AT810 series User Manual

Version: 1.3

2014-03-31

Release note

Version	Changed note	Editor	Date
1.0	First Release	Stefen	2013-1-25
1.1	Page 9: Packing List 1. Add Ethernet Cable(1.5m)	Jessie	2013-5-13
1.2	Web Settings -> Account Add parameters: 1.Session Timer Enable 2.Early Update Enable 3.Caller ID Display 4.AutoSubscribeMWBEnable 5.Server List 6.Dns Mode 7.Blf List Uri 8.G711A 9.G711U 10.AVT 11.G722 12.G723 13.G729ab 14.iLBC 15.G726-16 16.G726-24 17.G726-32 18.G726-40	Jason	2013-8-29
1.3	Web Settings -> Account Add parameter: 1.SIP Redundancy Server	Aimee	2014-3-31

Content

Contact ATCOM

The Introduction of ATCOM

ATCOM is the leading VoIP hardware manufacturer in global market. We have been keeping innovating with customer's needs oriented , working with partners to establish a total solution for SMB VoIP with IP phone , IP PBX and Asterisk cards.

With over 10 years' experience of R&D, manufacturing and service in network and VoIP filed, mission of creating the biggest value for IP terminals, we commit ourselves in supplying the competitive IP phone and other terminals for IP PBX, softswitch, IMS, NGN providers and carriers; supplying the competitive total VoIP solution for SMB market. We keep improving the customer's experience and creating the bigger value with our reliable products. Until now, our VoIP products has been sold to over 60 countries and used by millions of end users.

Contact Sales

Address	Area C, A2F , Block 3 ,Huangguan Technology Park , #21 Tairan 9th Rd, Chegongmiao , Futian District , Shenzhen China
Tel	+ (86) 755-83018618-8888
Fax	+ (86) 755-83018319
E-mail	sales@atcomemail.com

Contact Technical Support

Tel	+ (86) 755-83018618-8011
E-mail	Support@atcomemail.com

Website Address: http://www.atcom.cn/

Download Center: http://www/atcom/cn/download.html

1. AT810 Series Overview

AT810 is almost the same with AT810P without PoE.

AT810/AT810P

Type	POE	Power adapter
AT810	No	Standard accessory
AT810P	Yes	Optional accessory

1.1 Interfaces

➢ Power input: DC 12V, 500mA.

➢ WAN: RJ45 port.

➢ PC: RJ45 port.

➢ Headset port: RJ9 port.

➢ Handset port: RJ9 port.

1.2 Hardware

➢ LCD: Block code LCD

➢ FLASH: 8M.

➢ RAM: 16M.

➢ CPU: 262MHz Dual Core.

➢ LED indicator: 1 Status Light

1.3 Software

➢ Supports Sip 2.0 (RFC3261) and other related SIP RFCs.

➢ Supports web control.

➢ Supports Syslog.

➢ Supports G.711A/U, G722, G.723, G.726, G.729, iLBC Codec.

➢ Supports STUN.

➢ Supports 1 SIP account.

➢ Supports SIP domain，SIP authentication.

➢ Supports DTMF (Inband, RFC2833 and Info).

➢ Supports voicemail.

➢ Supports dial rule, IP to IP call.

➢ Supports network time synchronization (NTP).

➢ Supports speed dial.

➢ Supports Call Forward, Call transfer, 3-way conference, Call hold, Call waiting, Redial.

➢ Supports VAD, CNG，Jitter Buffer.

➢ Supports DND (Do Not Disturb), Auto Answer, Blacklist, Block Call ID, Block Anonymous Call.

➢ Supports HTTP, FTP, TFTP updating the configuration and firmware.

➢ Supports auto provision for config file upgrade.

➢ Supports phone book with 100 whitelist records and 50 blacklist records.

➢ Supports call log with 50 answered calls 50 missed calls, 50 dialed calls.

1.4 Network

➢ WAN port: Static IP, DHCP and PPPoE.

➢ PC port: Bridge mode

➢ Supports VLAN (DATA VLAN and VOICE VLAN).

➢ Supports L2TP VPN.

➢ Supports Qos, Qos supports Diffserv.

➢ Supports NAT.

1.5 Management and Maintenance

➢ Supports safe mode and firmware updating under safe mode.

➢ Supports different level user management.

➢ Configuration via web, keyboard.

➢ Firmware and configuration file auto provision.

➢ Firmware and configuration updating via HTTP, FTP and TFTP.

➢ Supports system log and call log.

1.6 Protocol

➢ IEEE 802.3 /802.3u 10 Base T/100Base TX.

➢ PPPoE：PPP over Ethernet.

➢ SIP RFC3261, RFC3262, RFC3263, RFC3264, RFC3265, RFC2543, RFC3489, RFC3842, RFC3515, RFC2976, RFC3428, RFC2327, RFC2782, RFC1889.

➢ TCP/IP: Transfer Control Protocol/Internet Protocol.

➢ RTP: Real-time Transport Protocol.

➢ RTCP: RTP Control Protocol.

➢ SRTP: Secure Real-time Transport Protocol

➢ VAD/CNG: Voice Activation Detection/Comfort Noise Generaror.

➢ DNS: Domain Name Server.

➢ TFTP: Trivial File Transfer Protocol.

➢ HTTP:Hypertext Transfer Protocol.

➢ FTP: File Transfer Protocol.

1.7 Compliant Standard

➢ CE：EN55024, EN55022.

➢ Comply with ROHS in EU.

➢ Comply with ROHS in China.

➢

Explanation: The letter "e" is the first letter of "environment" and "electronic". The rim is a round with two arrows, stands for recycles. The number 20 stands for the years of environment protection. Please note the years of environment protection is not discarding year nor usage life.

1.8 Operating Requirement

➢ Operation temperature: 0 to 45° C (32° to 113° F)

➢ Storage temperature: -5° to 55° C (23° to 131° F)

➤   Humidity: 10 to 90% no dew

1.9 Packing List

Model	Phone	Handset	Handset line	Power adapter	Ethernet Cable	Manual CD	Feet underprop
AT810	1	1	1	1,Standard accessory	1,1.5m	1	2
AT810P	1	1	1	Optional accessory	1,1.5m	1	2

Note: Power adapter Input: AC 100~240V, 50/60Hz; Output: 12V, 500mA.

1.10 Installation

Use Ethernet cable to connect the phone's WAN port and computer on the same switch, the phone defaults to use DHCP to obtain IP from switch. The following is the connection diagram of application:
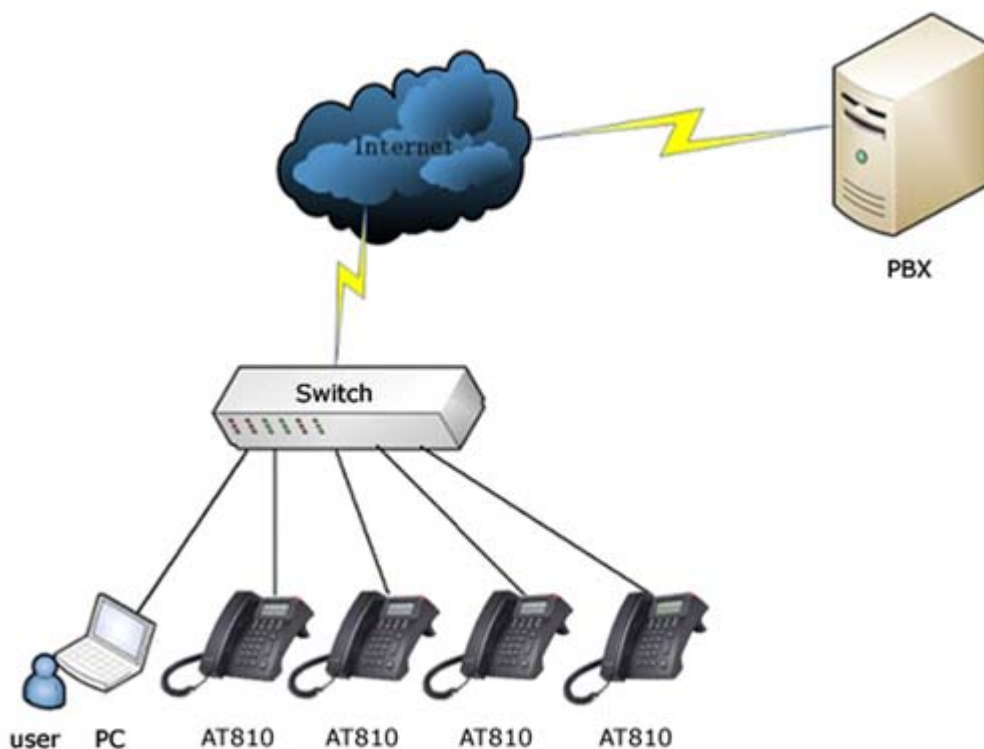
Steps to configure the phone via web:

➤ Press "Info" button for several times, user will see the information in sequence: IP address, Gateway and Account number.

