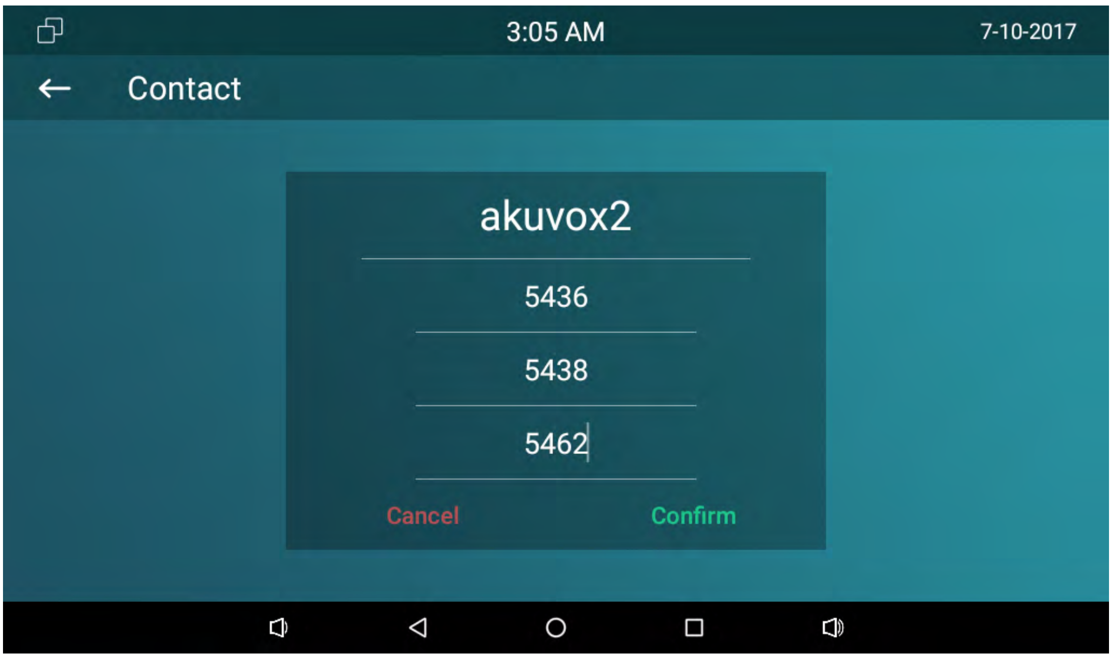


3.5.2 Modify the contact

1. Choose the existed contact, click to edit.

2. Then modify the contact you need.

3. Click CONFIRM to save.

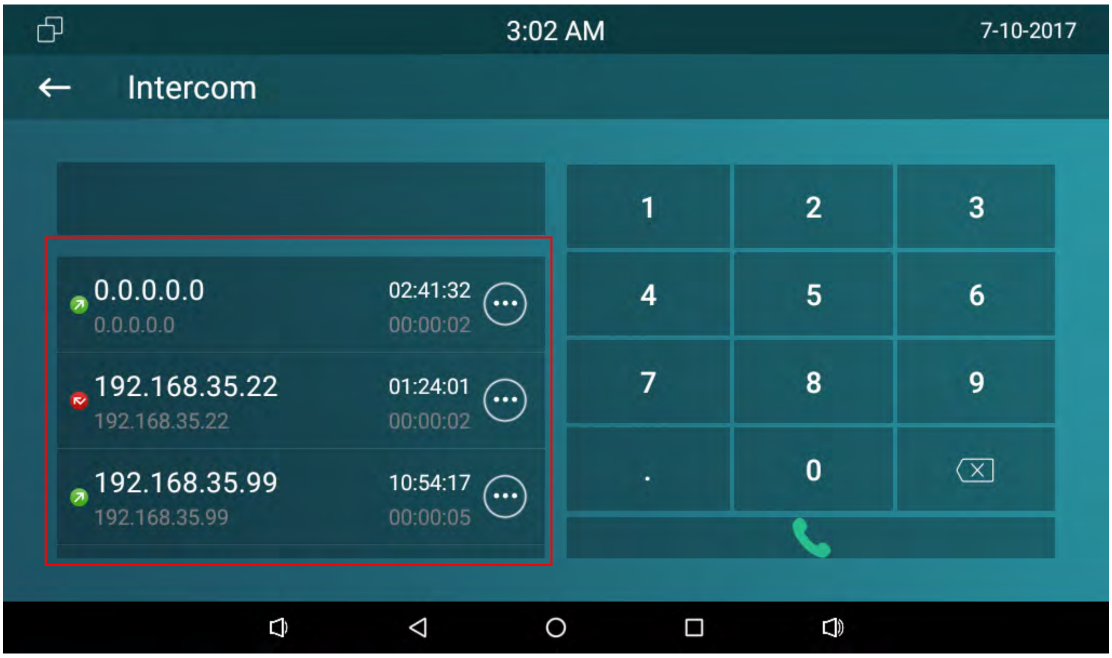


3.5.3 Delete the contact

Choose the the existed contact you need to delete. Click  to delete.

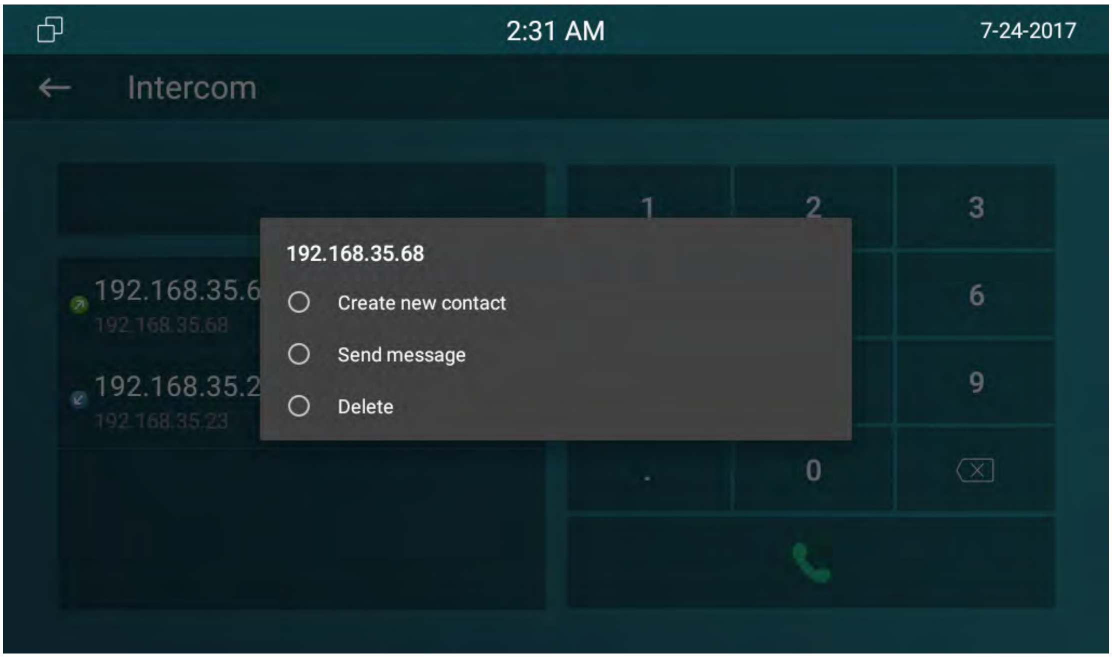
3.6 Call log

Click Intercom, slide up and down to check the all call log in the marked box as shown below.



3.6.1 Modify the call log

Choose one call log, click  to modify the log. Users can create a new contact or send the message to this call log .



3.6.2 Delete the log

Click  in the corresponding log, choose Delete to remove the call log number.

3.7 Capture

IT82 supports capture the visitor photo when incoming preview. IT82 series can auto capture if ringing for 3 seconds without answer.

3.8 Message

Users can receive and write the message.

3.9 Live View

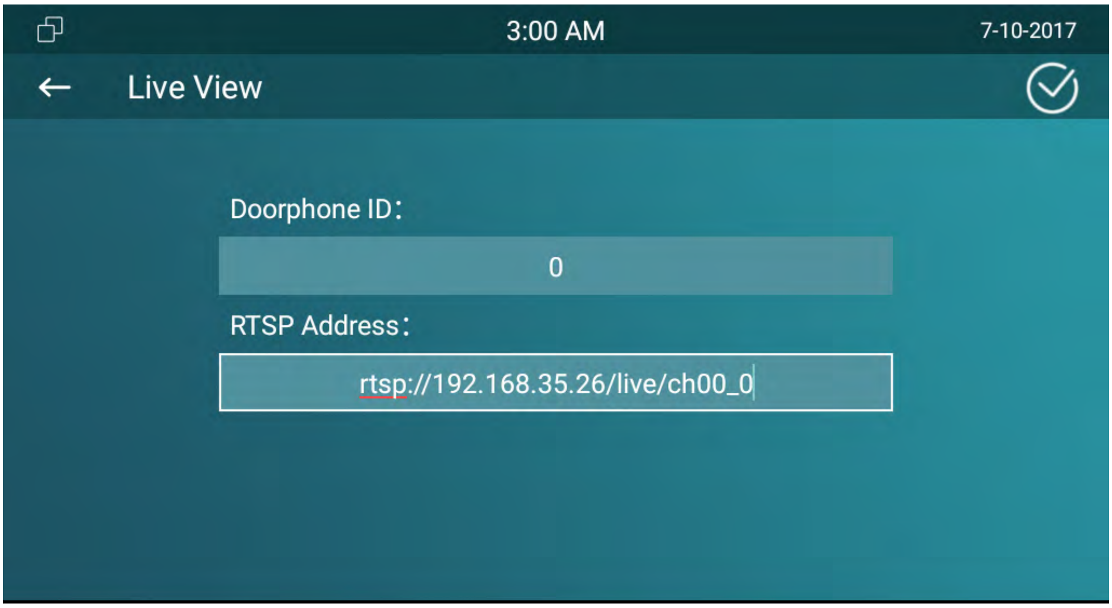
Live view is used to monitor the door phone via using RTSP.

1.Path: Settings->More->Live View

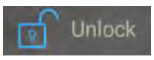
2.Enter the door phone RTSP address.(please make sure the address format is right)

3.Click CONRIM to save.

Then users can check the live video from the door phone any time.



