



SP-R55P/G IP Phone User Manual

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1. Overview

1.1. Introduction



The Akuvox R55P/G is a featured 3-line IP phone with full duplex hands-free speakerphone. It can be directly connected to an Internet Telephony Service Provider or to an IP PBX.

Based on the SIP standard, the Akuvox R55P/G 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 R55P/G 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

- HD Voice
- Up to 3 SIP Accounts
- 2.3" 132x64 Graphical LCD with Backlight
- Support 3-way Conference
- Full Compatible with Asterisk, BroadSoft Platform

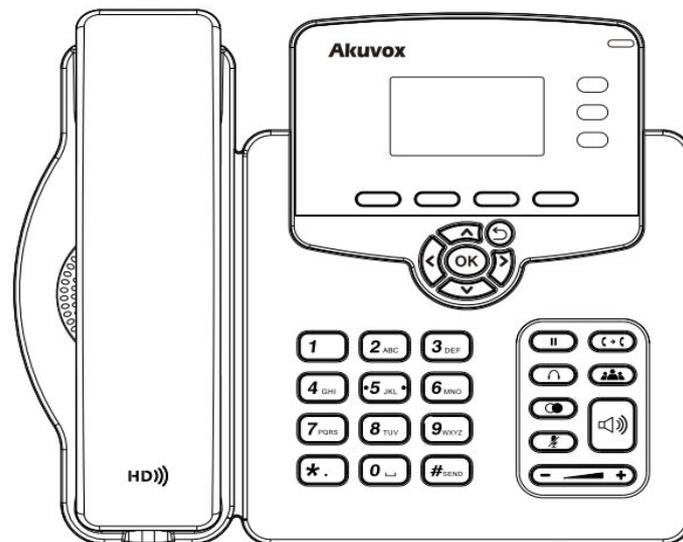
➤ Phone Features

- 3 Lines(support 3 SIP accounts)
- Support Call Waiting, Call Forwarding, Call Transfer
- Call on Hold, Mute, Auto-answer, Redial, DND
- Local 3-Way Conference
- Volume adjustable, Ring tones Selectable
- Speed Dial, Hotline
- Daylight Saving Time
- Network Packet Capture
- Country Ringtone Signal
- Direct IP call
- Auto Redial, Call Return
- Dial Plan
- XML Browser
- Hot Desking
- Keypad Lock
- Action URL/URI
- Phonebook (500 groups), Blacklist (100 groups), call logs (100 entries)
- 5 Remote Phone Book URL supported
- LDAP
- Multi-Language Support
- Multicast listening

➤ IP-PBX Feature

- SMS, Voicemail, MWI Message Notification
 - Music on hold, Intercom
 - BLF(Busy Lamp Field)
 - Call Pickup, Group Call Pickup
 - Call Recording
 - Anonymous Call, Anonymous Call Rejection
- **Network Features**
- 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
 - Dual-port Gigabit Ethernet (only R55G)
- **Administration Features**
- 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 Features**
- Support HTTPS (SSL)
 - Support SRTP for Voice Data Encryption
 - Support Login for Administration
 - SIP Over TLS

1.3. Keypad

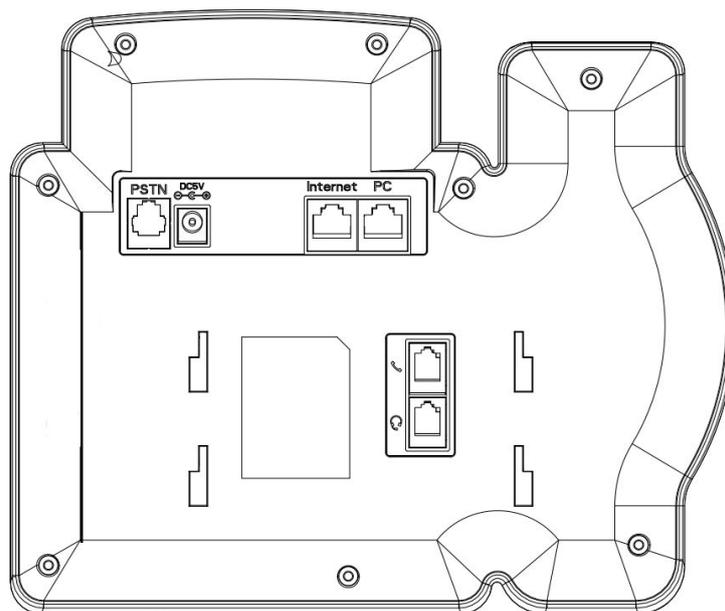


➤ Keypad Description

Key	Key name	Function Description
	Navigation	Assists you in selecting an item that you want to process under the menu by pressing the Up, Down, Right or Left key. Press the center key to save.
	Soft Keys 1/2/ 3/4	Key combination includes functions such as History/Favorites/Redial/CallReturn/HotDesking/ XML Browser/DND/Menu/MSG/Status/Book/FWD/PickUp/Group PickUp/Intercom/Speed Dial/and so on.
	Line Keys 1/ 2/3	Key combination includes functions such as History/Favorites/Redial/Account/ACD/BLF/BLFList/Call Return/HotDesking/Record/XML Browser/DTMF/DND/ Menu/MSG/Status/Book/Fwd/PickUp/Group PickUp/Intercom/Speed Dial/and so on.
	Hold	Use the Hold key to the current conversation.
	Transfer	Use the Transfer key to enter the transfer interface or transfer the conversation.

- 
Headset
Use the headset key to call out or call in.
- 
Conference
Use the Conference key to establish 3-way conference.
- 
Redial
View the Missed Calls, Incoming Calls and Dialed Calls.
- 
Mute
Press this key in calling mode and you can hear the other side, but the other side cannot hear you.
- 
Volume +/-
Turn down or turn up the volume by pressing the “-“ key or the “+“ key.
- 
Handfree
Make the phone into hands-free mode.
- 
Digital keyboard
Inputting the phone number or DTMF.
- 
Indicator light
Blinking light indicates there is an incoming call.

➤ Rear view and panel descriptions



Port	Port name	Description
	Power switch	Input: 5V DC 1A
	Internet port	10/100M(R55P) or 10/1000M(R55G) Connect it to Network;
	PC port	10/100M(R55P) or 10/1000M(R55G) Connect it to PC
	Handset	Port type: RJ-9 connector
	Headset	Port type: RJ-9 connector

1.4. Icon introduction

Registered		Network disconnection	
Register failed		Ring off	
Registering		Headset mode	
Deactivated account		New voice message	
Forward		New text message	
Auto answer		Missed calls	
Do Not Disturb			

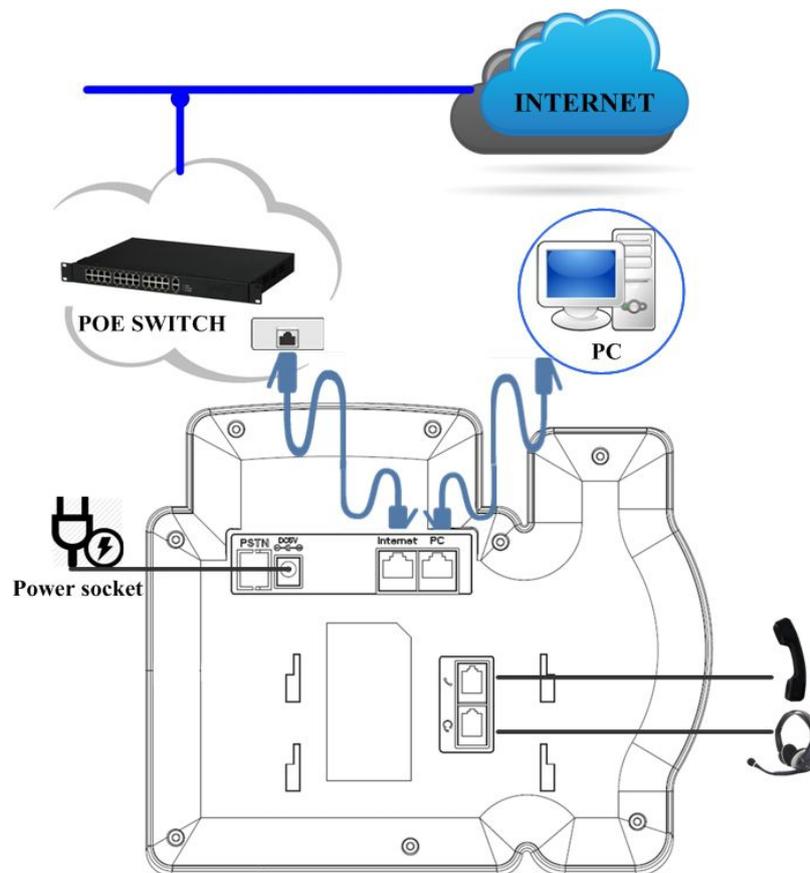
2. Installation

2.1. Check package content

Please refer to the package list below to check the completeness of package

Name	Quantity
SIP IP Phone unit	1
handset	1
RJ-9 Cable	1
Power Adapter	1
RJ-45 Cable	1
Stand	1
Quick installation guide	1

2.2. Connection diagram



2.3. 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 SP-R55P/G

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 SP-R55P/G 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 SP-R55P/G and the other end to the Ethernet port on your 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.

NOTE: Each phone has its own IP address, you can check it by press the OK key on the keyboard when the phone is idle.

3. Functions

3.1. Make a call

3.1.1. Call Device

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.

3.1.2. Call method

User can press an available line key if there is more than one account, then there are 4 ways to make a outgoing call:

- 1) Dial the number user wants to call.
- 2) Press History softkey. Use the navigation keys to select user choice (press Left/Right key to choose Missed Calls, Dialed Calls, Received Calls and Forwarded Calls.
- 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 programmable keys which are set as speed dial key. Then press the

Functions

speed dial programmable key to make the call if necessary.

3.2. Answer a call

- 1) If there are only one incoming call, pick up the handset or press the Speaker key/ Answer soft key to answer directly. You can also press headset key to answer if you use headset.
- 2) If there are more than one incoming call, press the answer soft key 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.

3.3. Mute

You can press the Mute key  to make the user NOT be heard by the other party, but user can hear the other party, and mute icon  will be shown on the LCD, and press the Mute key  again to recover.

3.4. Call Hold/Resume

1. Press the Hold soft key or Hold key  to put User active call on hold.
2. If there is only one call on hold, press the hold soft key or Hold key  to retrieve the call.
3. If there are more than one call on hold, press Up/Down button to highlight one call, and then press the Resume button or Hold key  to retrieve the call.

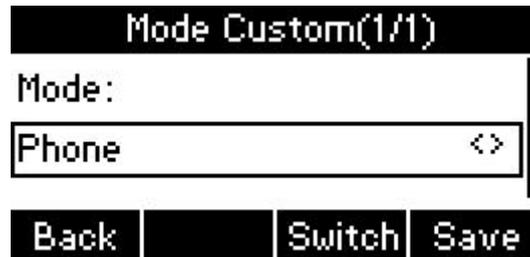
3.5. Do Not Disturb(DND)

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



To configure DND feature via phone.

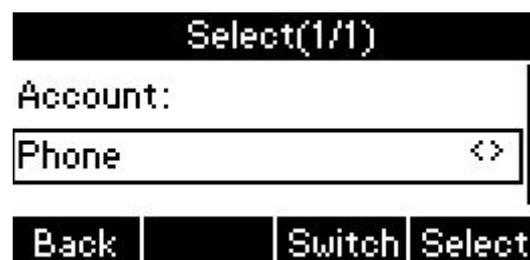
Go to the path: Menu->Features->DND Code->Mode Custom.



Mode Custom: There are two types Phone and Custom.

- 1) If you set the phone mode, enable the DND feature, all accounts enable the DND feature. You can not specify an account to open the DND feature.
- 2) If you set the custom mode, you can chose the specified account to preform the DND feature. Also you can remote control the status of DND from SIP server.

Go to the path: Menu->Features->DND Code->DND Code. Enter the Select interface. The default Phone mode, Switch soft key is unavailable. If you select Custom mode, you can press Left/Right button or Switch soft key to chose the account you want to configure.

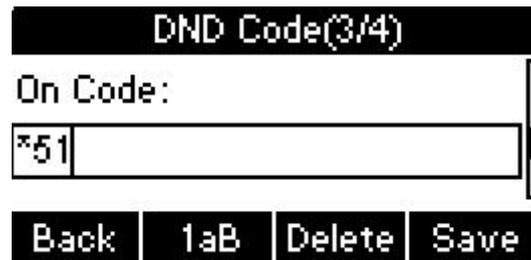


Functions

Go to the path: Menu->Features->DND Code->DND Code->Select soft key. The default Phone mode shown as below. If you select Custom mode, the corresponding account will be displayed. The Switch soft key is unavailable in this interface. You can press down/up button on the phone to enter the next item interface.



- 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



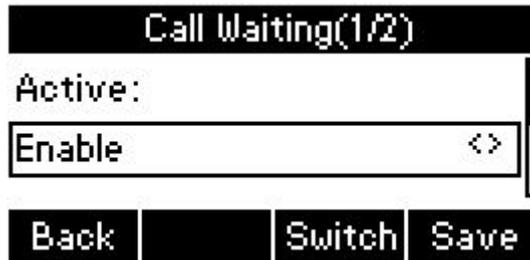
3.6. Call Waiting

Call Waiting feature allows phone to receive a new incoming call when there is already an active call.

Functions

To configure Call Forward via Phone interface.

- 1) Go to the path: Menu ->Features-> Call Waiting.
- 2) You can press Left/Right button or press Switch soft key to activate or deactivate call waiting feature.
- 3) Then press the Save key to save the changes. Press Up/Down button to enter the next item interface.



3.7. Call completion

To monitor the status of the other phone and send a prompt when the status changing. For instant, if users make a call and the callee is unavailable to answer the call, the feature will notify the user when the callee is available to receive the call. Moreover, if the phone enable the voice message, the function is also unavailable.

To configure Call completion via Phone interface.

- 1) Go to the path: Menu-->Feature--> Call Completion.
- 2) You can press Left/Right button or press Switch soft key to enable/disable call completion feature.
- 3) Press the Save soft key to save the changes.



Note: Not all servers can support call completion.

3.8. Call Forward

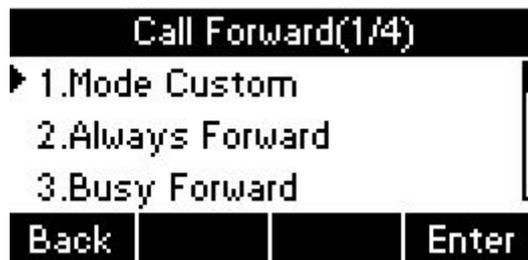
You can set the static forward to switch all the incoming calls to specified number; Also you can use dynamic forward to switch all the incoming calls forward to the number inputted when the phone is ringing.

There are three kinds of static forward.

- 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 tones is time out without answer.

To configure Call Forward via Phone interface:

- 1) Go to the path: Menu ->Features->Call Forward.