➤ Open web browser and key in the IP address to enter the configuration page.

➢ Click on "admin" to configure under admin mode.

The default username is admin for administrator and user for user. They were defaults with no password. User can set it in the webpage.

1.11 Feet installation instruction

1.11.1 Desktop position

Put the bottom side of the IP phone upside and press the plate with letter "PUSH" into the slot, please refer the picture as below:
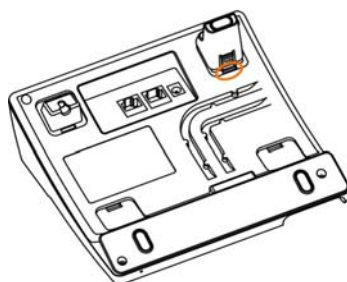
A. Press the other plate into the slot in accordance with the direction of the arrow.
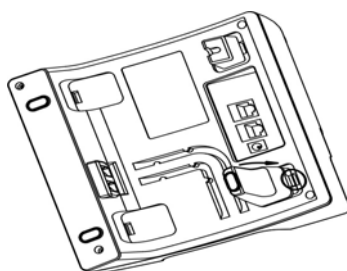
B. Repeat A and B. It is the right picture of putting on desk after fixing the two feet below:

C. Disassemble the feet:

Press the plate with word "PUSH" and pull the feet with the direction of arrow. When the plate is pull out of the slot (there will be a sound of "pa") user can take off the feet.
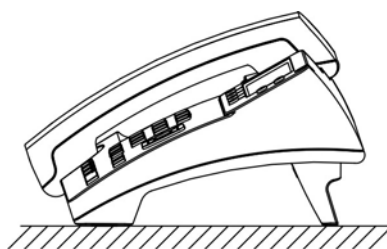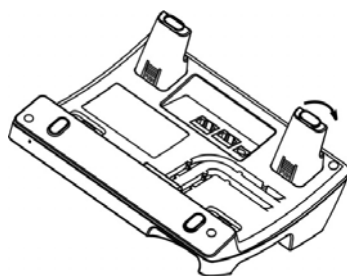
1.11.2 Wall position
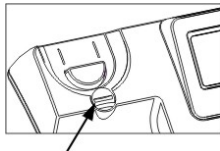
A. Put the bottom side of the IP phone upside and push the plate with letter "PUSH" into the slot, please refer the picture as below:

B. Push the other plate into the slot in accordance with the direction of the arrow.
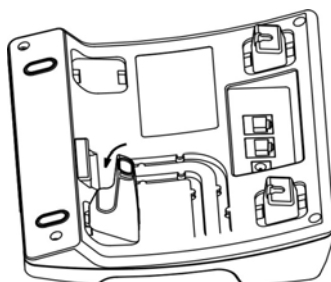
C. Repeat A and B. It is the picture of wall mounting after fixing the two feet below:
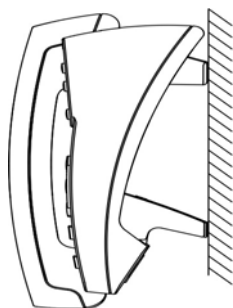
Attention: Please rotate the hook to the position as in picture with a coin or other tools
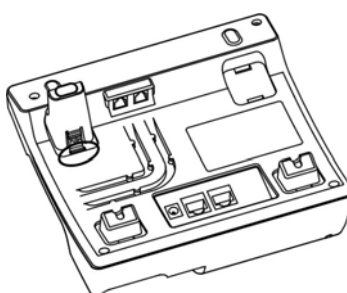
D. Disassemble the feet way:

Press the plate with word "PUSH" and pull the feet with the direction of arrow. When the plate is pull out of the slot (there will be a sound of "pa") user can take off the feet.
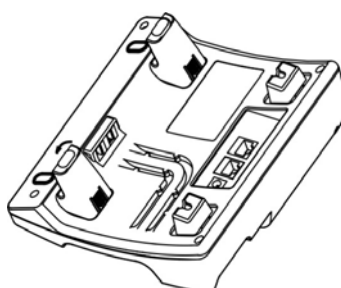
2. Keypad of IP Phone
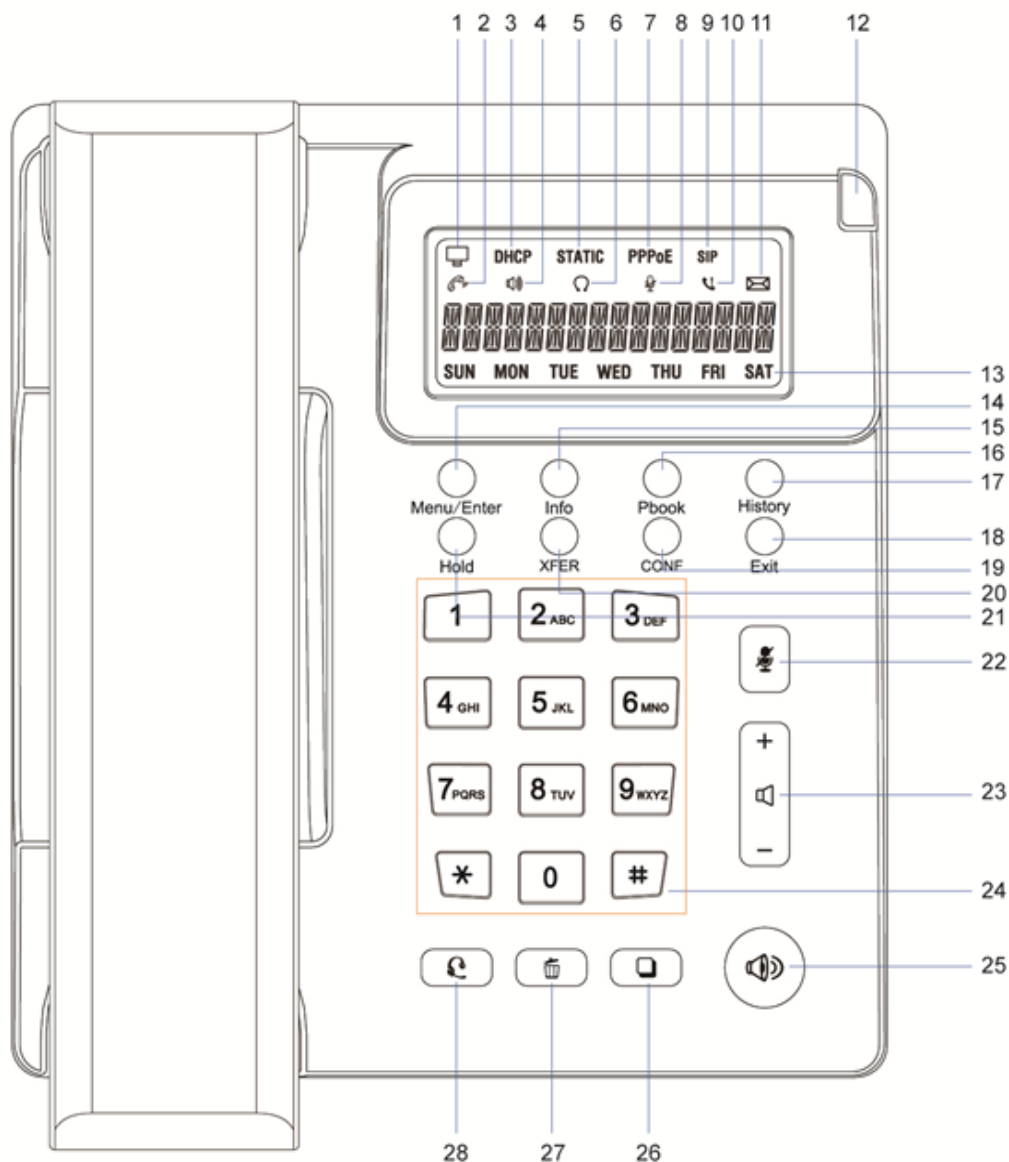
Keypad Map of AT810/AT810P

Description of indicators and buttons：

Number	Function description
1	Network status indicator: ➢ When the network connection is successful, it's on; ➢ When the network connection is unsuccessful, it blinks.
3. 5. 7	Network mode indicator: ➢ Number 3 stands for DHCP, When the IP phone gets IP address in DHCP mode, the indicator is on, or it blinks; ➢ Number 5 stands for Static, when the IP phone gets IP address in Static mode, the indicator is on, or it blinks;