3.10 Unlock

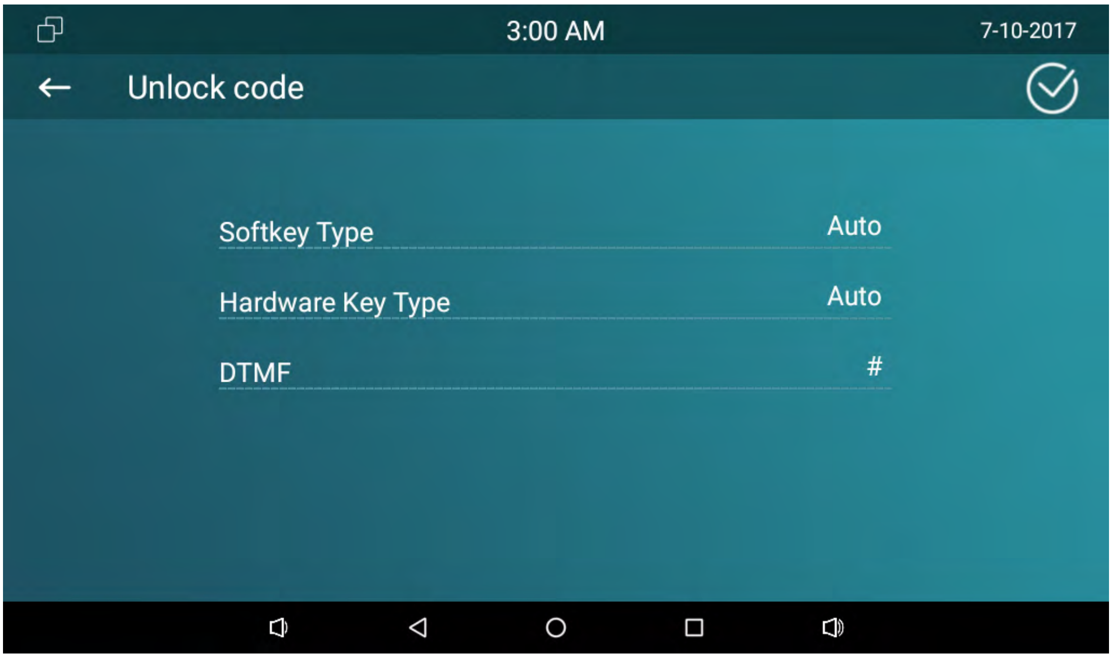
IT82 series supports remote unlock the door phone via using DTMF code. During the call, users can press Open Lobby label 

or Unlock touch key to open the door during the call.

Setup Unlock

IT82 support unlock local door lock or remote unlock. And there are two unlock keys-softkey during the talking, physical unlock key. If you want to unlock the local door lock, choose the type as Relay. Otherwise ,setup the type as DTMF, then choose the DTMF code.

Go to the path: Setting-> More(123456)-> Unlock Code



Note: IT82 series default DTMF code is #. Users need to predefine the same DTMF code of door phone and IT82. Configuration DTMF code, please consult your administrator.

3.11 SOS

SOS key is used to call out the emergency number in case of emergency. IT82 series will call out for three predefined numbers in a loop, each number will be called for 60s (by default).

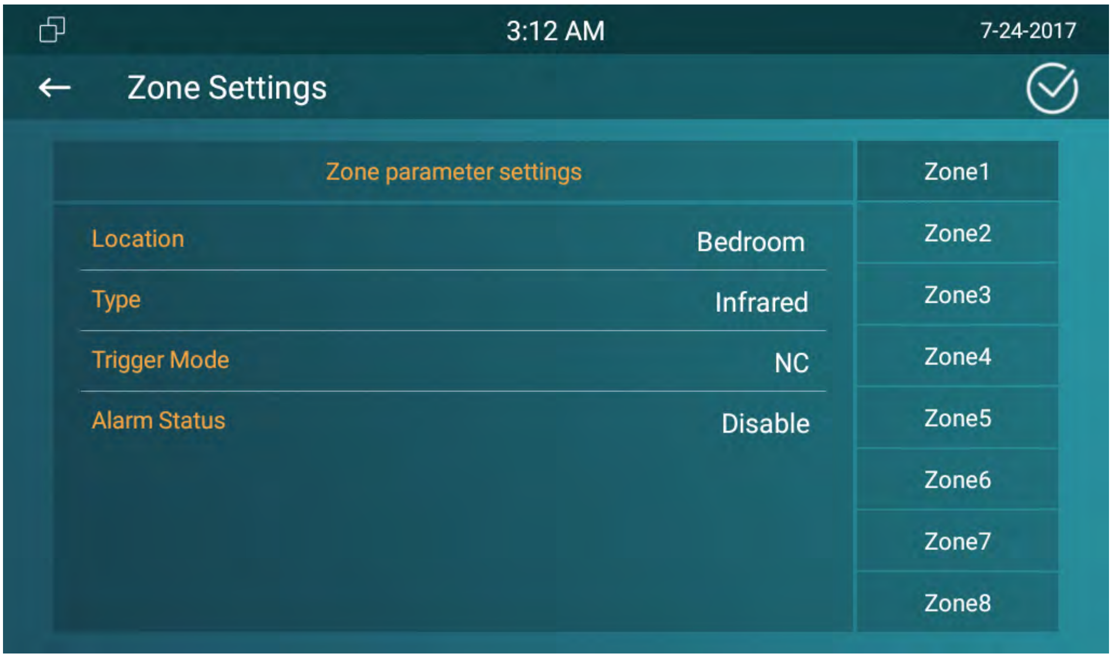
1. Go to Setting-> More-(password)>SOS
2. Set up 3 emergency number
3. Setup the call Timeout
4. Set up the loop time
5. Click CONFIRM to save

3.12 Security

IT82 series support connect 8 alarm zones via IO1-8 interface. Up to 5 Alarm Type - Infrared, Drmagnet, Smoke,Gas and Urgency and two trigger mode - NC(normal close) and NO(normal open) .

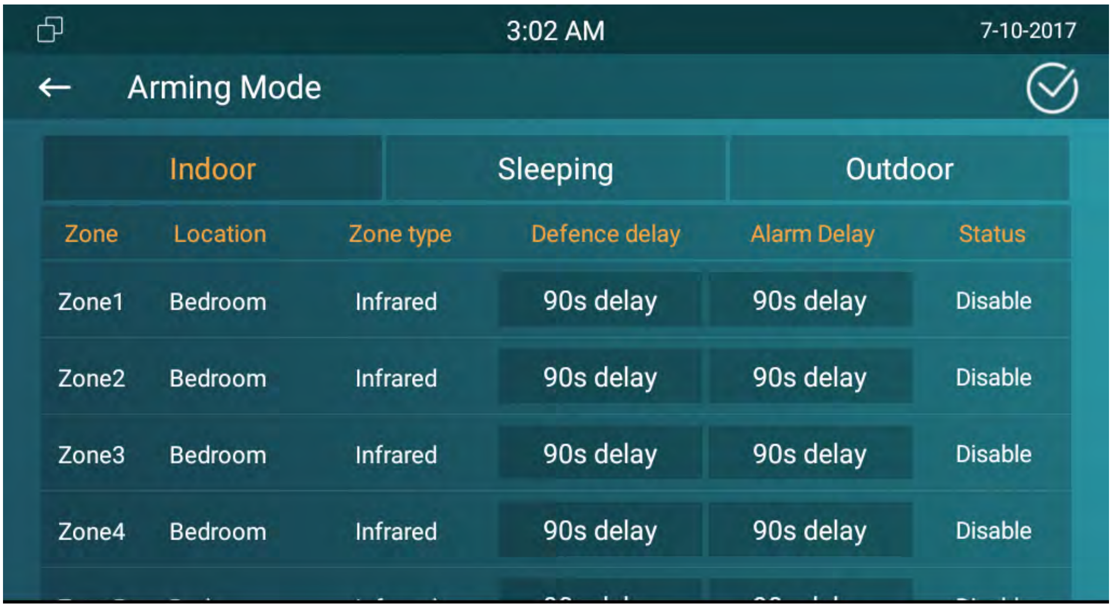
Go to the path: Settings-> More(Password:123456)-> Arming

- Different alarm sensor is suitable for different type. For example , if you use the smoke detector in your bedroom(zoon1) , connect the sensor in IO1 and GND contact in IT82, then you can setup the type as Smoke.
- Trigger Mode depends on connecting mode .
- Alarm Status includes 3 types: Disable, Enable,24H. Disable: alarm function is invalid. Enable: Alarm function is valid after you choose the Arming mode(indoor, sleep, outdoor). 24H: Alarm function is working whether you choose the mode or not.

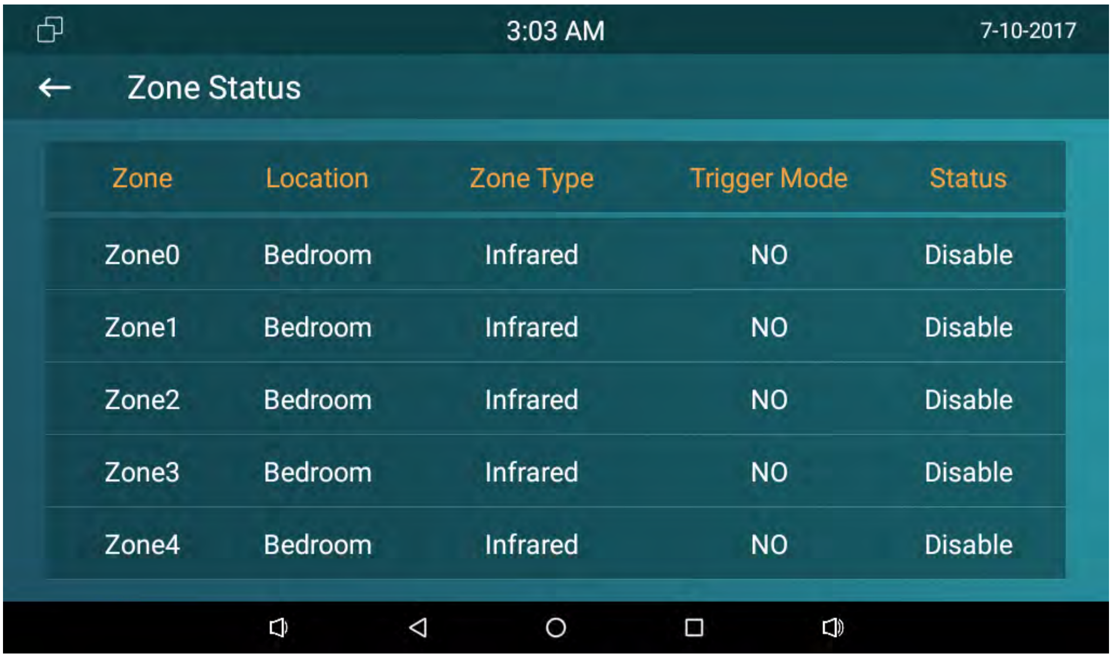


Return to the main interface, click Arming.