- 2) Mode Custom: There are two types Phone mode and Custom mode. If you set the phone mode, Call Forward is valid to phone system. If you set the custom mode, Call Forward can be configured for each or all accounts. Also in Custom mode, you can remote control the status of Call Forward from SIP server.
- 3) There are three types: Always Forward, Busy Forward, No Answer Forward.
- 4) If User chooses one of them, enter the phone number User wants to forward to receiving party. Press Save to save the changes.

3.9. Call Transfer

You can use the following two ways to transfer talking to the other party:

- Blind Transfer: Transfer a call directly to the other party without any negotiation.
- Semi-consultation Transfer: Transfer a call when the other party is ringing.
- Consultation Transfer: Transfer a call to the other person with previous consultation.

3.9.1. Blind Transfer

1. Press the Trans soft key or Transfer key  during the talking.
2. Enter the Trans number interface, and then Input the number you will transfer to.
3. Press the Trans soft key or Transfer key  to transfer the hold talking to the number you want to transfer to.
4. The phone will return to the Idle automatically.

Note: The UI will display Hold status interface when the number you want to transfer to is not existed.

3.9.2. Semi-consultation Transfer

1. Press the Trans soft key or Transfer key  to enter the number you want to transfer to during the talking. Input the number you want to transfer to.
2. Press the OK button on the phone keyboard or the Dial key to make a call.
3. The third party is ringing, then press Trans soft key or Transfer Key .
4. The phone will return to idle automatically.

3.9.3. Consultation Transfer

1. Press the Trans soft key or Transfer key  to enter the number you want to transfer to during the talking. Input the number you want to transfer to.
2. Press the OK button on the phone keyboard or the Dial key to make a call.
3. The third party answered the call, then press the Trans soft key or press Transfer Key  or hang up the handset to transfer.
4. The phone will return to idle automatically.

3.10. Conference

You can establish a 3-way conference with other two parties by pressing the Conf soft key or Conference Key  using local conference. Akuvox SP-R55P/G network conference feature can support multiple parties conference (more than 3-way conference).

3.10.1. Local Conference

Akuvox SP-R55P/G can invite other two parties at most to attend local conference. The default conference type of Akuvox SP-R55P/G is local conference.

- To set up a local conference
 1. Create a conversation with first party.
 2. Press the New soft key, then input the number of second party and press the OK key on the phone keyboard or the Dial key or the Send soft key to make a call.
 3. Press the Conference soft key or Conference Key  to start 3-way conference.
 4. You can press the Split soft key to split to two lines standalone talking, then this two parties talking are under Hold status.

- To make two calls attend local conference
 1. Use two different accounts to create two lines talking in the phone(For example, use account 1 to create line 1 talking and account 2 for line 2).
 2. Press the Up or Down key on the phone keyboard to select the talking you will create for conference, the talking must be on activated status (For example, select the talking in account 1)
 3. Press the conference soft key to add the two lines talking to conference.

- To separate a local conference
 1. You can press the Split soft key to split a conference to two lines standalone talking, then this two parties talking are under Hold status.
 2. Press the Resume soft key to resume the current talking.
 3. Press the Cancel soft key or Cancel button  to cancel the conference talking and return to Idle.

The conference interface shown as below.



3.10.2. Network Conference

Akuvox SP-R55P/G network conference feature can support multiple parties conference(more than 3-way conference).

Go to the path: Account->Advanced->Conference. Select the type as Network, enter the Conference URI. Then click Submit to save.



Conference

Type	Network ▾
Conference URI	conference@as.iop1.bi

Note: Network conference is not available on all servers. For more information, you can ask to administrator for help. You can configure it only via web interface.

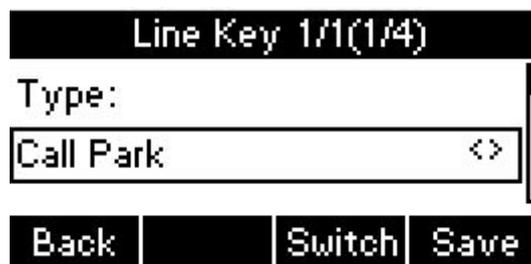
3.11. Call Park

You can use Call Park feature to hold the current conversation, and then resume the conversation in another phone (For example, in another phone of another office or conference).

Note: Not all server can support Call Park feature.

To configure Call Park via Phone interface:

- 1) Go to the path: Menu->Features->Programmable Keys->Line Keys->line Key 1.
- 2) Press Enter into the edit interface. Press Left/Right key or Switch soft key to setup Type as Call Park. Shown as below.



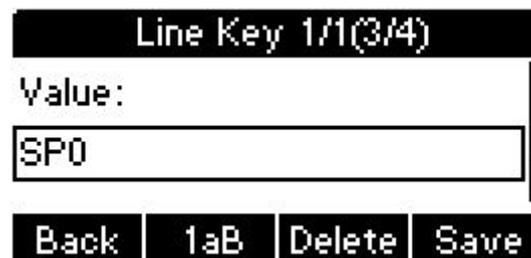
Line Key 1/1(1/4)

Type:

Call Park <>

Back Switch Save

- 3) You can press the Down key to set Label and Value. You can enter the reserved server extension/ call park number. Shown as below.

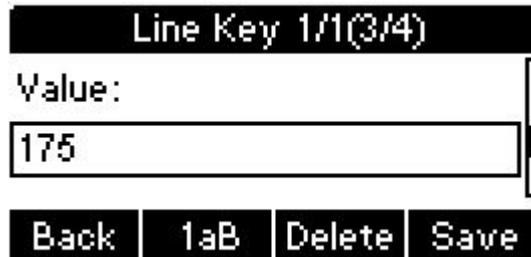


Line Key 1/1(3/4)

Value:

SP0

Back 1aB Delete Save



- 4) Press the Save to save the modification. The call park key led will be light up. The idle shown as below.



- 5) If you enter reserved server extension number in the Value item. During the conversation, press Call Park key. If success, the Call Park key led will be slow flash. Dial the reserved extension number in another phone to resume the Parking conversation.
- 6) If you enter call park number in the Value item. During the conversation, press Call Park key. If success, the Call Park led will be fast flash. The call will be directly transferred to the call park number.

3.12. Pickup

You can use pickup to answer other users' incoming call. The IP phone Akuvox SP-R55P/G supports specified pickup and group pickup.

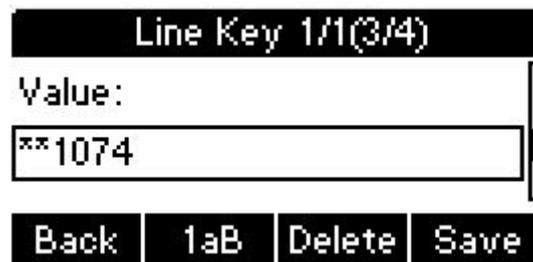
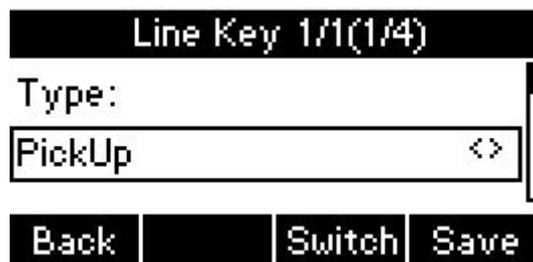
Note: Press the group pickup only to answer line 1 incoming call if there are many lines incoming calls in group.

3.12.1. Specified Pickup

Specified pickup can answer specified user's incoming calls.

Functions

- Set specified pickup key via phone interface,
 - 1) Go to the path: Menu->Features->Programmable keys->Line Keys/Soft Keys/Function Keys
 - 2) Press Left/Right button or press Switch soft key to select the type as PickUp. Press Down key to set label/Value/Account. The Value format you can ask the administrator for more information. The Value item is to enter the pick up code, the Account item is to select the suitable account to pick up the call. Press the save to save the modify. Shown as below.



- 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.



3.12.2. Group Pickup

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

Functions

group members.

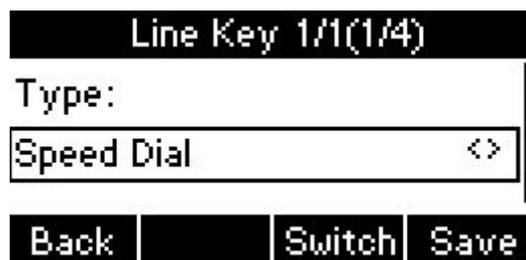
- Set group pickup via phone interface
 - 1) Go to the path: Menu->Features->Programmable keys->Line Keys/Soft Keys/Function Keys.
 - 2) Press Left/Right button or press Switch soft key to select the type as PickUp. Press Down key to set label/Value. The Value format you can ask the administrator for more information. The Value item is to set the group pick up code, the Account item is to select the suitable account to pick up the call. The more information can ask for your administrator for help.
- Use group pickup feature

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

3.13. Speed Dial

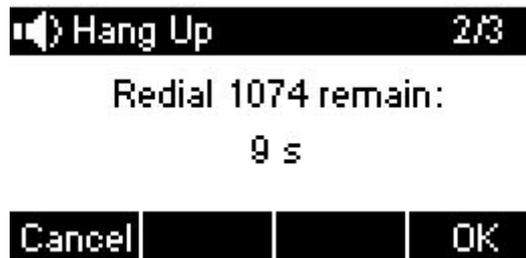
You can use the Speed Dial feature to dial the specified contact speedily

- 1) Go to the path: Menu->Features->Programmable keys->Line Keys/Soft Keys/Function Keys
- 2) Press Left/Right button or Switch soft key to select Speed Dial, then press Down key to set label/Value/Account, the Value item should input the target number you want to call. The Account be selected is to apply this feature. Press Save soft key to save the modify.
- 3) Press the speed dial key, it will directly dial out the target number.



3.14. Auto-redial

You can use auto redial feature to redial the number automatically when the callee is unreachable. When call failure during the calling, the phone will enter the auto-redial screen, and begin to count. Press OK for redial now directly. Or do nothing to make it call out. The phone returns to the standby screen until the times of redial attempts is exhausted.



To configure Auto Redial via Phone interface:

- 1) Go to the path: Menu ->Features->Auto Redial->Enter.
- 2) Use the Left or Right key to activate or deactivate Auto Redial. Auto redial feature is disable by default.
- 3) Use the Up or Down key to configure Interval and Times. The Auto Redial Interval column is to set the redial time interval. The Auto Redial Times is to set the redial times.
- 4) Then press the Save key to save the changes.
- 5) Dial out a number, the callee is busy. The phone interface will be shown as above. Press the Ok soft key to dial out directly, press Cancel soft key to stop the redial.

3.15. Hot Line

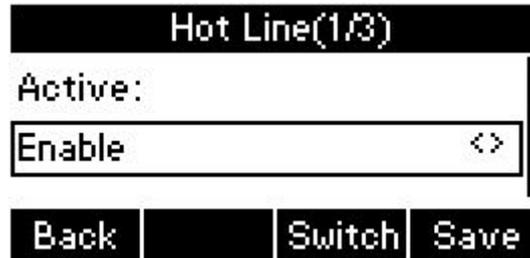
The Hot line refers to the number you often dial. You can set hot lines in the phone, the phone will dial the hot line number automatically when you pick up the handset, press the hand-free key or the line key. Also you can set the timeout of dialing the hot line number, then the phone will dial the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

Functions

timeout.

To configure Hot line via Phone interface:

- 1) Go to the path: Menu ->Features->Hot line.
- 2) Use the Left or Right key to activate or deactivate Hot line feature.



- 3) Use the Up or Down key to configure Number. Timeout column is to select the delay time to dial out the hot line number. The range is 0 to 5 seconds.
- 4) Then press the Save soft key to save the changes. Return to standby interface, pick up handfree or press the handfree or press line key which is registered account successfully.

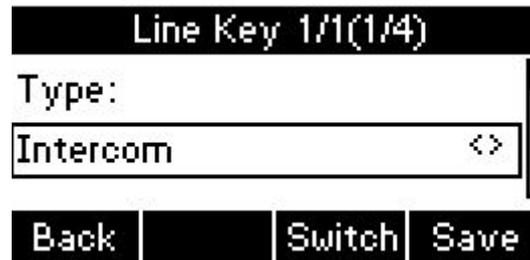
3.16. Intercom

Intercom feature is used to quickly connect with the target number. When you press the Intercom key to connect with a target number, the target extension will automatically answer the incoming intercom by default.

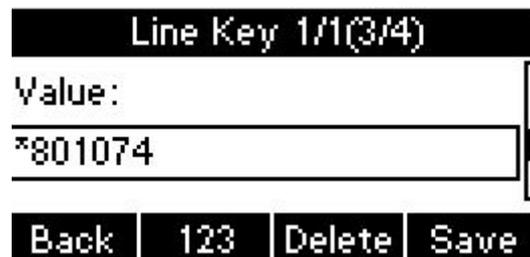
- Configure Intercom key

To configure Intercom key via Phone interface:

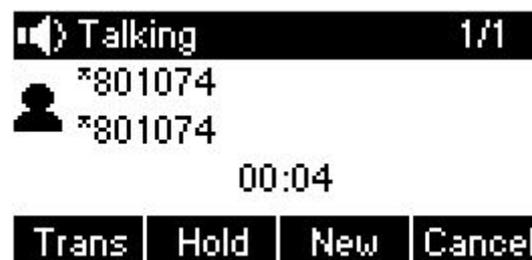
- 1) Go to the path: Menu->Features->Programmable Keys->Line Keys/Soft Keys/Function Keys.
- 2) Enter to select the type as Intercom. Press Left/Right button or Switch Soft key to enable this feature. Press Down key to set the Label/ Value/Account.



- 3) The Value column is to input the intercom extension number. The specific intercom number format need to ask for your administrator for more information. The Account column is to select to apply this feature. The default is Account 1. You can press Left/Right button or Switch soft key to modify. Press Save soft key to save the changes.



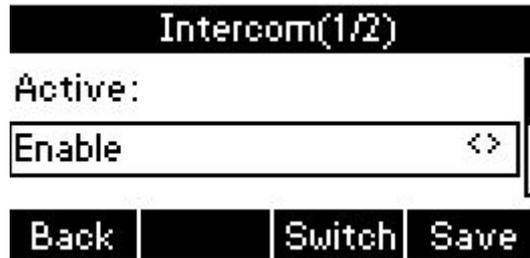
- 4) Press the Intercom key when the phone is available. The phone will connect the extension number of remote user automatically. Press Cancel soft key to end the intercom connection.



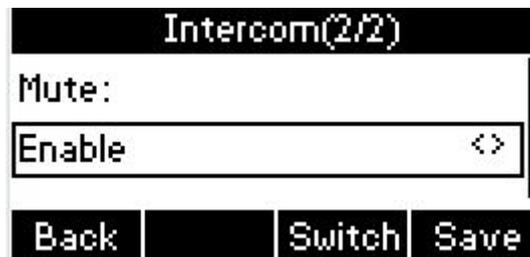
- The acceptor of the Intercom connection
- 1) About the intercom called party. In default situation, the IP phone Akuvox SP-R55P/G will answer the intercom incoming call automatically and can conversation with intercom calling party. You can set the phone to enable silent mode when picking up the intercom call so that the caller will not hear you.
 - 2) About the intercom callee, you can disable the intercom incoming call. Go to the path: Menu->Features->Intercom. Press the Left/Right button or Switch soft key

Functions

to change the Intercom active or deactivate.