	➢ Number 7 stands for PPPoE, when the IP phone gets IP address in PPPoE mode, the indicator is on, or it blinks.
2、4、6	Answering mode indicator: ➢ Number 2 stands for answering call with handset; ➢ Number 4 stands for answering call in handfree mode; ➢ Number 6 stands for answering call with headset.
8	Mute indicator: When the phone is muted, this indicator is on, or it's off.
9	Protocol indicator: ➢ When the register is enabled and the phone successfully register to SIP server, this indicator is on; ➢ When the register is enabled but the phone does not successfully register to server, this indicator blinks; ➢ When the register is disabled, this indicator is off.
10	Missed call indicator: When there is a missed call, this indicator will be on, and it will turn off when user checks the missed call information.
11	Voicemail indicator: When there is a new voice mail, this indicator will be on, only after all the voicemail have been checked, this indicator will turn off.
12	incoming call indicator ➢ When the phone is in standby without any incoming call, the indicator is off. ➢ When there is a missed call, this indicator blinks; ➢ When there is an incoming call, this indicator blinks; ➢ When there is a new voicemail, the indicator blinks.
13	Monday to Sunday indicator : The correct day will be on while others will be off.
14	This is the multi-function key ➢ In standby mode , press "Menu/Enter" to enter the Menu; ➢ In functional menu, press "Menu/Enter" to confirm selection.
15	Press "Info" to check network and registration status.
16	Press "Pbook" to enter the phone book and add or delete or edit.
17	Press "History" button to check call history including incoming calls, outgoing calls and missed calls.
18	Exit button: ➢ When dialing , press "Exit" to delete the dialed number and so user can dial a new number; ➢ In functional menu, press "Exit" to exit current menu and return to the upper one.
19	conference key : When user want to realize conference call, press "CONF" button and key in the third part number.
20	Transfer key, when need to transfer the current call to others,

	press it.
21	Press "Hold" to hold the call when talking, once more to recover.
22	Mute key: Press "Mute", user could hear the voice from the other end but he could not get any voice from user.
23	Voice volume key: ➢ "+" for turning up the volume unless the volume is the loudest yet. ➢ "-" for turning down the volume unless the volume is the lowest yet.
24	Numeral keys.
25	Handfree button: Press to enter handfree mode.
26	Dial button: ➢ When finish dialing the telephone number, press this button to send out the phone number. ➢ After lifting the phone, this button can act as redial key.
27	Del button: Press to delete a single number or letter.
28	Headset button: Press to enter headset answering mode.

3. Menu Operation

Press "Menu/Enter" to enter menu, there will be two modes to select when it displays "INPUT PASSWORD" on LCD:

1. If user input the password (which defaults to 123), it's the admin mode;

2. If user do not input password but press "Menu/Enter" directly, it's the user mode, user have no rights to configure anything but browse them.

If the password is error, it will display "INPUT AGAIN 2" on LCD. Please press Menu/Enter key to input the password again. If user input the incorrect password three times, the phone will return to idle state automatically.   If user wants to modify the configurations via phone, "De" button is needed to be pressed firstly, and then user can set a new value, and press "Menu/Enter" to save it.

Level 1 Menu	Level 2 Menu	Level 3 Menu	Level 4 Menu
NETWORK			
	NET MODE		
		DHCP ON	
		PPPOE	

		STATIC	
	WAN		
		STATIC NET	
			IP
			DNS
			DNS2
			GATEWAY
			NETMASK
		PPPoE	
			USER NAME
			PASSWORD
SIP			
	SIP SERVER		
	PORT		
	NUMBER		
	PASSWORD		
	ACCOUNT		
	REGISTER		
		1 ON	
		0 OFF	
SYSTEM	TIME		
	REBOOT		
	SET DEFAULT		
DSP	HANDSET VOL		
	HEADSET VOL		
	SPEAKER VOL		
	RING VOL		
	RING TYPE		

4. Basic functions and operations

4.1 Answer the calls

When there is an incoming call, the phone will remind user with ring. There are 3 ways to answer the call:

✓ Answer by handset

Pick up the handset and talk with the caller. If user wants to hang up, just put back the handset. If user wants to switch in handset mode, headset mode and handfree mode, just press the handfree button or the headset button. Be attention: only after done the switch can user put the handset back, or the call will be ended.

✓ Answer by handfree

Press the handfree button in the phone and talk with callers by built-in Micro-phone and Speaker. If user wants to switch in handset mode, headset mode and handfree mode, pick up the handset to switch to handset mode, and press the headset button to switch into headset mode.

✓ Answer by headset

Keep the headset connected with the RJ9 headset jack, when there is an incoming call, press the headset button on the IP phone and then user can talk with the caller by the headset. If user wants to switch in handset mode, headset mode and handfree mode, pick up the handset to switch to handset mode, and press the handfree button to switch into the handfree mode.

4.2 Make Calls

✓ Use the handset

1. Pick up the handset then input the number with the keyboard and press "#" or "Dial" to send the number. After hearing the tones of "du~~du~~" with "CALLING" showed on the LCD, it means that the callee's phone is ringing. If the callee answers the call, the phone call will be established and the LCD will show the calling time and the callee's number.

2. Input the number then pick up the handset, the phone will dial the number automatically. After hearing the tones of "du~~du~~" with "CALLING" showed on the LCD, it means that the callee's phone is ringing. If the callee answers the call, the phone call will be established and the LCD will show the calling time and the callee's number.

✓ Use the handfree button

1. Press the Speaker Phone button then input the number with the keyboard and press "#" or "Dial" to send the number. After hearing the tones of "du~~du~~" with "CALLING" showed on the LCD, it means that the callee's phone is ringing. If the callee answers the call, the phone call will be established, and the LCD will show the calling time and the callee's number.

2. Input the number then press the handfree button, the phone will dial the number automatically. After hearing the tones of "du~~du~~" with "CALLING" showed on the LCD, it means that the callee's phone is ringing. If the callee answers the call, the phone call will be established and the LCD will show the calling time and the callee's number.

✓ Use the headset

1. Press the headset then input the number with the keyboard and press "#" or "Dial" to send the number. After hearing the tones of" du~~du~~" with "CALLING" showed on the LCD, it means that the callee's phone is ringing. If the callee answers the call, the phone call will be established and the LCD will show the calling time and the callee's number.

2. Input the number then press the headset button, the phone will dial the number automatically. After hearing the tones of "du~~du~~" with "CALLING" showed on the LCD, it means that the callee's phone is ringing. If the callee answers the call, the phone will be established and the LCD will show the calling time and the callee's number.

✓ Use the phone book

1. Press "Pbook" button to enter the content, select the item "CURRENT", then use the keys "+" and "-" to find the certain contact. When the LCD shows the message of the right contact, dial to the number by pressing the "Menu/Enter" button twice, or pressing button "Dial", or making one of the three modes (handset, handfree and headset) active.

2. Press "Pbook" button to enter the content, select the item "SEARCH", then input the name saved in the phone book earlier who user want to dial, press button "Menu/Enter" to ensure the operation. When the LCD shows the message of the right contact, user could dial to the number wanted by pressing the "Menu/Enter" button twice, or pressing button "Dial", or making one of the three modes (handset, handfree and headset) active.

✓ Use the History

Press "History" key to enter the content, select one of the three kinds of record: DIALED, RECEIVED or MISSED. Press "Menu/Enter" key to enter the content, use the keys "+" and "-" to find the certain contact. After finding out the certain number, user can dial to the number wanted by pressing the "Menu/Enter" button twice, or pressing button "Dial", or making one of the three modes (handset, handfree and headset) active.

4.3 Hang up

✓ Handset hang up

When calling under handset mode, put the handset back to hang up.

✓ Handfree hang up

When calling under handfree mode, press button "handfree" to hang up.

✓ Headset hang up

When calling under headset mode, press button "headset" to hang up.

4.4 Call Transfer

✓ Attended transfer:

Attended transfer allow user to speak to the third part before transfer.

If A is using AT810 talking with B, B wants to speak to C with A's help of transferring. The operation steps are:

1. A presses "Hold".

2. A dials C's number to ask whether C would answer the call from B.

3. If C agrees, then A presses "XFER" to transfer B's call to C.

✓ Blind transfer:

Blind transfer do not allow user to talk with the third part instead of ending the call directly, it means to user that the call is over after the blind transfer.

