AT820 series User Manual

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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

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

1. AT820 Series Overview

AT820 is almost the same with AT820P without PoE.

AT820/AT820P

Type	PoE	Power adapter
AT820	No	Standard accessory
AT820P	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: 128×64 dot matrix.
- FLASH: 8M.
- RAM: 16M.
- CPU: 262MHz Dual Core.
- LED indicator: 1 Status Light , 2 line indicators, 1 voicemail indicator, 1 headset indicator, 1 mute indicator, 1 hand-free indicator.

1.3 Software

- Supports Sip 2.0 (RFC3261) and other related SIP RFC.
- Supports web control.
- Supports Syslog.
- Supports G711A/U, G722, G.723, G.726, G.729, iLBC Codec.
- Supports STUN.
- Supports 2 SIP accounts.
- 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, Pickup, Call Park.
- Supports VAD, CNG，Jitter Buffer.
- Supports DND (Do Not Disturb), Auto Answer, Blacklist, Block Call ID, Block Anc Call.
- Supports HTTP, FTP, TFTP updating the configuration and firmware.
- Supports auto provision for config file upgrade.
- Supports phone book with 100 records and 50 blacklist records.
- Supports call log with 50 answered, 50 missed and 50 dialed call records respectively.
- Supports multi-language

1.4 Network

- Supports bridge mode and route mode.
- Supports static IP, DHCP and PPPoE.
- Supports VLAN (DATA VLAN and VOICE VLAN).
- Supports L2TP VPN.
- Supports Qos, Qos supports Diffserv.
- Supports basic NAT and NAPT.

1.5 Management and Maintenance

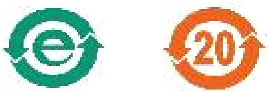
- Supports firmware updating under safe mode.

- Supports different level user management.
- Configuration via web, keyboard.
- Firmware and configuration file auto provision.
- Firmware and configuration updating via HTTP, FTP and TFTP.
- Support system log and call log.

1.6 Protocol

- IEEE 802.3 /802.3 u 10 Base T / 100Base TX.
- PPPoE：Point to Point Protocol over Ethernet.
- DHCP: Dynamic Host Configuration Protocol
- 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 Generator.
- 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.

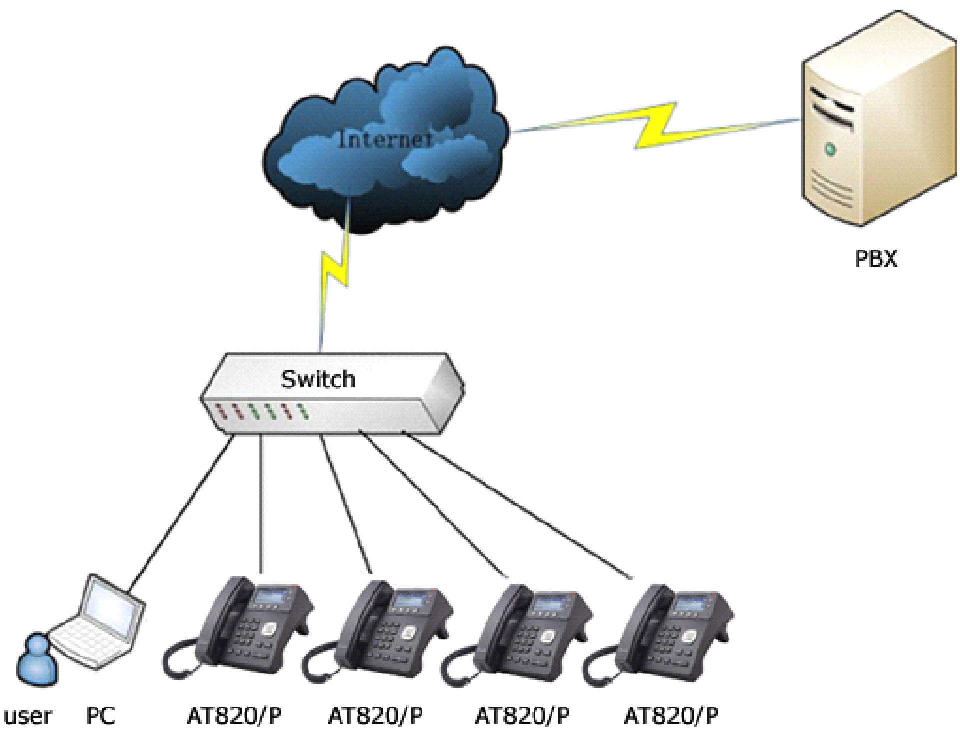
1.9 Packing List

Model	**Phone**	**Handset**	**Handset line**	**Power adapter**	**Manual CD**	**Feet underprop**
AT820	1	1	1	1,Standard accessory	1	2
AT820P	1	1	1	Optional accessory	1	2

