Akuvox Big Button SIP Phone

R15P User Manual

Content

1 Overview

1.1 Introduction

Akuvox R15P is a big button phone with full duplex hands-free speakerphone. It also can be called Akuvox Big Button SIP phone. Specially for the elder, R15P provides many handy features, like emergency call, remote pendant control and speed dial. For instant, when an alone elder needs help, he can press the pendant to call out the predefined number for emergency help.

Based on the SIP standard, the Akuvox R15P has been tested to ensure comprehensive interoperability with equipments from VoIP infrastructure leaders enabling service providers to quickly roll-out competitive, feature rich services to their customers.

Akuvox R15P is very easy to understand, configure, and deploy. The web interface is designed to provide a clean and user-friendly configuration window so that users won't get lost in complicated menus and maintenance.

1.2 Features

✧ Highlights

- Easy to use
- HD Voice with High Adjustable Volume
- Adjustable Tone
- Hearing Aid Compatible (HAC)
- Big Buttons, SOS key
- Up to 10 Wireless Emergency Pendants Supported(Optional)
- Support PoE
- Fully Compatible with Asterisk, BroadSoft Platform

✧ Phone Feature

- 1 Line (Support 1 SIP account)
- Support Call Waiting, Call Forward, Call Transfer
- Call on Hold, Mute, Auto-answer, Redial, DND
- Local 3-Way Conference
- Volume Adjustable, Tone Adjustable
- Ring tones Selectable
- Speed Dial, Hotline
- Network Packet Capture
- Country Tone Signal
- Direct IP call
- Auto Redial, Call Return
- Dial Plan
- Action URL/URI
- Phonebook (500 entries), Blacklist (100 entries),Call logs (100 entries)
- Remote mode switch
- Health Care
- Multi-Language Support

✧ Audio Feature

- HD Voice with high volume
- Hearing Aid Compatible (HAC)
- Wideband Codec: G.722
- Narrowband Codec: PCMA, PCMU, G.729, G723_53, G723_63,
- G726_16, G726_24, G726_32, G726_40
- VAD, CNG , Echo Canceller
- Full-Duplex Speakerphone

✧ Network Feature

- SIP V1(RFC2543), V2(RFC3261)
- Static IP/DHCP for IP configuration
- 3 DTMF modes: In-Band, RFC2833, SIP INFO
- HTTP/HTTPS Web Server for Management
- NTP for Auto Time Setting
- TFTP/FTP/HTTP/HTTPS Protocols
- 802.1Q VLAN

✧ Administrator Feature

- Auto provisioning using FTP/TFTP/HTTP/HTTPS/PnP
- Dial through IP PBX Using Phone Number
- Dial through IP PBX Using URL Address
- Configuration Managements with Web, Keypad on the phone and Auto Provisioning
- SNMP
- TR069

✧ Security Feature

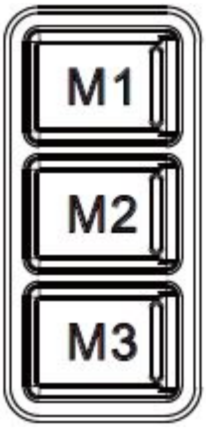
- Support HTTPS (SSL)
- Support SRTP for Voice Data Encryption
- Support Login for Administration
- SIP Over TLS

✧ Physical Feature

- Audiocodes Chipsetsss
- 2.9’’132x64 Graphical LCD with Backlight
- 25 Keys (with 1 SOS key, 4 Soft Keys, 3 Image Programmable Keys)
- 3 LED Lights (1 Power Light, 1 Handfree Light and 1 SOS Light)
- Up to 10 Wireless Emergency Pendants Supported(Optional)
- RJ9 Handset Jack and Headset Jack
- 2 RJ45 10/100 Ethernet Jacks
- AC Power Adapter: Input: AC 100-240V; Output: DC 5V/1A
- PoE: IEEE 802.3af
- Gift Box Size:248 x 204 x 107 (mm),weight: 1.19 kg

1.3 Keypad

Key	Key name	Function Description
	Soft key	Key combination includes functions such as History/Favorites/Redial/Call Return/HotDesking/XML Browser/DND/Menu/MSG/Status/Book/FWD/PickUp/Group /PickUp/Intercom/Speed Dial/and so on.
	Tone key	Turn down or turn up the tone by pressing the "-" key or the "+" key.
	Volume key	Turn down or turn up the volume by pressing the "-" key or the "+" key.

	SOS key	Dial out the specified emergency number
	Return/Recall/Transfer key	Back to the previous page or dial out the latest incoming call.
	Redial key	View the Missed Calls, Incoming Calls and Dialed Calls.
	Speaker key	Make the phone into hands-free mode.
	Memory key	Dial out the pre-configuration relative number.
	Digital Keyboard	Inputting the phone number or DTMF.

1.4 Installation

Check package contents

Name	Quantity
SIP IP Phone unit	1
Pendant (optional)	1

handset	1
RJ-9 Cable	1
Power Adapter	1
RJ-45 Cable	1
Stand	1
Quick installation guide	1

1.5 Installation Steps

Step 1 – Connect the power

Connect the provided power adapter to the Power port and plug the adapter into an available power outlet. The LCD will display “Initializing, Please Wait…”

Note1: Never use a power adapter other than the one provided with Akuvox R15P

Note2: Only Internet port supports POE.

Step 2 – Connect to the Internet

Connect one end of the RJ-45 Ethernet cable to the Internet port at the back of the Akuvox R15P and the other end to wall network jack.

Step 3 – Connect the computer

Connect one end of the RJ-45 Ethernet cable to the PC port at the back of the Akuvox R15P and the other end to the Ethernet port on you computer.

Step 4 – Configure the device

Launch the web browser on your computer, and enter the IP address of the phone into the address bar. The login screen will appear if the address is correct. Enter the user name and password to log into the web console.

Step 5 - Using with a pendant

Pendant needs to match with IP Phone. After learning, users can use pendant to call out the emergency number. Please refer to the chapter 3.7 for detail.

NOTE: 1.Each phone has its own IP address, you can check in Status interface.

2. Pendant uses button battery.

2 Functions

2.1 Make a call

User can make a phone call via the following methods：

1. Pick up the handset, icon will be shown on the idle screen.

2. Press the Handfree key, icon will be shown on the idle screen.

3. Press the Headset key if the headset is connected to the Headset Port in advance. The icon will be shown on the idle screen.

User can also dial the number first, and then choose the method user will use to speak to the other party.

2.2 Call Method

User can press the available line key, then

1. Dial the number User wants to call.

2. Press softkey. Use soft key to choose the call, click to dial out.

3. Press the Redial key twice to call the last number called or press Redial key to enter All Calls interface to choose the number to dial out.

4. Press the pendant to dial out the predefined number. Please refer to chapter 3.7 for detail.

2.3 Answer a call

1. If User is not on another phone, lift the handset to use, or press the Speaker key to answer using the speaker phone, or press the headset key to answer the headset.

2. If User is on another call, press softkey to answer new incoming and hold the current talking. During the conversation, User can alternate between Headset, Handset and Handfree by pressing the corresponding keys.

Note: The will flash during the Incoming interface

2.4 Call Hold/Resume

1. Press softkey to put User active call on hold.

2. If there is only one call on hold, press the hold softkey to retrieve the call.

3. If there are more than one call on hold, press the line button, and select the call, and then press the Resume button to retrieve the call.

2.5 DND

If you enable DND mode, the phone will reject to answer all calls automatically and play busy tone, the UI will present missed calls at the same time.

- 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.

2.6 Call waiting

Enable Call Waiting to ensure that the third party incoming call can be received while you are talking with another one. To configure Call Forward via web interface:

1. Phone->Call Feature->Call Waiting.

2. Enabled or disabled call waiting.

3. Then click Submit to save the changes.

2.7 Call forward

You can set the static forward to redirect all the incoming calls to specified number;

Also you can use dynamic forward to redirect all the incoming calls forward to the number you input when the phone is ringing.

Forward: Enable call forward feature, Options as follows：

- Always forward: All the incoming calls will be forwarded unconditionally to specified number.
- Busy Forward: The incoming calls will be forwarded to specified number when the phone is busy.
- No answer Forward: The incoming calls will be forwarded to the specified number when the ring tone is time out without answer.

2.8 Call transfer

R15P can only support one way to transfer the call during the conversation.

Blind Transfer: Transfer talking directly to the other party without any negotiation. Press transfer softkey or Key during the call, enter the number you want, then click transfer key again.

2.9 Conference

You can use the local conference feature to hold a 3-way conference by pressing the softkey to invite the current talking and one line talking held to attend conference. The Network conference feature allows you to add or delete the party who attend the conference.

The local conference feature of IP phone Akuvox R15P can invite two parties at most to attend conference. The conference type of IP phone Akuvox R15P is Local conference by default.

2.10 Call pickup

You can use pickup to answer other users' incoming call. The IP phone Akuvox R15P supports specified pickup and group pickup.

Specified pickup can answer specified user's incoming calls

1. Set specified pickup key via website

PATH: Phone->Key/Display->Soft key/Function Key, setup the Type as Pickup,enter the pickup number in Value and select the corresponding account.

2. Use specified pickup feature

When the user of specified pickup number is off or busy, you can press the pickup key to answer incoming call.

2.11 Group pickup

Group pickup can answer group's user incoming calls. Group pickup needs to set group members.

1. Set group pickup via phone interface

PATH: Phone->Key/Display->Soft key/Function Key, setup the Type as Group Pickup, enter the pickup number in Value and select the corresponding account.

2. Use group pickup feature

When anyone in group receives an incoming call, you can press the group pickup key to answer.

Note: Press the group pickup only to answer line 1 incoming call if there are more than 1 incoming calls in group.

2.12 Speed dial

You can use the Speed Dial feature to dial the specified contact directly. R15 often uses Memory Key as Speed Dial.

Setup Speed Dial in website

PATH: Phone->Key/Display->Soft Key/Function Key, setup Type as Speed Dial, and enter the corresponding value.

2.13 Auto redial

When hang-up by the other party, call failure during the calling, the phone will enter the auto-redial screen, and begins to count. Press ✔ soft key or wait for the time is up. After trying the predefined times of setting of auto-redial, the phone will hang-up automatically.

To configure Auto Redial in website;

1. Phone->Call Feature->Auto Redial;
2. Enabled or Disabled Auto Redial. Disabled by default;
3. To configure Interval and Times;
4. Then click Submit to save the changes.

2.14 Hotline

The Hot line refers to the number you often dial. You can set hot lines in the phone, the phone will dial the hot line number automatically when you pick up the handset, press the hand-free or the account key. Also you can set the delay time of dialing the hot line number, then the phone will dial the hot line number automatically after the delay time.

To configure Hot line in website:

1. Phone->Call Feature->Hot line
2. Enabled or Disabled Hot line. Disabled by default.
3. To configure Number and Delay time.
4. Then click Submit to save the changes.

2.15 Intercom

To configure Intercom in website:

Please go to the path: Phone->Call Feature->Intercom to enable the Intercom feature first.

PATH: Phone->Key/Display->Soft Key/Function Key->Intercom

Press the Intercom key when the phone is available. The phone will connect the extension number of remote user automatically.

1. Press the Intercom key or the softkey to end the intercom.
2. Answer the intercom incoming calling.
3. In default situation, the IP phone Akuvox R15P will answer the intercom incoming calling automatically and make a noise. You can set the phone to enable silent mode when picking up the intercom call so that the other will not hear you.

2.16 Emergency Call

This feature is specially designed for the elders. When the user needs help, he/she can press SOS key or pendant(Please refer to the chapter 3.7) to dial out for emergency help. R15P can call out for three predefined numbers in a loop, each number will be called for 60s (by default). When called side receives Emergency call, a prompt voice message will be played repeatedly, and it is needed to press the number 5 during the call to confirm that Emergency call is well received.

Setup Emergency call in website:

1. Go to the path: Phone->Emergency call->Emergency call
2. Enter three different phone numbers or IP addresses you need.
3. Setup the call timeout
4. Click Submit to save.

2.17 RF Number(optional)

This function is similar with Emergency call. But different from Emergency call number, users can predefine only one RF number. After configuration, user can press the pendant to dial out the RF Number for help.

Setup RF Number in website:

1. Go to the path: Phone->Emergency call->RF Key
2. Setup the RF Key Type as RF Number
3. Enter a target phone number or IP address in Parameter.
4. Click Submit to save.

2.18 Health care-- Auto Answer Number

Users can preconfigure many auto answer numbers in website. If the incoming number matches a stored AA number, handfree will be activated on R15P. Then the elder can answer the call without moving.

Note: Please use comma to separate these numbers.

2.19 Remote Mode Switch

During the specified calling, like Emergency call or RF Number call, the other side can control the conversation via numeric key 4,5,6. This feature ensures that the two sides can hear more clearly in a particular case.

Numeric key 4: The other side is mute

Numeric key 5: Normal mode

Numeric key 6: R15P is mute

3 Settings

3.1 Language

You can change the Phone language through below method:

Press Menu->Settings->Basic Setting->Language

3.2 Data&Time

1. The IP phone displays Time and Date in Idle status. You can set the Time and Date obtain from SNTP server automatically or you can set the time and date manually.