If A is using AT810 talking with B, B want to speak to C with A's transfer. The operation steps are:

1. A presses "XFER".

2. A dials C's number, then A will be hang up no matter C receives the call or not.

4.5 Mute

Once the "Mute" key is been pressed, no voice will be transported to the other part in the call anymore. Such state will be kept, no matter user switches the mode among handset mode and handfree mode and earphone mode, until the button "Mute" is been pressed again.

4.6 Call Hold

User can hold the current call by pressing "Hold". To resume the call, user needs to press soft "Hold" again. The functions are available under 3-way conference mode the same as under normal mode. When the phone is in the situation of "Hold", the call won't be hang up, instead, the call will be kept holding.

4.7 3-way conference call

Steps of starting a 3-way conference call are as follows:

1. When A is using AT810 talking with B and they want to make conference call with C. A or B just needs to press soft key "Conf", the other one in the call will be held, and then the initiator dials C's number, presses "#" or "Dial" to make the call.

2. After C answers the call, the initiator presses "Conf" to begin the 3-way conference call.

3. If the initiator hangs up directly before the 3-way conference call established, it will be ended, and the others will be hanged up, too.

4. Anyone in the 3-way conference call can act as an initiator of a new 3-way conference call. This initiator obeys the above rule number 3.

4.8 Call History

AT810 series phone supports 50 missed calls, 50 incoming calls and 50 dialed calls record. The latest call will update the history if the storage is full. When the phone updates or is factory resets, all the call history records will be cleared.

A. Missed call

1. When there is a missed call, the LCD screen will display the Missed call icon and the status LED will be 2s on, 500ms off.
2. Press history button, then press "Vol+" or "Vol-" to select "MISSED".
3. Press "Menu/Enter" to make AT810 display the records of missed call.
4. Press "Vol+" or "Vol-" to browse missed call history.
5. If there is no missed record, the LCD screen will display "EMPTY".

B. Received call

1. Press "History" button, then press "Vol+" or "Vol-"to select "RECEIVED".
2. Press "Menu/Enter" to make AT810 display the records of received call.
3. Press "Vol+" or "Vol-" to browse received call records.
4. If there is no record, the LCD screen will display "EMPTY".

C. Dialed call

1. Press "History" button and press "Vol+" or "Vol-"to select "DIALED".
2. Press "Menu/Enter" to make AT810 display the records of dialed call.
3. Press "Vol+" or "Vol-" to browse dialed call records.
4. If there is no record, the LCD screen will display "EMPTY".

4.9 Date and Time

1. Press "Menu/Enter" and input right password to enter the phone's menu.
2. Use "Vol+" or "Vol-" to find "System" item, press "Menu/Enter" to confirm.
3. Press "Menu/Enter" to confirm again when the LCD shows "Time".
4. Press "Del" to enter edit mode, when the phone shows "INPUT VALUE", input "month*day*year*hour*minute" such as "12*21*2012*12*21" which means December 21st, 2012, 21 past 12.
5. Press "Menu/Enter" to save, and press "Exit" to exit the menu and check the time.

4.10 Access Mode

There are two kinds of web access mode for AT810 series: admin mode and user mode. User could view and configure all items in admin mode, while user couldn't change the SIP configuration as well as server address and port but only access and view the

information. User would enter different mode after input different web address and user name and password:

A. Administrator mode

    Web address: the phone's IP/index.asp.

    User Name: admin.

    Password: default password is none.

B. User mode

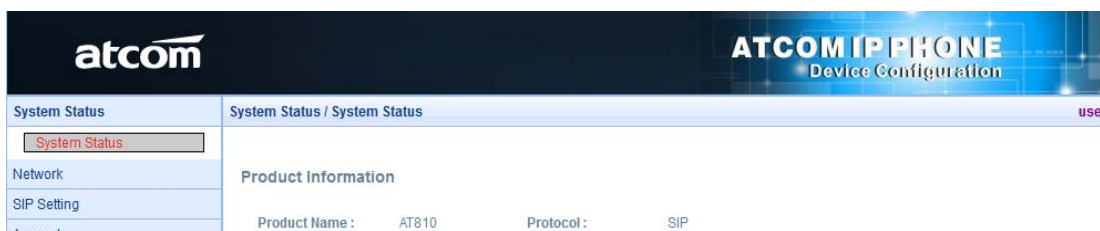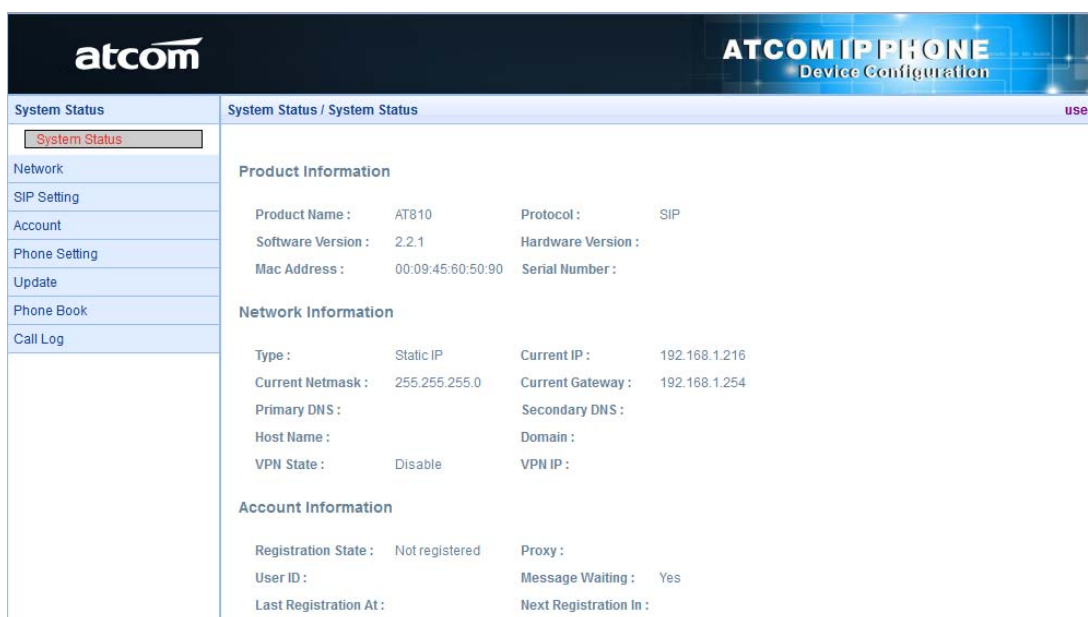   Web address: the phone's IP/user.asp.

    User Name: user.

    Password: default password is none.

5. Web settings

Input phone's IP addresses in the web browser and press the "Menu/Enter" button, it will go to the web interface of the user mode .Click on "admin" at right of the web page, user can config the phone from the web then.

5.1 System status

This page shows the current basic information of the IP phone like the product information, network information and account information.

Product information: Shows the product name, protocol, software version, hardware version, MAC address and serial number.
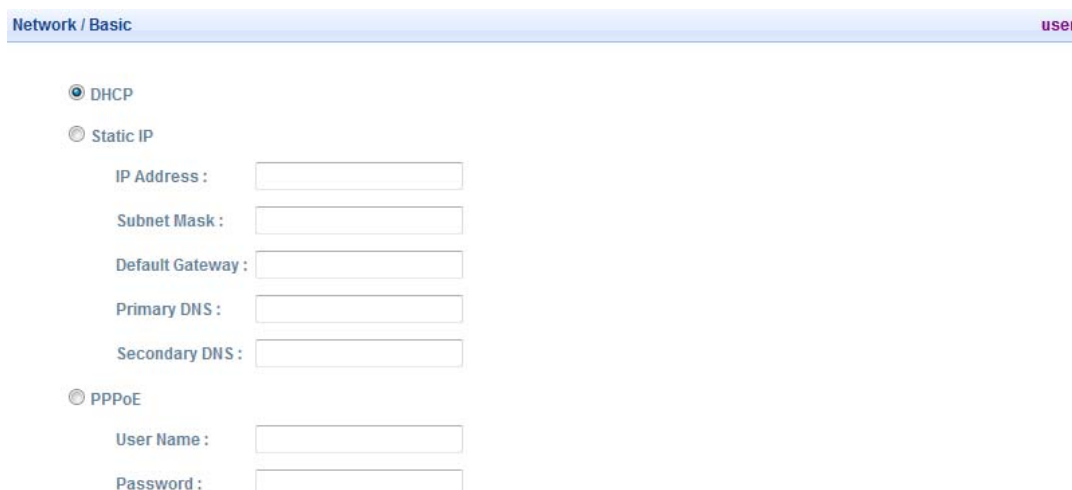
Network information: Shows the type of web connection, the current IP, the current netmask, current gateway, primary DNS, secondary DNS, host name and the domain.