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

1.10 Installation

Use Ethernet cable to connect AT820's WAN port and computer on the same switch, AT820 default to use DHCP to obtain IP from switch. The following is the connection diagram of application:



1.10.1 Start the phone

For AT820, it can start only with DC. But AT820P can also make it with POE.

A. Start through DC

Connect the phone with the power source by using a power adapter.

B. Start through POE

Connect the LAN port with a switch or a hub which compatibles the standard of IEEE02.3af.

1.10.2 SIP registration

The phone needs to register to SIP server before using it.

Steps to register:

A. Check the parameters of network:
   1. Press "MENU".
   2. Input the password which defaults to 123 to login.
   3. Move the cursor to "9.Network" and press "Select" soft key to check the phone's IP.

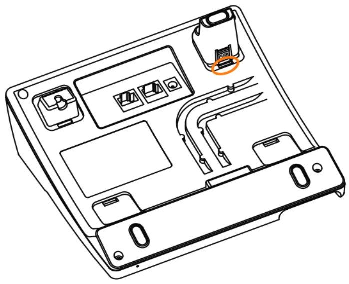
B. Configure the phone via web:
   1. Make sure the phone and the computer is in the same network segment.
   2. Open web browser and key in the phone's IP to login to the phone's webpage.

C. Configure SIP parameters:
   1. Click on "admin" at the top right corner of the webpage to enter administrator mode. The default username is admin for administrator and user for user, no passwords were set for them, but user can set password in the webpage.
   2. Select Account item.
   3. Configure the parameters refers to Account/SIP.

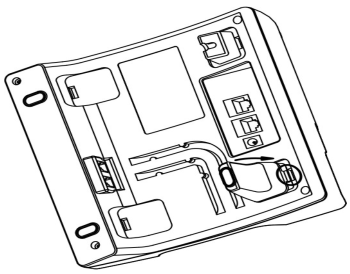
1.11 Feet installation instruction

1.11.1 Desktop position

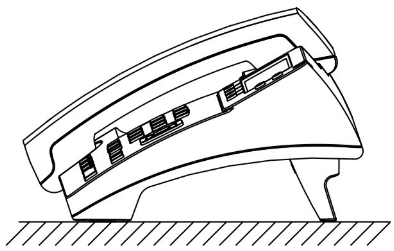
Put the bottom side of the IP phone upside and press the plate with letter “PUSH” into the slot, please refer the picture as below:



A. Press the other plate into the slot in accordance with the direction of the arrow.

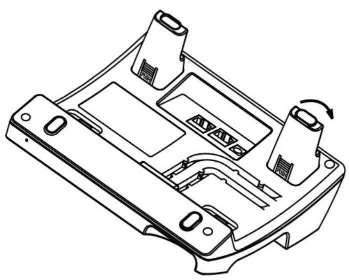


B. Repeat A and B. It is the right picture of putting on desk after fixing the two feet below:



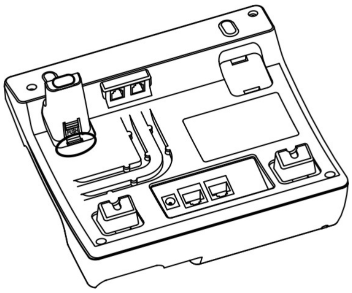
C. Disassemble the feet:

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.