2. Set SNTP via phone interface: Access Menu->Settings->Basic Setting->Date & Time ->SNTP Setting.

3. To set the date & time format via the phone interface, access Menu->Settings-> Basic Setting->Date & Time->Format Setting:

- Access the Time Format in Format Setting interface, then press the Left or Right Soft key to select the time format (12Hour or 24Hour).
- In the Date &Time Format interface, press the ▲ or ▼ key on the phone keyboard to access the Date Format. Press ◀▶ softkey to select the date format to process setting.
- The phone supports six Date formats. The selected date format will appear in the Idle. For example, if the time was “2015-11-18”, the date formats in the menu and the corresponding formats displayed in the Idle as follows:

Date Format	**Example(2015/11/18 Wed)**
YYYY-MM-DD	2015-11-18
YYYY/MM/DD	2015/11/18
DD-MM-YYYY	18-11-2015
DD/MM/YYYY	18/11/2015

WW-DD-MM	Wed 18 Nov
WW-MM-DD	Wed Nov 18

3.3 Ringtones

1. The Ring Tone refers to incoming ring tone, which remind the user that there is a new incoming call. The IP phone Akuvox R15P supports phone ring tone to distinguish the incomings from other near phones’ ring tone; Besides, the IP phone Akuvox R15P also support setting specific incoming ring tone for contacts.

2. To set the ring tone via the phone interface, access Menu -> Settings -> Basic Setting -> Ring Tones.

3.4 Phone Volume

1. The Volume key can be used to adjust the volume of handset, hands-free or headset during a call. Also, the key can be used to adjust the ring tones volume in the Idle mode.

2. Adjust the volume via the phone interface, access Menu -> Settings -> Basic Setting -> Phone Volume. In the Volume Setting interface, access the Handset Volume, Hand-free Volume or Headset Volume interface, then press the - or + softkey to adjust the volume. Press the ✔ softkey to save the operation or press the ↩ softkey to cancel operation.

3.5 Phone Tone

R15P supports different sound frequencies. To adapt to different people by adjusting different levels. For example, in order to hear clearly, using high frequency is more suitable for the elders.

Please press the Tone key in the keypad to adjust an appropriate phone tone.

3.6 Backlight

Set the screen backlight level and duration.

Press Menu->Settings->Basic Setting->Backlight

3.7 RF Keys Status(optional)

To show the learned RF key status. Enable the RF Key alarm, it will show the KEY ID, last receive time and battery power. Pendant will send the heartbeat message while it is used, then the phone will receive the time of last heartbeat message. If the pendant is over 26 hour without any operation, the phone will alarm and the Last Recv Time will time again.

Note: If the pendant powers on again , it will also send the heartbeat message.

3.8 RF Keys Learning(optional)

The RF key is the small size of phone on your hand, which enabled multiple features gives user remote control ability to control the phone. Once users use this feature, it needs to match with the phone. R15P can support up to 10 pendants.

To configure RF keys:

1. Go to the path: Settings->Advanced Settings->RF Keys Learning
2. Choose one RF Key and click ✔ Soft key
3. Press the pendant and click soft key on the phone.
4. It will show up “ RF Key1 Learned”

5. Go to phone webpage->Phone->Emergency Call->RF Key.
6. To enter the RF key Type and value
7. Click Submit to save.
8. Users can choose the learned RF keys, then click 🗑 soft key to delete it.

3.9 Phone

3.9.1 Local Phone Book

The Local Phone Book is used for storing the contacts names and number. The Akuvox R15P can store up to 500 entries contacts. You can add, edit, delete, search, or call any contact from the Local Phone Book.

Add contact manually

Add contacts manually from the Local phone book via Phone interface:

Press Phone book->Local phone book->All Contacts.

Select the relevant group (For example: contacts) and Press the ✔ soft key to enter All Contacts.

1. Press the ✔ soft key to enter the Add Contact interface.
2. Press the **+** soft key, then Input name in the relevant area.
3. Press the ▼ soft key to input the office number in the relevant area.
4. Press the ▼ soft key to input mobile number in the relevant area.
5. Press the ▼ soft key to input other number in the relevant area.
6. Press the ▼ soft key to enter RingTone selection; Press ◀▶ key to adjust the ringtone you need;
7. Press the ▼ soft key to enter Group selection; Press ◀▶ key to select the group you have build before, or you can use the default group.
8. Press ✔ soft key to save the contact.

Add contact from all call history

Add contact from All Calls History in the phone interface:

1. Press soft key;
2. Press soft key to select the contact you want to add;
3. Press soft key to add to contacts.

Search contacts

1. Enter the Local Phone Book interface
2. Press soft key, then enter the keywords to search contacts.

Input keywords such as name, any character of number or whole phone number, press softkey to enter the Search Contacts interface.

3.9.2 Blacklist

100 Blacklists contacts are available with Akuvox R15P IP phone. You can add, edit, delete, search or call contact. Any calls from the number in the blacklists will be rejected.

PATH: Press Phone book->Blacklist, press + softkey to add the contact into blacklist.

3.10 History Management

The History management of IP phone Akuvox R15P supports 100 logs storage at most. You can check the history, make calls from the calls history and delete the calls history.

1. Press the soft key, the LCD will display all the recent calls；
2. Press the soft key to select the log；
3. Press the soft key and select the detail. The LCD will display the detailed information of this log; Press the soft key, to make a call from the History;
4. Press the soft key to add to contacts(Add to Blacklists) from the History;

5. Press the soft key to select “Delete” to delete calls log from the History;
6. Press the soft key to select “Delete all” to delete all the call logs from the History.

3.11 SIP account management

Register an account

Register an account via phone interface：

1. Press soft key to enter setting interface to select advanced setting, input password (password: admin) to select account;
2. Press softkey;
3. Select “Enable” in the account activation status area;
4. Input the label, display name, register name, account, password and SIP server separately；
5. Press softkey to save or softkey to cancel；

Disable an account

1. Access Menu->Settings->Advanced setting->Account (password: admin).
2. Select the account you want to disable and press softkey.
3. Select ”Disable” in the account active status area.
4. Press softkey to save or softkey to cancel.

3.12 Basic Network Settings

3.12.1 LAN Port

DHCP Mode

1. In the Network Settings interface, press the softkey to enter LAN Port.
2. In the LAN Port interface, press or key on the phone keyboard to select DHCP (default is DHCP).

3. Press ✔ soft key to enter the DHCP switch interface, it will auto return to last interface after seconds.

Static Mode

1. In the LAN Port interface, press the ▲ or ▼ key on the phone keyboard to select Static IP, then Press ✔ soft key to enter Static IP Setting interface and input IP address.
2. Input the IP address, Subnet mask, Gateway, DNS 1 and DNS 2 in the corresponding area, Press ✔ softkey to save.

3.12.2 PC Port

1. In the Network Settings interface, press ▲ or ▼ soft key to select PC Port, press ✔ softkey to enter PC Port configuration interface;
2. In the PC Port configuration interface, press ▲ or ▼ soft key to select Bridge mode or Router mode;
3. Configured Bridge mode, there will pop-up “Reboot Phone”；Press ✔ key to reboot;（PS：Setting will take effect after reboot）
4. If cancel the reboot, the Settings will be saved but not take effect;
5. Configured Router mode, enter router setting interface, in the values in the corresponding position;
6. Press ✔ key after configuration, the phone will reboot.

3.12.3 VLAN Port

LAN Port

1. In the VLAN Port interface, press ▲ or ▼ soft key to select LAN Port, press ✔ soft key to enter LAN Port.
2. In the LAN Port interface, press ▼ soft key to configure the functionality Enable,

VID, Priority.

3. When the VID is not empty， press ✔ softkey to save it.

PC Port

1. In the VLAN Port interface, press ▲ or ▼ soft key to select PC Port, press ✔ softkey to enter PC Port.

2. In the PC Port interface, press ▼ soft key to configure the functionality Enable, VID, Priority.

3. When the VID is not empty， press ✔ softkey to save it.

3.13 Webserver

In the Advanced Setting interface, press ▲ or ▼ soft key to select “WebServer,” press ◀▶ softkey to access the disable/enable WebServer settings.

3.14 Reset to factory

In the Advanced Setting interface, press ▲ or ▼ soft key on the phone keyboard to select “Reset to factory”. Press ✔ soft key to access the reset to factory interface.

3.15 Password setting

To setup new password of advanced setting in phone interface. The default password is admin. Users can configure the new password and confirm it again. Then click ✔ soft key to save it.

3.16 Reboot

This is a function to reboot the phone.

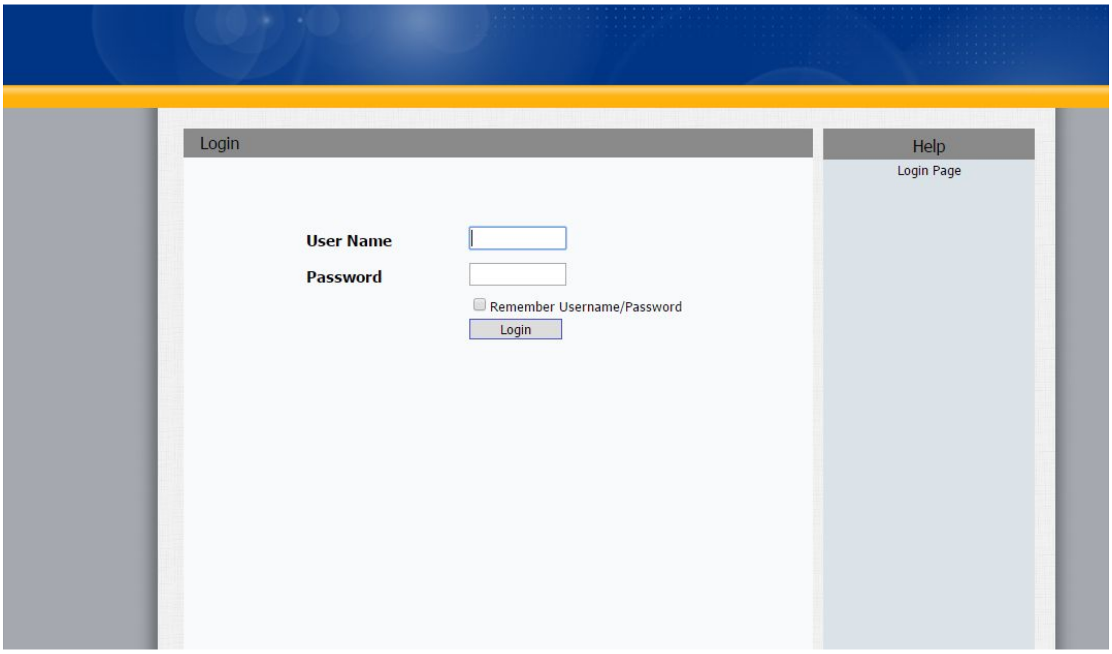
1. In the Advanced Setting interface, press ▲ or ▼ soft key to select Reboot;

2. Press ✔ soft key to reboot the phone.

4 Web interface

Web user interface (we will used Web UI for short in the following context) which is used for user or administration to check or change the IP SIP phone’s settings.

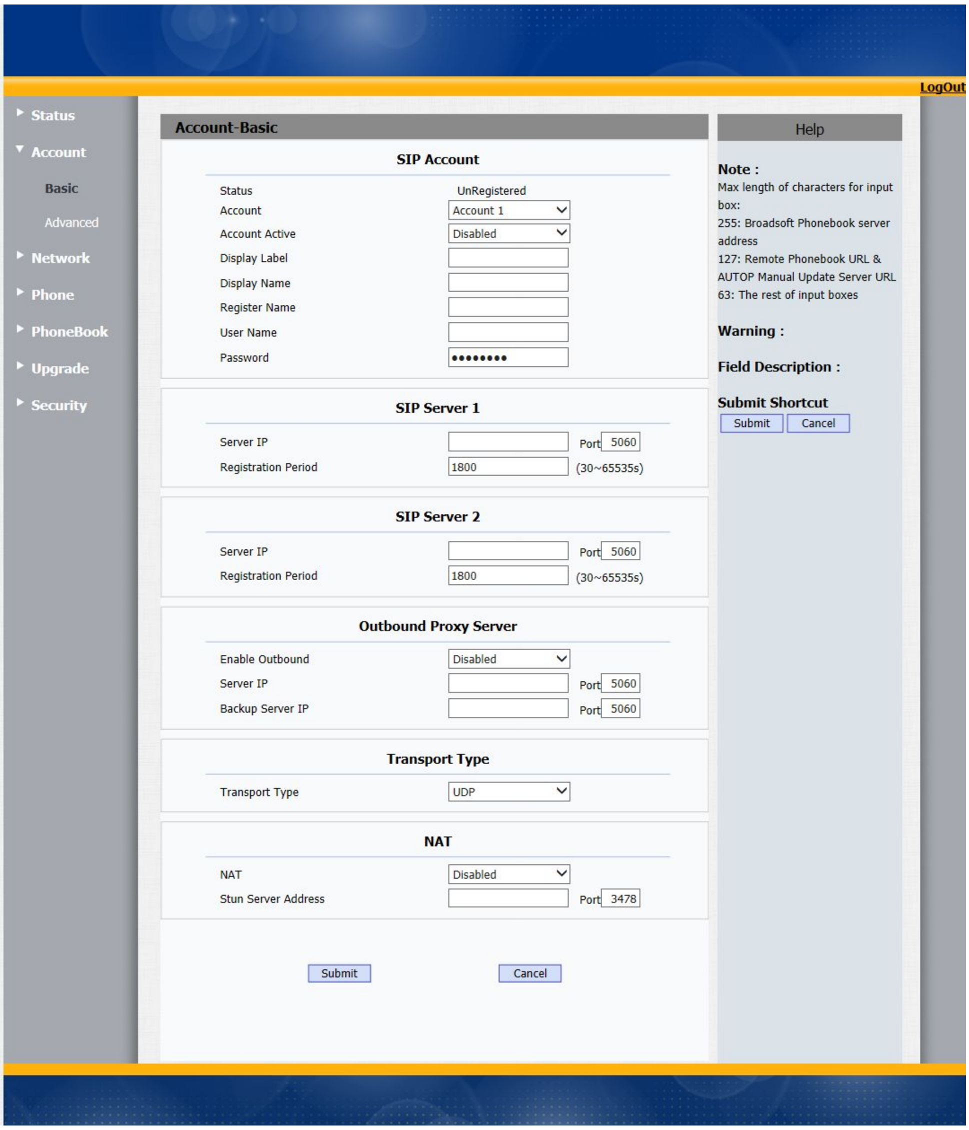
Enter the Phone IP into the web address bar, then input user name and password to login in.(user name/password: admin/admin by default)



4.1 Static-> Basic

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).