- 3) Press the Down button to enter the Mute interface. The default is disable mode. You can press Left/Right button or Switch soft key to enable it. If enable the Mute feature, when pick up intercom incoming call, your voice can not be heard by intercom caller.



3.17. Hot Desking

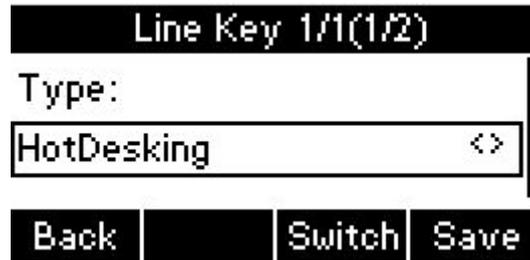
In some working place, the people are always walking around. HotDesking feature will make the staffs login his account on any computer in the company. In some public places, the working people is not fixed, anyone can use HotDesking for logging his account, and setting the phones to the familiar mode. Such as the remote function of the computer.

3.17.1. Set the HotDesking Key

To configure Intercom via Phone interface.

- 1) Go to the path: Menu->Features->Programmable keys->Line Keys/Soft Keys/Function Key. For example, enter the Line Key 1 programmable setting interface.

- 2) Press Left/Right or press Switch to select Type as HotDesking.



- 3) Press Down key to set label. Press Save soft key to save the modify. The phone standby interface shown as below.



3.17.2. HotDesking Feature

- 1) After setting the HotDesking on Line Key 1, back to the idle screen.
- 2) Pressing the HotDesking key, it will pop up the prompt information.



- 3) If you press Clear soft key on the screen, registration configurations of all accounts on the phone will be cleared immediately. The default is to clear previous account information. If you press Cancel soft key, it will exit the Hot Desking pages.
- 4) If you press Clear soft key, after clear the setting, the phone will enter the Account 1 setting screen. Include Register Name, User Name, Password column by default.

Account 1(1/3)
Register Name:
171
Cancel 123 Delete Save

- 5) After entering the account 1 information , press Save soft key to save the modify. Then back to the home screen, and begin to use the new account. The new account 1 information you input here should keep the same server ip address with the previous account 1. If not, the new account 1 may not register successful.

Note: Not all servers can support Hot Desking feature.

3.18. XML Browser

XML Browser allows the users to develop and deploy custom services on the server. Users can use XML Browser key to quickly access it. Users need to pre-configure a custom service functions on the server, such as news, weather report, stock information. The user receives and displays the service information on the IP phone from the server, and all service information are transmitted in XML object.

To configure XML Browser via Phone interface.

- 1) Go to the path: Menu->Features->Programmable keys-->Line Keys/Soft Keys/Function keys.
- 2) Press the Left/Right button or press Switch soft key to select Type as XMLBrowser. Press the Down button to set Label and Value. Shown as below.

Line Key 1/1(1/3)
Type:
XML Browser <>
Back Switch Save



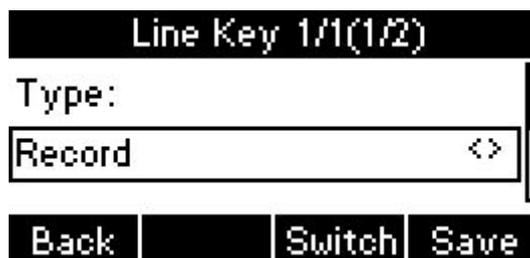
- 3) Then press Save soft key to save the changes. Press the XML Browser key to access the XML browser specified in the Value field.

3.19. Call Recording

You can record calls by pressing a record key on the phone. The phone sends SIP INFO message containing a specific header "Record: on/off " to trigger a recording. Call record is not available on all servers. Contact your system administrator for more information.

To configure a record key via phone user interface:

- 1) Go to the path: Menu->Features->Programmable keys->Line Keys.
- 2) Press Left/Right button or Switch soft key to select the Type field as Record.
Press Down key to set the Label. Press the Save soft key to save the setting.



Functions

- 3) Press the Record key during a call. If the recording starts successfully, the recording icon will appear on the LCD screen. The Record key LED will be slow flash. Shown as below.



- 4) Press the Record key again to stop recording. The recording icon disappears and Record key LED will be turn off.

Note: The way in which you listen to the recordings may be different on different servers. Contact your system administrator for more information.

3.20. Keypad Lock

You can lock the keypad of your phone when you are not using it. This feature helps to protect your phone from unauthorized use. You can enter the password to use it. Keypad Lock can be set to ON or OFF, how long to enable this function and you can choose to lock the function keys or all keys.

- **Function Keys:** The function keys are locked. You cannot use the LIINE KEYS, MESSAGE KEY, SOFT KEYS, NAVIGATION KEYS, and FUNCTION KEYS until unlocked.
- **All Keys:** All keys are locked.

This function can only be configured through the web UI.

- 1) Go to the path: Phone->Call Feature->Keypad Lock. Select the key type you want to lock. Enter the password to unlock the keypad in the Keypad Unlock PIN field. The Keypad Lock Timeout field is to set the lock delay time when you not operate the phone keypad.

Keypad Lock

Keypad Lock Type	All Keys ▾
Keypad Unlock PIN (0~15)
Keypad Lock Timeout	5 (0~3600s)

- 2) If set success, the lock icon will be displayed on the left screen.



- 3) Anyone need to enter the password to unlock the keypad. If the password is incorrect, it will pop up a prompt box “Wrong PIN!”



3.21. Busy Lamp Field

Busy Lamp Field (BLF) is used to monitor a specific user for status changes on IP phones. For example, you can configure a BLF key on a supervisor’s phone to monitor the phone user status (busy or idle). When the monitored user receives a call, a busy indicator on the supervisor’s phone indicates that the user’s phone is in using. SP-R55P/G phone support line key be configured as BLF key. The line key BLF LED status description shown as below.

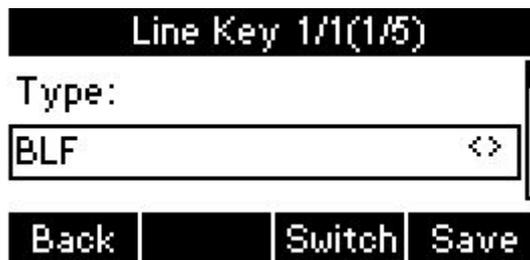
LED Status	Description
Solid green	The monitored user is idle.
Fast flashing green	The monitored user receives an incoming call.

Functions

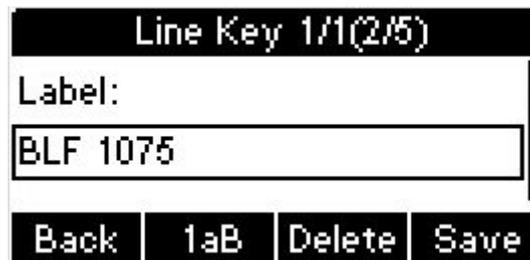
Slow flashing green	The monitored user is dialing. The monitored user is talking.
Off	The monitored user does not exist.

To configure a BLF key via phone user interface:

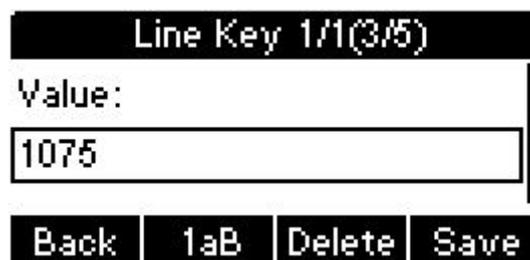
- 1) Go to the path: Menu->Features->Programmable keys->Line Keys->Line Key 1.
- 2) Press Left/Right button or Switch soft key to set the Type as BLF.



- 3) (Optional.) Press Down button to set Label. The Label will appear on the screen.



- 4) Press Down button to set Value. The Value column is to set the monitor phone number.



- 5) Press Down key to select Account ID. The Account column is to select the desired account to apply this feature.
- 6) (Optional.) press Down key to set the pick up code in Extension column. If configured, you can pick up the call that are received on the monitored phone.
- 7) Press Save soft key to save the setting. The standby status shown as below.



- 8) When the monitored user receives an incoming call, the BLF LED will be fast flashing green. If needed, you can press the BLF key to pick up the incoming call.
- 9) You can press BLF key to dial out the monitored phone number when the monitored phone is in standby status.
- 10) When there is an active call in the SP-R55P/G phone, you can transfer the current call to monitored phone by pressing the BLF key.

Note: Different server have different pick up code. The more information, you can ask your administrator for help.

3.22. BLF List

You can use the BLF List feature to monitor a group of phone number pre-configured by your server administrator. For example, your server platform support BLF List feature, and your administrator creates a BLF List URI (e.g. BLFList@test.com) contains a group of user1, user2,user3 on the server point. If configured successful, the BLF List keys on the SP-R55P/G phone will present the Label and status of user1, user2, user3. The BLF List LED will flashing or solid green when the status of those users be changed.

To configure BLF List feature.

- 1) Go to the path: Menu->Features->Programmable Keys->Line Keys->Line Key1/2/3. Press the Left/Right button or Switch soft key to select the type as BLF List. Press Down key to set the label/Account field. For example, set line key1 to line key 3 as BLF List. The idle screen shown as below.



- 2) Enter the SP-R55P/G phone web page. Go to the path: Account->Advanced->BLF List. Enter the BLF List URI. Enter the BLF List Pick up/Barge in code to pick up/barge in the call.

BLFList

BLFList URI	4721_blf@as.iop1.broa
BLFList PickUp Code	*97
BLFList BargeIn Code	*33

- 3) Click the Submit label to save the changes. The server will send the monitored phone number information to phone. The phone idle screen shown as below.



- 4) Enter the SP-R55P/G phone web page. Go to the path: Phone->Key&Display->Line Keys. The Value field is automatically filled in. Shown as below.

Line Key

Key	Type	Label	Value	Account	Extension
Line Key 1	BLFList ▼	2404984724àèù	2404984724	Account 1 ▼	**
Line Key 2	BLFList ▼	Zeng Raymond	2404984722	Account 1 ▼	
Line Key 3	BLFList ▼	24049847	RingslinkUser3	Account 1 ▼	

- 5) When the monitored user is in standby status, press the BLF list key can dial out the phone number.

Functions

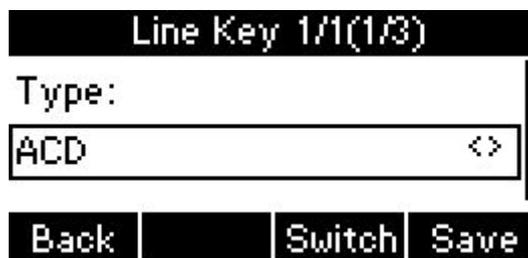
- 6) When there is already an active call on the SP-R55P/G phone, you can transfer the active call to the monitored user by pressing the BLF List key.
- 7) When the monitored user receives an incoming call, you can use the BLF List key to pick up the incoming call when the IP phone is idle.
- 8) When the monitored user is during an active call, you can use BLF List key to interrupt the active call and set up a 3-way conference.

3.23. Automatic Call Distribution(ACD)

Automatic Call Distribution (ACD) enables organizations to manage a large number of phone calls on an individual basis. ACD enables the use of IP phones in a call-center role by automatically distributing incoming calls to available users, or agents. ACD depends on support from a SIP server. ACD is disabled on the phone by default. You need to enable it on a per-line basis before logging into the ACD system.

To configure a ACD key via phone user interface:

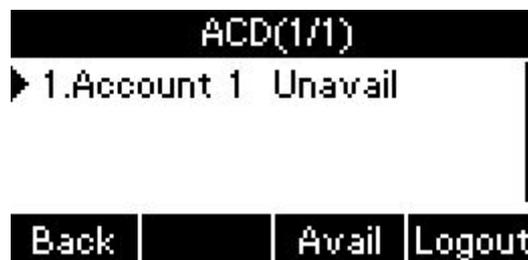
- 1) Go to the path: Menu->Features->Programmable keys->Line Keys->Line Keys 1.
- 2) Press Left/Right button or Switch soft key to select type as ACD. Press Down button to set Label and select account ID. Press Save soft key to save the changes.



- 3) Press ACD key to enter the below interface, the press Login soft key to login the ACD system. After logging in, your phone can receive calls from the ACD system.



- 4) When the ACD account is in Avail status, the ACD key LED displays solid green, the server begins automatically distributing incoming calls to the available phone. When the ACD account is in Unavail status, the ACD key LED flashes green, the server stops distributing incoming calls to your phone. You can press the Avail/Unavail soft key to change the ACD account status.



- 5) If you want to log out of the ACD system , press the Logout soft key. ACD account status synchronizes on both IP phone and ACD system.

3.24. Hoteling

Hoteling function enables the customer to login the own sip account on the Host IP phone, after login to the phone, the customer can use his own guest account on the host IP phone. Hoteling is not supported by all server. Please consult your administrator further information.

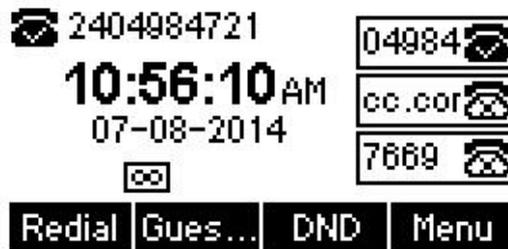
Application case

- Remote Work
 - 1) User goes to the branch office, his own extension number is 4723 in head office.
 - 2) User uses the remote work function, find an idle host IP phone.
 - 3) User can login the extension number 4723 on this host IP phone, to call in and

Functions

out using his own extension number.

- Work on different time division
 - 1) Users A and B work on different time division at a same table with a same host IP phone, their extension numbers are 4722 and 4723.
 - 2) A logs the extension number 4722 in the morning, logout after leave.
 - 3) B logs the extension number 4723 in the evening, using the number 4723 to call in and out, logout after leave.
- To enable the Hotling feature by Autop mode.
 - 1) For example, the host IP phone number is 2404984721. Shown as below.



- 2) Press GuestIn soft key to login the extension number. Enter the User Name: 2404984723. Press Down button to enter the guest user password. Press Enter soft key to login the extension number.



- 3) The extension number 2404984723 is ready for use.



3.25. Application

3.25.1. Text Message

The IP phone Akuvox SP-R55P/G can send and answer text message. The phone will make a “Du” sound and present “N piece of new message” on the LCD(For example: 1 new message), and a twinkling message icon will appears



Note: Not all servers support message feature.

- **Read Text Message**

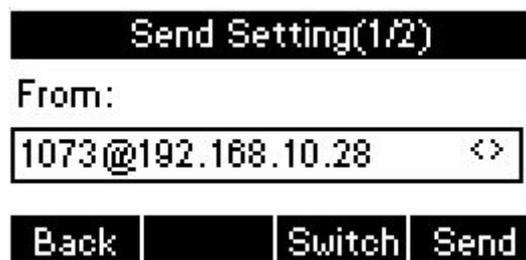
- 1) Access Menu->Message->Text Message-> In box
- 2) Press the OK key on the phone keyboard or the Enter softkey to enter the Text Message interface, Press the OK key on the phone keyboard or the Enter softkey to enter the in-box interface.
- 3) Select the message you will read and Press the OK key on the phone keyboard or the Enter softkey to read.