Account information: Shows the registration state, proxy, user id, the message waiting status, last registration time and next registration time.

5.2 Network

5.2.1 Basic

There are 3 ways for AT810 series to connect to the internet: DHCP, Static and PPPoE, please choose one according to the actual situation.

A. DHCP: the IP phone will get IP address from DHCP server, user do not have to fill in the data of IP address, net mask etc. Just choose DHCP and submit.

B. Static IP: If the ISP provides user with the fixed IP address, please choose static and fill in the correct information of IP Address, Net mask, Gateway, Primary DNS etc. If user does not know them, please refer to the ISP provider or the network management stuff.

Parameters:

1. IP Address: fixed IP address.

2. Subnet Mask: WAN net mask.

3. Default Gateway: Set Default gateway IP address.

4. Primary DNS: Set primary DNS address.

5. Secondary DNS: Set alternative DNS address.

C. PPPoE: When user want to use PPPoE, select "PPPoE"，and makes the phone's WAN interface connected to the Ethernet interface of the modem, and then configures the

user name and password provided by ISP.

Parameters：

    1. Username: Set account user name.

    2. Password: Set account password.

Notice:

    1. After configuration, please click on "submit" to effect the change.

    2. If the IP address is changed after effecting the configuration, user will have to get to the webpage with the new IP address.

5.2.2 Advance

5.2.2.1 Web Server

Web Server			
Admin Password :		User Password :	
HTTP Port :	80		

A. Admin Password: To set admin webpage access password, the default username for admin webpage is admin.

B. User Password: To set user webpage access password, the default username for user webpage is user.

C. HTTP Port: To set port for HTTP, the default is 80. If change it, user will have to input http://ip-address to open the interface again. For example, AT810's IP is "192.168.1.201", and HTTP port was set as 8080. Then user has to type "http://192.168.1.201:8080" in web browser to enter AT810 webpage.

5.2.2.2 VPN

VPN			
VPN Enable :	No	VPN Type :	L2TP Tunnel
Server Address :			
User Name :		Password :	

A. VPN Enable: Whether enable VPN or not.

B. VPN Type: L2TP VPN (without IPsec authentication).

C. Server address: VPN server address.

D. Username: VPN account's username.

E. Password: VPN account's password.

After applying the settings, the phone will reboot. The VPN IP will show on the System Status page.

5.2.2.3. VLAN

VLAN

You should receive the tagging values of a virtual LAN from your provider. Wrong settings will require factory reset.

Use VLAN Tagging :	No
Voice VLAN Identifier(1..4094) : 1	Voice VLAN Priority : 0
Data VLAN Identifier(1..4094) : 1	Data VLAN Priority : 0

A. Use VLAN Tagging: Whether to use VLAN (virtual local area network) or not.

B. Voice VLAN Identifier (1..4094): Set voice VLAN ID, range from 1 to 4094.

C. Voice VLAN Priority: Set the voice VLAN level, range from 0 to 7.

D. Data VLAN Identifier (1..4094): Set data VLAN ID, range from 1 to 4094.

E. Data VLAN Priority: Set the data VLAN level, range from 0 to 7.

5.2.2.4. Port Link

Port Link

WAN Port Link :	Auto negotiate	PC Port Link :	Auto negotiate

A. WAN Port Link: Auto negotiate, Full duplex 10Mbps, Full duplex 100Mbps, Half duplex 10Mbps, Half duplex 100Mbps.

B. PC Port Link: Auto negotiate, Full duplex 10Mbps, Full duplex 100Mbps, Half duplex 10Mbps, Half duplex 100Mbps.

5.2.2.5. Qos

Qos

SIP Qos(0..63) :	0	Voice Qos(0..63) :	0

A.   Qos：Quality of service.

B.   SIP Qos: Quality of service for SIP (Diffserv), integer ranges from 0 to 63.

C.   Voice Qos: Quality of service for RTP (Diffserv), integer ranges from 0 to 63.

5.2.2.6. Syslog

A.   Enable Syslog: Whether to enable syslog or not.

B.   Log level: None，Alert，Critical，Error，Warning，Notice，Info，Debug. Debug level is the most detailed.

C.   Syslog Server: Syslog Server address, which is a domain name or an IP address.

D.   Port: Syslog port, which usually is 514.

5.3 Sip Setting

5.3.1. Sip Timer Values

A.   Sip T1: RFC 3261 T1 value (RTT Estimate). Range from 0 to 64. The default value is 0.5.

B.   Sip T2: RFC 3261 T2 value (Maximum retransmit interval for non-INVITE requests and INVITE responses). Range from 0 to 64. The default value is 4.

C.   Sip T4: RFC 3261 T4 value (Maximum duration a message will remain in the network). Range from 0 to 64. The default value is 5.

D.   Reg Retry Intvl: Interval to wait before the phone retries registration again after encountering a failure condition during last registration. Range from 0 to 65535.The default value is 8.

E.   Sub Retry Intvl: Subscribe message's re-sending interval after a failure. Range from 0 to 65535.The default value is 10.

5.3.2 RTP Parameters

A、 RTP Port Min: Minimum port number for RTP transmission and reception. Range from 1 to 65535. Default to 16384.

B、 RTP Port Max: Maximum port number for RTP transmission and reception. Range from 1 to 65535. <RTP port Min> and <RTP Port Max> should define a range that contains at least 2 even number ports. Default to 16482.

C、 RTP Packet Size (ms): Packet size in milliseconds, which can be 10, 20, 30, 40, 60.

5.3.3 SDP Payload Types

AVT Dynamic Payload :	101	G729ab Dynamic Payload :	18
G726 Dynamic Payload :	108	G726 Mode :	32 kBit/s
iLBC Dynamic Payload :	98	iLBC Mode :	13.3 kBit/s
G711a Codec Name :	PCMA	G711u Codec Name :	PCMU
AVT Codec Name :	telephone-event	G722 Codec Name :	G722
G723 Codec Name :	G723	G723 Mode :	6.3 kBit/s
G726 Codec Name :	G726	G729ab Codec Name :	G729
iLBC Codec Name :	iLBC		

The configured dynamic payloads are used for outbound calls only where the SPA presents the SDP offer. For inbound calls with a SDP offer, SPA will follow the caller's dynamic payload type assignments.

A. AVT Dynamic Payload---AVT dynamic payload type. Default to 101.

B. G729ab Dynamic Payload---G729ab dynamic payload type. Default to 18.

C. G726 Dynamic Payload---G726 dynamic payload type. Default to 108.

D. G726 Mode---G726 codec code rate, which can be 16kBit/s, 24kBit/s, 32kBit/s and 40kBit/s.

E. iLBC Dynamic Payload---iLBC dynamic payload type. Default to 98.

F. iLBC Mode---iLBC codec code rate, which can be 13.3kbit/s, 15.2kbit/s.

G. G711a Codec Name---G711a codec name used in SDP. Default to PCMA.

H. G711u Codec Name---G711u codec name used in SDP. Default to PCMU.

I. AVT Codec Name---AVT codec name used in SDP. Default to telephone-event.

J. G722 Codec Name---G722 codec name used in SDP. Default to G722.

K. G723 Codec Name---G723 codec name used in SDP. Default to G723.

L. G723 Mode---G723 codec code rate, which can be 6.3kbit/s, 5.3kbit/s.

M. G726 Codec Name---G726 codec name used in SDP. Default to G726.

N. G729ab Codec Name---G729ab codec name used in SDP. Default to G729.

O. iLBC Codec Name---iLBC codec name used in SDP. Default to iLBC.

5.3.4 NAT Support Parameters

NAT Support Parameters

Enable Stun :	No ▼	Stun Server :	

A. Enable Stun: Whether to use stun to discover NAT mapping or not.

B. Stun Server: Stun server to contact for NAT mapping discovery.

5.4 Account

5.4.1 SIP

SIP

Display Name :		User ID :	5006
Authenticate ID :		Password :	••••
SIP Server :	172.16.0.55	SIP Port :	5060
SIP Redundancy Server :	172.16.0.54		
Use Outbound Proxy :	No ▼		
Outbound Proxy Server :		Outbound Proxy Port :	5060
Register Expires :	300	Subscribe Expires :	3600
Transport Type :	UDP ▼	SIP 100Rel Require :	No ▼
Session Timer Enable :	No ▼	Early Update Enable :	No ▼
Caller ID Display :	No ▼	AutoSubscribeMWIEnable :	No ▼
Server List :	Common ▼	Dns Mode :	A Record ▼
BLF List URI :			

✓ Display Name: The phone's display name, it will be shown on the far end when dialing. It will prefer to show the name set by the server if the server has set a name for account. It will show the account instead if set the display name as blank.