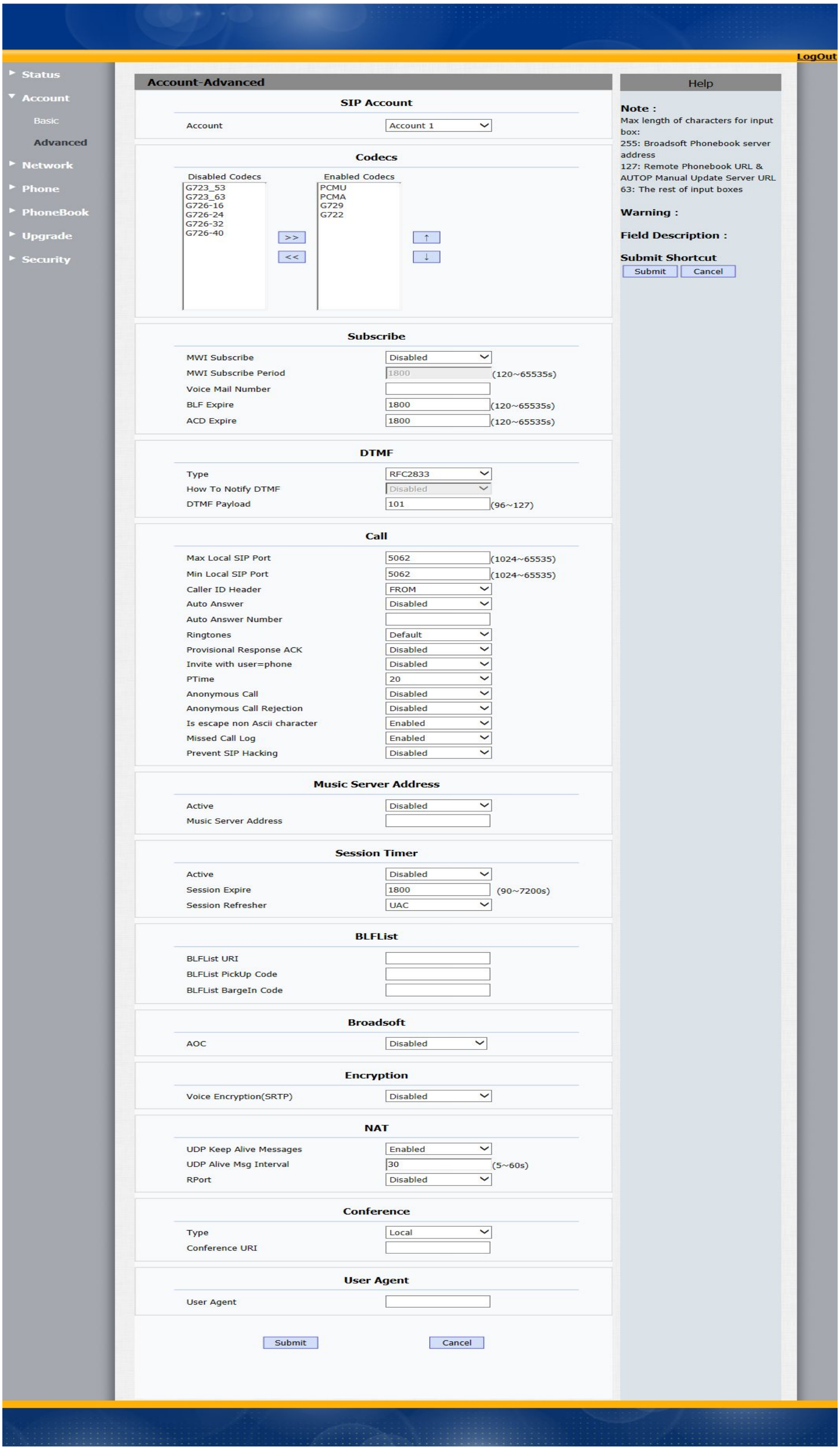
4.2 Account-> Basic



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.
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 Account-> Advanced

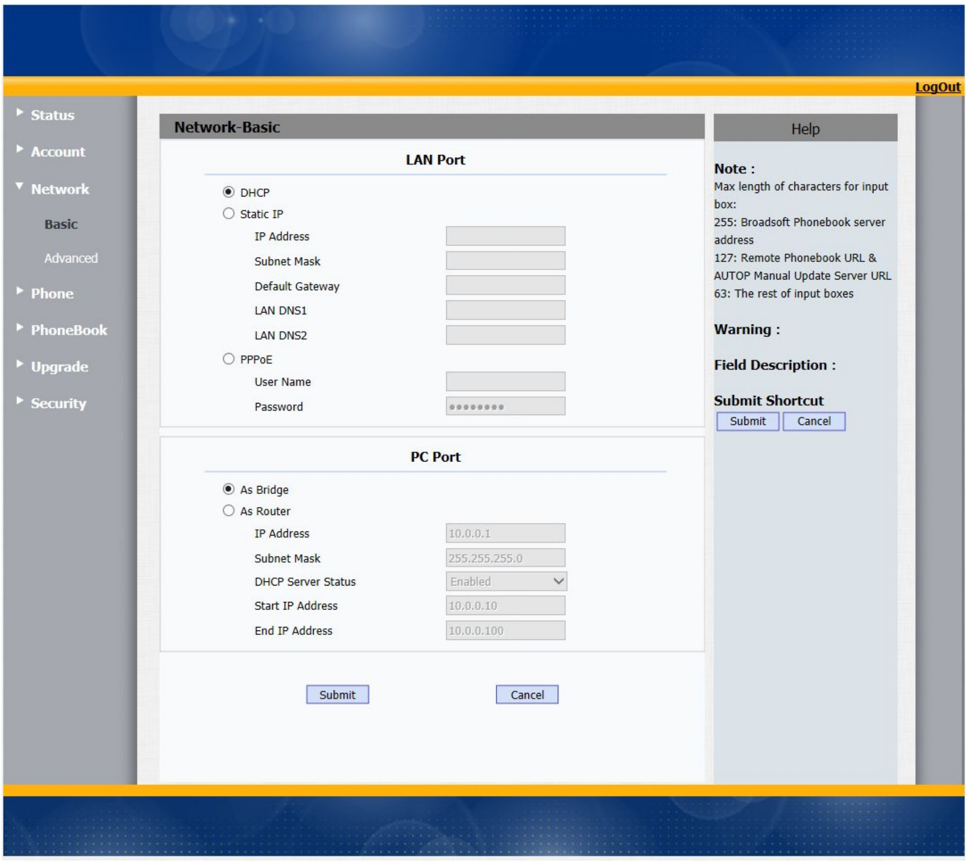

LogOut
Status
Account
Basic
Advanced
Network
Phone
PhoneBook
Upgrade
Security
Account-Advanced
SIP Account
Account
Account 1
Codecs
Disabled Codecs
G723_53
G723_63
G726-16
G726-24
G726-32
G726-40
Enabled Codecs
PCMU
PCMA
G729
G722
Subscribe
MWI Subscribe
Disabled
MWI Subscribe Period
1800
(120~65535s)
Voice Mail Number
BLF Expire
1800
(120~65535s)
ACD Expire
1800
(120~65535s)
DTMF
Type
RFC2833
How To Notify DTMF
Disabled
DTMF Payload
101
(96~127)
Call
Max Local SIP Port
5062
(1024~65535)
Min Local SIP Port
5062
(1024~65535)
Caller ID Header
FROM
Auto Answer
Disabled
Auto Answer Number
Ringtones
Default
Provisional Response ACK
Disabled
Invite with user=phone
Disabled
PTime
20
Anonymous Call
Disabled
Anonymous Call Rejection
Disabled
Is escape non Ascii character
Enabled
Missed Call Log
Enabled
Prevent SIP Hacking
Disabled
Music Server Address
Active
Disabled
Music Server Address
Session Timer
Active
Disabled
Session Expire
1800
(90~7200s)
Session Refresher
UAC
BLFList
BLFList URI
BLFList PickUp Code
BLFList BargeIn Code
Broadsoft
AOC
Disabled
Encryption
Voice Encryption(SRTP)
Disabled
NAT
UDP Keep Alive Messages
Enabled
UDP Alive Msg Interval
30
(5~60s)
RPort
Disabled
Conference
Type
Local
Conference URI
User Agent
User Agent
Submit
Cancel
Help
Note :
Max length of characters for input box:
255: Broadsoft Phonebook server address
127: Remote Phonebook URL & AUTOP Manual Update Server URL
63: The rest of input boxes
Warning :
Field Description :
Submit Shortcut
Submit
Cancel

Sections	Description
SIP Account	To display current Account settings or to select which account to display.
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), G723,G726,G729 and so on.
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.
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 use 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 anonym-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.
Music Server Address	To display or configure third-party MOH (music-on-hold) server. ● Active: To enable or disable this MOH server, If enabled, the IP phone will play MOH from configured server. ● Music Server Address: To configure MOH server address.
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.
Broadsoft	To display or configure Broadsoft AOC feature. ● AOC: A feature used to be accounting on Broadsoft platform. **Note**: Please consult your administrator further information.
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.

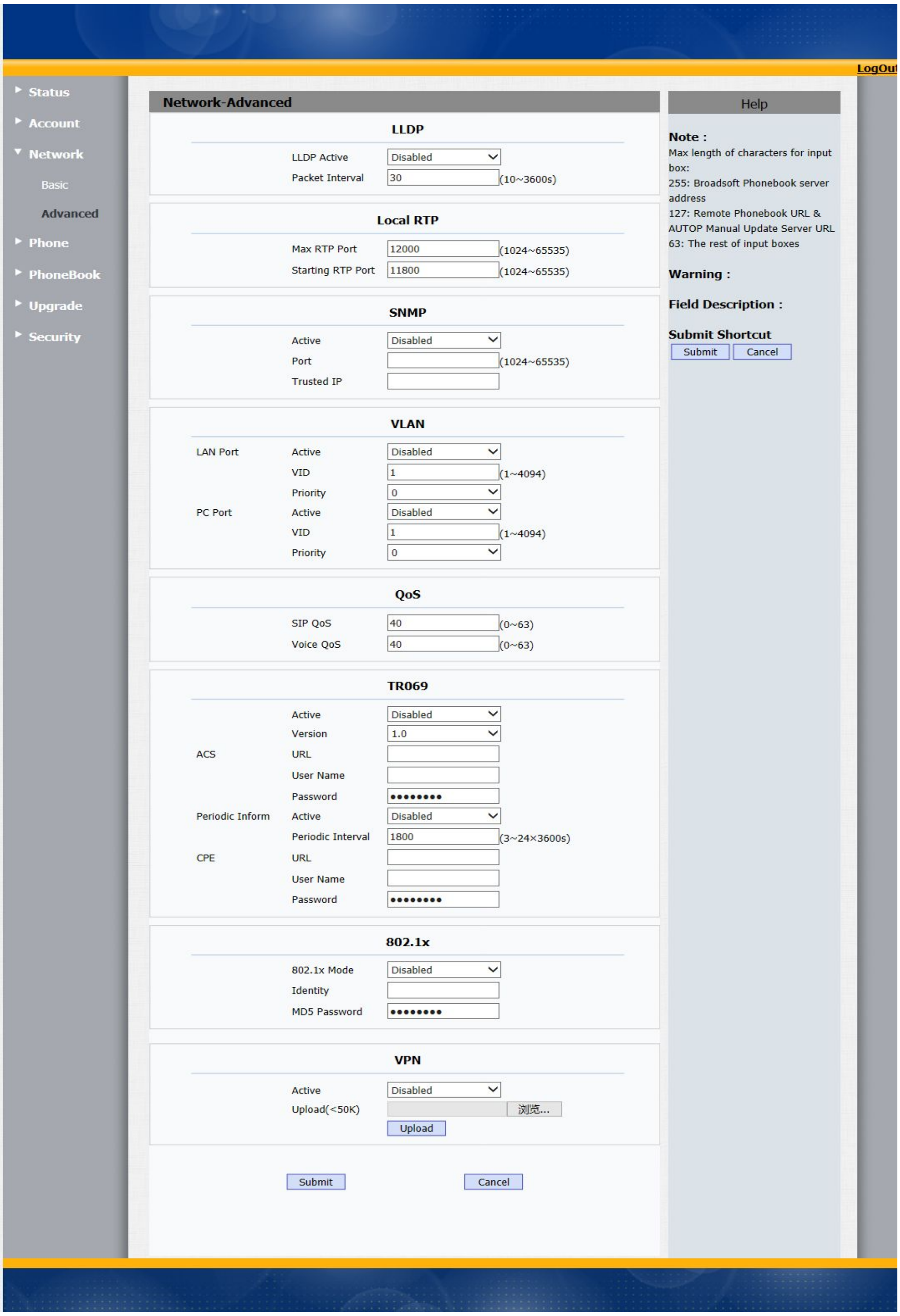
	● UDP Keep Alive message: If enabled, IP phone will send UDP keep-alive message periodically to router to keep NAT port alive. ● UDP Alive Msg Interval: Keepalive message interval. ● Rport: Remote Port, if enabled, it will add Remote Port into outgoing SIP message for designated account.
Conference	To 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-> 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.
PC Port	To display and configure PC Port settings. ● As Bridge: If selected, IP phone will act as a switch to route all incoming and outgoing packets from PC port. ● As Router: If selected, IP phone will act as a router to route all incoming and outgoing packets from PC port.