Setup alarm delay, the alarm will ringing after the alarm has been triggered. Open or close Alarm Status to enable or disable this function.



Enter Zone Status to check 8 alarm zones working mode.



When the alarm is triggered, click Cancel and input “0000” to disable it.

4 Website

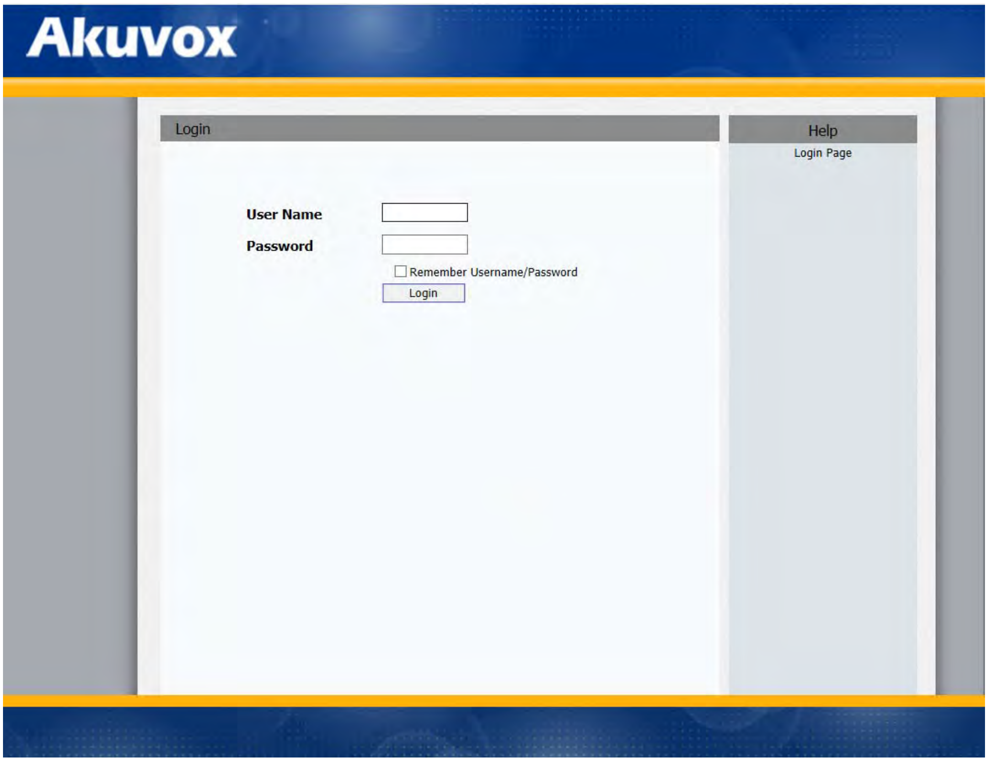
4.1Web login

The Akuvox IT82 series ses DHCP IP address by default, go to the path: Settings-System Info-Network to check the IP address.

Open a Web Browser, enter the corresponding IP address. Then, type the default user name and password to log in. The default User Name and Password are as below:

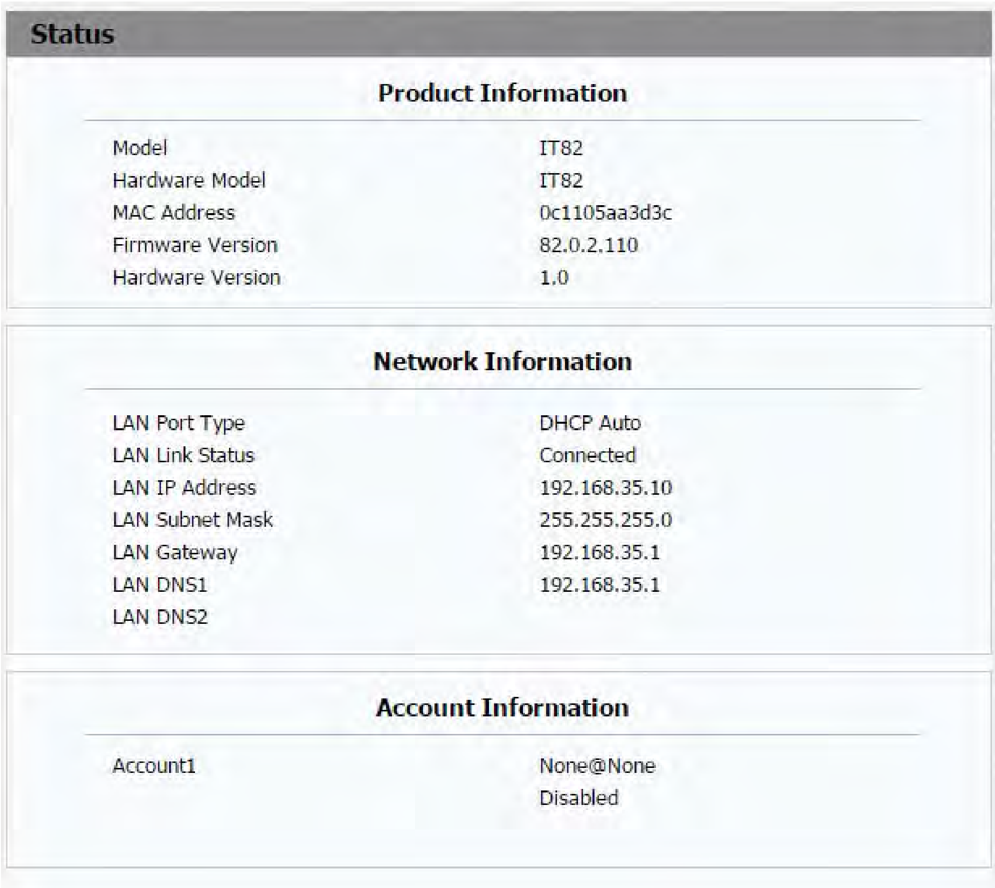
User name: admin

Password: admin



4.2 Status

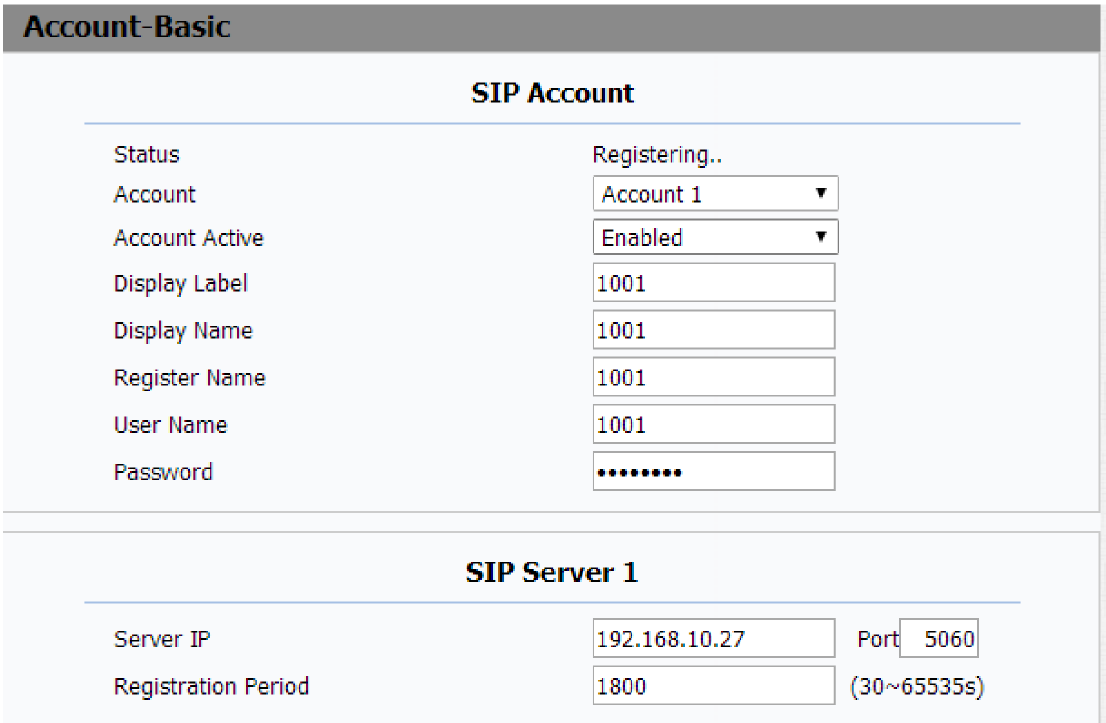
4.2.1Basic



Sections	Description
Product Information	To display the device's information such as Model name, MAC address (IP device's physical address), Firmware version and Hardware firmware.
Network Information	To display the device's Networking status(LAN Port),such as Port Type(which could be DHCP/Static/PPPoE), Link Status, IP Address, Subnet Mask, Gateway, Primary DNS server, Secondary DNS server, Primary NTP server and Secondary NTP server(NTP server is used to synchronize time from INTERNET automatically).
Account Information	To display device's Account information and Registration status (account username, registered server's address, Register result).