- **Send Text Message**

- 1) Go to the path: Menu->Message->Text Message->New Message.
- 2) Press the OK key on the phone keyboard or the Enter soft key to enter new message and edit it, press the “abc” soft key to switch the input methods.



- 3) Press the OK key on the phone keyboard or the Send soft key to send message. Press the Left or Right key on the phone keyboard or the Switch soft key to switch to the relevant receiver.



- 4) Input the number of receiver. Press the Send soft key to send message. You can enter the Sentbox to view the sending records.



- Delete Text Message
 - 1) Go to the path: Menu->Message->Text Message->Inbox.
 - 2) Press the OK key on the phone keyboard or the Enter soft key to enter the in-box interface.
 - 3) Select the message you want to delete and press the Delete soft key;
 - 4) Delete all the text messages in the in-box. Press the Delete soft key and select "Delete All", press the OK soft key then all the messages in the in-box will be deleted.

3.25.2. Voice Message

The IP phone R55P/G can send or answer voice message. The phone will make a “Du Du” sound as well as the LED light of message flashes green, and the LCD presents “New Voice Message” on the LCD with a twinkling voice message icon.



Note: Not all servers support voice message.

Voice Message

You can leave a message when the user who you call is busy or unavailable. Leave a message according to the voice prompt of server, and then hang up after leaving the message.

Set Visit account number of voice message via phone interface .

- 1) Go to the path: Menu->Message->Voice Message.
- 2) In the Message interface, Press the Down key on the phone keyboard to select the voice message, Press the OK key on the phone keyboard or the Enter soft key to enter the Voice Message interface.
- 3) Select the Voice Message Setting.
- 4) Press the OK key on the phone keyboard or the Enter soft key to set account 1, input voice message visit account number(For example: *97), press 123 soft key to switch the input methods.



- 5) Press the Down key on the phone keyboard to set account 2, input the Visit account number of voice message(For example: *99), press 123 softkey to switch the input methods:



- 6) Press the Down key on the phone keyboard to set account 3, input voice message visit account number(For example: *100), press 123 softkey to switch the input methods.



- 7) Press the OK key on the phone keyboard or the Save softkey to save and return to message interface.

4. Setting

4.1. Basic Settings

4.1.1. Language

You can change the language through below method:

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



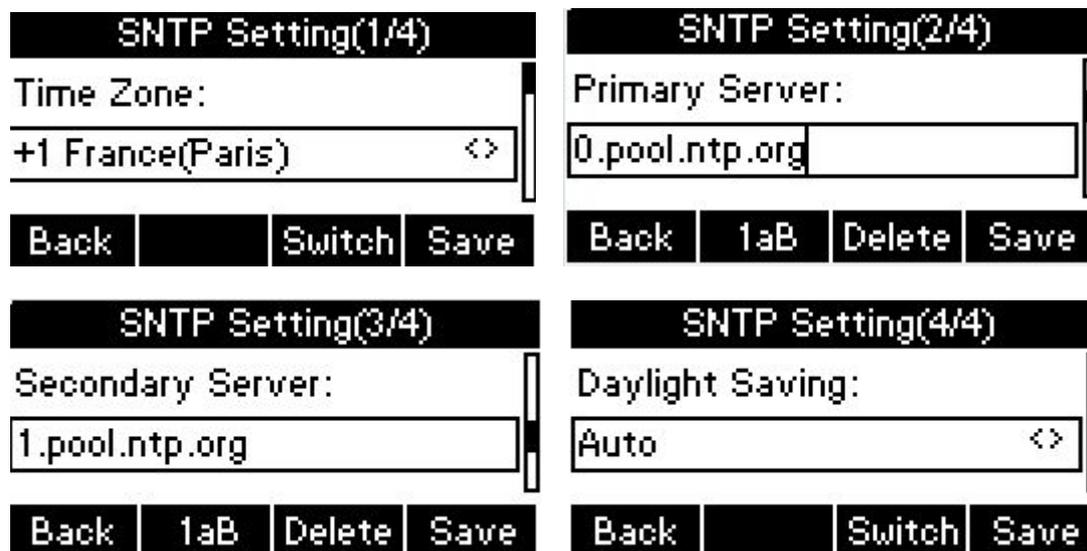
4.1.2. Date&Time

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.

Set SNTP via phone interface:

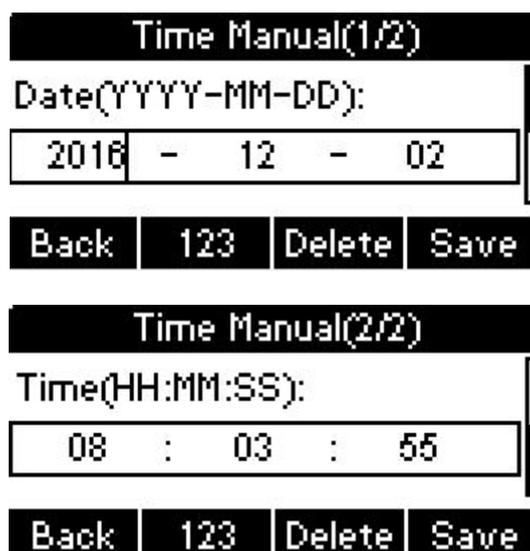
1. Set SNTP via phone interface: Access Menu -> Settings -> Basic Setting -> Date & Time -> SNTP Setting. Press Enter soft key to enter the SNTP interface.
2. Press Switch key to modify the local time zone for SNTP server. Input the Primary server you need . The secondary server will take effect while the primary server

is invalid. Daylight Saving is Auto by default. Users can also disable it. Press Save soft key to save the settings.



Manual setting:

1. Go to the path: Menu -> Settings -> Basic Setting -> Date & Time -> Manual Setting.
2. In Data area, users can delete the original data, enter the new date. Users can also modify the correct time .

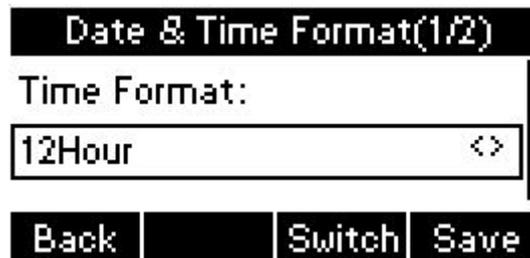


Format setting:

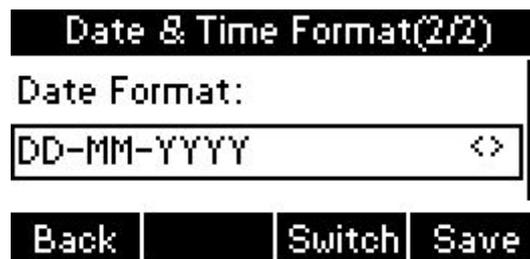
To set the date & time format via the phone interface, access Menu -> Settings ->

Basic Setting -> Date & Time -> Format Setting.

- Access the Time Format in Format Setting interface, then press the Left or Right key on the phone keyboard, or the Switch soft key to select the time format (12Hour or 24Hour).



- In the Date & Time Format interface, press the Up or Down key on the phone keyboard to access the Date Format. Press the Left or Right key on the phone keyboard or the Switch soft key to select the date format to process setting.



- The phone support four 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

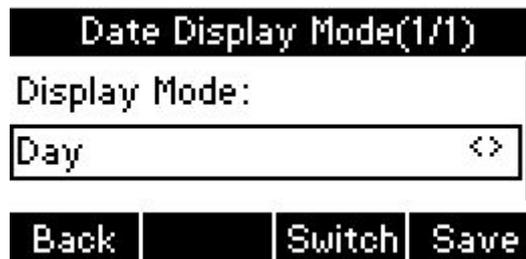
WW-MM-DD	Wed Nov 18
----------	------------

Display mode setting:

To setup which mode will show in the main interface.

Go to the path: Menu->Settings->Basic Setting->Date & Time->Display Mode Setting.

There are 4 modes - Day, Date, Rolling, Disable. Press Switch soft key to change the mode you need.



4.1.3. Backlight

Set the screen backlight level and duration of backlight.

1. Go to the path: Press Menu -> Settings -> Basic Setting ->Backlight.
2. To press left or right key or Switch softkey to adjust the backlight level. The default level is 4. The range of the value is 1 to 5.



3. Press Down button to set the backlight time. Press Left or Right Key or Switch softkey to modify the backlight time. 30 seconds by default.
4. Press Save soft key to save the modify.

4.1.4. Password Setting

This function is to set into the advanced Settings password.

1. Go to the path: Menu -> Settings -> Advanced Setting (admin by default password)->Password Setting.
2. Input the currently password, press Down button to set the new password, then to set confirm new password to modify the current password. Press Save soft key to save the modify.

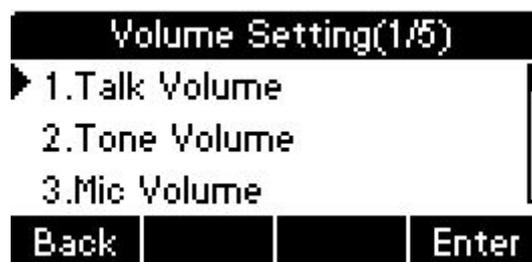


4.2. Sound settings

4.2.1. Phone Volume

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.

1. Adjust the volume via the phone interface, access Menu -> Settings -> Basic Setting -> Phone Volume.
2. In the Volume Setting interface, access the Handset Volume, Hand-free Volume or Headset Volume interface, then press the + or - soft key or Left or Right key to adjust the volume. Press the Save soft key to save the operation or press the Back soft key to cancel operation.



3. User can setup the handset, headset or hand-free volume in different situation. For example, setup the handset talk volume. Enter the Talk Volume interface, choose Handset Volume. Press - or + soft key to adjust. Press Save soft key to save the configuration.

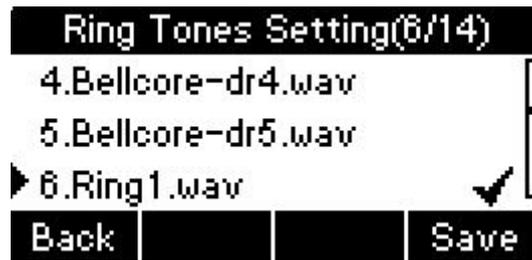


Note: when you modify the Mic Volume, save the change, it will prompt you to reboot the phone.

4.2.2. Ring Tones

The Ring Tone refers to incoming ring tone, which remind the user that there is new incoming call. The IP phone Akuvox SP-R55P/G supports phone ring tone to distinguish the incoming call from other near phones' ring tone. At the same time, the IP phone Akuvox SP-R55P/G also supports setting specific incoming ring tone for contacts.

1. To set the ring tone via the phone interface, access Menu -> Settings -> Basic Setting -> Ring Tones.
2. Press Up or Down key on the keyboard to choose a suitable ring tone. Press Save soft key to save the configuration.



4.3. Phone Book

4.3.1. Local Phone Book

The Local Phone Book is used for storing the contacts names and number. The Akuvox SP-R55P/G can store up to 500 entries contacts. You can add, edit, delete, search, or call any contact from the Local Phone Book.

4.3.1.1.1. Add contact manually

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

1. Go to the path: Menu -> Phone book -> Local phone book.
2. Select the relevant group (For example: Test1) or enter All Contacts. Press the Add softkey to enter the Add Contact interface.Press the Add soft key to enter the Add Contact interface.



3. Input name in the relevant area.



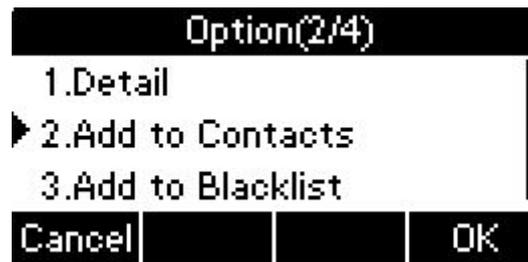
4. Press the Down key on the phone keyboard to input the office number in the relevant area.
5. Press the Down key on the phone keyboard to input mobile number in the relevant area.
6. Press the Down key on the phone keyboard to input other number in the relevant area.
7. Press the Down key on the phone keyboard to enter Ring Tone selection, choose a specified ring tone for this contact.
8. Press the Down key on the phone keyboard to enter the Group selection.
9. Press the Down key to enter Account selection, press the Left or Right key on the phone keyboard or the Switch soft key to select the relevant account, if Auto selected, the phone will select the current available account automatically when the contact called from Local phone book.



4.3.1.1.2. Add contact from All Calls History

Add contact from All Calls History in the phone interface:

1. Press the History soft key.
2. Press the Up or Down key on the phone keyboard to select the contact you want to add.
3. Press the Option soft key to add to contacts.



4. Select Add to Contacts, then press OK soft key to enter edit interface. After finish edit, press Save soft key to save the modify.



4.3.1.1.3. Search Contacts

1. Press the Book soft key in the Idle interface to enter the Phone Book menu.
2. Select the Local Phone Book, Press the OK key on the phone keyboard or the Enter soft key to enter the Local Phone Book.
3. Press the Search soft key to search contacts.



4. Input keywords such as name, any character of number or whole phone number, press the Search softkey or the OK key to enter the Search Contacts interface. For example, input "A", press Search soft key . It will show the name which contains A letter in the contact.



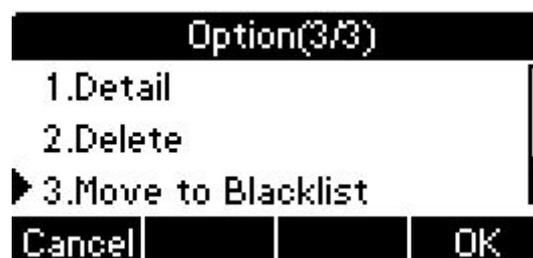
4.3.2. Blacklist

100 Blacklists contacts are available with Akuvox R53P IP phone. You can add, edit, delete, search or call contact. The phone will reject to answer automatically within the blacklists contacts' incoming call. User can add blacklist directly or add from the contact.

1. Go to the path: Phone book -> Blacklist -> Add.
2. Enter the corresponding information, press Save soft key to save.



3. Or you can choose a existed contact. Go to the pah: Menu->Phone Book->Local Phone Book->All Contacts. Press Option soft key, choose Move to Blacklist.



- It pop up the prompt “Move to Blacklist?”, press OK soft key to confirm.



4.3.3. Remote Phone Book

Access the remote phone book, add the contacts to the local phone book from the remote phone book or make calls from the remote phone book. 5 URLs of remote phone book is available to set.

- Set the remote phone book via web interface:
 - Go to the path: Phone Book-> Remote Phone Book.
 - Input URL of phone book. Input the phone book name.
 - Click the Submit key to save.

Remote Book		
Remote Book		
Index	Local Book URL	Local Book Name
1	ftp://192.168.10.151/remote_contact2.xml	Akuvox110
2		
3		
4		
5		

- Access the remote phone book via phone interface.
 - Go to the path: Menu->Phone Book->Remote phone book.
 - Select the relevant Remote Group and press the Enter/Update soft key. The phone will load the remote group information, and the LCD will display the contacts of this remote group.