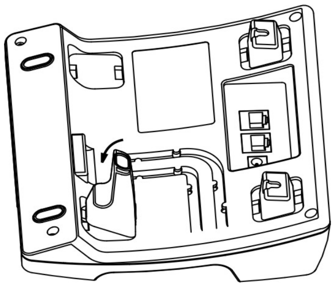


1.11.2 Wall position

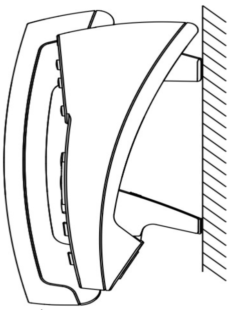
A. Put the bottom side of the IP phone upside and push the plate with letter "PUSH" into the slot, please refer the picture as below:

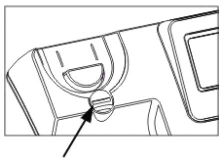


B. Push the other plate into the slot in accordance with the direction of the arrow.



C. Repeat A and B. It is the picture of wall mounting after fixing the two feet below:

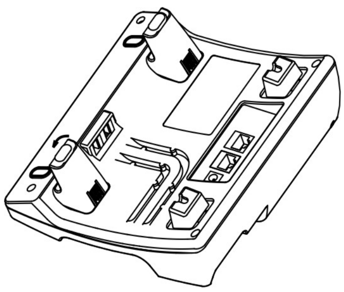




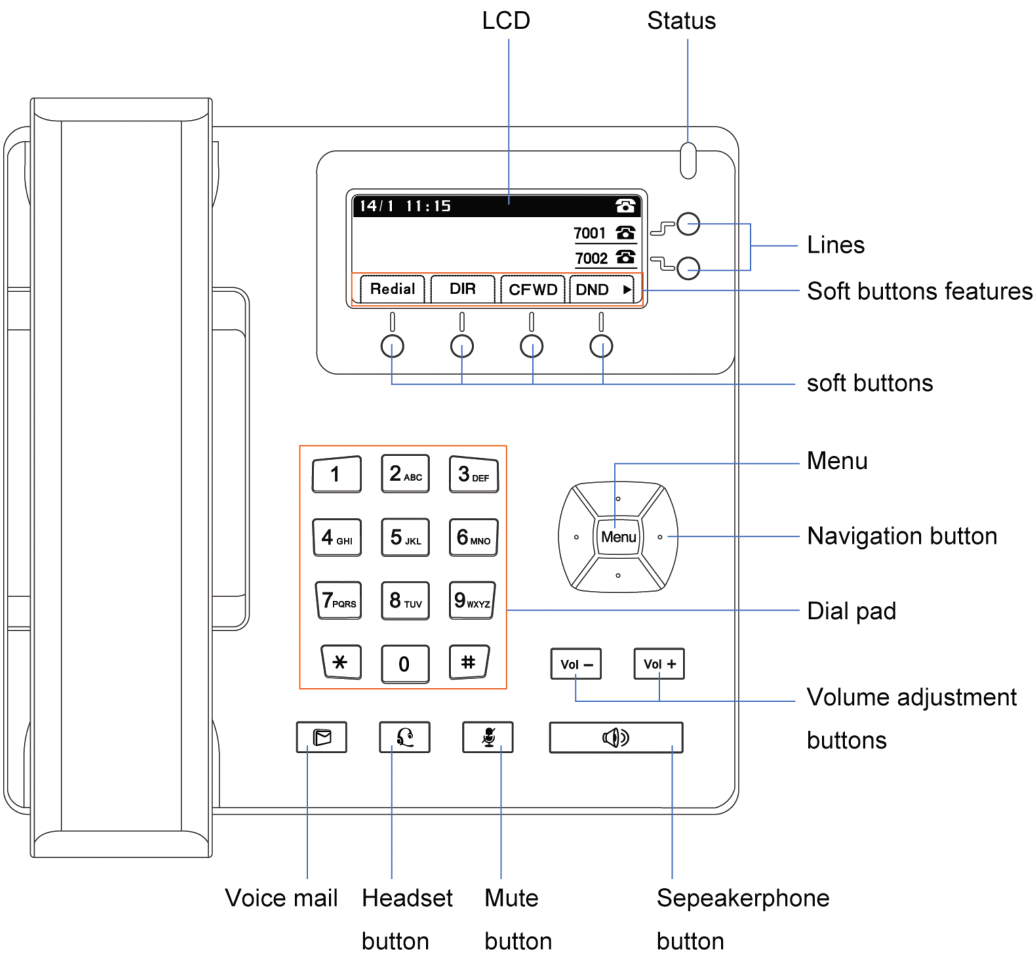
Attention: Please rotate the hook to the position as in picture with a coin or other tools

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



Sketch Map of AT820/AT820P

Description of indicators and buttons:

LCD Screen	Display screen for the phone: It shows the date, time, phone number, incoming caller's ID (if available), line/call status, extension numbers and the soft key features.
Status	Shows the phone status ➢ If the phone is starting, the LED will flicker. ➢ If the phone is standby, the LED will be off. ➢ If there is income calling, the LED will flicker. The frequency is 500ms off, 500ms on.
Lines	Shows extension number and status. There are three colors for LED: red, yellow and orange. ➢ If the line is registered, the LED will show yellow.