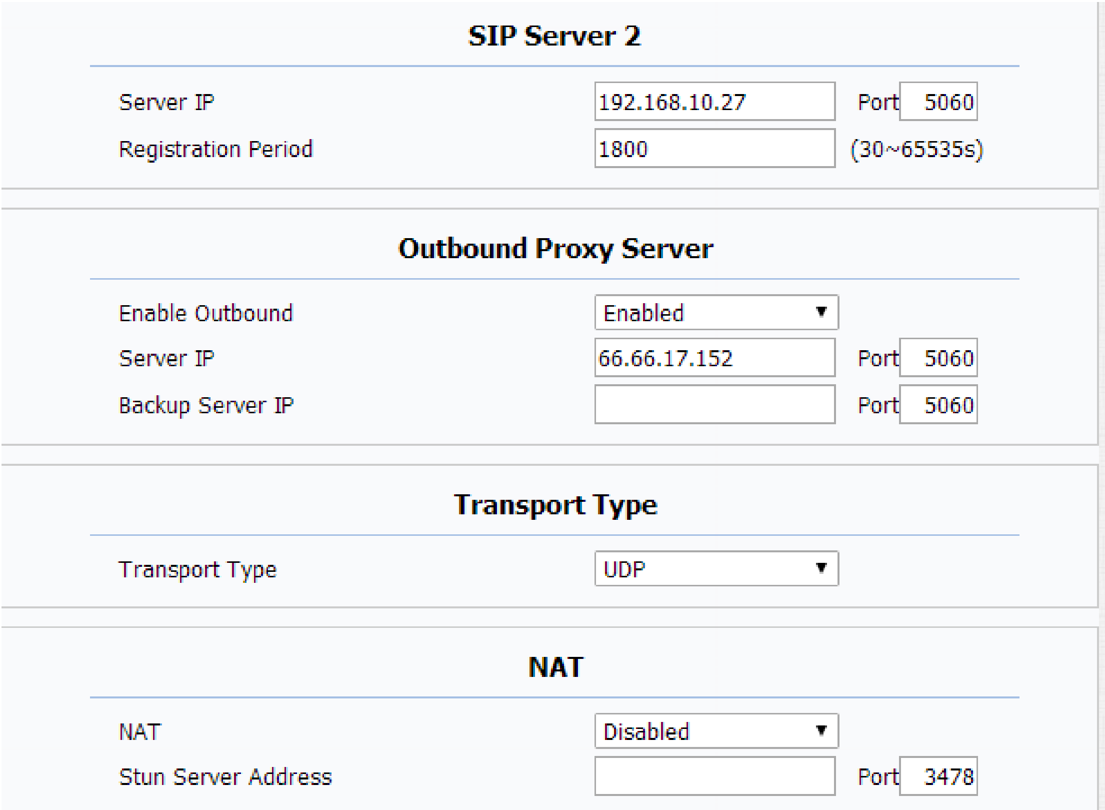
4.3 Account

4.3.1 Basic



Sections	Description
SIP Account	To display and configure the specific Account settings. ● Status: To display register result. ● Display Label: Which is displayed on the phone's LCD screen. ● Display Name: Which is sent to the other call party for displaying. ● Register Name: Allocated by SIP server provider, used for authentication. ● User Name: Allocated by your SIP server provide, used for authentication. ● Password: Used for authorization.
SIP Server 1	To display and configure Primary SIP server settings. ● Server IP: SIP server address, it could be an URL or IP address. ● Registration Period: The registration will expire after Registration period, the IP phone will re-register

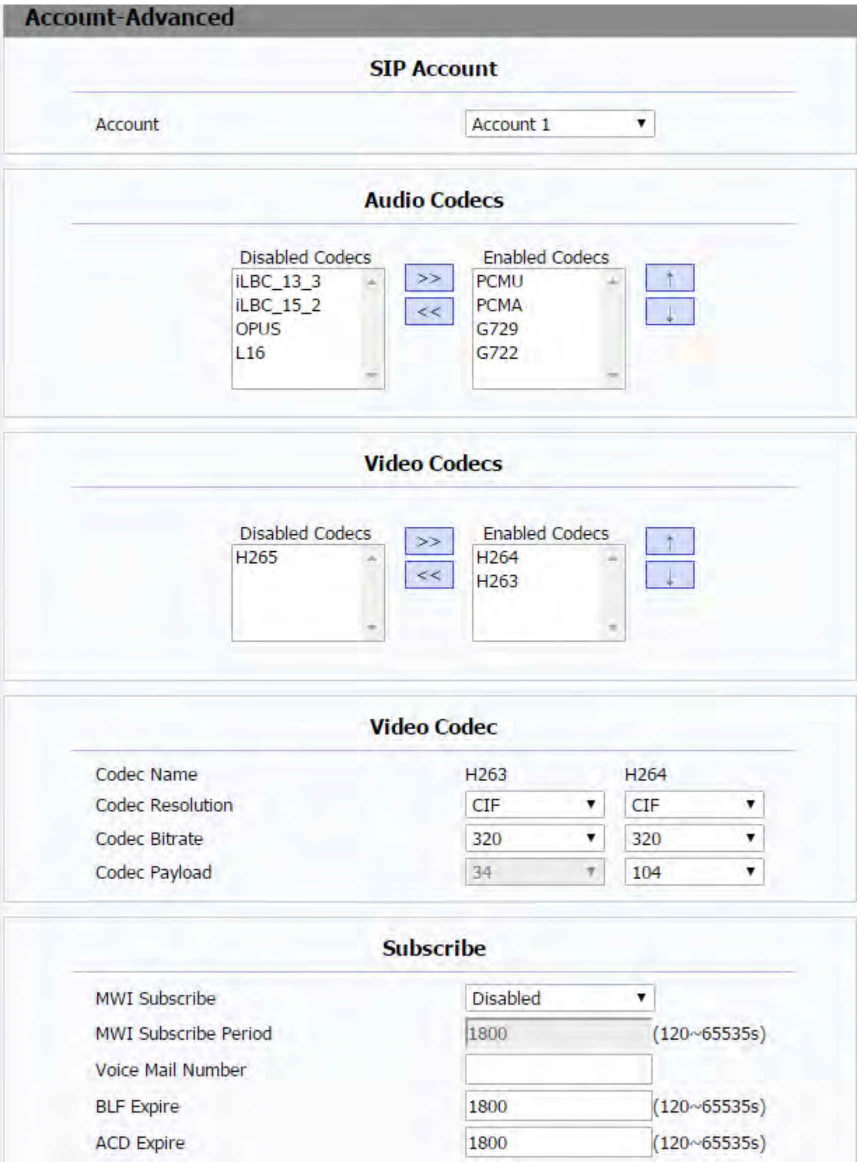
	automatically within registration period.



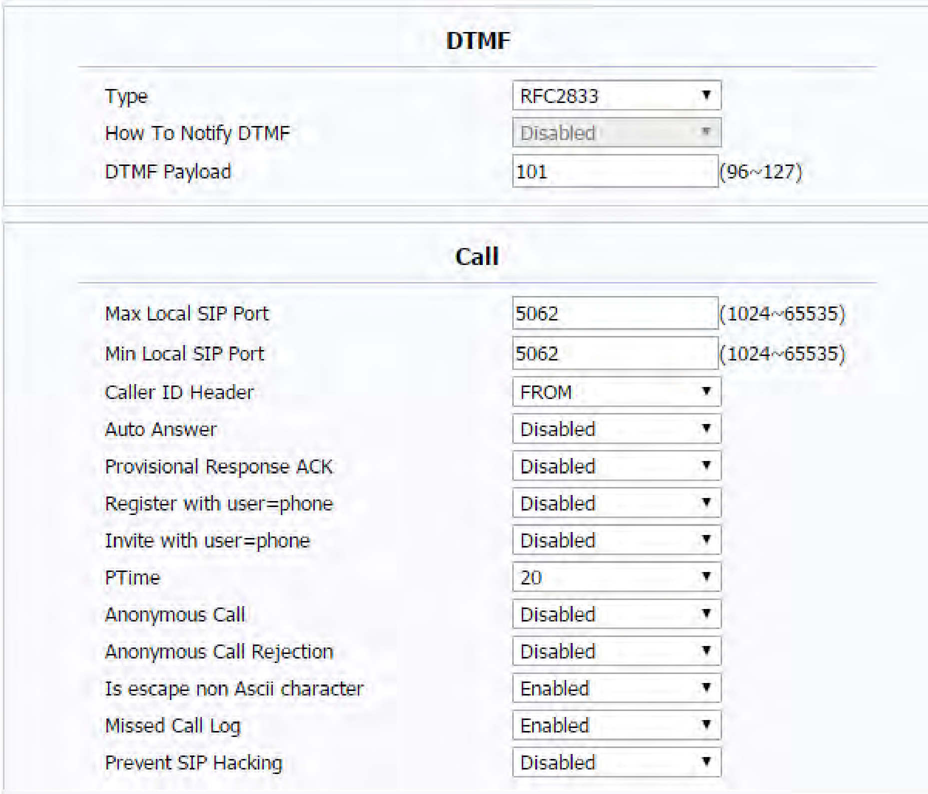
Sections	Description
SIP Server 2	To display and configure Secondary SIP server settings. This is for redundancy, if registering to Primary SIP server fails,the IP phone will go to Secondary SIP server for registering. **Note**: Secondary SIP server is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.
Outbound Proxy Server	To display and configure Outbound Proxy server settings. An outbound proxy server is used to receive all initiating request messages and route them to the designated SIP server. **Note**: If configured, all SIP request messages from the IP phone will be sent to the outbound proxy server forcefully.
Transport Type	To display and configure Transport type for SIP message ● UDP: UDP is an unreliable but very efficient transport layer protocol.

	● TCP: Reliable but less-efficient transport layer protocol. ● TLS: Secured and Reliable transport layer protocol. ● DNS-SRV: A DNS RR for specifying the location of services.
NAT	To display and configure NAT(Net Address Translator) settings. ● STUN: Short for Simple Traversal of UDP over NATS, a solution to solve NAT issues. **Note**: By default, NAT is disabled.