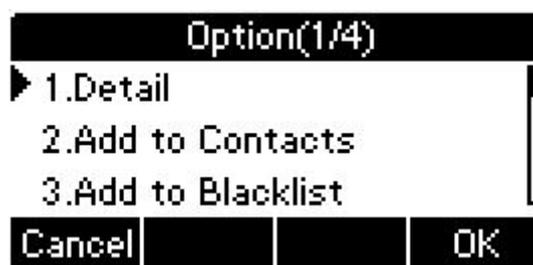
4.4. History Management

The History management of IP phone Akuvox SP-R55P/G contains dialed calls, received calls, missed calls and forwarded calls and support 100 logs storage at most. You can check the history, make calls from the calls history and delete the calls history.

1. Press the History key, the LCD will display all the recent calls.
2. Press the Left or Right key on the phone keyboard to switch the lists of All Calls, Dialed Calls, Received Calls, Missed Calls and Forwarded Calls.



3. Press the Up or Down key on the phone keyboard to select the log.
4. Press the Option soft key and select the detail. The LCD will display the detailed information of this log; Press the Dial, to make a call from the History.
5. Press the Option soft key to add to contacts (Add to Blacklists) from the History.
6. Press the Delete soft key to delete calls log from the History.
7. Press the Option soft key to select "Delete all" to delete all the calls log from the History.



4.5. System Customization

4.5.1. Programmable Keys

1. Press the Menu soft key in the Idle interface, access Menu->Features->Programmable keys.
2. Select the programmable key you will set and press the Enter soft key.
3. Select key style in the type area.
4. Input suitable value in the label area.
5. (Optional) Select the relevant account in the account ID area.
6. (Optional) Input suitable value in Value blank.
7. (Optional) Press the Save soft key to save or the Cancel soft key to cancel.
8. Press the Save soft key to save or the Cancel soft key to cancel.

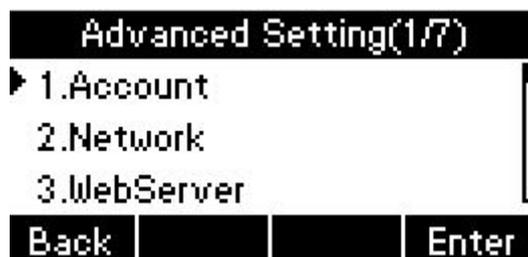
4.5.2. SIP Account Management

You can register one account or multi-account, but also can set one account key or multi-account keys for one account.

4.5.2.1.1. Register an Account

Register an account via phone interface:

1. Go to the path: Menu->Settings->Advanced setting(password: admin)->Account.



2. Select the account you want to set and press the Enter soft key.
3. Select "Enable" in the account activation status area.



4. Press Down to enter the next item. Input the label, display name, register name, account, password and SIP separately.
5. Choose Auto Answer or not.
6. Press the Save soft key to save or the Back soft key to cancel.
7. Repeat step 2 to 5 to finish all the account register.
8. If the account successful registered. The account status will be shown as below.



4.5.2.1.2. Disable an Account

1. Go to the path: Menu->Settings->Advanced setting(password: admin)->Account.
2. Select the account you want to disable and press the Enter soft key.
3. Select "Disable" in the account active status area.



4. Press the Save soft key to save or the Back soft key to cancel. The account status shown as below.

**Note:**

- 1) The operation and interface of account 2 and 3 is the same as account 1.
- 2) You can configure multi-account keys for one account. The incoming calls of this account will be divided equally to different account key; Analogously, the dialings also will be divided equally.

4.6. Basic Network Settings

Through the Basic Network setting, you can set the IP Phones to get the IP address by three ways: DHCP, static IP and PPPOE, also can set the VLAN, PC port mode.

PATH: Menu -> Settings -> Advanced Setting -> Network

4.6.1. DHCP Mode

To configure the DHCP mode via phone user interface.

1. In IP Phone idle interface, press soft key Menu->Settings->Advanced Setting(default password:admin)->Network->LAN Port.
2. Phone user interface shown: DHCP by default.



3. Press soft key Enter or OK button to enter the DHCP setting interface, after obtain the IP address, it will automatically back to the previous interface.

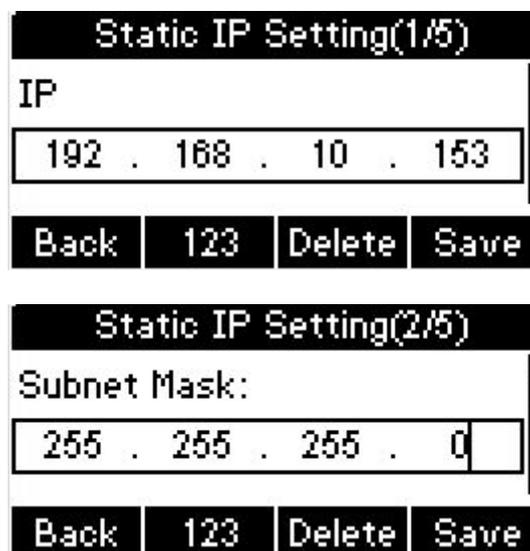


Note:Default DHCP mode after reset to factory.

4.6.2. Static IP Mode

To configure the Static IP mode via phone user interface.

- 1) In IP Phone idle interface,press soft key Menu-> Settings -> Advanced Setting (default password:admin)->Network->LAN Port.
- 2) Press navigation key Down on the phone keypad to select Static IP,then press OK button or soft key Enter to the Static IP Setting interface,input the IP address,Subnet Mask,Gateway,DNS 1 and DNS 2 in the corresponding area,press OK button or soft key Save to save the change.



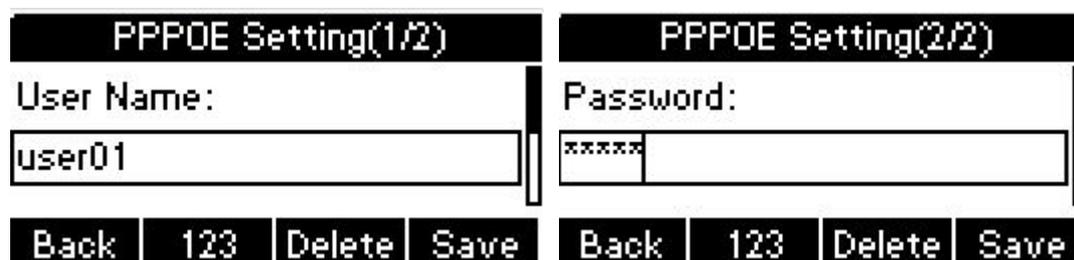
4.6.3. PPPoE Mode

To configure the PPPOE mode via phone user interface.

1. In IP Phone idle interface,press Menu->Settings->Advanced Setting(default

password:admin)->Network->LAN Port.

2. Press twice navigation key Down on the phone keypad to select PPPOE,then press OK button or Enter soft key to enter PPPOE Setting interface,input the User Name and Password,press OK button or Save soft key to save the configuration.



4.6.4. Configure PC Port Mode

Bridge mode

1. In IP Phone idle interface,press Menu->Settings->Advanced Setting(default password:admin)->Network->PC Port.
2. In the PC Port configuration interface, press the navigation key Up or Down to select Bridge mode. press OK button or Enter soft key , enter Warning interface prompt “Reboot Phone?”; Press OK soft key to reboot.(PS: Setting will take effect after reboot.If cancel the reboot, the Settings will be saved but not take effect.)



Router mode

1. In the PC Port configuration interface, press the navigation key Up or Down to select Router mode, input valid value in the corresponding position;
2. Press OK button or Save soft key after configuration, the phone will reboot.

Router Setting(1/5)				Router Setting(2/5)			
IP				Subnet Mask:			
10	.	0	.	0	.	0	.
255	.	255	.	255	.	0	
Back	123	Delete	Save	Back	123	Delete	Save
Router Setting(3/5)				Router Setting(4/5)			
DHCP Server:				Start IP:			
Enable <>				10 . 0 . 0 . 10			
Back		Switch	Save	Back	123	Delete	Save
Router Setting(5/5)							
End IP:							
10	.	0	.	0	.	100	
Back	123	Delete	Save				

4.6.5. Configure VLAN

To configure the VPN mode via phone user interface.

1. In IP Phone idle interface,press Menu->Settings->Advanced Setting(default password:admin)->Network->VPN.
2. Press Enter soft key to enter the VPN Setting interface,press Switch soft key to enable or disable the VPN status.press Save soft key to save the change.(about more detail information,you can refer to the following web configuration.)

VPN Setting(1/1)			
Active:			
Enable <>			
Back		Switch	Save

3. When configured VPN function success,in the phone idle interface,it will shown VPN  icon.

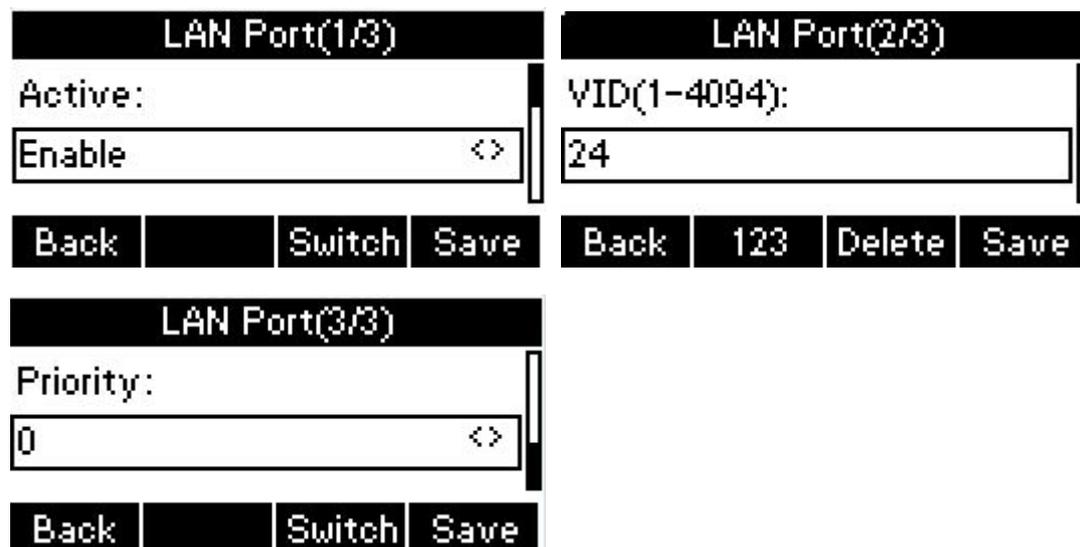
4.6.6. VLAN

To configure the VLAN mode via phone user interface.

In IP Phone idle interface,press Menu->Settings->Advanced Setting(default password:admin)->Network->VLAN Port.

LAN Port

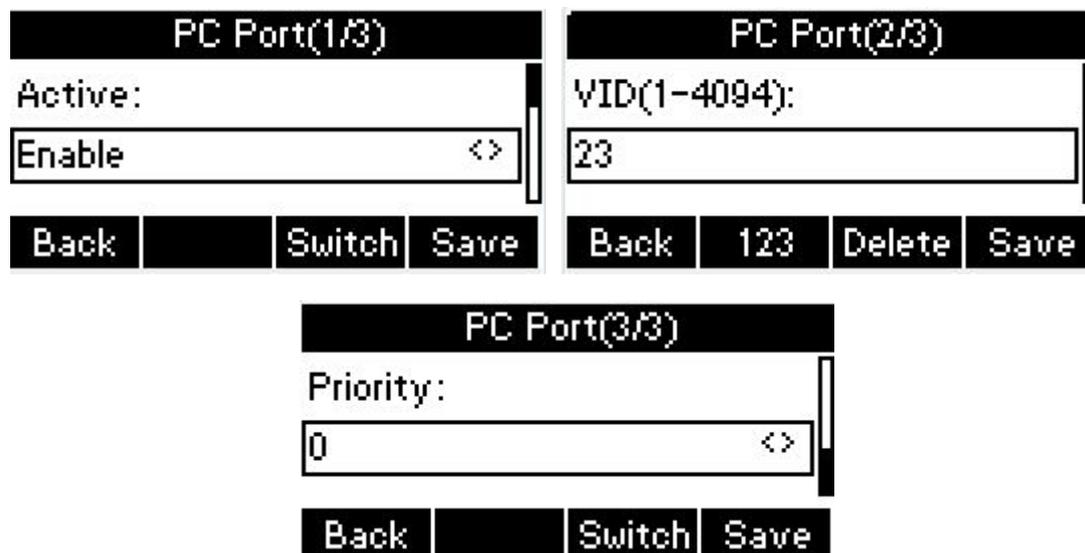
- 1) In the VLAN Port interface, press the navigation key Up or Down to select LAN Port, press the OK button or Enter soft key to enter LAN Port.
- 2) In the LAN Port interface, press the navigation key Up or Down to configure the functionality Enable, VID, Priority.press OK button or Save soft key.it will pop up a prompt "Setup completed! Restart to take effect?",press OK button or OK soft key to reboot.



PC Port

1. In the VLAN Port interface, press the navigation key Up or Down to select PC Port, press the OK button or the soft key Enter to enter PC Port.In the PC Port interface, press the Up or Down key on the phone keyboard to configure the functionality Enable, VID, Priority.
2. In the PC Port interface, press the Up or Down key on the phone keyboard to

configure the functionality Enable, VID, Priority.press OK button or soft key Save.it will pop up a prompt “Setup completed! Restart to take effect ?”,press OK button or soft key OK to reboot.



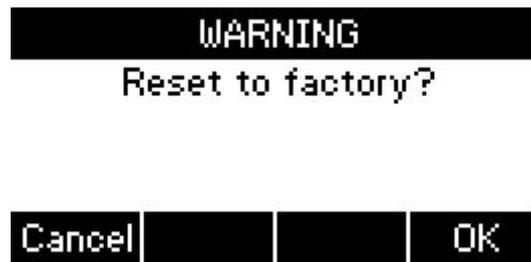
4.7. WebServer

To configure the WebServer via phone user interface.

1. In IP Phone idle interface,press Menu->Settings->Advanced Setting(default password:admin)->WebServer.
2. Press OK button or Enter soft key to access the disable or enable WebServer settings.Enable the WebServer ,user can change the configure by website.

4.8. Reset to Factory

In the Advanced Setting interface, press the Up or Down key on the phone keyboard to select “Reset to factory”, Press the OK key on the phone keyboard or the Enter soft key to access the reset to factory interface.



4.9. Password Setting

User can modify the password, go to the path: Advanced->Setting.(Default password :admin.)

1. In IP Phone idle interface, press Menu->Settings->Advanced Setting(default password:admin)->Password Setting.
2. Enter the Current Password,New Password,Confirm Password.Press OK button or Save soft key to save the change.

4.10. Autoprovision

To configure the Autoprovision via phone user interface.

1. In IP Phone idle interface,press Menu->Settings->Advanced Setting(default password:admin)->Autoprovision.
2. Enter the URL,User Name,Password to update the configuration,also user can modify the provision mode and time.press OK button or Save soft key the change.