4.5 Network-> Advanced

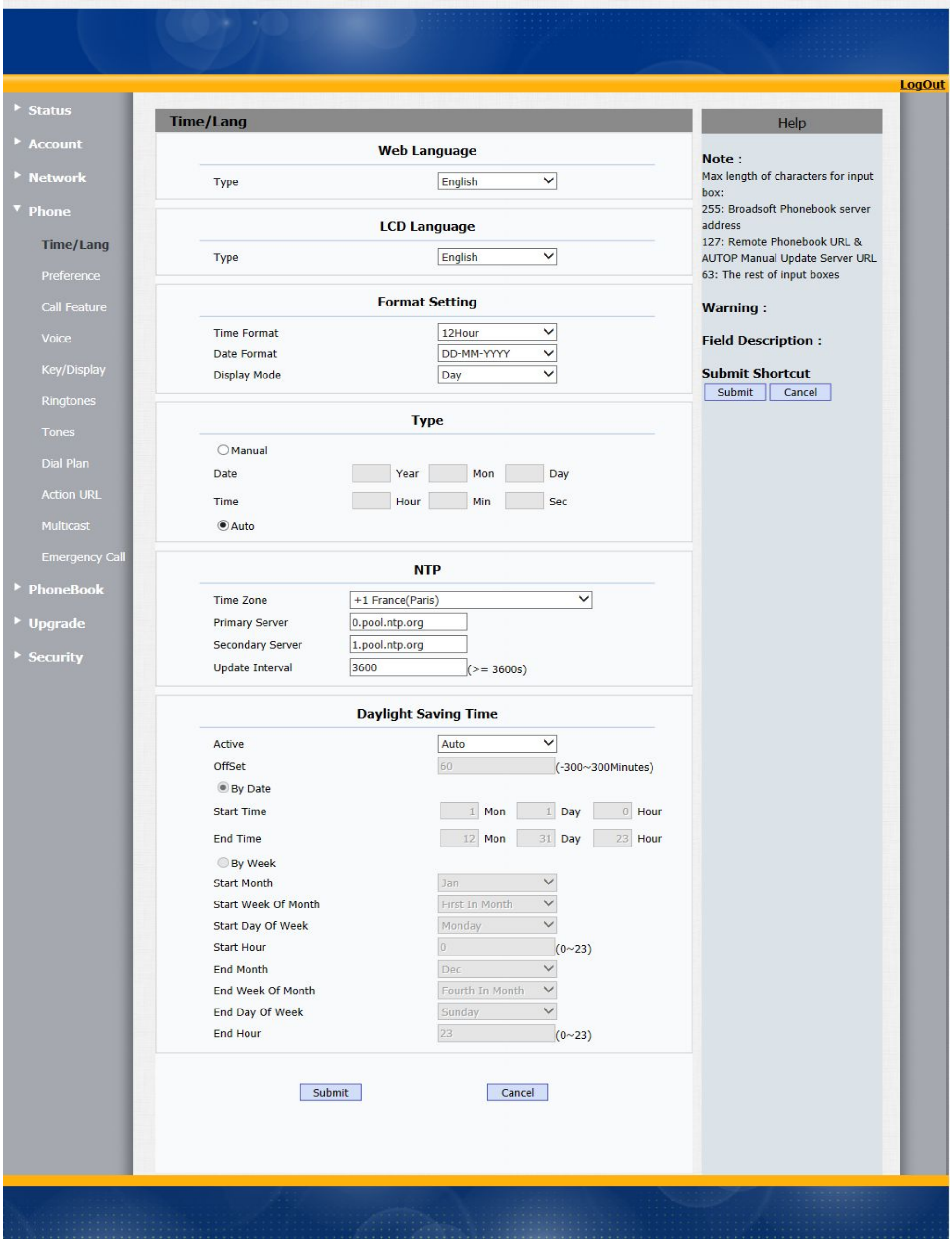


Sections	Description
LLDP	To display and configure LLDP settings. ● LLDP Active: To enable or disable LLDP feature. ● Packet interval: To configure the interval for LLDP admin message.

	Note: LLDP stands for Link Layer Discovery Protocol, it's used to exchange device information between any two directly-connected devices. LLDP is often used to configure Voice Vlan automatically for IP phone.
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.
SNMP	To display and configure SNMP settings. ● Active: To enable or disable SNMP feature. ● Port: To configure SNMP server's port. ● Trusted IP: To configure allowed SNMP server address, it could be an IP address or any valid URL domain name. **Note**: SNMP (Simple Network Management Protocols) is Internet-standard protocol for managing devices on IP networks.
VLAN	To display and configure VLAN settings. ● LAN Port/PC Port: You can configure VLAN setting for both ports respectively. ● Active: To enable or disable VLAN feature for designated port. ● Vid: To configure VLAN id for designated port. ● Priority: To select VLAN priority for designated port. **Note**: Please consult your administrator for specific VLAN settings in your networking environment.
QoS	To display and configure QoS settings. ● SIP QoS: To configure QoS value for all SIP message. ● Voice QoS: To configure QoS value for all audio stream(RTP streams).
TR069	To display and configure TR069 settings. ● Active: To enable or disable TR069 feature. ● Version: To select supported TR069 version (version 1.0 or 1.1). ● ACS/CPE: ACS is short for Auto configuration servers as server side, CPE is short for Customer-premise equipment as client side devices. ● URL: To configure URL address for ACS or CPE. ● User name: To configure username for ACS or CPE. ● Password: To configure Password for ACS or CPE. ● Periodic Inform: To enable periodically inform. ● Periodic Interval: To configure interval for periodic inform.

	Note: TR-069(Technical Report 069) is a technical specification entitled CPE WAN Management Protocol (CWMP).It defines an application layer protocol for remote management of end-user devices.
802.1x	To display and configure 802.1x settings. ● 802.1x Mode: To enable or disable 802.1x. ● Identity: To input identity if 802.1x is enabled. ● MD5 password: To input MD5 password if 802.1 is enabled.
VPN	To display and configure VPN settings. ● Active: To enable or disable VPN feature. ● Upload: To upload VPN client configuration file which is used to connect to VPN server. **Note**: For now, IP phone can only support OpenVPN.

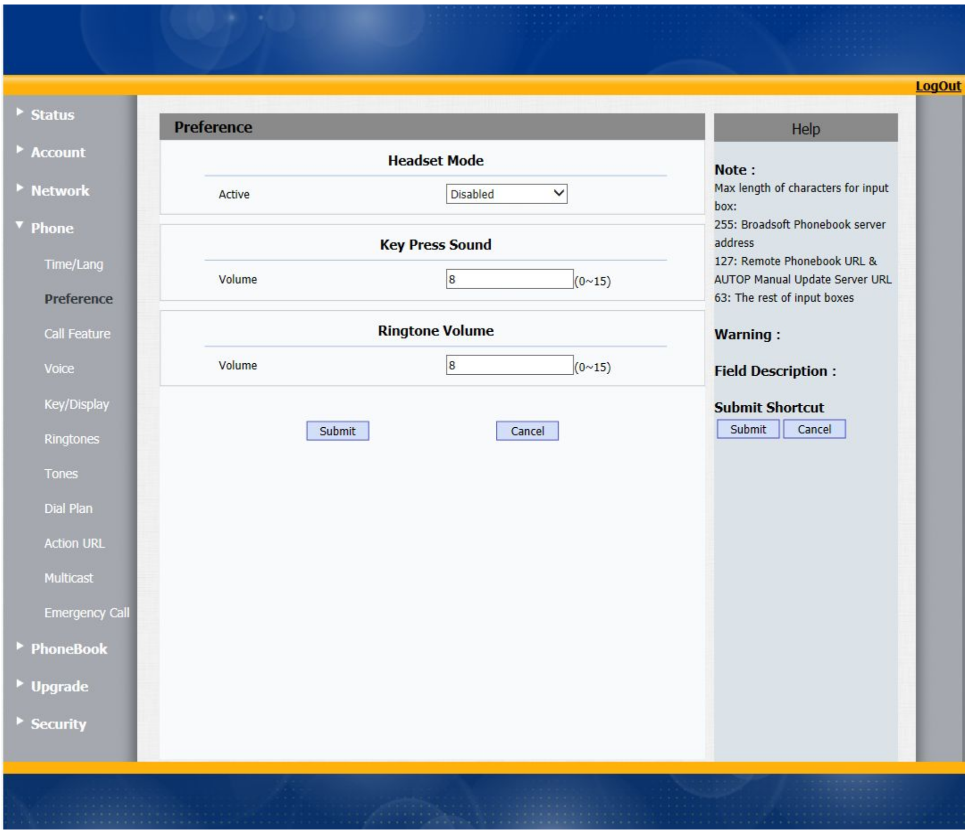
4.6 Phone-> Time/Language



Sections	Description
Web Language	To choose the web language.
LCD Language	To choose the phone language.
Format Setting	To configure time display settings. ● Time Format: Determine what format to display on Phone UI(12 hour/24 hour). ● Date Format: Determine what format to display on Phone UI for Date. ● Display Mode: Determine what mode to display Time&Date on Phone UI.
Type	To select how to configure time, it could be set by

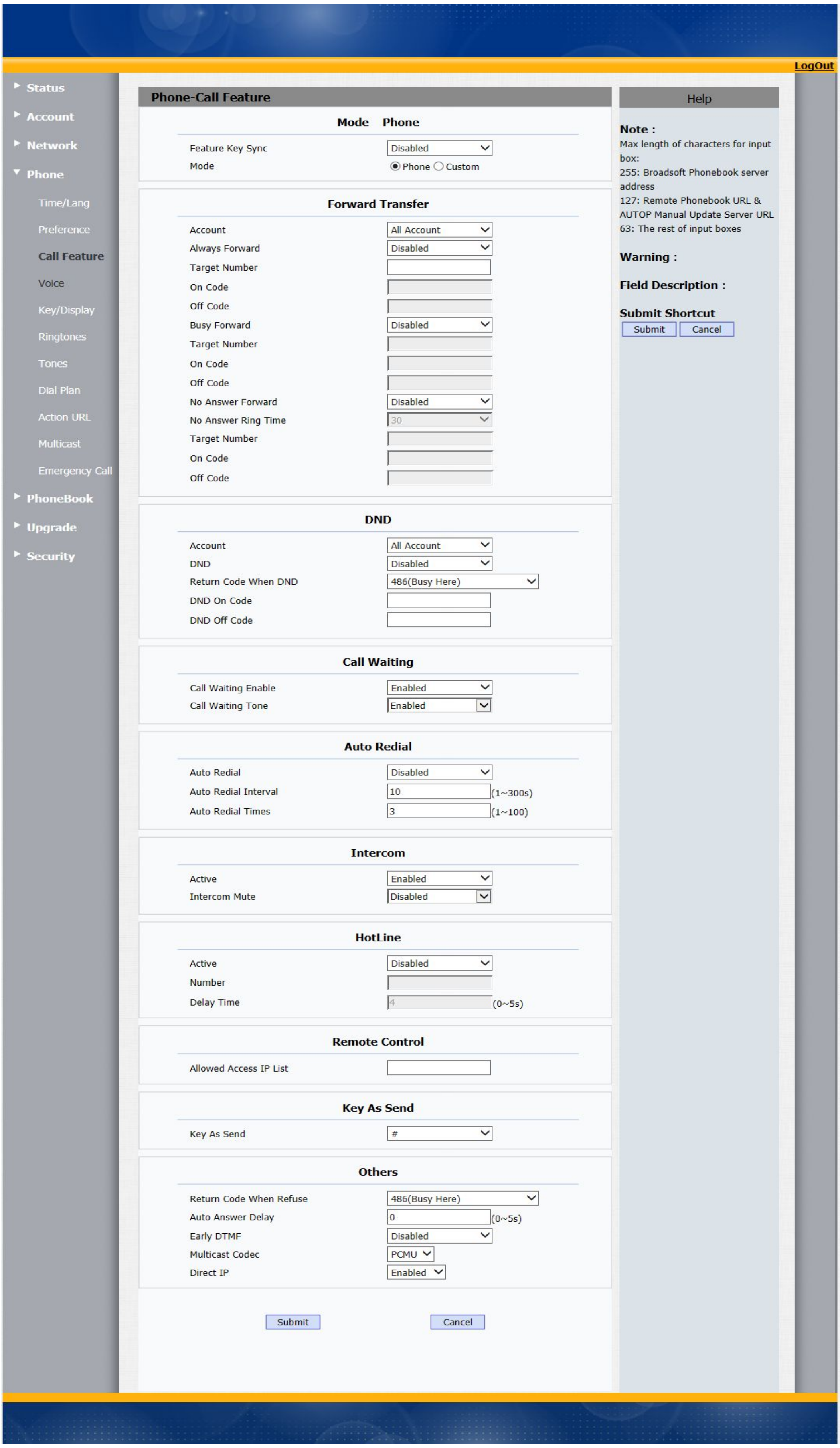
	manually or get from Internet automatically via NTP server. ● Manual: To set Time and Date manually. ● Auto: To get Time via NTP server. **Note**: If you set time to be Manually, it only takes effect till next reboot, after reboot, the phone will switch to Auto mode automatically, because there is no way for IP phone to record time during power off.
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.
Daylight Saving Time	To display or configure DST settings. **Note**: Here DST, is short for Daylight saving time, which stands for the time in the summer days when sun rises early will be adjusted forward to save daylight. The DST will take effects during the period that set by user. (all the settings for DST are all self-explanatory, please consult with your administrator for local DST details).