	➢ If enables the line but failed to register to server, the LED will show orange. ➢ If the line has income call, the LED will show red and flicker, the frequency is 500ms off, 500ms on. ➢ If the line is on the call, the LED will show red. ➢ If disables the line, the LED will be off.
Soft key features	Show available choices based on current phone function displayed on the last line of LCD screen.
Soft keys	Keys for pressing to select a feature shown in the soft key features.
Menu	Press it to enter the menu, and once more to exit it.
Navigation button	Allows users to navigate(left, right, up, down), under the standby mode, press up and down to change the line, right and left to change soft key features.
Dial pad	For entering numbers, letters or characters
Volume adjustment buttons	Adjust the volume ➢ "+" for turning up the volume unless the volume is the loudest yet. ➢ "-" for turning down the volume unless the volume is the lowest yet.
Voicemail button	Check the voicemail status, if there is unread voicemail, the LED button will blink.
Headset button	Pick up and hung up under headset mode. When pick up by headset, the LED button will be on.
Mute button	➢ Mute the handset, headset or speakerphone by press the mute button. It prevents the person on the active call from hearing what user or someone else in the room is speaking. ➢ To cancel the mute function, press the mute button again. ➢ If mute the voice, the LED will light on this button. ➢ If the WAN link is disconnected, the LED will blink.
Speakerphone button	Pick up and hung up on the speakerphone mode, when pick up by speakerphone, the LED of the button is on

3. Menu Operation

Press Menu/Enter to enter menu, user will see the list of items in turn as follows.

Level 1 Menu	**Level 2 menu**	Level 3 Menu
1.Directory		
	Personal Directory	
2.Speed Dial		

	2<Not Assigned>	
	3<Not Assigned>	
	4<Not Assigned>	
	5<Not Assigned>	
	6<Not Assigned>	
	7<Not Assigned>	
	8<Not Assigned>	
	9<Not Assigned>	
3.Call History		
	Dialed Calls	
	Answered Calls	
	Missed Calls	
4.Ring Tone		
	Ext1:Ring2.wav	
	Ext2:Ring1.wav	
5.Preferences		
	Block Caller ID	
	Block Anonymous Call	
	Do Not Disturb	
	Call Waiting	
	Dial Assistance	
	Preferred Audio Device Speaker	
	Auto Answer	
6.Call Forward		
	CFWD All Number	
	CFWD Busy Number	
	CFWD No Ans Number	
	CFWD No Ans Delay	
7.Time/Date		
	Time Mode	
	Manual Settings	
	NTP Settings	
	Date Format	
	Time Format	
8.Voice Mail		
	Voice Mail Number	
9.Network		
	Connect Type	
	Current IP	
	Host Name	
	Domain	
	Current Netmask	

	Current Gateway	
	Enable Web Server	
	Non-DHCP IP Address	
	Non-DHCP Subnet mask	
	Non-DHCP Default Rout	
	Non-DHCP DNS 1	
	Non-DHCP DNS 2	
	PPPoE Login Name	
	PPPoE Login Password	
10.Product Info		
	Product Name	
	Serial Number	
	Software Version	
	Hardware Version	
	MAC Address	
	Protocol	
11.Status		
	Phone	
	Ext 1	
	Ext 2	
	LineKey 1	
	LineKey 2	
12.Reboot		
13.Factory Reset		
14.Set Password		
	New Password:	
	New Password:	
15.LCD Contrast		

4. Basic functions and operations

4.1 Answer calls

When there is an incoming call, AT820 will remind user with ring. There are 3 ways to answer the call:

A. 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 hand-free mode, just press the hand-free button or the headset button. Be attention: only after done the switch can user put the handset back, or the call will be ended.

B. 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.

C. 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 hand-free mode, user just need to pick up the handset to switch to handset mode, and press the hand-free button to switch into the hand-free mode.