4.3.2 Advance

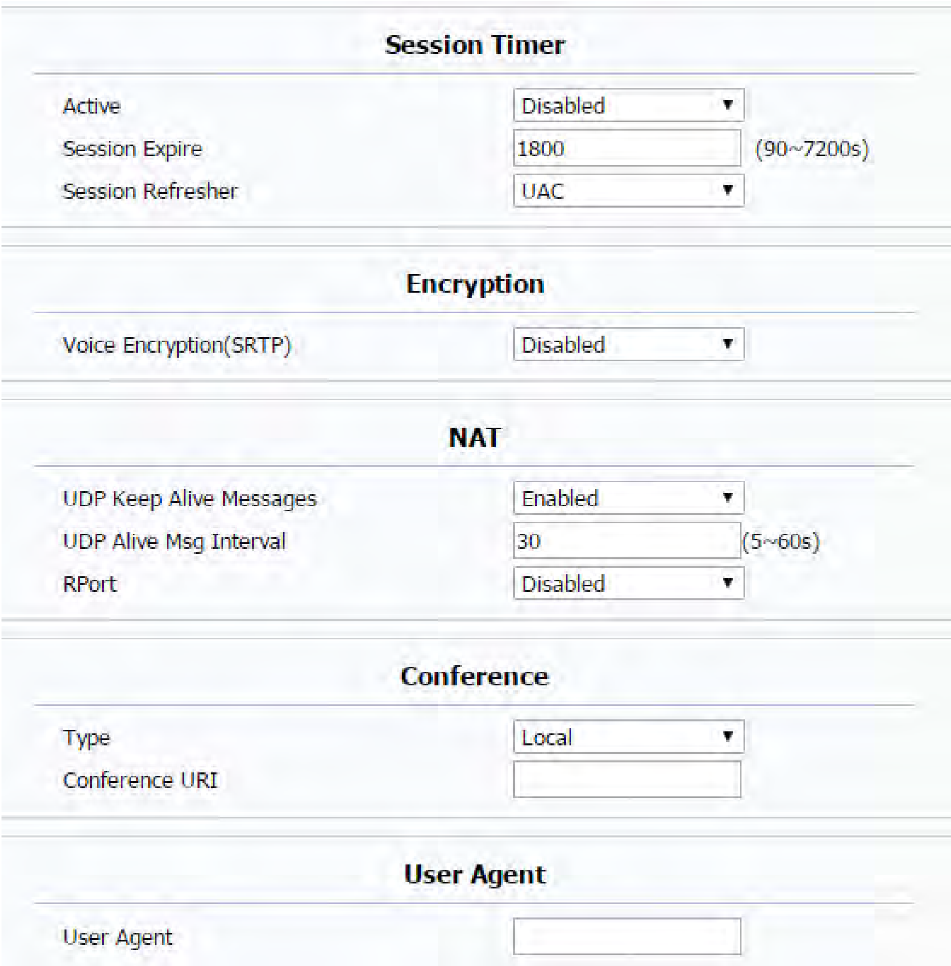


Sections	Description
SIP Account	To display current Account settings or to select which account to display.
Audio Codecs	To display and configure available/unavailable codecs list. Codec means coder-decoder which is used to transfer analog signal to digital signal or vice versa. Familiar codecs are PCMU(G711U), PCMA(G711A), G722 (wide-bandth codecs), G729 and so on.
Video Codecs	To configure the video quality. ● Codec Name: The default video codec is H264. ● Codec Resolution: It can support QCIF, CIF, VGA, 4CIF, 720P. ● Codec Bitrate: The lowest bitrate is 128, the highest bitrate is 2048. ● Codec payload: From 90-119.
Subscribe	To display and configure MWI, BLF, ACD subscription settings. ● MWI: Message Waiting Indicator which is used to indicate whether there is unread new voice message. ● BLF: BLF is short for Busy Lamp Field which is used to monitor the designated extension status. ● ACD: Automatic Call Distribution is often used in offices for customer service, such as call center. The setting here is to negotiate with the server about expire time of ACD subscription.



Sections	Description
DTMF	To display and configure DTMF settings. ● Type:Support Inband,Info, RFC2833 or their combination. ● How To Notify DTMF: Only available when DTMF Type is Info. ● DTMF Payload: To configure payload type for DTMF. **Note**: By default, DTMF type is RFC2833 which is the standard. Type Inband uses inband frequency to indicate DTMF tone which is most used to be compatible to traditional telephone server. Type Info uses SIP Info message to indicate DTMF message.
Call	To display and configure call-related features. ● Max Local SIP Port: To configure maximum local sip port for designated account. ● Min Local SIP Port: To configure minimum local sip port for designated account. ● Caller ID Header: To configure which Caller ID format to fetch for displaying on Phone UI. ● Auto Answer: If enabled, IP phone will be auto-answered when there is an incoming call for designated account. ● Ringtones: Choose the ringtone for each account. ● Provisioning Response ACK: 100% reliability for all

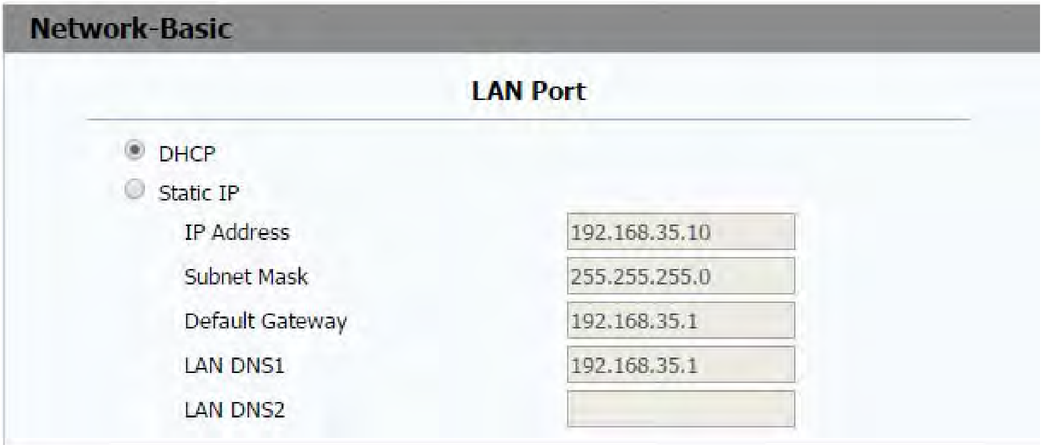
	provisional messages, this means it will send ACK every time the IP phone receives a provisional SIP message from SIP server. ● User=phone: If enabled, IP phone will send user=phone within SIP message. ● PTime: Interval time between two consecutive RTP packets. ● Anonymous Call: If enabled, all outgoing call for the designated account will be anonymous number. ● Anonymous Call Rejection: If enabled, all incoming anonymous-out call for the designated account will be rejected. ● Is escape non Ascii character: To transfer the symbol to Ascii character. ● Missed Call Log: To display the miss call log. ● Prevent SIP Hacking: Enable to prevent SIP from hacking.



Sections	Description
Session Timer	To display or configure session timer settings. ● Active: To enable or disable this feature, If enable, the ongoing call will be disconnected automatically once the session expired unless it's been refreshed by UAC or UAS. ● Session Expire: Configure session expire time. ● Session Refresher: To configure who should be response for refreshing a session. **Note**: UAC means User Agent Client, here stands for IP phone. UAS means User Agent Server, here stands for SIP server.
Encryption	To enable or disabled SRTP feature. ● Voice Encryption(SRTP): If enabled, all audio signal (technically speaking it's RTP streams) will be encrypted for more security.
NAT	To display NAT-related settings. ● UDP Keep Alive message: If enabled, IP phone will send UDP keep-alive message periodically to router to keep NAT port alive. ● UDP Alive Msg Interval: Keepalive message interval. ● Rport: Remote Port, if enabled, it will add Remote Port into outgoing SIP message for designated account.
Conference	To select Local or network conference. ● Type: To select desired conference type ● Conference URI: If network conference is selected, a network conference URI is needed to be input.
User Agent	One can customize User Agent field in the SIP message; If user agent is set to specific value, user could see the information from PCAP. If user agent is not set by default, user could see the company name, model number and firmware version from PCAP

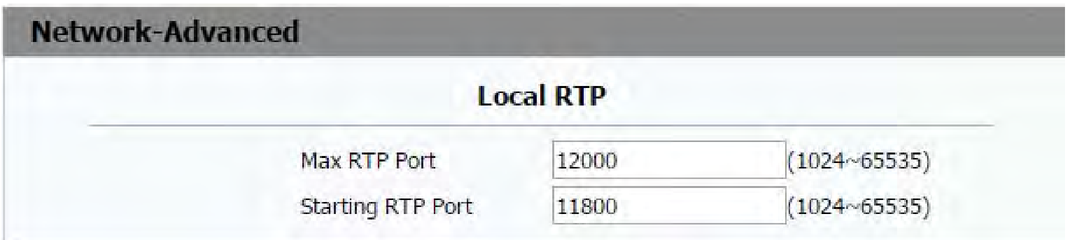
4.4 Network

4.4.1 Basic



Sections	Description
LAN Port	To display and configure LAN Port settings. ● DHCP: If selected, IP phone will get IP address, Subnet Mask, Default Gateway and DNS server address from DHCP server automatically. ● Static IP: If selected, you have to set IP address, Subnet Mask, Default Gateway and DNS server manually. ● PPPoE: Use PPPoE username/password to connect to PPPoE server.

4.4.2 Advance



Sections	Description
Local RTP	To display and configure Local RTP settings. ● Max RTP Port: Determine the maximum port that RTP stream can use. ● Min RTP Port: Determine the minimum port that RTP

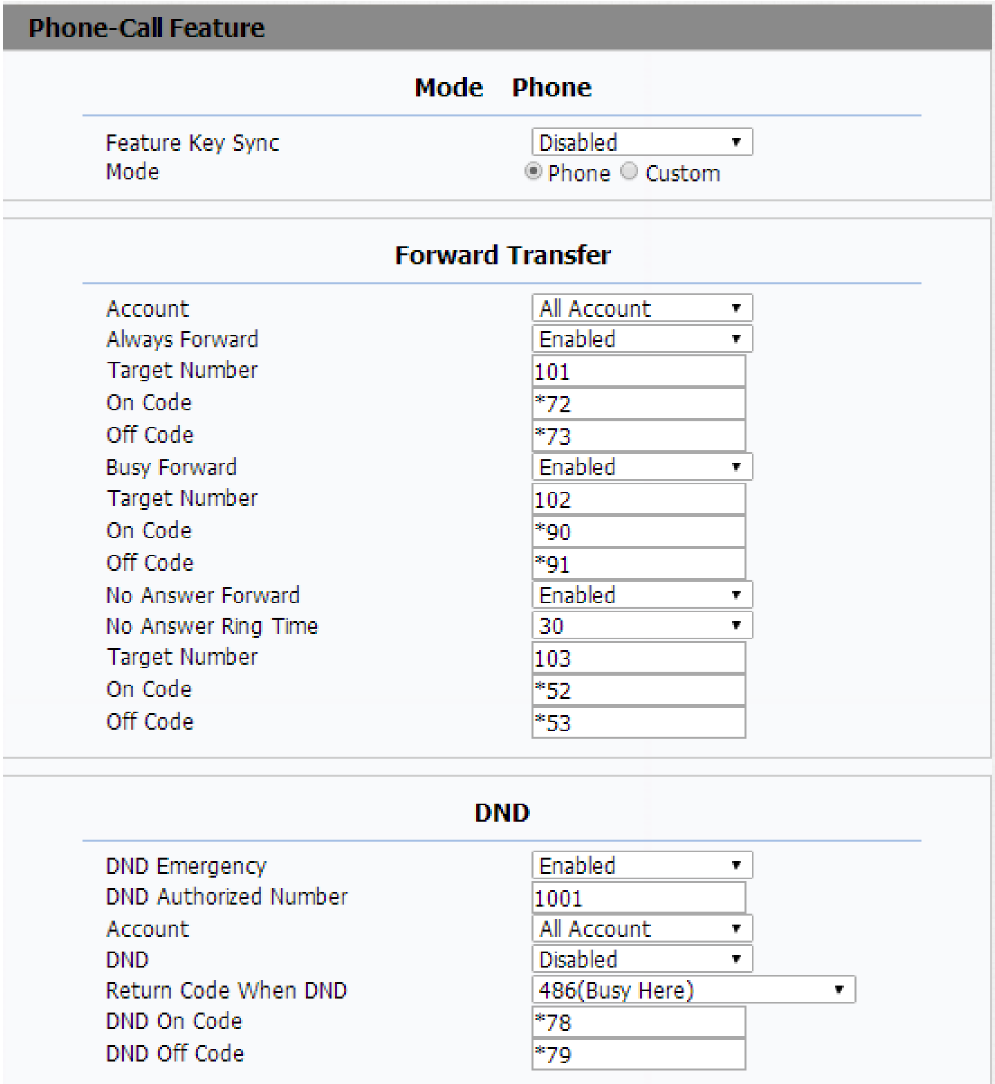
	stream can use.