✓ User ID: Set user ID for sip account.

✓ Authenticate ID: Normally is the same with User ID, can also leave it as blank.

✓ Password: Set password for sip account.

✓ SIP Server: Set SIP server, supports both IP address and domain name.

✓ SIP Port: Set SIP server port, default is 5060.

✓ SIP Redundancy Server: SIP redundancy server address, support both IP address and domain name. It can be configured manually and also can be auto configured as

the server address carried in the DNS SRV record if there is a DNS which supports specifying the location of the server or domain for SIP protocol. Only when AT810 fails to register to the SIP Server, it will try to register to the SIP Redundancy Server.

✓ Use Outbound Proxy: Whether to use outbound proxy or not.

✓ Outbound Proxy Server：Normally the Proxy server is the same as SIP server. If they are different then fill in the correct information that provided by ISP.

✓ Outbound Proxy Port: Set outbound proxy server port.

✓ Register Expires: Set register expires time, default is 300s. AT810 will auto configure this expires time to the server recommended setting if it is different from the SIP server.

✓ Subscribe Expires: Set subscribe expires time, default is 3600s.

✓ Transport Type: UDP/TCP/TLS.

✓ SIP 100Rel Require: Whether to enable SIP 100rel require function or not. When set as "No", the phone will support PRACK selectively which means the phone can dial out successfully no matter the UAS support PRACK or not. But for "Yes", the phone will ask for supporting PRACK by the strong hand, and then the phone can't dial out successfully if the UAS does not support PARCK.

✓ Session Timer Enable: Whether to enable session timer extension for this account. Default is no.

✓ Early Update Enable: this parameter control the account to support receiving Early Update.

✓ Caller ID Display: this parameter is used to supported the P-Asserted-Identity header.

✓ AutoSubscribeMWI Enable: Enable this account to subscribe for message summary automatically.

✓ Server List: the SIP server list supported, can be common and Broadsoft, default to common.

✓ DNS Mode: enable the account to use DNS SRV query. Default to no which use DNS A query.

✓ BLF List URI: the Broadworks BLF uri to request.

5.4.2 Codec Configuration

Codec Configuration

G711A :	Yes	G711U :	Yes
AVT :	Yes	G722 :	Yes
G723 :	Yes	G729ab :	Yes
iLBC :	Yes	G726-16 :	Yes
G726-24 :	Yes	G726-32 :	Yes
G726-40 :	Yes		
Prefer Codec :	G711u	User Prefer Codec Only :	No
DTMF Tx Method :	RFC2833		

- ✓ G711A: enable or disable G711A codec support in this account.
- ✓ G711U: enable or disable G711U codec support in this account.
- ✓ AVT: enable or disable AVT codec support in this account.
- ✓ G722: enable or disable G722 codec support in this account.
- ✓ G723: enable or disable G723 codec support in this account.
- ✓ G729ab: enable or disable G729ab codec support in this account.
- ✓ iLBC: enable or disable iLBC codec support in this account.
- ✓ G726-16: enable or disable G726-16 codec support in this account.
- ✓ G726-24: enable or disable G726-24 codec support in this account.
- ✓ G726-32: enable or disable G726-32 codec support in this account.
- ✓ G726-40: enable or disable G726-40 codec support in this account.
- ✓ Prefer Codec: Select a preferred codec for all calls, such as G711u, G711a, G729, G722, G7223, G726, iLBC and so on. However, the actual codec used in a call still depends on the outcome of the codec negotiation protocol.
- ✓ User Prefer Codec Only: Whether to use the preferred codec only for all calls or not. If select "yes", the call will fail if the far end does not support this codec.
- ✓ DTMF Tx Method: Select the method to transmit DTMF signals to the far end: InBand, RFC2833, and INFO. Defaults to RFC2833.

5.4.3 Call Feature Setting

Call Feature Setting

Message Waiting :	Yes ▾
Pickup Service Code :	*8
UDP Keep Alive Enable :	No ▾
Default Ring :	1 ▾
SRTP :	Off ▾

UDP Keep Alive Intvl : 15

✓   Message Waiting: Whether to prompt if there is a voicemail.

✓   Pickup Service Code: To set the code to pick up calls. Default to "*8".

✓   UDP Keep Alive Enable: Once enabled, the phone will keep sending UDP packet to the register port of the server to make sure the server is staying available.

✓   UDP Keep Alive Intvl: To set the interval time to send UDP packet.

✓   Default Ring: Select a default ring for the phone when there's an incoming call.

✓   SRTP: Turn on or off the Secure Real Time Control Protocol. If on, support from the server will be needed unless make IP calls.

5.4.4 Dial Plan

Dial Plan

Dial Plan :   (xxxxxxxxxxx.)

✓ Submit   ↩ Reset

Dial Plan: Configure the dial plan for the SIP account.

The Dial Plan defines the way that how to transform the number user dialed to the real dialed characters sequentially, the syntax of the dial plan is similar to MGCP's and MEGACO's.

✓ Functions of dial plan are defined by the configurable parameters as follows:

    A. Interdigit Long Timer---the details please refer to <u>Interdigit Long Timer</u>.

    B. Interdigit Short Timer---the details please refer to <u>Interdigit Short Timer</u>.

    C. Dial plan---Dial plan contains a series of digit sequences and rules, each rule is been separated by "|", "(" stands for start and ")" stands for end. The default dial plan is "(xxxxxxxxxxxx.)".

✓ The phone will take actions as follows after the user input the number:

    1. The call will be rejected if the number does not match the dial plan. For example, according to the default rule, it only receives digits. If we input characters or symbols, such dials will be rejected.

    2. It will wait for input if there are more than one rule match user's enters.

3. It will send the number directly if the input is timeout.

4. It will send the number directly if user presses the key "#".

✓ Syntax

1. "x": Stands for a digit range from 0 to 9.

2. "[]": Stands for a range.

   For examples: [389] stands for 3 or 8 or 9;

   [3-6] stands for 3 or 4 or 5 or 6;

   [235-8*] stands for 2 or 3 or 5 or 6 or 7 or 8 or *.

3. ".": Stands for the previous one digit could be repeated. For example "01.", the numbers match this rule are 01, 011, 0111,…and so on.

4. "<>": The replace rule. For example: "<8:1650>xxxxxxx" stands for any numbers begin with 8 and the length is 7, the first 8 will be replaced by 1650 and dialed out then. Such as dialing 8551212, the phone will dial 1650551212 at last.

5. ",": Stands for playing an outside dial tone. For example "9,1xxxxxxxxxx", user will hear an outside dial tone when user press the first key "9" until the following key "1" is been pressed.

6. "!": Stands for the number which is forbidden to dial. For example "137xxxxxxxx!" stands for forbidding user to dial numbers whose length is 11 and begin with 137.

For example:

Set the dial plan as (xxxxxxx|[*#]xxxx|9,1xxxxxxxxxx|00xxx!).

The syntax above contains 4 rules:

1. Allow dialing numbers whose length is 7.

2. Allow dialing numbers whose length is 4 and begin with * or #.

3. User will hear an outside dial tone when user press the first key "9", and then a number whose length is 11 and begin with 1 will be dialed out.

4. Forbid dialing Numbers whose length is 5 and begin with 00.

The following dial plan accepts only US style 1 + area code + local number, with no restrictions on the area code and number.

(1 xxx xxxxxxx)

The following also allows 7-digit US-style dialing, and automatically inserts a 1 + 212 (local area code) in the transmitted number.

(1 xxx xxxxxxx | <:1212> xxxxxxx)

For an office environment, the following plan requires a user to dial 8 as a prefix for local calls and 9 as a prefix for long distance. In either case, an "outside line" tone is played after the initial 8 or 9, and neither prefix is transmitted when initiating the call.

(<9, :> 1 xxx xxxxxxx | <8,:1212> xxxxxxx)

The following allows only placing international calls (011 call), with an arbitrary number of digits past a required 5 digit minimum, and also allows calling an international call operator (00). In addition, it lengthens the default short interdigit timeout to 4

seconds.

S: 4, (00 | 011 xxxxx x.)

The following allows only US-style 1 + area-code + local-number, but disallows area codes and local numbers starting with 0 or 1. It also allows 411, 911, and operator calls (0).

(0 | [49]11 | 1 [2-9] xx [2-9] xxxxxx)

The following allows US-style long distance, but blocks 9xx area codes.

(1 [2-8] xx [2-9] xxxxxx)

The following allows arbitrary long distance dialing, but explicitly blocks the 947 area code.