4.2 Make Calls

A. 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 dialed number showed on the LCD, it means that the callee's phone is ringing. If the callee answers the call, the call will be established and the LCD will show the calling time and 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 dialed number 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 callee's number.

B. Use the hand-free 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 dialed number 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 callee's number.
2. Input the number then press the hand-free button, the phone will dial the number automatically. After hearing the tones of "du~~du~~" with dialed number 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 callee's number.

C. 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 dialed number 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 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 dialed number 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 callee's number.

D. Use the phone book

Press " MENU" button to enter the menu, after inputting the password, select the first item: "1 Directory", then select the "Personal Directory". Use "up" and "down" keys to find the contact person. After finding out the certain contact person, press "Dial" to make call.

E. Use the History

Press "MENU" key to enter the menu, after inputting the password, select the third item: "3 Call History", then choose one of the three kinds of record: "Dialed" or "Answered" or "Missed", use the keys "+" and "-" to find the certain contact. After finding out the certain number, user could dial to the right number by pressing button "Dial", or making one of the three modes (handset, hand-free and headset) active.

4.3 Hang up the phone

A. Handset hang up

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

B. Handfree hang up

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

C. Headset hang up

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

4.4 Call Transfer

A. Attended transfer:

Attended transfer allows user to speak to the third part before transfer. If A is talking with B, A wants to transfer the call to C. The operation steps are:

1. A presses "Xfer".
2. A dials C's number to ask whether C would answer the call from B or not.
3. If C agrees, A presses "Xfer" again to transfer B's call to C.

B. Blind transfer:

Blind transfer doesn't 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 talking with B, B wants to speak to C with A's help of transferring. The operation steps are:

1. A presses "Bxfer"
2. A dials C's number, then A will be hang up no matter C receive the call or not.

4.5 Mute

Once the "Mute" key is been pressed, no voice will be transported to the other port anymore. Such state will be kept no matter user switches the mode among handset mode and handfree mode and earphone mode till 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 hanged 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:

A. When A is 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.

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

C. 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.

D. 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

AT820 supports 50 missed calls, 50 incoming calls and 50 dialed calls record. When the storage is full, the latest call will update the history. Press "Redial" soft key when standby, all the received (<-), dialed (->) and missed calls (!) will be listed. When the phone updates or is factory reset, all the call history records will be cleared.

A. Missed call

When there is a missed call, the LCD screen will display the amount of missed calls, press "Missed" to check the records. There is another way to obtain the missed call:

1. Press the "MENU" button, input right password.
2. Press the navigation button to choose “3 Call History” or press digit button “3”, then press “Select” soft key.
3. Press the navigation button to choose “Missed Calls” or press digit button “1”, then press “Select” soft key.
4. Press the navigation button to browse the missed call record. If there is no record, the LCD screen will display “List is Empty”.

B. Received call
1. Press the "MENU" button, input right password.
2. Press the navigation button to choose “3 Call History” or press digit button “3”, then press “Select” soft key.
3. Press the navigation button to choose “Answered Calls” or press digit button “2”, then press “Select” soft key.
4. Press the navigation button to browse the received call records. If there is no record, the LCD screen will display “List is Empty”.

C. Dialed call
1. Press the “MENU” button.
2. Press the navigation button to choose "3 Call History" or press digit button “3”, then press “Select” button.
3. Press the navigation button to choose “Dialed calls” or press digit button “3”, then press “OK” soft key.
4. Press up or down navigation button and check the received calls, LCD will show “List is Empty” if there is no received incoming call.

4.9 SMS function

A. Create new SMS
1. Press right navigation key.
2. Press “SMS” soft key.
3. Press “New” soft key.
4. Edit SMS context, user can switch the input method by press right navigation key then press "Alpha" to change to "Num", press "Num" to change to "IP", and press “IP” turn back to “Alpha”.
5. After finishing editing, press "Send" and input receiver’s phone number, name or IP. Press “Send” again to send out the message.
6. Press “Cancel” to exit.

B. Check new SMS

When there is a new SMS, voicemail indicator will flicker.
1. Press right navigation key.

2. Press “SMS” soft key.
3. Press “View” to check the new SMS.
4. Press “Reply” to reply the SMS, or press “Delete” to delete this SMS.
5. Press “Cancel” to exit.

4.10 Access Mode

There are two kinds of web access mode for AT820 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.