4.11. Reboot

To reboot the phone via phone user interface.

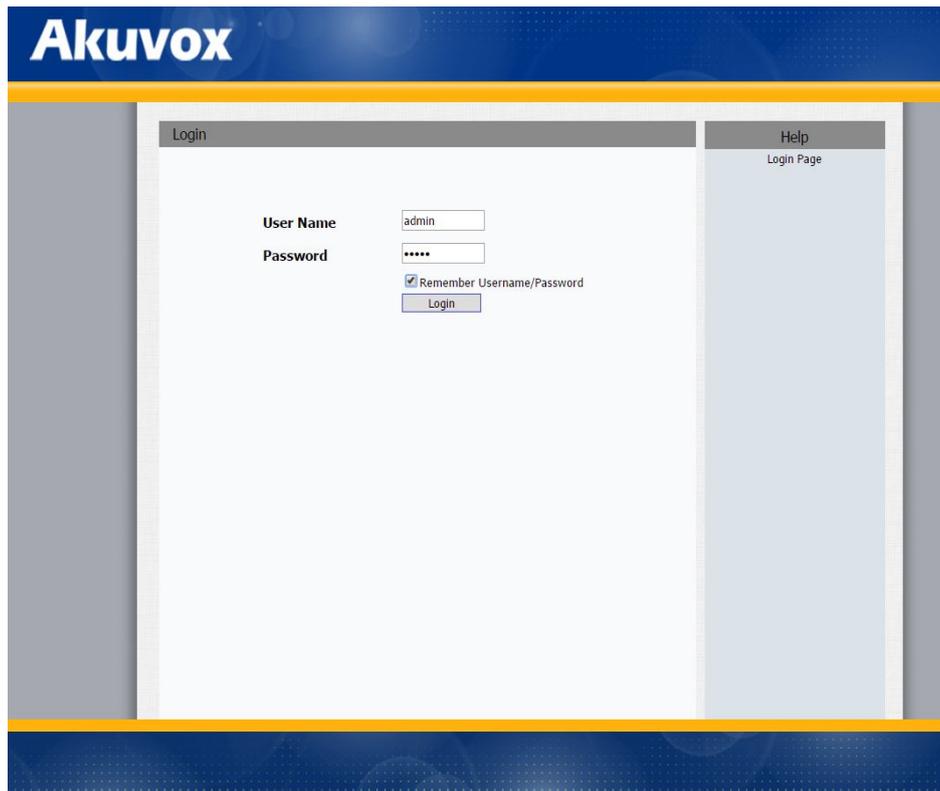
1. In IP Phone idle interface,press Menu->Settings->Advanced Setting(default password:admin)->Reboot.

2. Press the OK button or Enter soft key to enter the Warning interface, prompt "Reboot Phone?", press OK button or Save soft key to reboot the phone.

5. Web interface

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

- Press the OK key on the phone keyboard to check the Phone's IP address.
- Type the IP address on IE, input default User Name and Password: admin/admin to login the web interface.



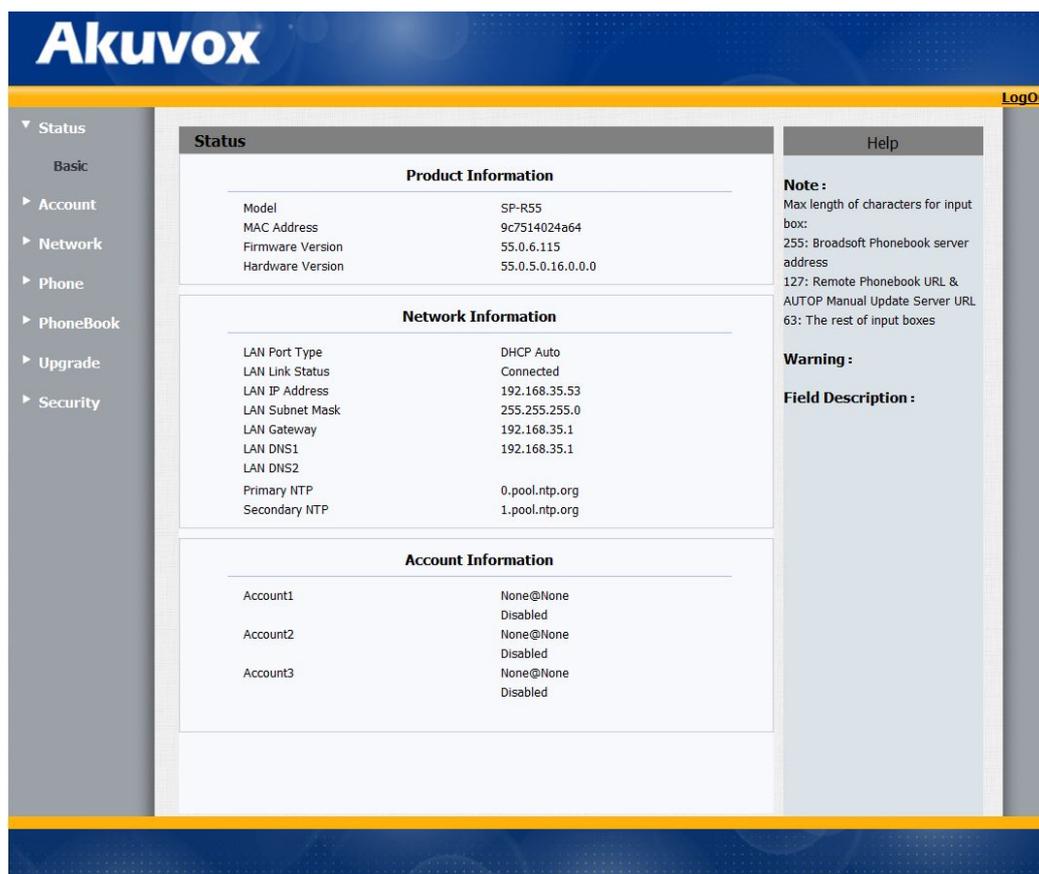
The screenshot shows the Akuvox web interface login page. The page has a dark blue header with the Akuvox logo. Below the header, there is a white content area with a grey border. The content area is divided into two sections: 'Login' on the left and 'Help' on the right. The 'Login' section contains the following fields and controls:

- User Name**: A text input field containing the text 'admin'.
- Password**: A password input field containing six dots.
- Remember Username/Password
-

The 'Help' section contains the text 'Login Page'.

5.1. Status->Basic

Go to **Status->Basic** Page, as figure below shown:



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).

Note: Status->Basic page is used to display some basic information for IP Phone. Please refer to corresponding page for any further information.

5.2. Account->Basic

Path: Web UI -> Account->Basic

Account-Basic

SIP Account

Status	Registered	
Account	Account 1 ▼	
Account Active	Enabled ▼	
Display Label	<input type="text" value="173"/>	
Display Name	<input type="text" value="173"/>	
Register Name	<input type="text" value="173"/>	
User Name	<input type="text" value="173"/>	
Password	<input type="password" value="....."/>	

SIP Server 1

Server IP	<input type="text" value="192.168.10.31"/>	Port	<input type="text" value="5060"/>
Registration Period	<input type="text" value="1800"/>	(30~65535s)	

SIP Server 2

Server IP	<input type="text"/>	Port	<input type="text" value="5060"/>
Registration Period	<input type="text" value="1800"/>	(30~65535s)	

Sections	Description
SIP Account	<p>To display and configure the specific Account settings.</p> <ul style="list-style-type: none"> ● 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 display. ● 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	<p>To display and configure Primary SIP server settings.</p> <ul style="list-style-type: none"> ● 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	<p>To display and configure Secondary SIP server settings.</p>

	<p>This is for redundancy, if registering to Primary SIP server failed, the IP phone will go to Secondary SIP server for registering.</p> <p>Note: Secondary SIP server is used for redundancy, it can be left blank if there is not redundancy SIP server in user's environment.</p>
--	--

Outbound Proxy Server

Enable Outbound Enabled ▾

Server IP 66.66.17.152 Port 5060

Backup Server IP Port 5060

Transport Type

Transport Type UDP ▾

NAT

NAT Disabled ▾

Stun Server Address Port 3478

Sections	Description
Outbound Proxy Server	<p>To display and configure Outbound Proxy server settings.</p> <p>An outbound proxy server is used to receive all initiating request messages and route them to the designated SIP server.</p> <p>Note: If configured, all SIP request messages from the IP phone will be sent to the outbound proxy server forcefully.</p>
Transport Type	<p>To display and configure Transport type for SIP message</p> <ul style="list-style-type: none"> ● 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	<p>To display and configure NAT(Net Address Translator) settings.</p> <ul style="list-style-type: none"> ● STUN: Short for Simple Traversal of UDP over NATS, a solution to solve NAT issues. <p>Note: By default, NAT is disabled.</p>

5.3. Account->Advanced

Path: Web UI->Account->Advanced

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)

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. <ul style="list-style-type: none"> ● 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

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	▼
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	▼

Sections	Description
DTMF	<p>To display and configure DTMF settings.</p> <ul style="list-style-type: none"> ● 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. <p>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.</p>
Call	<p>To display and configure call-related features.</p> <ul style="list-style-type: none"> ● 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.

	<ul style="list-style-type: none"> ● 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.
--	--

Music Server Address

Active Disabled ▼

Music Server Address

Session Timer

Active Disabled ▼

Session Expire (90~7200s)

Session Refresher UAC ▼

BLFList

BLFList URI

BLFList Pickup Code

BLFList BargeIn Code

Broadsoft

AOC Disabled ▼

Sections	Description
Music Server Address	To display or configure third-party MOH (music-on-hold) server. <ul style="list-style-type: none"> ● Active: To enable or disable this MOH server, If enabled,

	<p>the IP phone will play MOH from configured server.</p> <ul style="list-style-type: none"> ● Music Server Address: To configure MOH server address.
Session Timer	<p>To display or configure session timer settings.</p> <ul style="list-style-type: none"> ● 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. <p>Note: UAC means User Agent Client, here stands for IP phone. UAS means User Agent Server, here stands for SIP server.</p>
BLF List	<p>To display or configure BLF List URI address.</p> <ul style="list-style-type: none"> ● BLF List URI: BLF List is short for Busy Lamp Field List. ● BLFList PickUp Code: To set the BLF pick up code. ● BLFList Bargain Code : To set the BLF barge in code.
Broadsoft	<p>To display or configure Broadsoft AOC feature.</p> <ul style="list-style-type: none"> ● AOC: A feature used to be accounting on Broadsoft platform. <p>Note: Please consult your administrator further information.</p>

Encryption

Voice Encryption(SRTP) Disabled ▼

NAT

UDP Keep Alive Messages Enabled ▼

UDP Alive Msg Interval 30 (5~60s)

RPort Disabled ▼

Conference

Type Network ▼

Conference URI conference@as.iop1.br

User Agent

User Agent

Sections	Description
Encryption	<p>To enable or disabled SRTP feature.</p> <ul style="list-style-type: none"> ● Voice Encryption(SRTP): If enabled, all audio signal (technically speaking it's RTP streams) will be encrypted for more security.

<p>NAT</p>	<p>To display NAT-related settings.</p> <ul style="list-style-type: none"> ● 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.
<p>Conference</p>	<p>To select Local or network conference.</p> <ul style="list-style-type: none"> ● Type: To select desired conference type ● Conference URI: If network conference is selected, a network conference URI is needed to be input.
<p>User Agent</p>	<p>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</p>

5.4. Network->Basic

Path: Web UI->Network->Basic

Network-Basic

LAN Port

DHCP
 Static IP

IP Address
 Subnet Mask
 Default Gateway
 LAN DNS1
 LAN DNS2

PPPoE

User Name
 Password

Sections	Description
<p>LAN Port</p>	<p>To display and configure LAN Port settings.</p> <ul style="list-style-type: none"> ● 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. <ul style="list-style-type: none"> ● PPPoE: Use PPPoE username/password to connect to PPPoE server.
--	---

PC Port

As Bridge
 As Router

IP Address
 Subnet Mask
 DHCP Server Status
 Start IP Address
 End IP Address

Sections	Description
PC Port	To display and configure PC Port settings. <ul style="list-style-type: none"> ● 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.

5.5. Network->Advanced

Path: Web UI->Network->Advanced

Network-Advanced

LLDP

LLDP Active
 Packet Interval (10~3600s)

Local RTP

Max RTP Port (1024~65535)
 Starting RTP Port (1024~65535)

SNMP

Active
 Port (1024~65535)
 Trusted IP

Sections	Description
LLDP	<p>To display and configure LLDP settings.</p> <ul style="list-style-type: none"> ● LLDP Active: To enable or disable LLDP feature. ● Packet interval: To configure the interval for LLDP admin message. <p>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.</p>
Local RTP	<p>To display and configure Local RTP settings.</p> <ul style="list-style-type: none"> ● 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	<p>To display and configure SNMP settings.</p> <ul style="list-style-type: none"> ● 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. <p>Note: SNMP (Simple Network Management Protocols) is Internet-standard protocol for managing devices on IP networks.</p>

VLAN

LAN Port	Active	<input type="text" value="Disabled"/>	
	VID	<input type="text" value="1"/>	(1~4094)
	Priority	<input type="text" value="0"/>	
PC Port	Active	<input type="text" value="Disabled"/>	
	VID	<input type="text" value="1"/>	(1~4094)
	Priority	<input type="text" value="0"/>	

QoS

SIP QoS	<input type="text" value="40"/>	(0~63)
Voice QoS	<input type="text" value="40"/>	(0~63)

TR069

	Active	<input type="text" value="Disabled"/>	
	Version	<input type="text" value="1.0"/>	
ACS	URL	<input type="text"/>	
	User Name	<input type="text"/>	
	Password	<input type="password" value="*****"/>	
Periodic Inform	Active	<input type="text" value="Disabled"/>	
	Periodic Interval	<input type="text" value="1800"/>	(3~24×3600s)
CPE	URL	<input type="text"/>	
	User Name	<input type="text"/>	
	Password	<input type="password" value="*****"/>	

Sections	Description
VLAN	To display and configure VLAN settings. <ul style="list-style-type: none"> ● 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. <p>Note: Please consult your administrator for specific VLAN settings in your networking environment.</p>
QoS	To display and configure QoS settings. <ul style="list-style-type: none"> ● 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. <ul style="list-style-type: none"> ● Active: To enable or disable TR069 feature. ● Version: To select supported TR069 version (version 1.0 or 1.1).