(1 947 xxxxxxx! | 1 xxx xxxxxxx)

The following implements a Hot Line phone, which automatically calls 1 212 5551234.

(S0 <:12125551234>)

The following provides a Warm Line to a local office operator (1000) after 5 seconds, unless a 4 digit extension is dialed by the user.

(P5 <:1000> | xxxx)

5.5 Phone Setting

5.5.1 Preference

5.5.1.1 Output Volume (1~8)

✓ Handset Volume: Specify handset volume grade, the higher the grade, the louder will user hear from the handset.

✓ SpeakerPhone Volume: Specify speakerphone volume grade, the higher the grade, the louder will user hear from the speakerphone.

✓ Headset Volume: Specify headset volume grade, the higher the grade, the louder will user hear from the headset.

5.5.1.2 Input Gain

✓ Handset Gain：Specify handset gain, the bigger the gain, the louder will the other end hear when the user is using the handset.

✓ SpeakerPhone Mic Gain: Specify speakerphone mic gain, the bigger the gain, the louder will the other end hear when the user is using the speakerphone.

✓ Headset Gain: Specify the headset gain, the bigger the gain, the louder will the other end hear when the user is using the headset.

5.5.1.3 LCD

LCD

| Backlight Level: | High ▾ | Backlight Time(Seconds): | 10 ▾ |
| keypad password : | ●●● | | |

✓ Backlight Level: Specify the level of the blacklight:Off, Low, High.

✓ Backlight Time(Seconds): Specify the duration that the blacklight will be on for.

✓ Keypad password: To set a password when press "MENU" button on keypad.

5.5.1.4 Control Timer Values (Seconds)

Control Timer Values(Seconds)

| Interdigit Long Timer : | 10 | Interdigit Short Timer : | 3 |
| Reorder Delay : | 5 | | |

✓ Interdigit Long Timer: If the numbers or characters input are not finished and they have not matched the full dial plan, the phone will wait for the long time that have been set and then dial them out automatically. It ranges from 1 to 64, the default value is 10.

✓ Interdigit Short Timer: If the numbers or characters input have matched the full dial plan, the phone will wait for the short time that have been set and then dial them out automatically.   It ranges from 1 to 64, the default value is 3.

✓ Reorder Delay: Delay after far end hangs up before reorder tone is played. 0 means playing it immediately. It ranges from 0 to 99, the default value is 5.

5.5.1.5 Data And Time

Date And Time

◉ NTP

    NTP Server :   pool.ntp.org

    Time Zone :   (GMT+08:00)Beijing,Chongqing,Hong Kong,Urumqi ▾

    Daylight Saving Time :   No ▾

    Daylight Saving Time Rule :

◯ Manual

    Set Local Date(YYYY/mm/dd) :

    Set Local Time(HH:mm:ss) :

✓ NTP

1. NTP Server: Network time protocol service server, default is pool.ntp.org.

2. Time Zone: Choose the time zone.

3. Daylight Saving Time: Whether to enable Daylight saving time or not.

4. Daylight Saving Time Rule: This parameter is a rule with three fields, each field is separated by semicolon, as show below:

   Start=<start-time>; end=<end-time>; save=<save-time>

   <start-time> and <end-time> specify the start and end date of daylight saving time, month/date/week and <save-time> is the amount of hour/min/sec to add to the current time during daylight saving period. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition.

   If <weekday > is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the <day> value must not be negative. If<weekday >is not zero, then the daylight saving starts or ends on the <weekday >on or after the given date if <day> is positive, or on or before the given date if <day> is negative. If <day> is -1, it means the <weekday> on or before the end-of-the-month (in other words the last occurrence of <weekday> in that month).

   Optional values inside [] are assumed to be 0 if they are not specified. Midnight means 0:0:0 of the given date.

   E.G. start=4/1/7/8:00:00;end=10/1/7/9:00:00;save=-1

       Start time is 1st April, Sunday, 8:00:00 am.

       End time is 1st October, Sunday, 9:00:00 am.

       The display time will be 1 hour early than the standard time.

✓ Manual

1. Set Local Date (YYYY/mm/dd): Manually set local date.

2. Set Local Time (HH:mm:ss): Manually set local time.

In manual mode, IP phone will resume to the manual time value after a reboot or power off.

5.5.2 Features

5.5.2.1 Speed Dial

Spead Dial

Speed Dial 2 :		Speed Dial 3 :	
Speed Dial 4 :		Speed Dial 5 :	
Speed Dial 6 :		Speed Dial 7 :	
Speed Dial 8 :		Speed Dial 9 :	

Set speed dial number for 2-9. For example, set 6750 for key 2, then user can call 6750 by call 2.

5.5.2.2 Call Forward

Call Forward

Always Target:		Busy Target:	
No Answer :			
After Ring Time(Seconds) :	5		

✓ Always Target: To set a target number that forward all the incoming call to.

✓ Busy Target：To set a target number that forward the incoming call when user's phone is busy.

✓ No Answer: To set a target number that forward the incoming call when no one answer it.

✓ After Ring Time (Seconds): Forward the incoming call to the no answer target after the time that set here. Default to 5 seconds. Range from 3~60 s.

If Always forward, Busy forward and No answer forward are both enabled, IP phone will implement always forward by priority. That is to say, all incoming calls will be forwarded to the always target number.

5.5.2.3 Call Settings

✓ Do Not Disturb: Whether to enable do not disturb function or not. The phone cannot receive any incoming call but show the missed call informations if specifies it as "Yes".

✓ Call Waiting: Whether to enable call waiting function or not. The phone can receive a new call even if it is in busy the time a new call is coming if specifies it as "Yes".

✓ Block Call ID: Whether to enable block call id function or not. The phone will call others as anonymous while enable it.

✓ Block Anc Call: Whether to enable block anc call function or not. The phone will reject anonymous caller while enable it.

✓ Auto Answer: Whether to enable auto answer function or not. The phone will answer the incoming call automatically through headset or handfree while enable it.

5.5.2.4 Others

Send Key: Specify the Key to send the number after pressing it, "#" and "*" are optional.

5.5.3 Voice

5.5.3.1 Echo Cancellation

✓ VAD: Whether to enable VAD (Voice Active Detection) or not. G729 Payload length can not be set over 20ms while enable it.

✓ CNG: Whether to enable CNG (Comfort Noise Generator) or not. If mute the phone and set VAD as yes, the phone will send comfort noise to remind the other one that the call is remain in force.

5.5.3.2 Jitter Buffer

Jitter Buffer

Type :	Fixed		
Min Delay :	0	Max Delay :	300
Normal Delay :	120		

✓   Type: Specify type for jitter buffer: Adaptive or Fixed. The SPA can buffer incoming voice packets to minimize out-of-order packet arrival. This process is known as jitter buffer.

✓   Min Delay: The minimum delay of the jitter buffer. Range from 0 to 997. The default value is 0.

✓   Max Delay: The maximum delay of the jitter buffer. Range from 2 to 999.The default value is 300.

✓ Normal Delay: It is used as fixed delay which should be between min delay and max delay. Range from 1 to 998. The default value is 120.

5.5.4 Ring

Notes: Ring tone must be wav file, 8k sampling rate ,8 Bit u-law compression. File size should < 200Kbytes

Upload Ring Tone	浏览…	Upload
Ring Tone Type	User Define 1	Delete

Administrator can upload 2 user define rings for AT810. The ring file should be wav (8k, 8bit, u-law) and no larger than 200kb.

5.5.5 Tone

Tone	
Country Stands :	Custom
Dial Tone :	350@-19,440@-19;10(*/0/1+2)
Outside Dial Tone :	420@-16;10(*/0/1)
Busy Tone :	480@-19,620@-19;10(.5/.5/1+2)
Reorder Tone :	480@-19,620@-19;10(.25/.25/1+2)
Off Hook Warning Tone :	480@-10,620@0;10(.125/.125/1+2)
Ring Back Tone :	440@-19,480@-19;*(2/4/1+2)
MWI Dial Tone :	350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2)
Holding Tone :	350@-19,440@-19;10(*/0/1+2)
Conference Tone :	350@-19;20(.1/.1/1,.1/9.7/1)

Set the Tone Script for each tone.

E.G.: MWI Dial Tone: 350@-19,440@-19;2(.1/,1/1+2);10(*/0/1+2).

Syntax: Frequency1, Frequency2; Cadence Section1; Cadence Section2.

➢ 350@-19: Frequency1 is 350HZ at -19dBm.

➢ 440@-19: Frequency2 is 440HZ at -19dBm.

➢ 2(.1/,1/1+2): Cadence Section length is 2s, 0.1s on, 0.1s off, with frequencies 1 and 2.