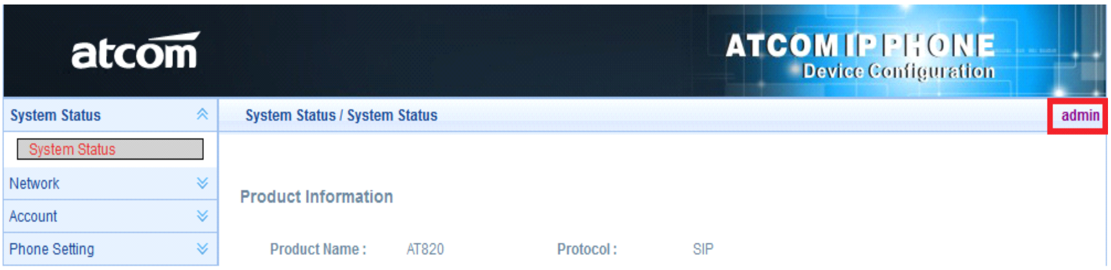
Password: default password is none.

4.11 Move the cursor

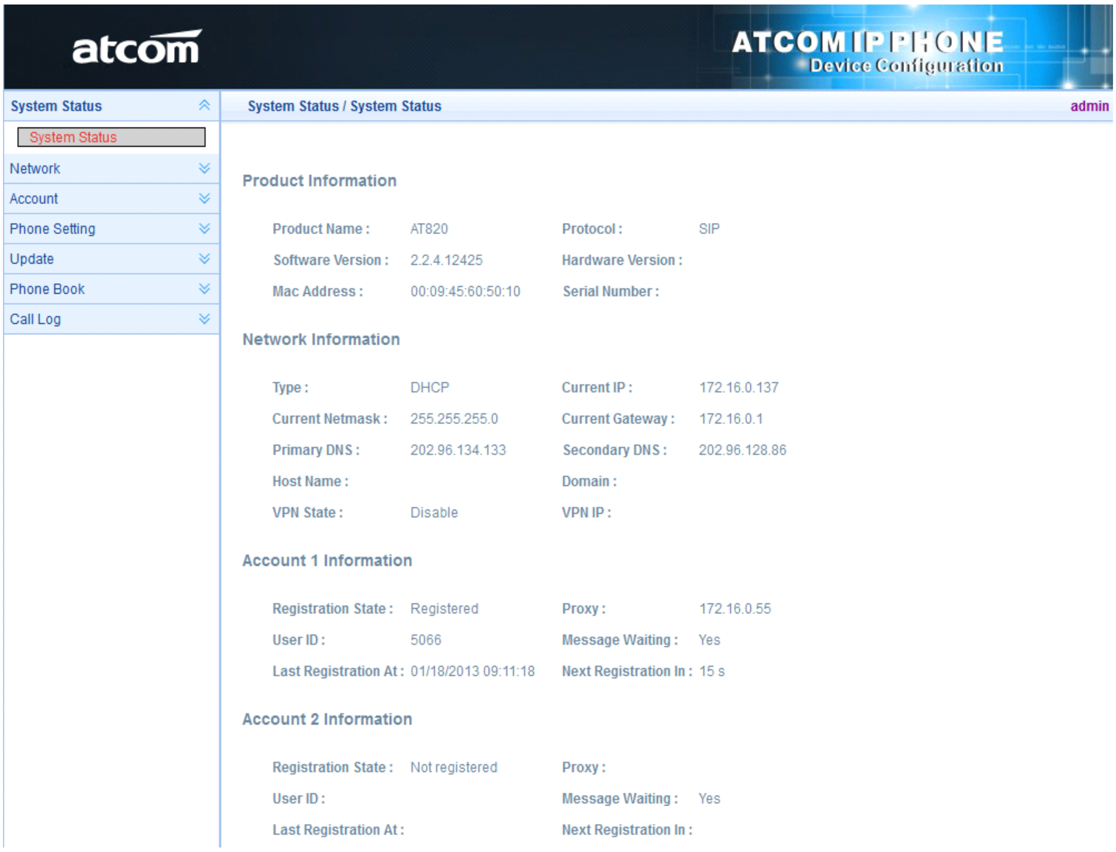
If user wants to modify the content when inputting information on the phone, please press navigation button “Right” at first, then user will see “<<”soft key and “>>”soft key. Use “<<” to recede the cursor and “>>” to linefeed.