4.7 Phone-> Preference



Sections	Description
Headset Mode	To enable or disable Headset Mode. ● Active: If enabled, the default audio track will be headset mode, if audio track is changed during a call, it will be back to headset mode after you hangup the call.
Key Press Sound	To configure the sound volume for key press. ● Volume: The valid volume range is from 0~15,by default it's 8.
Ringtone Volume	To configure the sound volume for ringtone. ● Volume: The valid volume range is from 0~15,by default it's 8.

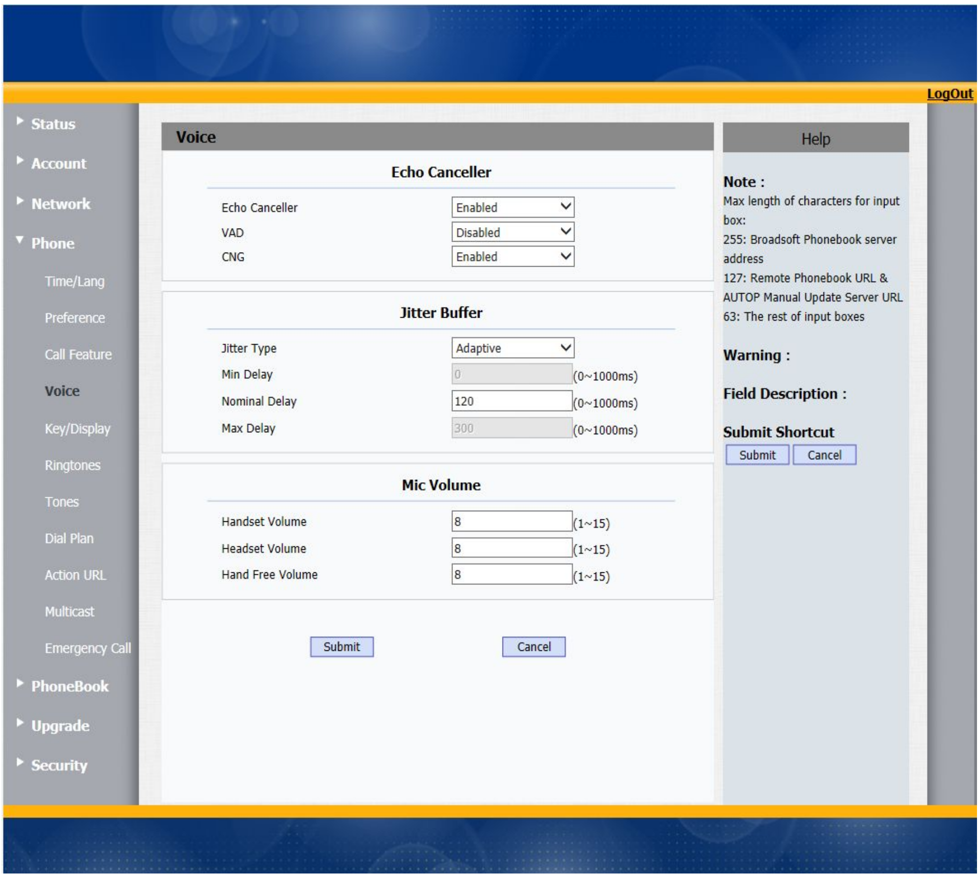
4.8 Phone-> Call Feature


LogOut
Status
Account
Network
Phone
Time/Lang
Preference
Call Feature
Voice
Key/Display
Ringtones
Tones
Dial Plan
Action URL
Multicast
Emergency Call
PhoneBook
Upgrade
Security
Phone-Call Feature
Mode Phone
Feature Key Sync
Disabled
Mode
Phone Custom
Forward Transfer
Account
All Account
Always Forward
Disabled
Target Number
On Code
Off Code
Busy Forward
Disabled
Target Number
On Code
Off Code
No Answer Forward
Disabled
No Answer Ring Time
30
Target Number
On Code
Off Code
DND
Account
All Account
DND
Disabled
Return Code When DND
486(Busy Here)
DND On Code
DND Off Code
Call Waiting
Call Waiting Enable
Enabled
Call Waiting Tone
Enabled
Auto Redial
Auto Redial
Disabled
Auto Redial Interval
10
(1~300s)
Auto Redial Times
3
(1~100)
Intercom
Active
Enabled
Intercom Mute
Disabled
HotLine
Active
Disabled
Number
Delay Time
4
(0~5s)
Remote Control
Allowed Access IP List
Key As Send
Key As Send
#
Others
Return Code When Refuse
486(Busy Here)
Auto Answer Delay
0
(0~5s)
Early DTMF
Disabled
Multicast Codec
PCMU
Direct IP
Enabled
Submit
Cancel
Help
Note :
Max length of characters for input box:
255: Broadsoft Phonebook server address
127: Remote Phonebook URL & AUTOP Manual Update Server URL
63: The rest of input boxes
Warning :
Field Description :
Submit Shortcut
Submit
Cancel

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: An incoming call will be forwarded if IP phone is busy. ● No answer Forward: An incoming call will be forwarded if it's no answer after a specific time.
Call Waiting	To enable or disable Call Waiting. ● Call Waiting Enable: If enabled, it allows IP phones to receive a new incoming call when there is already an active call. ● Call Waiting Tone: If enabled, it allows IP phones to play the call waiting tone to the waiting callee.
Auto Redial	Auto redial allows IP phones to redial an unsuccessful call for designated times within designated interval. ● Auto Redial: To enable or disable auto redial feature. ● Auto Redial Interval: Determine the interval between two consecutive attempts. ● Auto Redial Times: Determine how many times to redial.
DND	DND (Do Not Disturb) allows IP phones to ignore any incoming calls. ● Return Code when DND: Determine what response 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.
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.
HotLine	HotLine allows user to call out a defined number automatically after hearing the dial tone without dialing any number. ● Active: To enable or disable HotLine feature. ● Number: To set a defined HotLine number. ● Delay Time: To set the automatically call out interval after hearing the dail tone.
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
Key As Send	Key As Send allows you to disable send key or assign pound key as send key.
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. ● Early DTMF: To enable or disable early DTMF

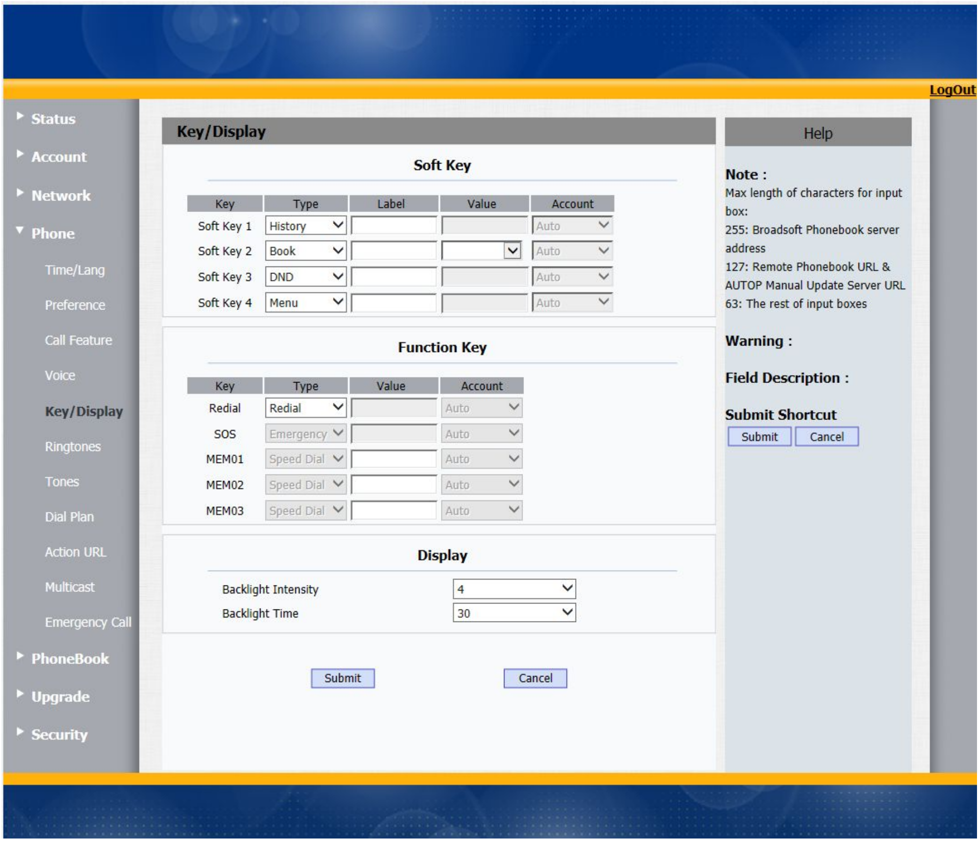
4.9 Phone-> Voice



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

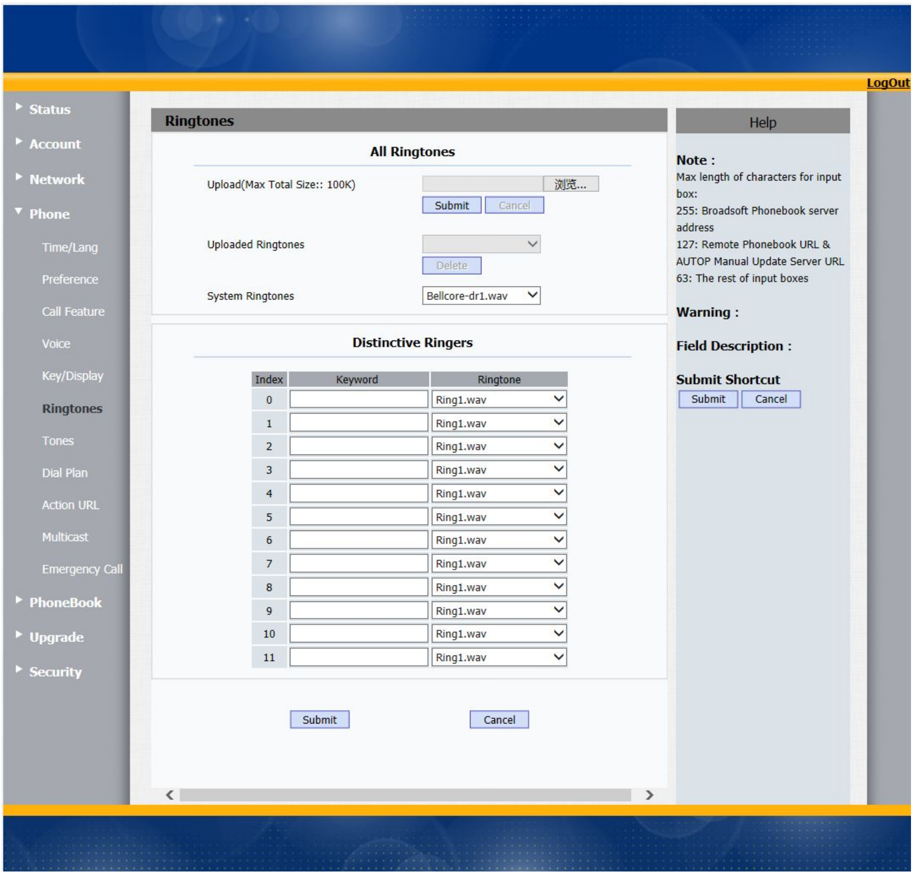
	noise until voice activity resumes. The insertion of artificial noise gives the illusion of a constant transmission stream, so that background sound is consistent throughout the call and the listener does not think the line has released.
Jitter Buffer	Jitter buffer is a shared data area where voice packets can be collected, stored, and sent to the voice processor in even intervals. Jitter is a term indicating variations in packet arrival time,which can occur because of network congestion, timing drift or route changes. The jitter buffer, located at the receiving end of the voice connection, intentionally delays the arriving packets so that the end user experiences a clear connection with very little sound distortion. IP phones support two types of jitter buffers: fixed and adaptive. Fixed: Add the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on IP phones. Adaptive: Capable of adapting the changes in the network's delay. The range of the delay time for the dynamic jitter buffer added to packets can be also configured on IP phones.
Mic Volume	To configure Microphone volume for headset, handset and speaker mode.