➢ 10(*/0/1+2): Cadence Section length is 10s, always on (* means always, 0 means never), with frequencies 1 and 2.

Here are the details of each item of the tone:

✓ Country stands: Select a country standard tone as default.

✓ Dial Tone：Played when prompting the user to enter a phone number.

✓ Outside Dial Tone：An alternative to <Dial Tone> usually used to prompt the user to enter an external phone number (versus an internal extension). This is triggered by a "," character encountered in the dial plan.

✓ Busy Tone：Played when a 486 RSC is received for an outbound call.

✓ Reorder Tone：Played when an outbound call has failed or after the far end hangs up during an established call.

✓ Off Hook Warning Tone: Played when the subscriber does not place the handset on the cradle properly.

✓ Ring Back Tone：Played for an outbound call when the far end is ringing.

✓ MWI Dial Tone：This tone is played instead of <Dial Tone> when there are unheard messages in the subscriber's mail box.

✓ Holding Tone：Indicate to the local user that the far end has placed the call on hold.

✓   Conference Tone：Plays to all parties when a 3-way conference is in progress.

5.6 Update

5.6.1 Manual Update

Bootloader Version: 2.3

Firmware Version: 2.2.1

Hardware Version:

Boot Loader: [   ] [浏览…] [💾 Upgrade]

Firmware: [   ] [浏览…] [💾 Upgrade]

Configuration: [   ] [浏览…] [💾 Upgrade] [⬇ Download]

Phone Book: [   ] [浏览…] [💾 Upgrade] [⬇ Download]

✓   Boot Loader: Download Boot file on ATCOM's webpage, then select the file from user's PC to upgrade.

✓   Firmware: Download firmware on ATCOM's webpage, then select the firmware file from user's PC to upgrade.

✓   Configuration: Upgrade or download config.xml file.

✓   Phone Book: Upgrade or download phonebook.xml.

5.6.2 Auto Provision

Configuration Profile

Provision Enable :   [No ▼]   Resync Random Delay :   [0   ]

Profile Rule :   [   ]

right format,like tftp://192.168.1.121/config.xml

Firmware Upgrade

Upgrade Enable :   [No ▼]   Error Retry Delay :   [0   ]

Upgrade Rule :   [   ]

right format,like tftp://192.168.1.121/firmware.fw

✓   Provision Enable: Whether to enable auto provision or not.

✓   Resync Random Delay: Set a random delay to resync.

✓   Profile Rule: It supports TFTP/FTP/HTTP auto provision. FTP only supports anonymous account login. After configuring the TFTP/FTP/HTTP server, set profile rule as below:

➢   TFTP rule: tftp://ip-address/config.xml.

- FTP rule: ftp://ip-address/config.xml.

- HTTP rule: http://ip-address/config.xml.

Click on "Submit" after configuration, the phone will reboot, and it will download the config file automatically and update during the time the phone is rebooting.

✓ Upgrade Enable: Whether to enable auto firmware upgrade or not.

✓ Error Retry Delay: Set a retry delay when error occurs.

✓ Upgrade Rule: It supports TFTP/FTP/HTTP auto provision. FTP only supports anonymous account login. After configuring the TFTP/FTP/HTTP server, set profile rule as below:

- TFTP rule: tftp://ip-address/AT810.xml.

- FTP rule: ftp://ip-address/AT810.xml.

- HTTP rule: http://ip-address/AT810.xml.

Click on "Submit" after configuration, the phone will reboot, and it will download the upgrade file automatically and update during the time the phone is rebooting.

5.6.3 Reset & Reboot

Reboot System Now

[Reboot]

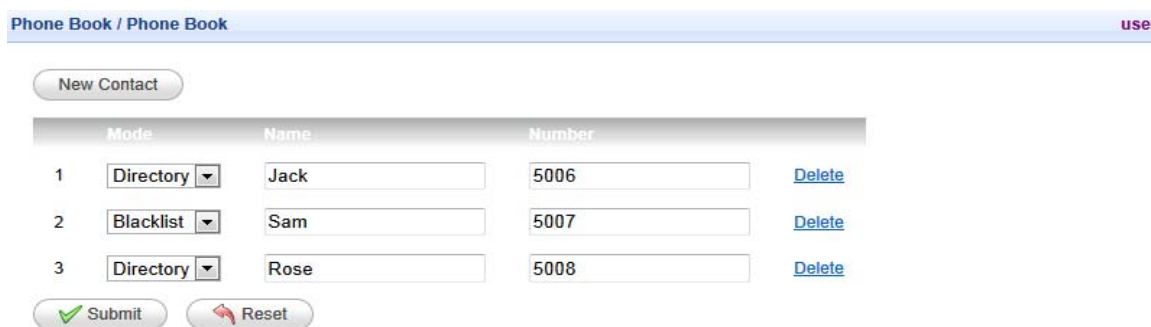Warning : Reboot system will terminate all active calls!

Reset to Factory Setting

[Reset]

Warning : A factory Reset will erase all configuration data on the system! Please do not power off the system. Any power interruption during this time could cause damage to the system!

✓ Reboot: Press to reboot the phone, such action will terminate all active calls!

✓ Reset: Press to set AT810 to factory default, please backup the config.xml, phonebook.xml and attendant_keypad.xml before reset.

5.7 Phone Book

Phone Book / Phone Book			user
New Contact			
	Mode	Name	Number
1	Directory	Jack	5006	Delete
2	Blacklist	Sam	5007	Delete
3	Directory	Rose	5008	Delete
Submit Reset			

✓ New Contact：Click on it to creat a new contact.

✓ Mode: Can be directory mode and blacklist mode. Maximum 100 directory recorders, and 50 blacklist recorders.

✓ Delete: Delete the current recorder.

5.8 Call Log

✓ Redial: The recorder of the dialed calls, maximum 50 items.

Call Log / Redial			user
	Name	Number	Time
1 :	8151	5008	22/11 14:43
2 :	8151	5008	22/11 14:41
3 :	5008	5008	22/11 14:15

✓ Received: The recorder of the received calls, maximum 50 items.

Call Log / Received			user
	Name	Number	Time
1 :	8151	5008	22/11 14:43
2 :	8151	5008	22/11 14:37
3 :	8151	5008	22/11 14:36

✓ Missed: The recorder of the missed calls, maximum 50 items.

Call Log / Missed			user
	Name	Number	Time
1 :	8151	5008	22/11 14:44
2 :	8151	5008	22/11 14:41

6. FAQ & Trouble Shooting

6.1 Frequently Asked Questions

6.1.1 How to make a factory reset

There are two ways to realize the factory reset:

    A. Factory reset from keyboard，steps are:

        1. Press key "Menu/Enter" and input the right password;

2. Use navigation key "+" or "-" to find "SYSTEM", then press "Menu/Enter";

3. Use navigation key "+" or "-" to find "SET DEFAULT", then press "Menu/Enter";

4. Press "Menu/Enter" to make factory reset after the LCD shows the message "ARE YOU SURE".

B. Factory reset through web, please refer to Reset & Reboot.

6.1.2 Upgrade firmware under safe mode

If the phone could not start up normally, upgrade firmware under safe mode. Steps are:

1. Build a TFTP server and set its IP address as 192.168.1.200;

2. Copy the firmware to the root directory of the TFTP server and modify the file name as AT810.FW;

3. Make sure the TFTP server and the phone are contacting with the same switch and reachable to eachother;

4. Long hold "#" key and start up the phone until the LCD display "Upgrading…";

5. The phone will download the firmware from the TFTP server;

6. After the download is done, the phone will upgrade automatically;

7. After the update is finished, the phone will start up with the configuration of factory reset.

6.1.3 How to make direct IP call

IP dial-up is almost the same as number dial-up, it just uses "*" to replace "." which is the characteristic part of the IP address when dialing. If you want to call 192.168.1.100, just press 192*168*1*100, and press # key to dial it out.

6.2 Trouble Shooting

6.2.1 The phone can't register successfully

✓ Check the IP address, and if the mode of WAN port is DHCP, please make sure the DHCP server is in service.

✓ Check the gateway.

✓ Check the DNS.

✓ Make sure the information of the account is consistent with which offered by the service supplier.

✓ Make sure the SIP server is on.

✓ Check the port of the SIP server whose default value is 5060.

6.2.2 The phone can't obtain IP address

✓ Make sure the cable has been connected to the WAN port of the phone.

✓ Make sure the cable and the switch's port that connected with the cable is available.

✓ Make sure the DHCP server is on and there are some assignable IP addresses in the address pool.

✓ Try to change the WAN port mode as Static.