5. Web settings

Input the phone’s IP addresses in the web browser and press “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 product information, network information and account information.

A. Product information: Shows the product name, protocol, software version, hardware version, MAC address and serial number.

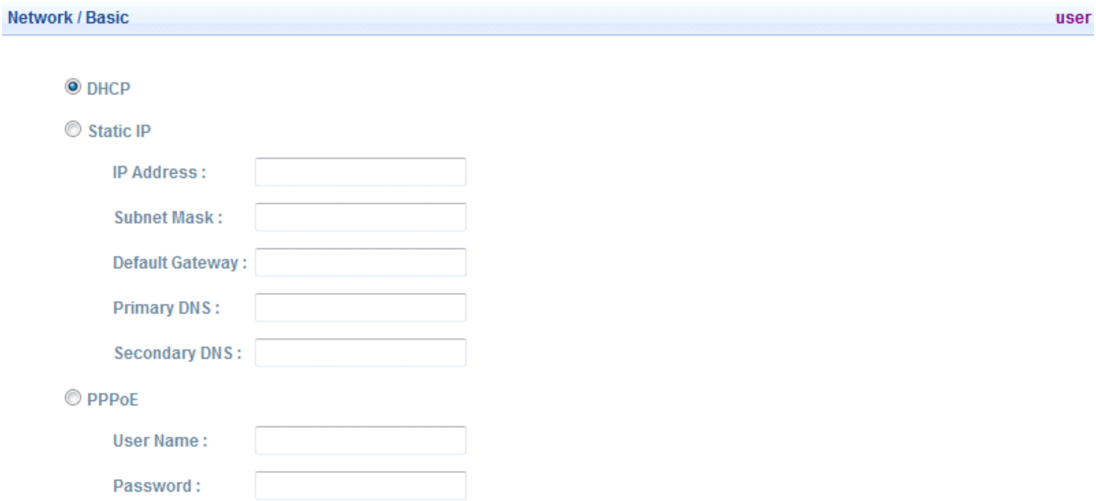
B. Network information: Shows the type of web connection, the current IP, the current netmask, current gateway, primary DNS, secondary DNS, host name, domain, VPN state and VPN IP.

C. Account 1/2 information: Shows the registration state, proxy, user id, message waiting status, last registration time and next registration interval.

5.2 Network

5.2.1 Basic

There are 3 ways for AT820 to connect to the internet: DHCP, Static IP and PPPoE, please choose one according to the actual situation.



A. DHCP: The IP phone will get IP address from DHCP server (such as router).

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 network management stuff.

   Parameters:

   1. IP Address: fixed IP address.
   2. Subnet Mask: LAN 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. When user wants to use PPPoE, selects "PPPoE", and makes the phone's LAN interface connected to the Ethernet interface of the modem, and then configures the user name and password provided by ISP.

   Parameters:

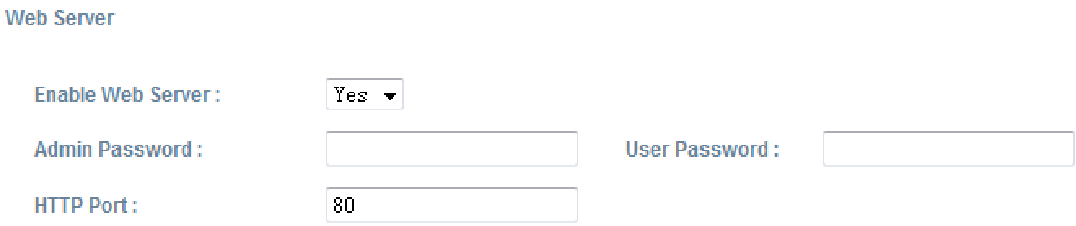
   1. Username: Set account user name.
   2. Password: Set account password.