	<ul style="list-style-type: none"> ● 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. <p>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.</p>
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802.1x

802.1x Mode

Identity

MD5 Password

VPN

Active

User Name

Password

Upload(<50K) 未选择任何文件

Sections	Description
802.1x	To display and configure 802.1x settings. <ul style="list-style-type: none"> ● 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. <ul style="list-style-type: none"> ● Active: To enable or disable VPN feature. ● Upload: To upload VPN client configuration file which is used to connect to VPN server. <p>Note: For now, IP phone can only support OpenVPN.</p>

5.6. Phone ->Time/Lang

Path: Web UI->Phone->Time/Lang

Time/Lang

Web Language

Type English ▼

LCD Language

Type English ▼

Format Setting

Time Format 12Hour ▼

Date Format DD-MM-YYYY ▼

Display Mode Day ▼

Sections	Description
Web Language	To choose the web language.
LCD Language	To choose the phone language.
Format Setting	To configure time display settings. <ul style="list-style-type: none"> ● 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

Manual
 Auto

Date Year Mon Day

Time Hour Min Sec

NTP

Time Zone +1 France(Paris) ▼

Primary Server 0.pool.ntp.org

Secondary Server 1.pool.ntp.org

Update Interval 3600 (>= 3600s)

Sections	Description
Type	<p>To select how to configure time, it could be set by manually or get from INTERNET automatically via NTP server.</p> <ul style="list-style-type: none"> ● Manual: To set Time and Date manually. ● Auto: To get Time via NTP server. <p>Note: If you set time to be Manually, it only take 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.</p>
NTP	<p>To configure NTP server related settings.</p> <ul style="list-style-type: none"> ● 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. <p>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.</p>

Daylight Saving Time

Active: Auto ▼

OffSet: 60 (-300~300Minutes)

By Date

Start Time: 1 Mon 1 Day 0 Hour

End Time: 12 Mon 31 Day 23 Hour

By Week

Start Month: Jan ▼

Start Week Of Month: First In Month ▼

Start Day Of Week: Monday ▼

Start Hour: 0 (0~23)

End Month: Dec ▼

End Week Of Month: Fourth In Month ▼

End Day Of Week: Sunday ▼

End Hour: 23 (0~23)

Sections	Description
Daylight Saving Time	<p>To display or configure DST settings.</p> <p>Note: Here DST, is short for Daylight saving time, which stands</p>

	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).
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5.7. Phone->Preference

Path: Web UI->Phone->Preference

Preference

Headset Mode

Active Disabled ▼

Key Press Sound

Volume (0~15)

Ringtone Volume

Volume (0~15)

Sections	Description
Headset Mode	To enable or disable Headset Mode. <ul style="list-style-type: none"> ● 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. <ul style="list-style-type: none"> ● Volume: The valid volume range is from 0~15,by default it's 8.
Ringtone Volume	To configure the sound volume for ringtone. <ul style="list-style-type: none"> ● Volume: The valid volume range is from 0~15,by default it's 8.

5.8. Phone->Call Feature

Path:Web UI->Phone->Call Feature

Phone-Call Feature

Mode Phone

Feature Key Sync

Mode

Phone Custom

Forward Transfer

Account

Always Forward

Target Number

On Code

Off Code

Busy Forward

Target Number

On Code

Off Code

No Answer Forward

No Answer Ring Time

Target Number

On Code

Off Code

Sections	Description
Mode	To enable or disable feature key sync. <ul style="list-style-type: none"> ● Feature Key Sync: To enable or disable feature key sync. ● Mode: Select the desired mode.
Forward Transfer	To display and configure Forward setting. <p>Note: There are three types of forward: Always Forward, Busy Forward and No answer Forward.</p> <ul style="list-style-type: none"> ● 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.

DND

DND Emergency Disabled ▼

DND Authorized Number

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 ▼

On Code

Off Code

Auto Redial

Auto Redial Disabled ▼

Auto Redial Interval (1~300s)

Auto Redial Times (1~100)

Sections	Description
DND	<p>DND (Do Not Disturb) allows IP phones to ignore any incoming calls.</p> <ul style="list-style-type: none"> ● 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.
Call Waiting	<p>To enable or disable Call Waiting.</p> <ul style="list-style-type: none"> ● Call Waiting Enable: If enabled, it allows IP phones to

	<p>receive a new incoming call when there is already an active call.</p> <ul style="list-style-type: none"> ● Call Waiting Tone: If enabled, it allows IP phones to play the call waiting tone to the waiting callee.
Auto Redial	<p>Auto redial allows IP phones to redial an unsuccessful call for designated times within designated interval.</p> <ul style="list-style-type: none"> ● 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.

Call Pickup

Visual BLF Pickup Alert Disabled ▼

Intercom

Active Enabled ▼
 Intercom Mute Disabled ▼

HotLine

Active Disabled ▼
 Number 1074
 Delay Time 4 (0~5s)

ACD

ACD Activated Auto Disabled ▼
 ACD Activated Auto Timer 90 (0~180s)

Sections	Description
Call Pickup	<ul style="list-style-type: none"> ● Visual BLF Pickup Alert: To enable the BLF pick up function.
Intercom	<p>Intercom allows user to establish a call directly with the callee.</p> <ul style="list-style-type: none"> ● Active: To enable or disable Intercom feature. ● Intercom Mute: If enabled, once the call established, the callee will be muted.
HotLine	<p>HotLine allows user to call out a defined number automatically after hearing the dailtone without dialing any number.</p> <ul style="list-style-type: none"> ● 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 dailtone.

ACD	<p>ACD (Automatic Call Distribution) is most used in call-center market, it allows IP phone to login the system with multiple status so that the system could distribute an incoming calls to available user or agents.</p> <ul style="list-style-type: none"> ● ACD Activated Auto: If enabled, Agent status will turn to available automatically within a designated time(decided by ACD Activated Auto Timer). ● ACD Activated Auto Timer: To configure interval when will Agent status becomes available automatically. ● ACD Pwd Required: To enable the ACD password.
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Remote Control

Allowed Access IP List

Keypad Lock

Keypad Lock Type Disabled ▼

Keypad Unlock PIN (0~15)

Keypad Lock Timeout (0~3600s)

Key As Send

Key As Send # ▼

SIP Config

SIP Session T1 (0.5~10s)

SIP Session T2 (2~40s)

Sections	Description
Remote Control	<p>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.</p> <ul style="list-style-type: none"> ● Allowed Access IP List: To configure the allowed host address. <p>Note: For now, IP phone can only support IP address, IP address list and IP address pattern as allowed hosts</p>
Keypad Lock	<p>Keypad Lock allows to lock the keypad of your phone temporarily when you are not using it. This feature helps to protect your phone from unauthorized use.</p> <ul style="list-style-type: none"> ● Keypad Lock Type: To lock the phone with function keys or all keys; ● Keypad Unlock PIN: To lock the phone with a password. ● Keypad Lock Timeout: The idle interval to lock the

	phone.
SIP Config	Setup the SIP protocol package interval. T2 is maximum. The interval should be larger the T1, but less then T2.

UACSTA

UACSTA Active Disabled ▾

Register Name

Password

Server IP Port

Control Account Account 1 ▾

Others

Return Code When Refuse 486(Busy Here) ▾

Auto Answer Delay (0~5s)

Early DTMF Disabled ▾

Display DTMF Enabled ▾

Multicast Codec PCMU ▾

Direct IP Enabled ▾

Sections	Description
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.
Others	<ul style="list-style-type: none"> ● 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 ● Multicast codec: To choose different multicast codec ● Direct IP: To enable or disable direct IP call.

5.9. Phone->Voice

Path: Web UI->Phone->Voice

Voice

Echo Canceller

Echo Canceller	<input type="text" value="Enabled"/>
VAD	<input type="text" value="Disabled"/>
CNG	<input type="text" value="Enabled"/>

Jitter Buffer

Jitter Type	<input type="text" value="Adaptive"/>
Min Delay	<input type="text" value="0"/> (0~1000ms)
Nominal Delay	<input type="text" value="120"/> (0~1000ms)
Max Delay	<input type="text" value="300"/> (0~1000ms)

Mic Volume

Handset Volume	<input type="text" value="7"/> (1~15)
Headset Volume	<input type="text" value="8"/> (1~15)
Hand Free Volume	<input type="text" value="8"/> (1~15)

Sections	Description
Echo Canceller	<p>Echo Canceller: To remove acoustic echo from a voice communication in order to improve the voice quality .</p> <ul style="list-style-type: none"> ● 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

	<p>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.</p> <p>IP phones support two types of jitter buffers: fixed and adaptive.</p> <p>Fixed: Add the fixed delay to voice packets. You can configure the delay time for the static jitter buffer on IP phones.</p> <p>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.</p>
Mic Volume	<p>To configure Microphone volume for headset, handset and speaker mode.</p>

5.10. Phone->Key/Display

Path: Web UI->Phone->Key/Display

Key/Display

Line Key

Key	Type	Label	Value	Account	Extension
Line Key 1	Account ▼	<input type="text"/>	<input type="text"/>	Account 1 ▼	<input type="text"/>
Line Key 2	Account ▼	<input type="text"/>	<input type="text"/>	Account 2 ▼	<input type="text"/>
Line Key 3	Account ▼	<input type="text"/>	<input type="text"/>	Account 3 ▼	<input type="text"/>

Sections	Description
Line Key	<p>Allows user to assign specific feature to the designated line key.</p> <p>For line key, the available feature list: DND, Menu, MSG, Status, Book, Fwd, Pickup, Group Pickup, Intercom, Speed Dial, History, Favorites, Redial, Account, ACD, BLF, BLFList, Call Return, Hot Desking, Record, XML Browser, DTMF,XML VoiceMail,XML PhoneBook, XML History,Multicast paging.</p>

Soft Key

Key	Type	Label	Value	Account
Soft Key 1	History ▼	<input type="text"/>	<input type="text"/>	Auto ▼
Soft Key 2	Book ▼	<input type="text"/>	all ▼	Auto ▼
Soft Key 3	DND ▼	<input type="text"/>	<input type="text"/>	Auto ▼
Soft Key 4	Menu ▼	<input type="text"/>	<input type="text"/>	Auto ▼

Function Key

Key	Type	Value	Account
OK	Status ▼	<input type="text"/>	Auto ▼
Cancel	N/A ▼	<input type="text"/>	Auto ▼
RD	Redial ▼	<input type="text"/>	Auto ▼
Mute	N/A ▼	<input type="text"/>	Auto ▼
Voice Message	Voice Mess. ▼	<input type="text"/>	Auto ▼
Transfer	N/A ▼	<input type="text"/>	Auto ▼
Hold	N/A ▼	<input type="text"/>	Auto ▼
Conference	N/A ▼	<input type="text"/>	Auto ▼

Display

Backlight Intensity	<input type="text" value="4"/> ▼
Backlight Time	<input type="text" value="30"/> ▼
LineKey Display Type	<input type="text" value="Rolling"/> ▼

Sections	Description
Soft Key	<p>Allows user to assign specific feature to the designated soft keys.</p> <p>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, Simple Menu.</p>
Function Key	<p>Allows user to assign specific feature to the designated function keys.</p> <p>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.</p>
Display	<ul style="list-style-type: none"> ● 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.

5.11. Phone->Ring tones

Path: Web UI->Phone->Ringtones

Ringtones

All Ringtones

Upload(Max Total Size: 100K) 选择文件 未选择任何文件

Uploaded Ringtones ▼

System Ringtones ▼

Distinctive Ringers

Index	Keyword	Ringtone
0	<input type="text"/>	Ring1.wav ▼
1	<input type="text"/>	Ring1.wav ▼
2	<input type="text"/>	Ring1.wav ▼
3	<input type="text"/>	Ring1.wav ▼
4	<input type="text"/>	Ring1.wav ▼
5	<input type="text"/>	Ring1.wav ▼
6	<input type="text"/>	Ring1.wav ▼
7	<input type="text"/>	Ring1.wav ▼
8	<input type="text"/>	Ring1.wav ▼
9	<input type="text"/>	Ring1.wav ▼
10	<input type="text"/>	Ring1.wav ▼
11	<input type="text"/>	Ring1.wav ▼

Sections	Description
All Ringtones	<p>Allow user to upload and view ringtone files or delete uploaded ringtone files.</p> <p>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.</p> <p>System ringtones files cannot be deleted thus user can only delete uploaded ringtones.</p>
Distinctive Ringers	<p>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</p>

	ringtones to play.
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5.12. Phone->Tones

Path: Web UI->Phone->Tones

Tone

Select Country	Default ▾
Ring Back	<input type="text"/>
Dial	<input type="text"/>
Call Waiting	<input type="text"/>
DTMF 0	<input type="text"/>
DTMF 1	<input type="text"/>
DTMF 2	<input type="text"/>
DTMF 3	<input type="text"/>
DTMF 4	<input type="text"/>
DTMF 5	<input type="text"/>
DTMF 6	<input type="text"/>
DTMF 7	<input type="text"/>
DTMF 8	<input type="text"/>
DTMF 9	<input type="text"/>
DTMF *	<input type="text"/>
DTMF #	<input type="text"/>

Sections	Description
Tone	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.

5.13. Phone->Dial Plan->Replace Rule

Path: Web UI->Phone->Dial Plan->Replace Rule

Dial Plan

Rules Replace Rule ▼

Index	Account	Prefix	Replace	<input type="checkbox"/>
1	Auto	171	172	<input type="checkbox"/>
2				<input type="checkbox"/>
3				<input type="checkbox"/>
4				<input type="checkbox"/>
5				<input type="checkbox"/>
6				<input type="checkbox"/>
7				<input type="checkbox"/>
8				<input type="checkbox"/>
9				<input type="checkbox"/>
10				<input type="checkbox"/>

Add
Edit
Delete

Rules Modify >>

Account Auto ▼

Prefix

Replace

Submit
Cancel

Area Code

Code

Min Length 1 (1~15)

Max Length 1 (1~15)

Account Auto ▼

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	<p>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.</p> <p>Note: There is only one area code rule supported.</p>

5.14. Phone ->Dial Plan->Dial Now

Path:Web UI->Phone->Dial Plan->Dial Now

Dial Plan

Rules Dial Now ▾

Index	Account	Dial Now Rule	
1	Auto	172	<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>
10			<input type="checkbox"/>

Add
Edit
Delete

Rules Modify >>

Account Auto ▾

Dial Now Rule

Submit
Cancel

Dial Now Delay

All Dial Delay Disabled ▾

Dial Now Delay (0~15s)

Area Code

Code

Min Length (1~15)

Max Length (1~15)

Account Auto ▾

Sections	Description
Rules	Allow user to select Replace rule or Dial-now to display or edit.
Dial Now Delay	Allow user 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.

5.15. Phone →Action URL

Path:Web UI->Phone->Action URL

ActionURL	
Active	Enabled
Setup Completed	Completed: http://192.168.10.11/help.xml?mac=\$rr
Registered	Registered: http://192.168.10.11/help.xml?mac=\$rr
Unregistered	
Registered Failed	
Off Hook	
On Hook	
Incoming Call	
Outgoing Call	
Established	
Terminated	
Open DND	
Close DND	
Open Always FWD	
Close Always FWD	
Open Busy FWD	
Close Busy FWD	
Open No Answered FWD	
Close No Answered FWD	
Transfer Call	
Blind Transfer	
Attended Transfer	
Hold	
UnHold	
Mute	
UnMute	
MissedCall	
IP Changed	
FWD Incoming Call	
Reject Incoming Call	
Answer New Call	
Transfer Finished	
Transfer Failed	
Idle To Busy	
Busy To Idle	

Sections	Description
Action URL	<p>To display and configure Action URL settings.</p> <p>Setup Completed: When the IP phone completes startup.</p> <ul style="list-style-type: none"> ● 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.