4.10 Phone-> Key/Display



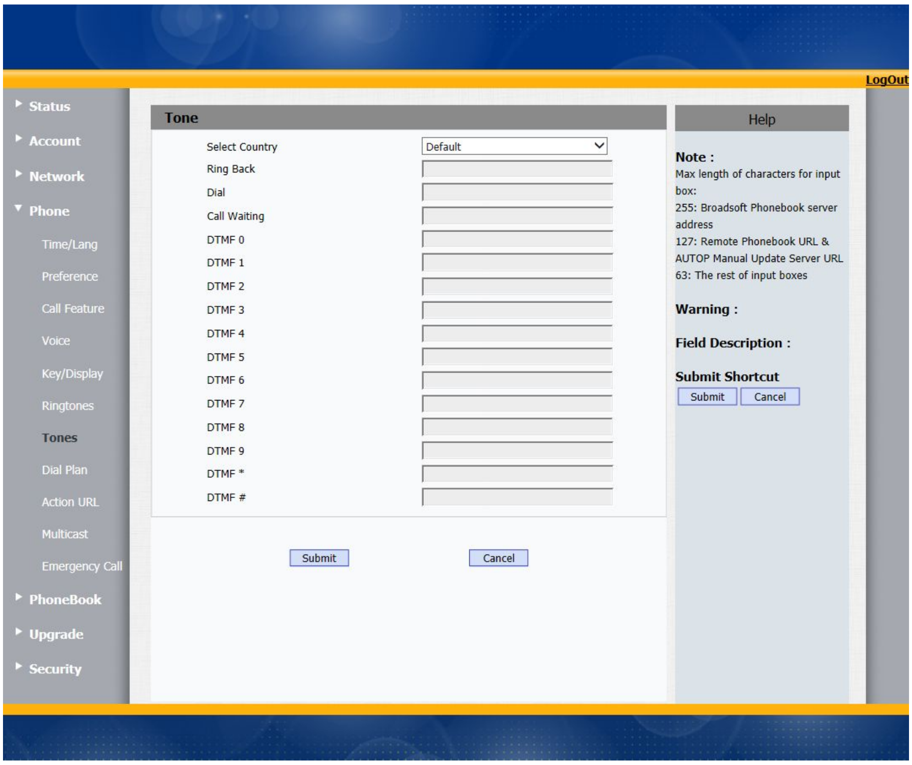
Sections	Description
Soft Key	Allows user to assign specific feature to the designated soft keys. For softkey, the available features list: DND, Menu, MSG, Status, Book, Fwd, PickUp, Group, Pickup, Intercom, Speed Dial, History, Favorites, Redial, Call Return, Hot Desking, XML Browser.
Function Key	Allows user to assign specific feature to the designated function keys. For function keys, the available features list: N/A, DND, Menu, MSG, Status, Book, Fwd, PickUp, Group PickUp, Intercom, Speed Dial, History, Favorites, Redial, Call Return, Hot Desking, XML Browser. **Note:** SOS Key only can be used as Emergency call and Memory Key can only used as speed dial key.
Display	● Backlight Intensity: To adjust the backlight intensity of Phone UI. ● Backlight Time: To adjust backlight on timer, once expired the backlight of Phone UI will go off.

4.11 Phone-> Ringtone



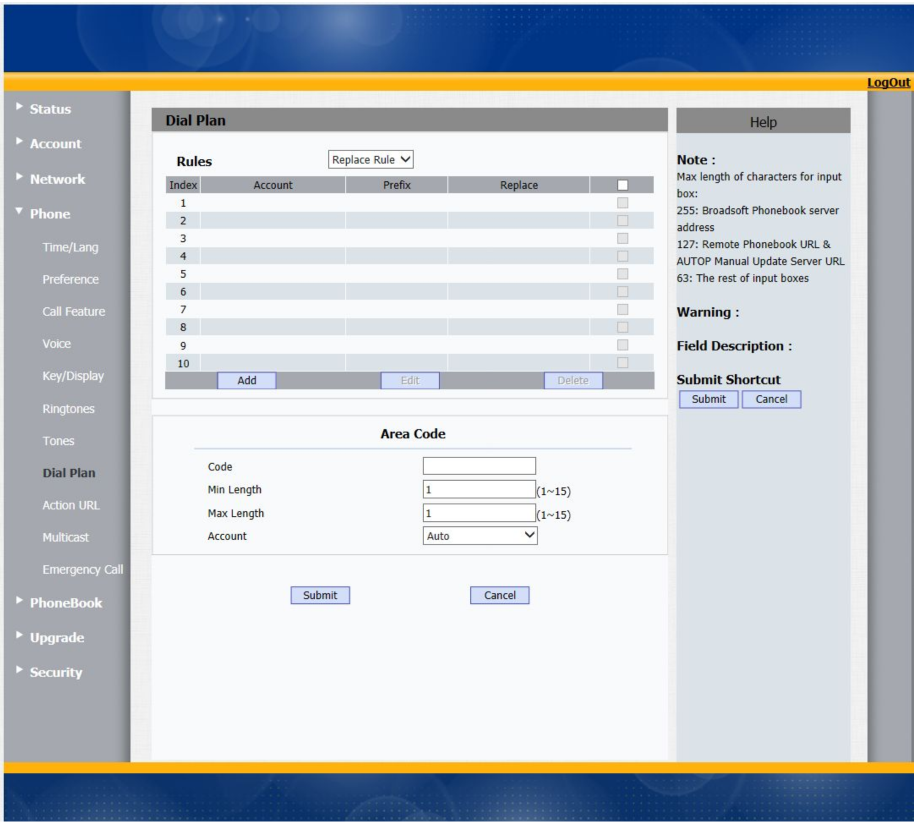
Sections	Description
All Ringtones	Allow user to upload and view ringtone files or delete uploaded ringtone files. **Note**: Ringtone files must be .wav format and has some specific requirement, please contact to Akuvox technical support team for instructions how to make ringtone files. System ringtones files cannot be deleted thus user can only delete uploaded ringtones.
Distinctive Ringers	Distinctive ringers allow different incoming calls to trigger distinctive ringtones. The IP phone will check "Alert-Info" header inside the incoming “invite” SIP message. And strip out the URL or keyword inside the “Alert-Info” header ,from the specific URL or keyword to match or download designated ringtones to play.

4.12 Phone-> Tones



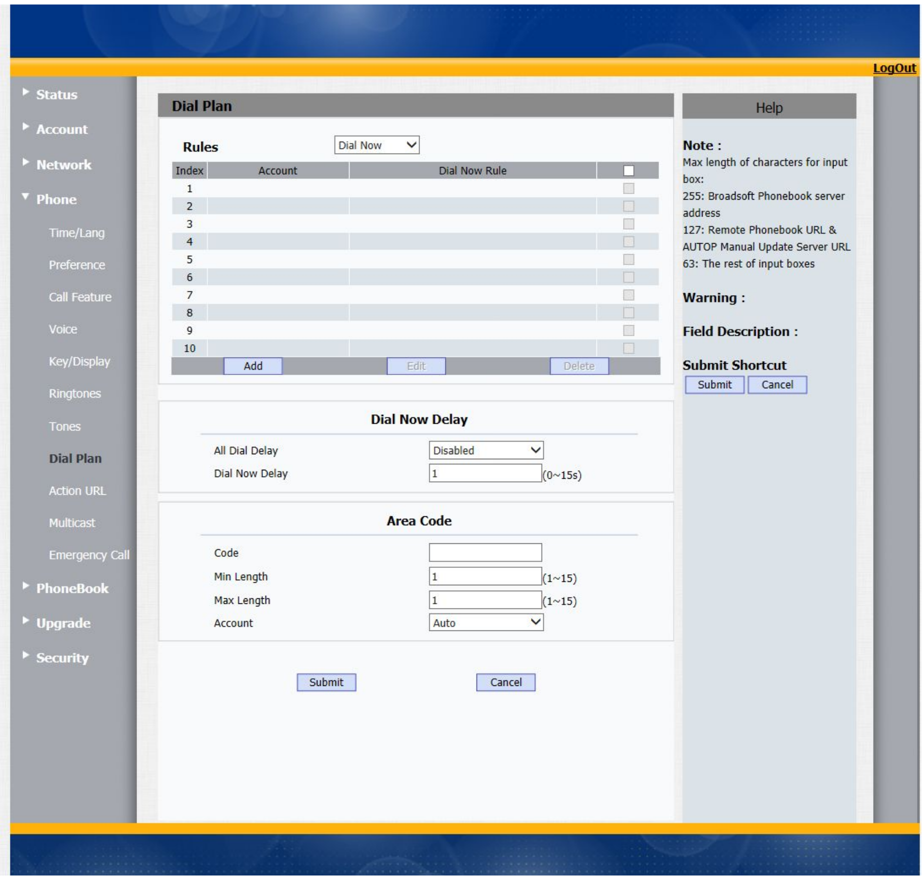
Sections	Description
Tones	Allows user to select a specialized tone sets (classified by countries) or to customize own tones. **Note**: Available country tones sets are: China,Spain,Luxembourg,Sweden,Taiwan,Belgium,Denmark,Finland,Germany,Netherlands,Norway,Portugal.

4.13 Phone-> Replace rule



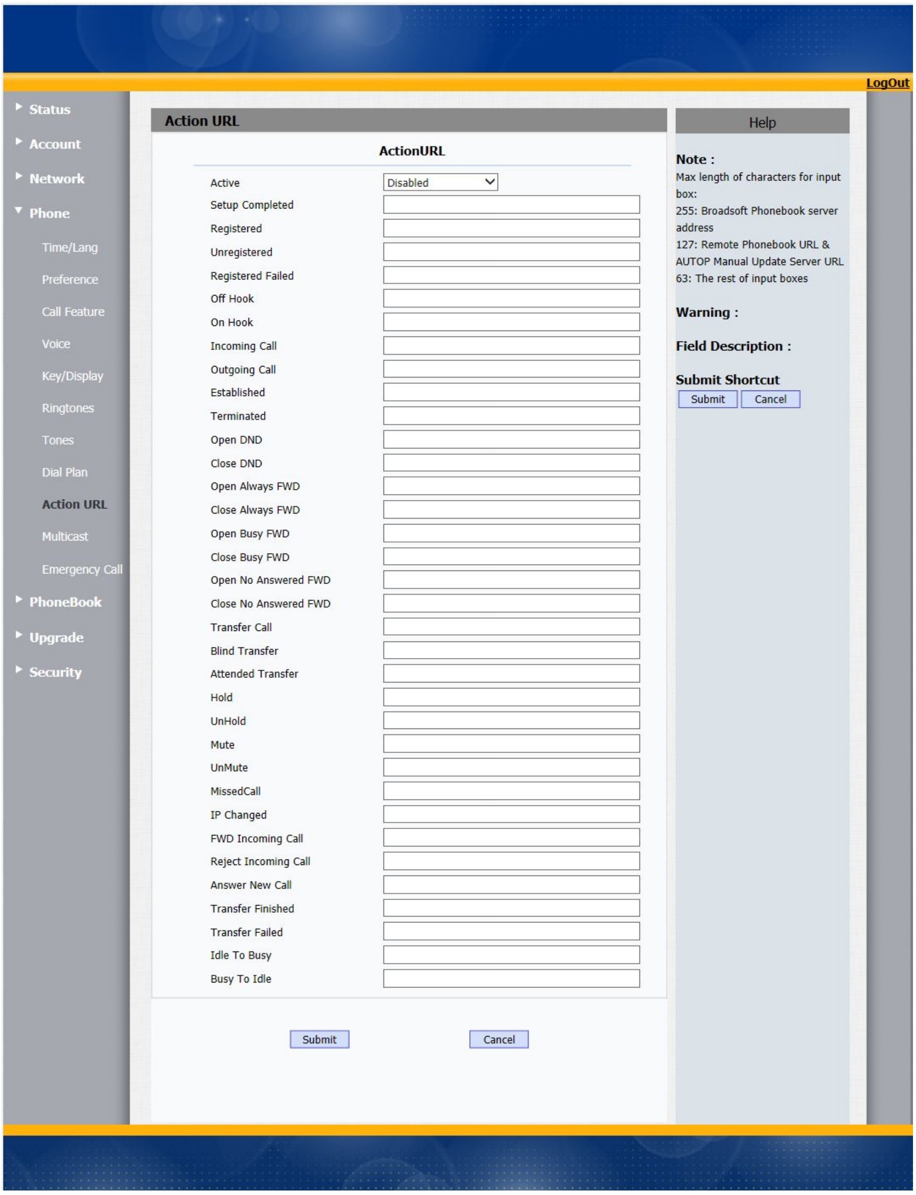
Sections	Description
Rules	Allow user to select Replace rule or Dial-now to display or edit.
Rules Modify	Allow user to modify selected rules information, for replace rule, you can modify related accounts, prefix and replace.
Area Code	Area codes are also known as NPAs (Numbering Plan Areas). They usually indicate different geographical areas within one country. If entered numbers match the predefined area code rule, the IP phone will automatically prefix outgoing number with area code. **Note**: There is only one area code rule supported.

4.14 Phone-> Dial Now



Sections	Description
Rules	Allow user to select Replace rule or Dial-now to display or edit.
Dial Now Delay	Allow user to configure dial now delay time for dial now. It means user can configure the IP phone to dial out the phone number automatically after the designated delay time if it match any dial now rule.
Rules Modify	Allow user to modify selected rules information, for dial-now rule, user can modify related accounts, Dial now Rule itself.
Area Code	Area codes are also known as NPAs(Numbering Plan Areas). They usually indicate different geographical areas within one country. If entered numbers match the predefined area code rule, the IP phone will automatically prefix outgoing number with area code. **Note**: There is only one area code rule supported.