4.5 Phone

4.5.1 Time/Language

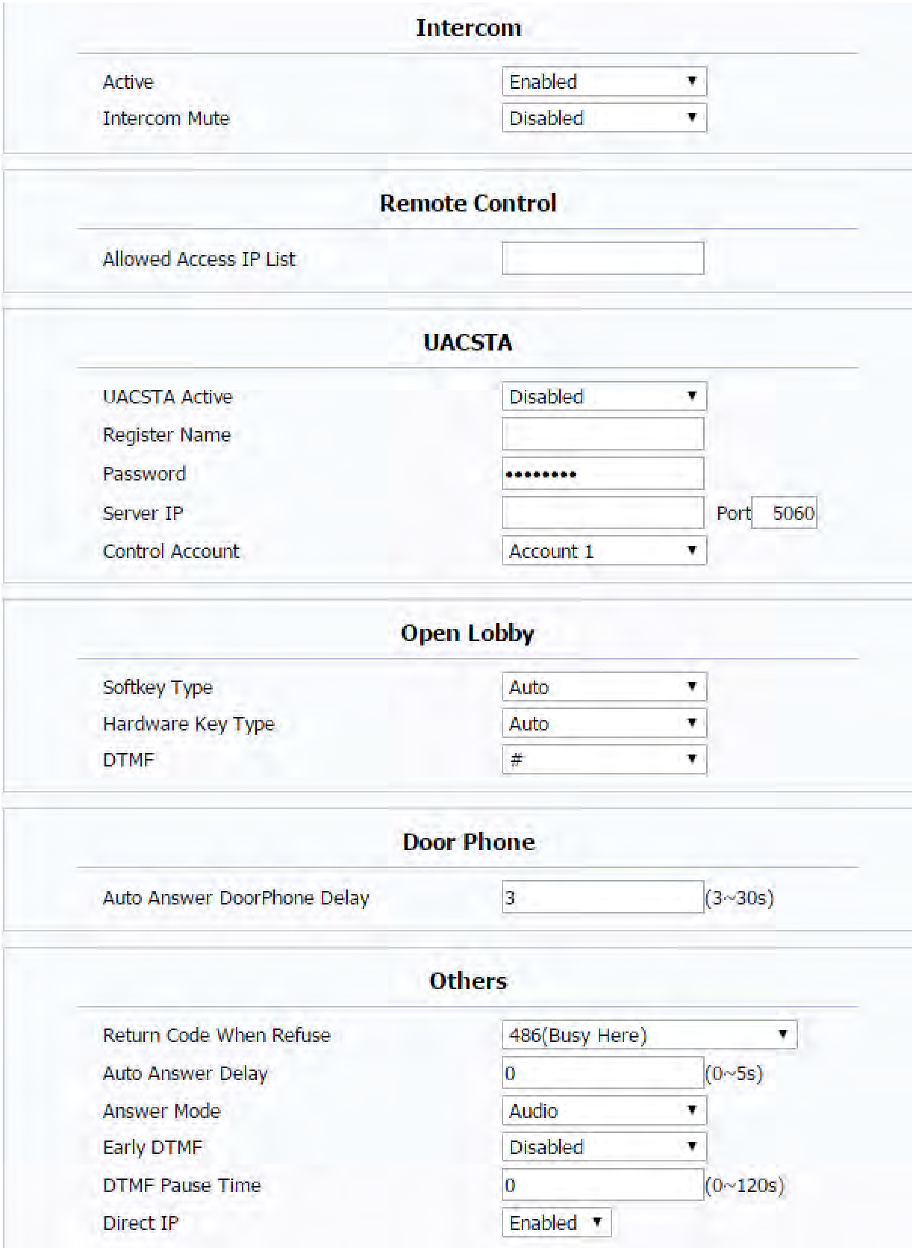
Sections	Description
Web Language	Choose the suitable web language you need. English by default.
NTP	To configure NTP server related settings. ● Time Zone: To select local Time Zone for NTP server. ● Primary Server: To configure primary NTP server address. ● Secondary Server: To configure secondary NTP server address, it takes effect if primary NTP server is unreachable. ● Update interval: To configure interval between two consecutive NTP requests. **Note**: NTP, Network Time Protocol is used to automatically synchronized local time with INTERNET time, since NTP server only response GMT time, so that you need to specify the Time Zone for IP phone to decide the local time.

4.5.2 Call Feature



Sections	Description
Mode	To enable or disable feature key sync. ● Feature Key Sync: To enable or disable feature key sync. ● Mode: Select the desired mode.
Forward Transfer	To display and configure Forward setting. **Note**: There are three types of forward: Always Forward, Busy Forward and No answer Forward. ● Always Forward: Any incoming call will be forwarded in any situation. ● Busy Forward: Any incoming call will be forwarded if IP

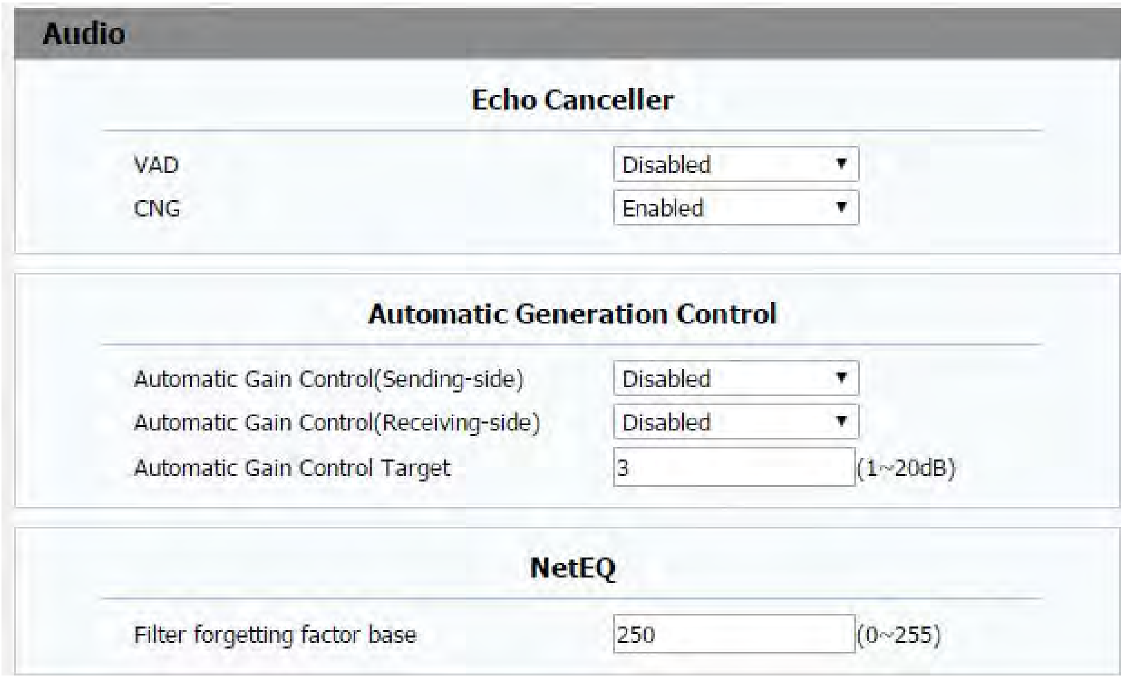
	phone is busy. ● No answer Forward: Any incoming call will be forwarded if it's no answer after a specific time.
DND	DND (Do Not Disturb) allows IP phones to ignore any incoming calls. ● DND Emergency: the phone from the Authorized number can still be received after enable this function. ● DND Authorized Number: Setup authorized numbers for DND Emergency. ● Account: Select an account for DND ● DND: Disable by default. ● Return Code when DND: Determine what responses code should be sent back to server when there is an incoming call if DND on. ● DND On Code: The Code used to turn on DND on server's side, if configured, IP phone will send a SIP message to server to turn on DND on server side if you press DND when DND is off. ● DND Off Code: The Code used to turn off DND on server's side, if configured, IP phone will send a SIP message to server to turn off DND on server side if you press DND when DND is on.



Sections	Description
Intercom	Intercom allows user to establish a call directly with the callee. ● Active: To enable or disable Intercom feature. ● Intercom Mute: If enabled, once the call established, the callee will be muted.
Remote Control	Remote Control allows specific host to interact with IP phone by sending HTTP or HTTPS requests. The specific action could be answering an incoming call, hangup an ongoing call and so on. ● Allowed Access IP List: To configure the allowed host address. ● **Note**: For now, IP phone can only support IP address, IP

	address list and IP address pattern as allowed hosts
UACSTA	Using CSTA for SIP phone user agents. It can control some features of calling. UACSTA is used to send ECMA-323(CSTA XML) information during SIP calling. The default status is disabled
Open Lobby	User can choose which types you need for each key. Akuvox IT82 supports 2 types to unlock-DTMF and Relay. DTMF is used to unlock the lobby door remotly, Relay is used to open the local door. Softkey: During the talking, user can press Unlock key to open the door. Hardware Key: User can also press hardware key to unlock the door. DTMF: If you choose DTMF code for one unlock key. Please setup the DTMF code.
Door Phone	When there is an incoming call from doorphone, setup the delay auto answer time, IT82 series will auto answer the call after the timeout. **Note:** if you enable Auto Answer function, this feature will be not available.
Others	● Return Code When Refuse: Allows user to assign specific code as return code to SIP server when an incoming call is rejected. ● Auto Answer Delay: To configure delay time before an incoming call is automatically answered. ● Auto Answer Mode: To set video or audio mode for auto answer by default. ● Early DTMF: Enable or disable early DTMF function ● Direct IP: Direct IP call without SIP proxy.