	<ul style="list-style-type: none"> ● 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.
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5.16. Phone->Multicast

Multicast

Multicast Setting

Paging Barge 1 ▾

Paging Priority Active Enabled ▾

Priority List

IP Address	Listening Address	Label	Priority
1 IP Address	<input type="text" value="224.1.6.16:22006"/>	<input type="text" value="test1"/>	1
2 IP Address	<input type="text"/>	<input type="text"/>	2
3 IP Address	<input type="text"/>	<input type="text"/>	3
4 IP Address	<input type="text"/>	<input type="text"/>	4
5 IP Address	<input type="text"/>	<input type="text"/>	5
6 IP Address	<input type="text"/>	<input type="text"/>	6
7 IP Address	<input type="text"/>	<input type="text"/>	7
8 IP Address	<input type="text"/>	<input type="text"/>	8
9 IP Address	<input type="text"/>	<input type="text"/>	9
10 IP Address	<input type="text"/>	<input type="text"/>	10

Sections	Description
Multicast Setting	To display and configure the Multicast setting. <ul style="list-style-type: none"> ● 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. <ul style="list-style-type: none"> ● Listening Address: Enter the IP address you need to listen ● Label: Input the label for each listening address

5.17. PhoneBook->Local Phone Book

Path:Web UI->PhoneBook->Local Book

Local Book

Contact

Search

Dial

All Contacts ▼

All Contacts
 Favorites
Black List

Auto ▼

Index	Name	Office Num	Mobile Num	Other Num	Group	Ring	Account	<input type="checkbox"/>
1	AAA	171	171	171	Default	Auto	Auto	<input type="checkbox"/>
2	DG	12	132	3	Default	Auto	Auto	<input type="checkbox"/>
3								<input type="checkbox"/>
4								<input type="checkbox"/>
5								<input type="checkbox"/>
6								<input type="checkbox"/>
7								<input type="checkbox"/>
8								<input type="checkbox"/>
9								<input type="checkbox"/>
10								<input type="checkbox"/>

Page 1 ▼

Move To
All Contacts ▼

Contact Setting

Name

Office Num

Mobile Num

Other Num

Group Default ▼

Ring Auto ▼

Account Auto ▼

Sections	Description
Contact	To display and select local contact type. <ul style="list-style-type: none"> ● 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.
Group Setting	To display or change Group name, related ringtone or description.
Import/Export	To import or export the contact or blacklist file.

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5.18. Phone Book->Remote Phone Book

Path:Web UI->PhoneBook->Remote Book

Remote Book

Remote Book

Index	Local Book URL	Local Book Name
1	ftp://192.168.10.53/remote phone book.xml	test2
2		
3		
4		
5		

Search Remote Phonebook Name Enabled ▾

Refresh Interval 3600 (120~2592000s)

Submit
Cancel

Sections	Description
Remote Book	<p>To display and configure Remote Book settings.</p> <ul style="list-style-type: none"> ● Index: To select desired Remote Book item to display and configure. ● Local Book URL: To configure remote book server address ● Local Book Name: To configure display remote book name on Phone UI <p>Search Remote Phonebook Name: To enable or disable search remote phonebook name</p> <p>Search Flash Interval: To set interval (Range from 120s to 2592000s)</p> <p>Note: IP phone support at most 5 remote books. Please refer to your administrator for how to establish a remote book server and how to create remote book xml file.</p>

5.19. Phone Book->Call log

Path:Web UI->PhoneBook ->Call Log

Call Log							
Call History		All	<input type="button" value="Hand Up"/>				
Index	Type	Date	Time	Local Identity	Name	Number	<input type="checkbox"/>
1	Forwarded	2016-09-07	14:06:26	192.168.10.1 91@192.168.1 0.191	192.168.10.230	192.168.10.2 30@192.168.1 0.230	<input type="checkbox"/>
2	Missed	2016-09-07	14:05:56	192.168.10.1 91@192.168.1 0.191	192.168.10.230	192.168.10.2 30@192.168.1 0.230	<input type="checkbox"/>
3	Received	2016-09-07	14:05:44	192.168.10.1 91@192.168.1 0.191	192.168.10.230	192.168.10.2 30@192.168.1 0.230	<input type="checkbox"/>
4	Dialed	2016-09-07	14:05:28	192.168.10.1 91@192.168.1 0.191	Unknown	192.168.10.2 14@192.168.1 0.214	<input type="checkbox"/>
5							<input type="checkbox"/>
6							<input type="checkbox"/>
7							<input type="checkbox"/>
8							<input type="checkbox"/>
9							<input type="checkbox"/>
10							<input type="checkbox"/>
11							<input type="checkbox"/>
12							<input type="checkbox"/>
13							<input type="checkbox"/>
14							<input type="checkbox"/>
15							<input type="checkbox"/>
Page 1		<input type="button" value="Prev"/>	<input type="button" value="Next"/>	<input type="button" value="Delete"/>	<input type="button" value="Delete All"/>		

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

5.20. Phone Book->LDAP

Path:Web UI->PhoneBook->LDAP

LDAP

LDAP

Name Filter	<input type="text" value="((cn=#)(sn=#))"/>
Number Filter	<input type="text" value="((telephoneNumber=#)(mobileNumber=#))"/>
Server	<input type="text" value="192.168.10.31"/>
Port	<input type="text" value="389"/> (1~65535)
Base DN	<input type="text" value="ou=Group1,o=RL,c=cn"/>
User Name	<input type="text" value="cn=admin,o=RL,c=cn"/>
Password	<input type="password" value="••••••••"/>
Name Attribute	<input type="text" value="sn cn"/>
Number Attribute	<input type="text" value="telephoneNumber mobile home"/>
Display Name	<input type="text" value="sn cn"/>
Max Hits	<input type="text" value="50"/> (1~500)
Search Delay Time	<input type="text" value="1000"/> (200~3000)ms

Sections	Description
LDAP	<p>To display and configure LDAP phonebook settings.</p> <ul style="list-style-type: none"> ● Name Filter: The settings used to tell LDAP server what name attributes to search. ● Number Filter: The settings used to tell LDAP server what number attributes to search. ● Server: To configure LDAP server’s address. ● Port: To configure LDAP server’s port. ● Base DN: To configure searching base DN on LDAP server. ● User Name: To configure user name for accessing LDAP server. ● Password: To configure password for accessing LDAP server. ● Name Attribute: To configure which name attributes should be feedback from LDAP server. ● Number Attribute: To configure which number attributes should be feedback from LDAP server. ● Display Name: To configure display name on Phone UI when there is any searching result from LDAP server. ● Max Hits: To configure the maximum size of result response from LDAP server. ● Search Delay Time: To configure delay time before initiate LDAP searching request after you input a value from Phone UI. <p>Note: For setting details, please consult with your system administrator for further information.</p>

5.21. Phone Book->BroadSoft

Path:Web UI->PhoneBook->Broadsoft

Broadsoft

Broadsoft PhoneBook

PhoneBook Item: ▼

Display Name: ▼

Server Address:

Server Port: (1~65535)

User Name:

Password:

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.

5.22. Upgrade->Basic

Path:Web UI->Upgrade->Basic

Upgrade-Basic

Firmware Version: 53.0.6.137

Hardware Version: 53.0.5.0.16.0.0.0

Upgrade: 未选择任何文件

Reset To Factory Setting:

Reboot:

Sections	Description
Upgrade	To select upgrading rom file from local or a remote server

	<p>automatically. Note: Please make sure it's right file format for right model.</p>
Firmware version	<p>To display firmware version, firmware version starts with MODEL name. For example, R53 firmware version should be like 53.xxx.xxx.xxx.</p>
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.

5.23. Upgrade->Advanced

Path: Web UI->Upgrade->Advanced

Upgrade-Advanced

PNP Option

PNP Config Enabled ▾

DHCP Option

Custom Option (128~254)
 (DHCP Option 66/43 is Enabled by Default)

Sections	Description
PNP Option	<p>To display and configure PNP setting for Auto Provisioning.</p> <ul style="list-style-type: none"> ● 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	<p>To display and configure custom DHCP option.</p> <ul style="list-style-type: none"> ● 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

URL	<input type="text" value="http://192.168.10.29"/>
User Name	<input type="text" value="administrator"/>
Password	<input type="password" value="••••••••"/>
Common AES Key	<input type="password" value="••••••••"/>
AES Key(MAC)	<input type="password" value="••••••••"/>

Automatic Autop

Mode	<input type="text" value="Power On"/>
Schedule	<input type="text" value="Sunday"/>
	<input type="text" value="22"/> Hour(0~23)
	<input type="text" value="0"/> Min(0~59)
Clear MD5	<input type="button" value="Submit"/>
Export Autop Template	<input type="button" value="Export"/>

Sections	Description
Manual Update Server	<p>To display and configure manual update server’s settings.</p> <ul style="list-style-type: none"> ● 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). <p>Note: AES is one of many encryption, it should be configure only configure filed is ciphered with AES, otherwise left blank.</p>
AutoP	<p>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.</p>

System Log

LogLevel 3 ▾

Export Log

Remote System Log Disabled ▾

Remote System Server

PCAP

PCAP

PCAP Auto Refresh Disabled ▾

Others

Config File(.tgz/.conf/.cfg) 未选择文件。

(Encrypted)

Sections	Description
System Log	<p>To display syslog level and export syslog file.</p> <ul style="list-style-type: none"> ● 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	<p>To start, stop packets capturing or to export captured Packet file.</p> <ul style="list-style-type: none"> ● Start: To start capturing all the packets file sent or received from IP phone. ● Stop: To stop capturing packets. <p>Note: IP phone will save captured packets file to a temporary file, this file maximum size is 1M(mega bytes), and will stop capturing once reaching this maximum size.</p>
Others	<p>To display or configure others features from this page.</p> <ul style="list-style-type: none"> ● Config file: To export or import configure file for IP phone.

Note: Please refer to Auto Provisioning manual for instructions how to do Auto Provisioning for Akuvox SIP-R5X series IP phone.

5.24. Security->Basic

Path:Web UI->Security->Basic

Security-Basic

Web Password Modify

User Name

Current Password

New Password

Confirm Password

admin ▾

Session Time Out

Session Time Out Value

300 (60~14400s)

Sections	Description
Web Password Modify	To modify user's password. <ul style="list-style-type: none"> 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	Over the session time out value, users need to login in the web again. <ul style="list-style-type: none"> Session Time Out Value: the ranger is from 60s to 14400s.

5.25. Security->Advanced

Path:Web UI->Security->Advanced

Advanced

Web Server Certificate

Index	Issue To	Issuer	Expire Time	Delete
1	Akuvox	Akuvox	Sun Oct 9 16:00:00 2034	<input type="button" value="Delete"/>

Web Server Certificate Upload

未选择文件。

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.

Client Certificate

Index	Issue To	Issuer	Expire Time	
1	AK	Akuvox	Sun May 28 06:21:54 2014	<input type="checkbox"/>
2				<input type="checkbox"/>
3				<input type="checkbox"/>
4				<input type="checkbox"/>
5				<input type="checkbox"/>
6				<input type="checkbox"/>
7				<input type="checkbox"/>
8				<input type="checkbox"/>
9				<input type="checkbox"/>
10				<input type="checkbox"/>

Client Certificate Upload

Index

未选择文件。

Only Accept Trusted Certificates

Auto ▾

Disabled ▾

Sections	Description
Web Server Certificate Upload	To upload a certificate file, this 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 which is used as client certificate. <ul style="list-style-type: none"> ● Only Accept trusted Certificates: If this option is enabled, only trusted certificates will be accepted.

6. Troubleshooting

Issue 1: The LCD does not light up

- Check the AC power adapter. Make sure it is the one provided in your package.
- Check the power outlet. Make sure that the power that outlet you are plugging your device into is working. Try to plug a different device into the socket to make sure it has power.

Issue 2: No signal tone heard from the handset

- Check the connection cord between the handset and the phone. Make sure it is connected properly.

Issue 3: Cannot access the web interface

- Check the connection between the PC port of the device and the network port of the computer. Make sure it is fine.
- Check whether the IP address of the device is correct.
- If it is LAN, please make sure there is no IP address collision with other devices on the network.

Issue 4: Cannot call out

- Please see the network connection status of device, if it is exception, and then check the connection of network.
- If the network connection is normal, please check whether the device has registered successfully.
- If the network connection and the registered are both normal, please confirm whether the dial rule is correct, or please communicate with the service operator.

7. Appendix : Time Zones

Time Zone	Time Zone Name
-11:00	Samoa
-10:00	United States-Hawaii-Aleutian
-10:00	United States-Alaska-Aleutian
-09:00	United States-Alaska Time
-08:00	Canada(Vancouver, Whitehorse)
-08:00	Mexico(Tijuana, Mexicali)
-08:00	United States-Pacific Time
-07:00	Canada(Edmonton, Calgary)
-07:00	Mexico(Mazatlan, Chihuahua)
-07:00	United States-Mountain Time
-07:00	United States-MST no DST
-06:00	Canada-Manitoba(Winnipeg)
-06:00	Chile(Easter Islands)
-06:00	Mexico(Mexico City, Acapulco)
-06:00	United States-Central Time
-05:00	Bahamas(Nassau)
-05:00	Canada(Montreal, Ottawa, Quebec)
-05:00	Cuba(Havana)
-05:00	United States-Eastern Time
-04:30	Venezuela(Caracas)
-04:00	Canada(Halifax, Saint John)
-04:00	Chile(Santiago)
-04:00	Paraguay(Asuncion)
-04:00	United Kingdom-Bermuda(Bermuda)
-04:00	United Kingdom(Falkland Islands)
-04:00	Trinidad&Tobago
-04:00	Curacao
-03:30	Canada-New Foundland(St.Johns)

Time Zone	Time Zone Name
-03:00	Denmark-Greenland(Nuuk)
-03:00	Argentina(Buenos Aires)
-03:00	Brazil(no DST)
-03:00	Brazil(DST)
-02:00	Brazil(no DST)
-01:00	Portugal(Azores)
0	GMT
0	Greenland
0	Denmark-Faroe Islands(Torshavn)
0	Ireland(Dublin)
0	Portugal(Lisboa, Porto, Funchal)
0	Spain-Canary Islands(Las Palmas)
0	United Kingdom(London)
0	Morocco
+01:00	Albania(Tirane)
+01:00	Austria(Vienna)
+01:00	Belgium(Brussels)
+01:00	Caicos
+01:00	Chatam
+01:00	Croatia(Zagreb)
+01:00	Czech Republic(Prague)
+01:00	Denmark(Kopenhagen)
+01:00	France(Paris)
+01:00	Germany(Berlin)
+01:00	Hungary(Budapest)
+01:00	Italy(Rome)
+01:00	Luxembourg(Luxembourg)
+01:00	Macedonia(Skopje)
+01:00	Netherlands(Amsterdam)
+01:00	Namibia(Windhoek)
+02:00	Estonia(Tallinn)

Time Zone	Time Zone Name
+02:00	Finland(Helsinki)
+02:00	Gaza Strip(Gaza)
+02:00	Greece(Athens)
+02:00	Israel(Tel Aviv)
+02:00	Jordan(Amman)