4.15 Phone-> Action URL

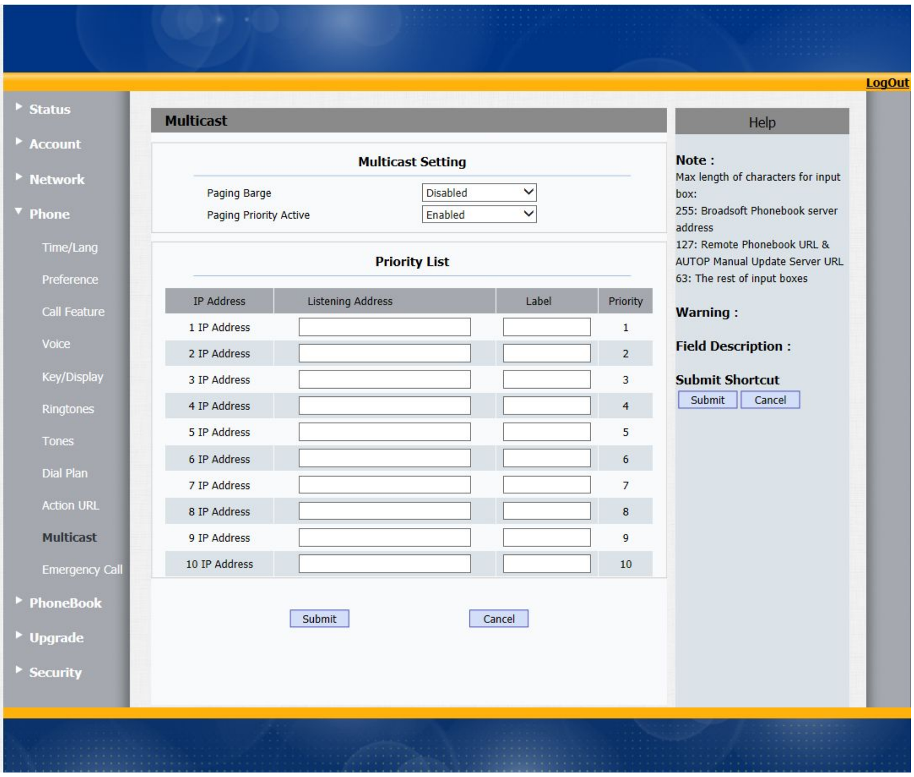


Sections	Description
Action URL	To display and configure Action URL settings. Setup Completed: When the IP phone completes startup. ● Registered: When the IP phone successfully registers an account. ● Unregistered: When the IP phone logs off the registered account. ● Register Failed: When the IP phone fails to register an account. ● Off Hook: When the IP phone is off hook. ● On Hook: When the IP phone is on hook.

	- Incoming Call: When the IP phone receives an incoming call. - Outgoing Call: When the IP phone places a call. - Established: When the IP phone establishes a call. - Terminated: When the IP phone terminates a call. - Open DND: When the IP phone enables the DND mode. - Close DND: When the IP phone disables the DND mode. - Open Always Forward: When the IP phone enables the always forward. - Close Always Forward: When the IP phone disables the always forward. - Open Busy Forward: When the IP phone enables the busy forward. - Close Busy Forward: When the IP phone disables the busy forward. - Open No Answer Forward: When the IP phone enables the no answer forward. - Close No Answer Forward: When the IP phone disables the no answer forward - Transfer Call : When the IP phone transfers a call. - Blind Transfer: When the IP phone blind transfers a call. - Attended Transfer: When the IP phone performs the semi-attended/attended transfer. - Hold: When the IP phone places a call on hold. - UnHold: When the IP phone retrieves a hold call. - Mute: When the IP phone mutes a call. - UnMute: When the IP phone un-mutes a call. - Missed Call: When the IP phone misses a call. - IP Changed: When the IP address of the IP phone changes. - FWD Incoming Call: When the IP phone forwards an incoming call. - Reject Incoming Call: When the IP phone rejects an incoming call. - Answer New Call: When the IP phone answers a new call. - Transfer Finished: When the IP phone completes to transfer a call. - Transfer Failed: When the IP phone fails to transfer a call. - Idle To Busy: When the state of the IP phone

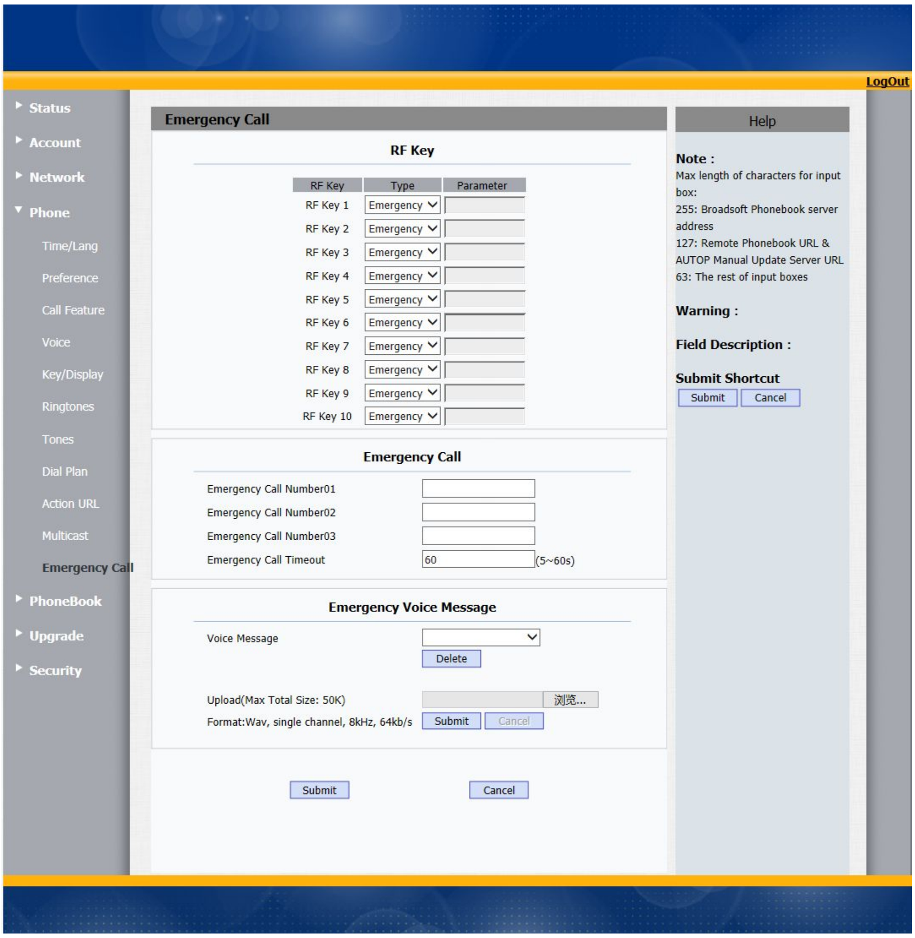
	changes from idle to busy. ● Busy To Idle: When the state of phone changes from busy to idle.

4.16 Phone-> Multicast



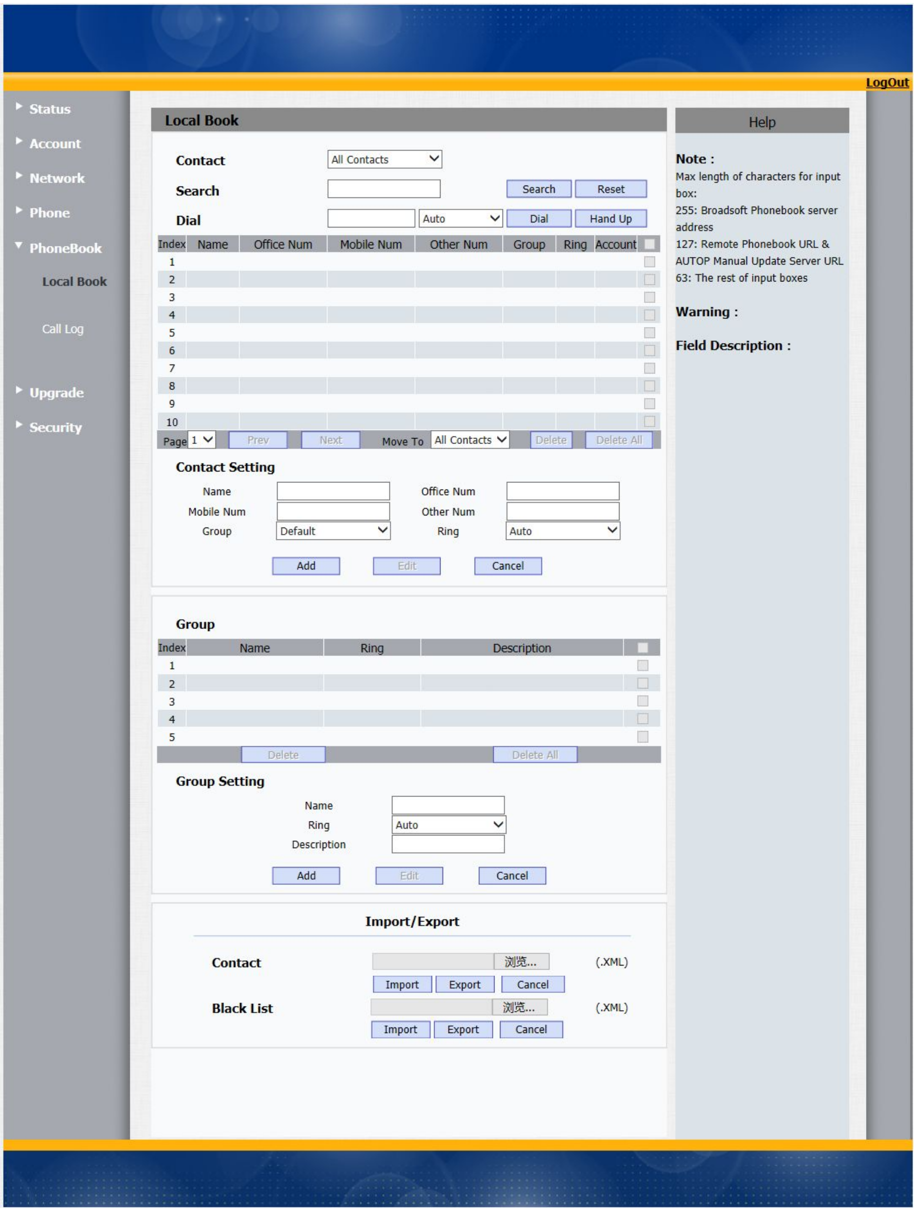
Sections	Description
Multicast Setting	To display and configure the Multicast setting. ● Paging Barge: Choose the multicast number ,the range is 1-10. ● Paging priority Active: Enable o disable the multicast.
Priority List	To setup the multicast parameters. ● Listening Address: Enter the IP address you need to listen ● Label： Input the label for each listening address

4.17 Phone-> Emergency Call



Sections	Description
RF Key(optional)	RF Key can be setup as Speed Dial and Emergency call. Enter the target number or IP address in value bar.
Emergency call	To ensure the emergency call is never lost. R15P can call out for three emergency numbers for every 60 seconds(by default) in a loop. ● Emergency Call Number: Enter phone number or IP address you need. ● Emergency Call Timeout: The range is from 5s to 60s. 60s by default.
Emergency Voice Message	When during emergency call, the callee party will hear the specified voice message every 5s. The callee party need to press 5 to ensure the emergency call is received. ● Voice Message: it will show the tone you uploaded. Users can delete it. ● Upload: Select and upload the specified music you need, click Submit to save.

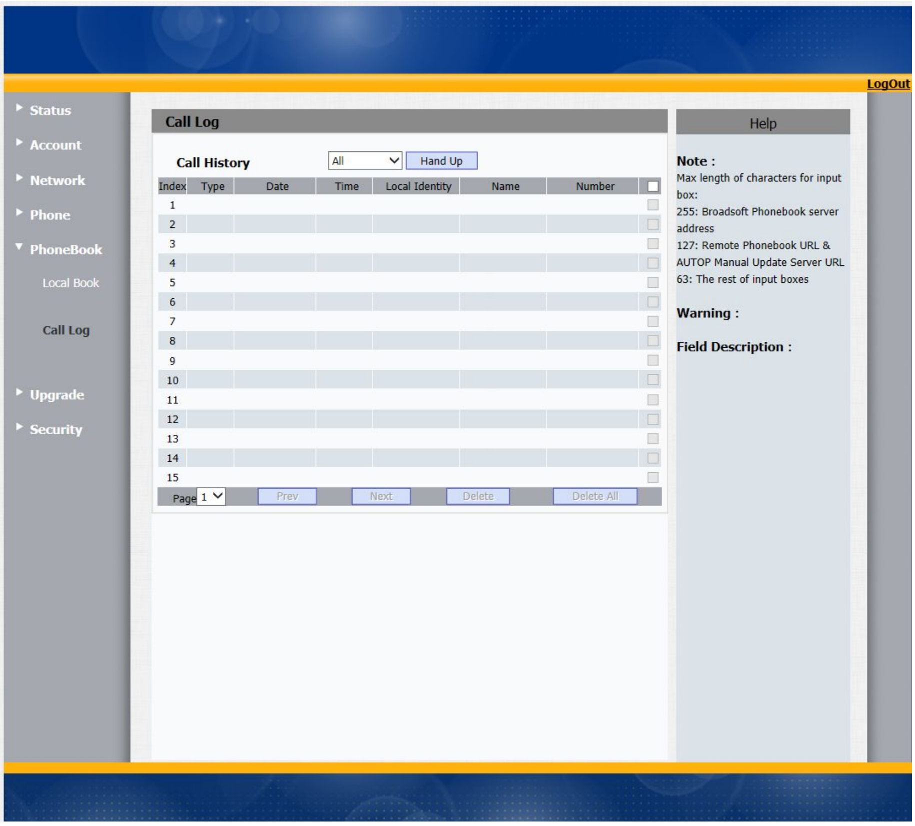
4.18 PhoneBook-> Local Phone Book



Sections	Description
Contact	To display and select local contact type. ● All Contacts: To display or edit all local contacts. ● Favorites: To display or edit favorites 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