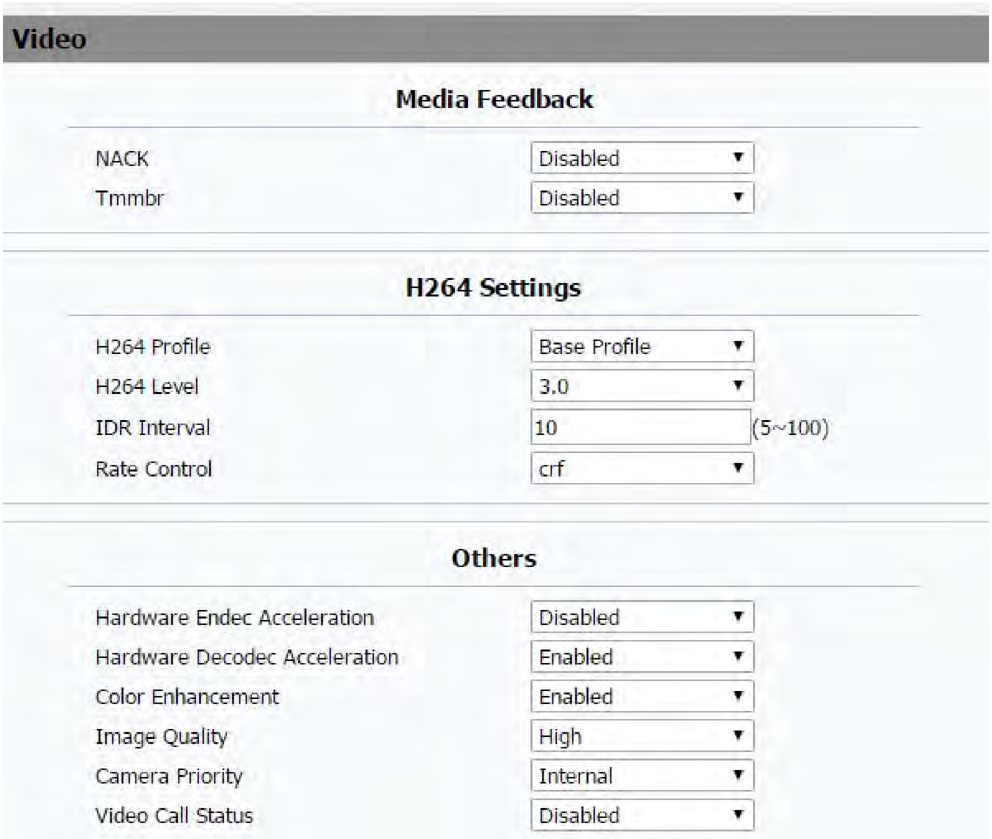
4.5.3 Audio



Sections	Description
Echo Canceller	Echo Canceller: To remove acoustic echo from a voice communication in order to improve the voice quality . ● VAD(Voice Activity Detection): Allow IP phone to detect the presence or absence of human speech during a call. When detecting period of "silence", VAD replaces that silence efficiently with special packets that indicate silence is occurring. It can facilitate speech processing, and deactivate some processes during non-speech section of an audio session. It can avoid unnecessary coding or transmission of silence packets in VoIP applications, saving on computation and network bandwidth. ● CNG(Comfort Noise Generation): Allow IP phone to generate comfortable background noise for voice communications during periods of silence in a conversation. It is a part of the silence suppression or VAD handling for VoIP technology. CNG, in conjunction with VAD algorithms, quickly responds when periods of silence occur and inserts artificial noise until voice activity resumes. The insertion of artificial noise gives the illusion of a constant transmission stream, so that

	background sound is consistent throughout the call and the listener does not think the line has released.
Automatical Generation Control	R48G will auto adjust amplification circuit gain via signal. ● Automatic Gain Control(Sending-side): Disabled by default ● Automatic Gain Control(Receiving-side): Disabled by default ● Automatic Gain Control Target: Range from 1 to 20 dB. 3dB by default.
NetEQ	Filter forgetting factor base: Range from 0~255. 250 by default.

4.5.4 Video



Sections	Description
Media Feedback	● NACK: Enabled it to filter mosaic. ● Tmmbr: Send the maximum temporary rate request. Disabled by default.
H264 Settings	H264: A video stream compression standard. Different from H263, it provides an approximately identical level of video stream quality but a half bit rate. This type of compression is

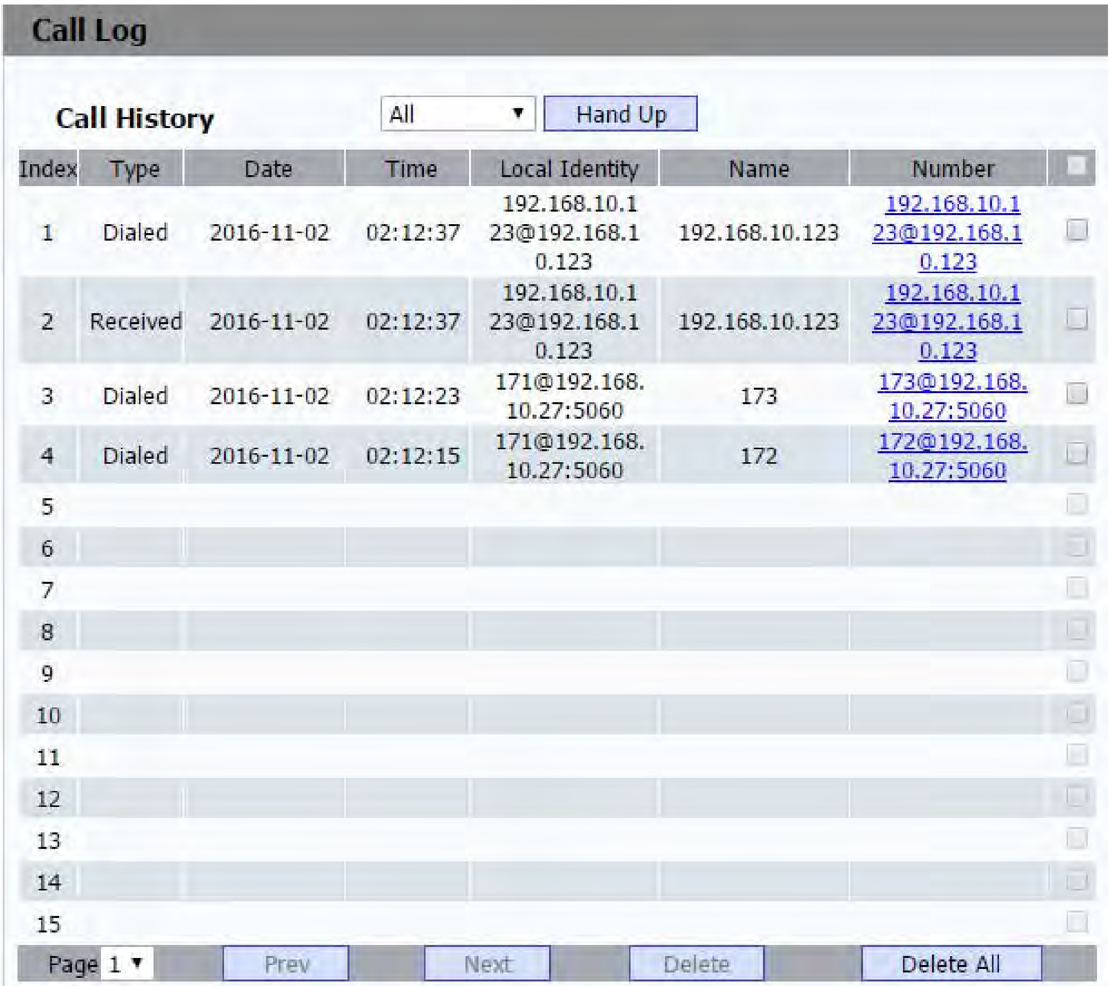
	sometimes called MPEG-4 part 10. To setup corresponding H264 video parameters. ● H264 Profile: There are 4 modes-Base、Main、High、Extend profile. Different profiles makes different coding function and video quality. ● H264 Level: Different profiles has corresponding Level value. ● IDR Interval:IDR means Instantaneous Decoding Refresh. It is used to control the process of coding and decoding. ● Rate Control: Choose one H264 video rate.
Others	● Hardware Endec Acceleration: This function is used to solve the image issue. User can enable the Hardware Endec Acceleration when you need. ● Hardware Decodec Acceleration: Disabled by default. ● Color Enhancement: To increase the phone display color. Enabled by default. ● Image Quality: User can select Low, Middle or High mode. ● Camera Priority: IT82 series can connect extra camera. If R48G has 2 cameras , please setup the priority for external camera or internal one. ● Video Call Status

4.6 PhoneBook

4.6.1 Local Book

Sections	Description
Contact	To display and select local contact type. ● All Contacts: To display or edit all local contacts. ● Black List: To display black list contacts.
Search	To search designated contacts from local phonebook.
Dial	To dial out a call or hangup an ongoing call from Web UI. **Note**: For this feature, you need to have the remote control privilege to control IP phone via Web UI. Please refer to section "Remote Control" in the Web UI->Phone->Call Feature page.
Group	To display or edit Group contacts.
Group Setting	To display or change Group name, related ringtone or description.
Import/Export	To import or export the contact or blacklist file.

4.6.2 Call Log



Index	Type	Date	Time	Local Identity	Name	Number	
1	Dialed	2016-11-02	02:12:37	192.168.10.123@192.168.10.123	192.168.10.123	192.168.10.123@192.168.10.123	
2	Received	2016-11-02	02:12:37	192.168.10.123@192.168.10.123	192.168.10.123	192.168.10.123@192.168.10.123	
3	Dialed	2016-11-02	02:12:23	171@192.168.10.27:5060	173	173@192.168.10.27:5060	
4	Dialed	2016-11-02	02:12:15	171@192.168.10.27:5060	172	172@192.168.10.27:5060	
5							
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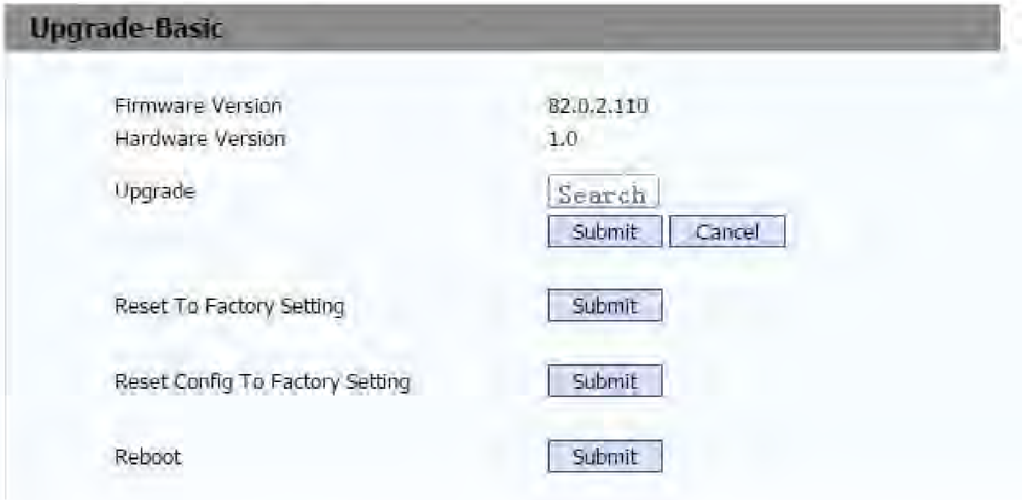
Sections	Description
Call History	To display call history records. Available call history types are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls. ● HangUp: To click to hangup ongoing call on the IP phone. **Note**: For “HangUp” feature, you need to have the remote control privilege to control IP phone via Web UI. Please refer to section “Remote Control” in the Web UI->Phone->Call Feature page.

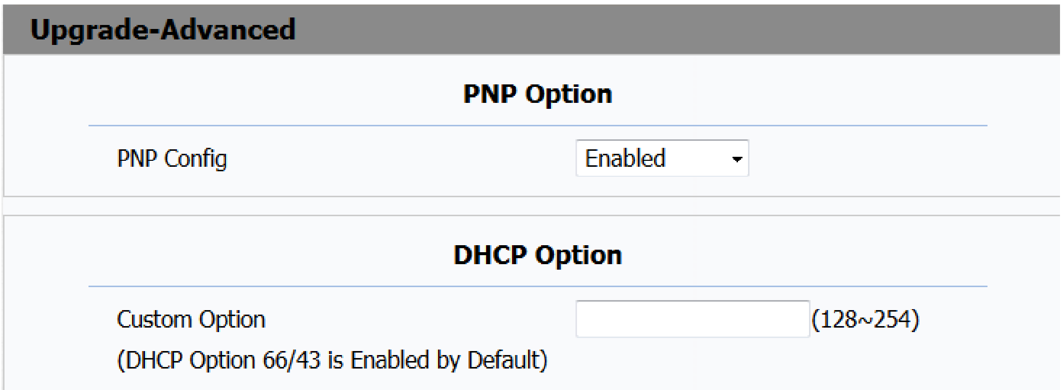
4.7 Upgrade

4.7.1 Basic

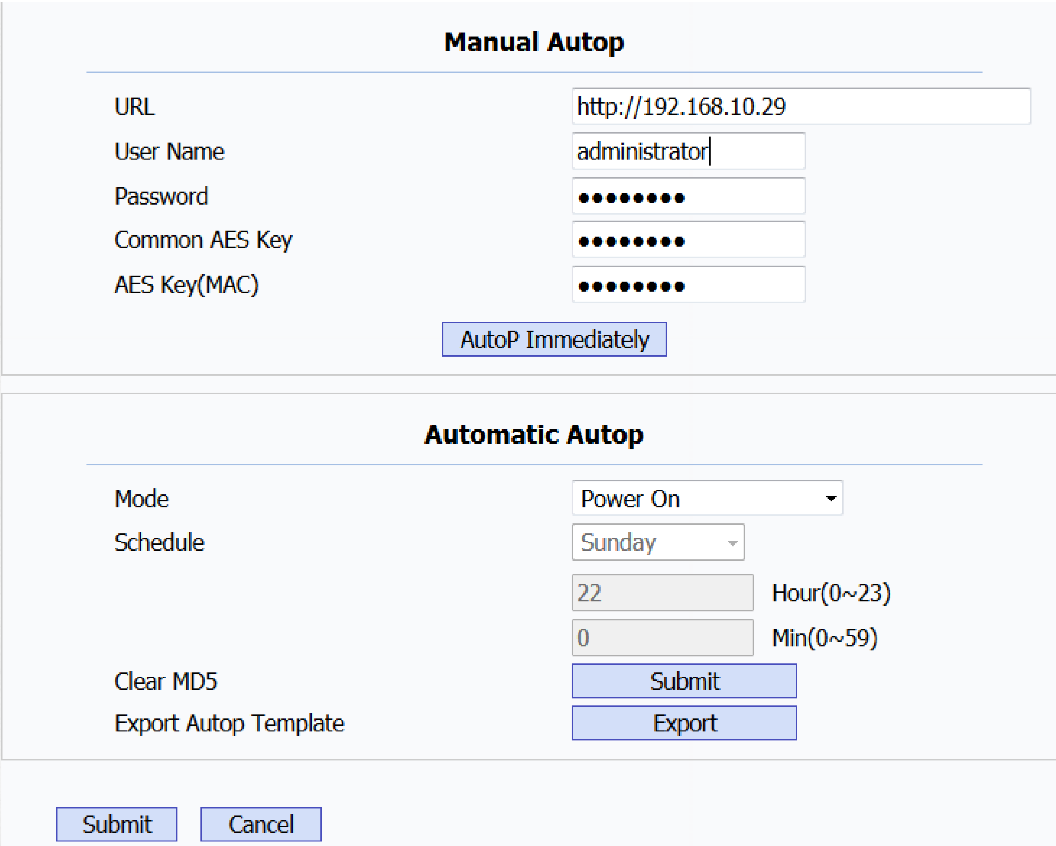


Sections	Description
Firmware version	To display firmware version, firmware version starts with MODEL name.
Hardware Version	To display Hardware version.
Upgrade	To select upgrading zip file from local or a remote server automatically. **Note:** Please make sure it's right file format for right model.
Reset to Factory Setting	To enable you to reset IP phone's setting to factory settings.
Reboot	To reboot IP phone remotely from Web UI.

4.7.2 Advance



Sections	Description
PNP Option	To display and configure PNP setting for Auto Provisioning. ● PNP: Plug and Play, once PNP is enabled, the phone will send SIP subscription message to PNP server automatically to get Auto Provisioning server's address. By default, this SIP message is sent to multicast address 224.0.1.75(PNP server address by standard).
DHCP Option	To display and configure custom DHCP option. ● DHCP option: If configured, IP Phone will use designated DHCP option to get Auto Provisioning server's address via DHCP. This setting require DHCP server to support corresponding option.



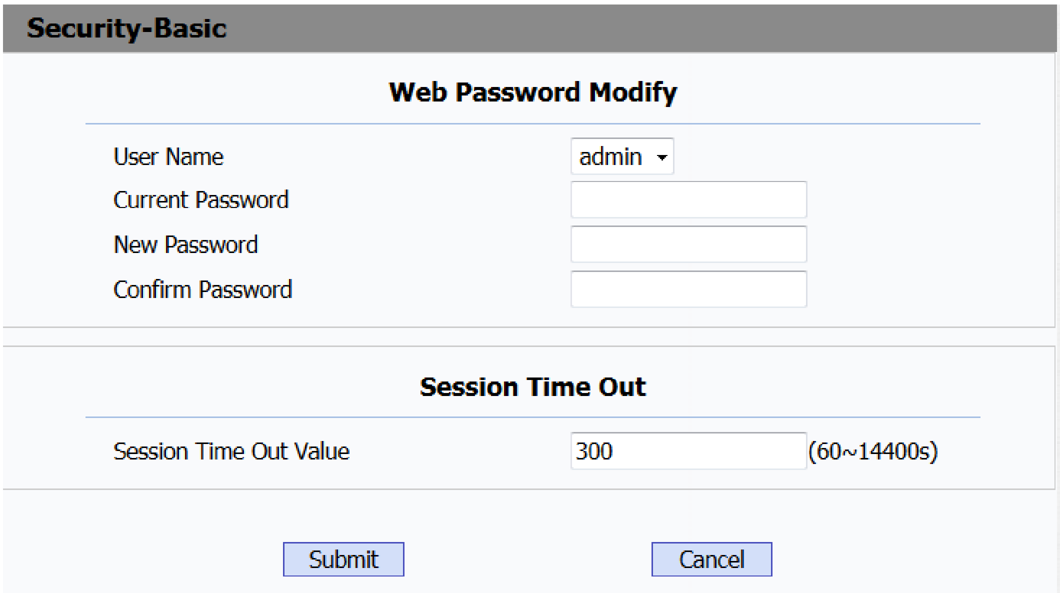
Sections	Description
Manual Auto	To display and configure manual update server's settings. ● URL: Auto provisioning server address. ● User name: Configure if server needs an username to access, otherwise left blank. ● Password: Configure if server needs a password to access, otherwise left blank. ● Common AES Key: Used for IP phone to decipher common Auto Provisioning configuration file. ● AES Key(MAC): Used for IP phone to decipher MAC-oriented auto provisioning configuration file(for example, file name could be 0c1105888888.cfg if IP phone's MAC address is 0c1105888888). Enter the URL address,then click the AutoP Immediately label ,the phone will according the URL to ask for configuration file to update. **Note:** AES is one of many encryption, it should be configure only configure filed is ciphered with AES, otherwise left blank.
Automatic AutoP	To display and configure Auto Provisioning mode settings. This Auto Provisioning mode is actually self-explanatory. For example, mode "Power on" means IP phone will go to do Provisioning every time it powers on.

Sections	Description
System Log	To display syslog level and export syslog file. ● Syslog level:From level 0~7.The higher level means the more specific syslog is saved to a temporary file. By default, it's level 3. ● Export Log: Click to export temporary syslog file to local PC. ● Remote System Log: To enable or disable Remote System Log. ● Remote System Server: To input the syslog server address.
PCAP	To start, stop packets capturing or to export captured Packet file. ● Start: To start capturing all the packets file sent or received from IP phone. ● Stop: To stop capturing packets. ● Export: To export the capture packet file,use capture tool to open the file. **Note:** IP phone will save captured packets file to a temporary file, this file maximum size is 1M(mega bytes), and will top capturing once reaching this maximum size.
Others	To display or configure others features from this page.

	● Config file: To export or import configure file for IP phone.

4.8 Security

4.8.1 Basic



Sections	Description
Web Password Modify	To modify user's password. ● Current Password: The current password you used. ● New Password: Input new password you intend to use. ● Confirm Password: Repeat the new password. **Note**: For now, IP phone can only support user admin.
Session Time Out Value	Over the session time out value, users need to login in the web again. ● Session Time Out Value: the ranger is from 60s to 14400s.

4.8.2 Advance

Sections	Description
Web Server Certificate	To display or delete Certificate which is used when IP phone is connected from any incoming HTTPs request. **Note**: The default certificate could not be deleted.

Sections	Description
Web Server Certificate Upload	To upload a certificate file which will be used as server certificate.
Client Certificate	To display or delete Certificates which is used when IP phone is connecting to any HTTPs server.
Client Certificate Upload	To upload certificate files, this is used as client certificate. ● Only Accept trusted Certificates: If this option is enabled, only trusted certificates will be accepted.