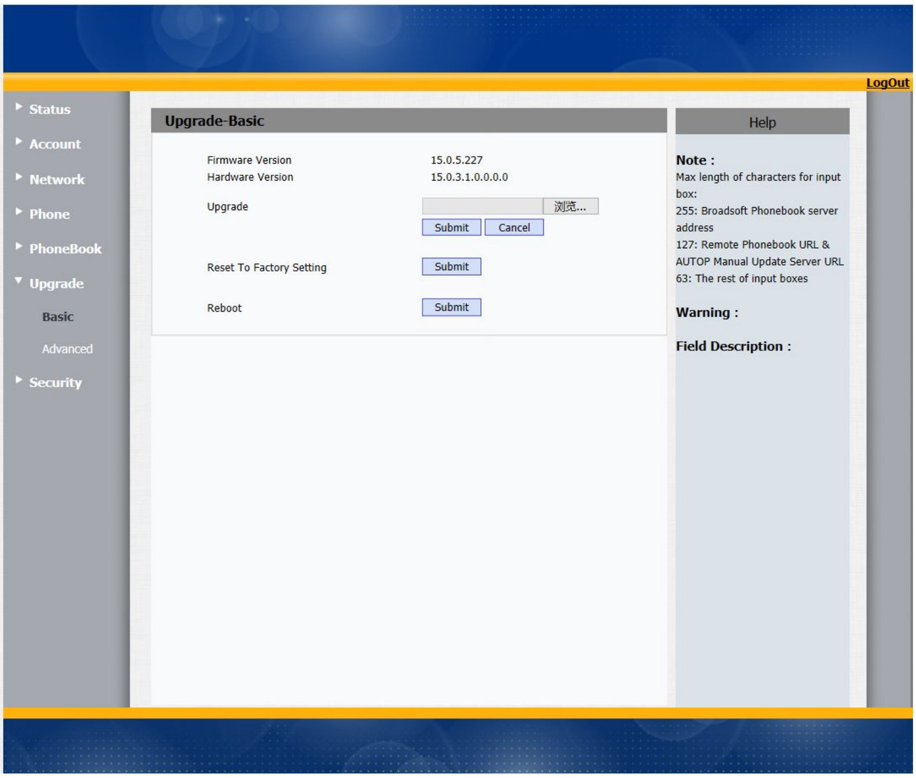
	refer to section “Remote Control” in the Web UI->Phone->Call Feature page.
Group	To display or edit Group contacts.
Contact Setting	To display or change Group name, related ringtone or description.
Import/Export	To import or export the contact or blacklist file.

4.19 Phone-> Call Log



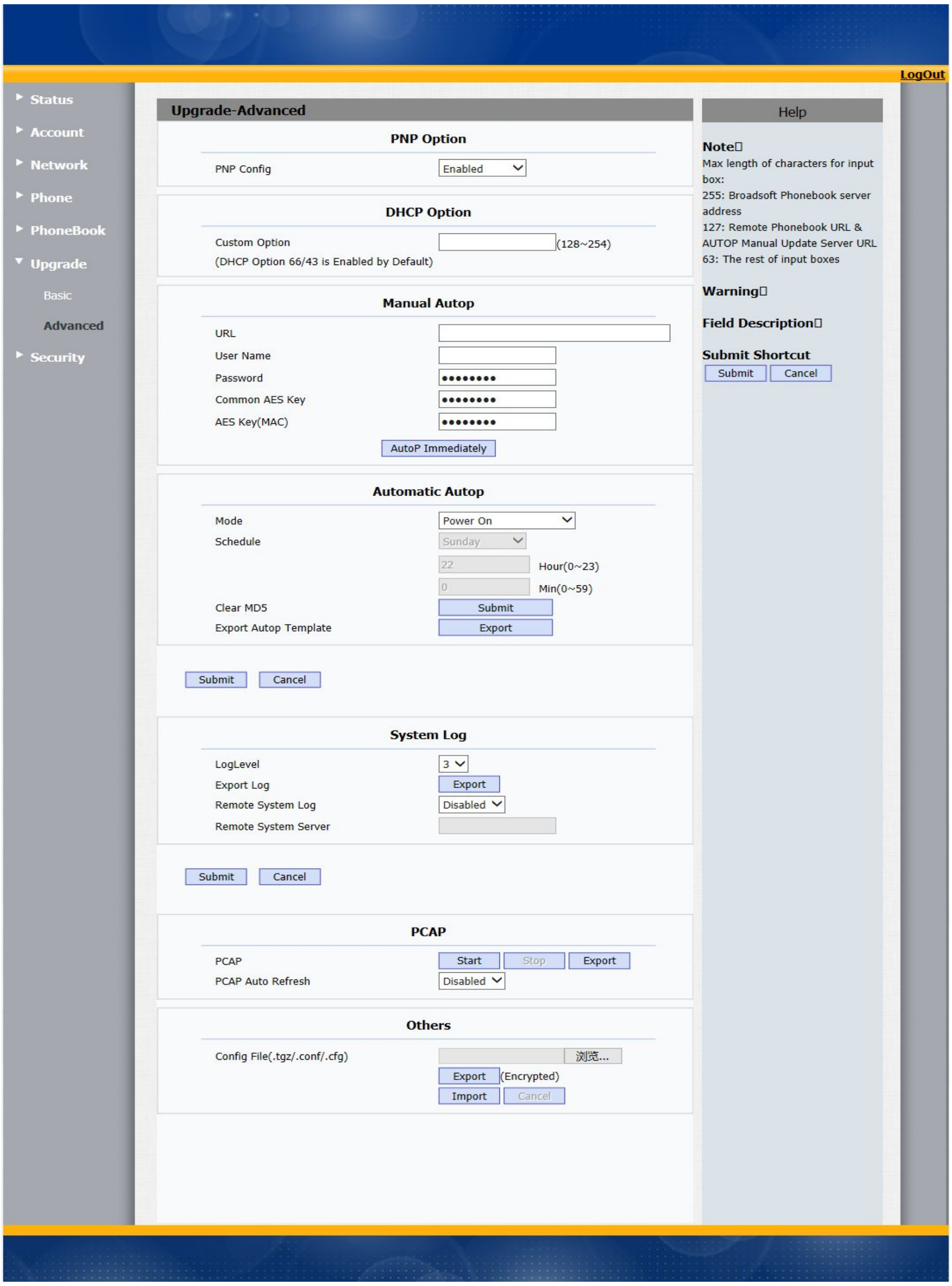
Sections	Description
Call History	To display call history records. Available call history type are All calls, Dialed calls, Received calls, Missed calls, Forwarded calls. HangUp: To click to hangup ongoing call on the IP phone.

4.20 Upgrade-> Basic



Sections	Description
Upgrade	To select upgrading rom file from local or a remote server automatically. **Note:** Please make sure it's right file format for right model.
Firmware version	To display firmware version, firmware version starts with MODEL name.
Hardware Version	To display Hardware version.
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.

4.21 Upgrade-> Advanced

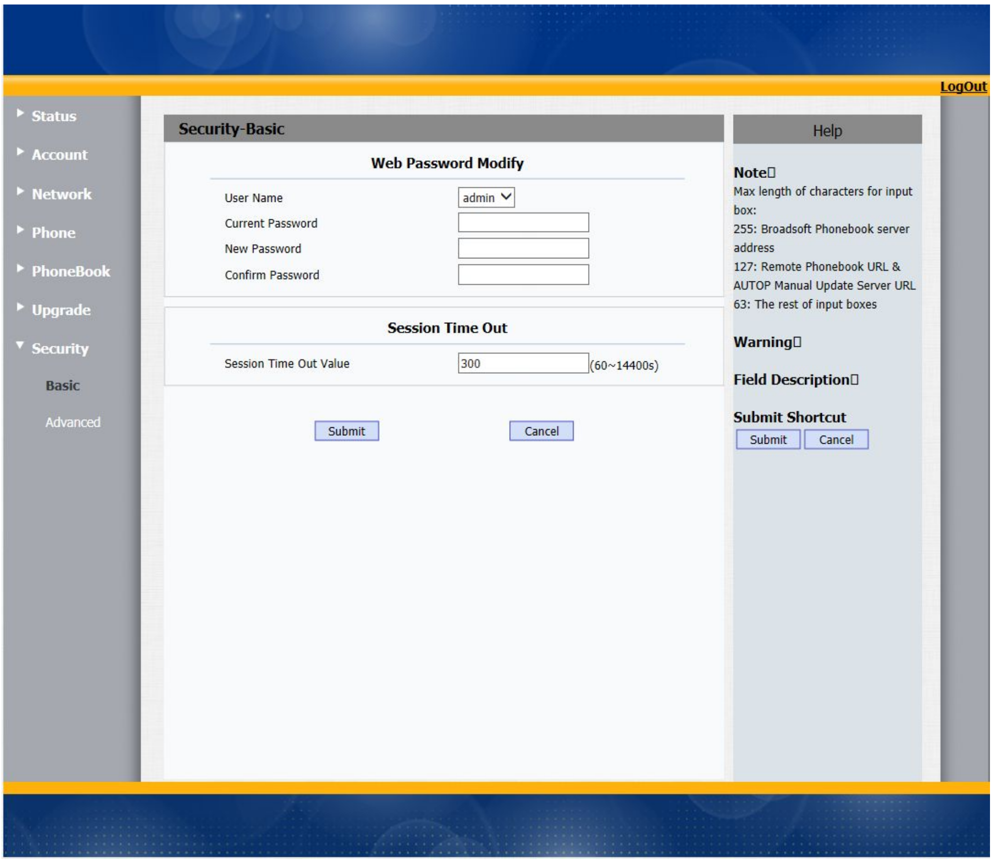


Sections	Description
PNP Option	To display and configure PNP setting for Auto Provisioning. ● PNP: Plug and Play, once PNP is enabled, the phone will send SIP subscription message to PNP server automatically to get Auto Provisioning server's address. By default, this SIP message is sent to multicast 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.
Manual Autop	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). **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.
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. **Note:** IP phone will save captured packets file to a temporary file, this file maximum size is 1M(mega

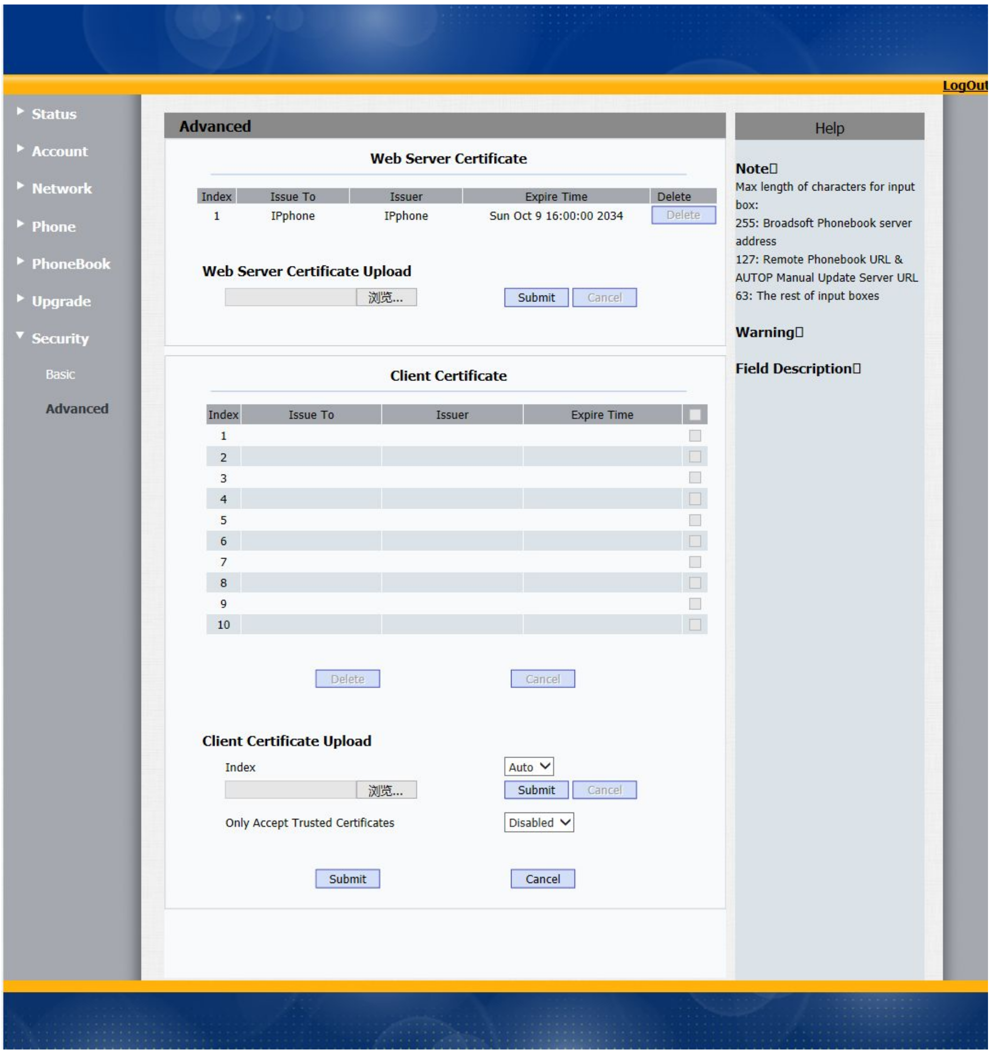
	bytes), and will top capturing once reaching this maximum size.
Others	To display or configure others features from this page. Config file: To export or import configure file for IP phone.

4.22 Security-> 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.

4.23 Security-> Advanced



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.
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.