   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



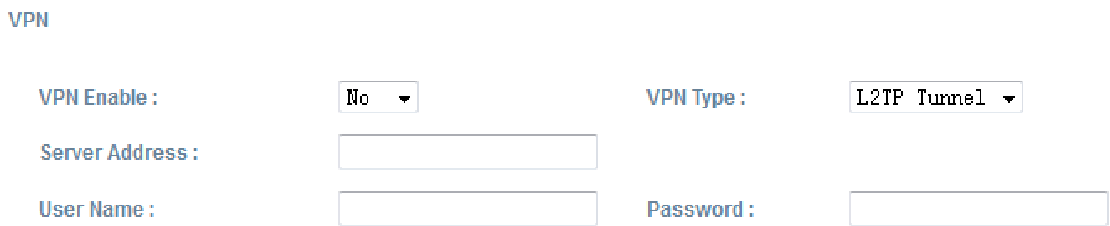
A. Enable Web Server: Enable or disable the web server. If set this item as "No", user will can't access the webpage. User can recover this function from the MENU: "9 Network"-> "7 Enable Web Server" or factory reset.

B. Admin Password: Set admin webpage access password, the default username for admin webpage is admin.

C. User Password: Set user webpage access password, the default username for user webpage is user.

D. HTTP Port: Set port for HTTP, the default is 80. If modify it, user will have to input http://ip-address:port to open the interface again. For example, the AT820'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 AT820's webpage.

5.2.2.2 VPN



A. VPN Enable: Whether enable VPN or not.

B. VPN Type: Do not support L2TP VPN with IPsec authentication.

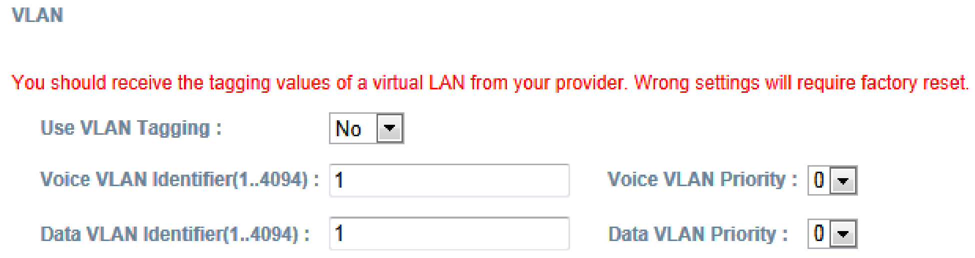
C. Server address: Set VPN server address.

D. Username: Set VPN account's username.

E. Password: Set VPN account's password.

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

5.2.2.3 VLAN



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

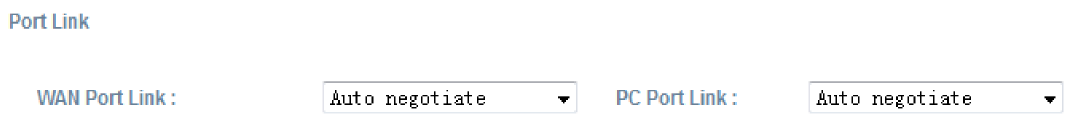
B. Voice VLAN Identifier (1..4094): Set voice VLAN ID, range from 1 to 4094.

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.

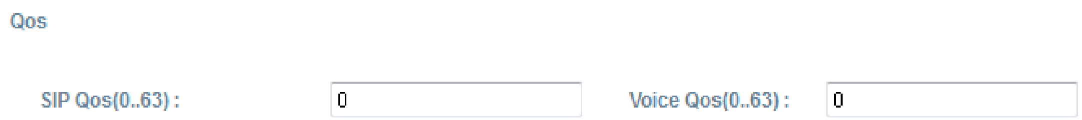
5.2.2.4 Port Link



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

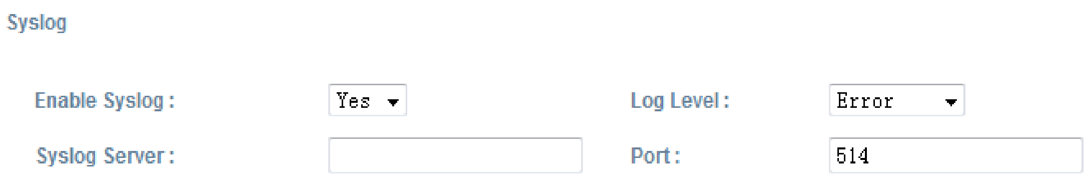


A. Qos：Quality of service.

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

C. Voice Qos: Set value for 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: Set syslog server address, which is a domain name or an IP address.

D. Port: Set 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). Ranges 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). Ranges 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). Ranges 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. Ranges from 0 to 65535. The default value is 8.

E. Sub Retry Intvl: Subscribe message's re-sending interval after a failure. Ranges 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

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

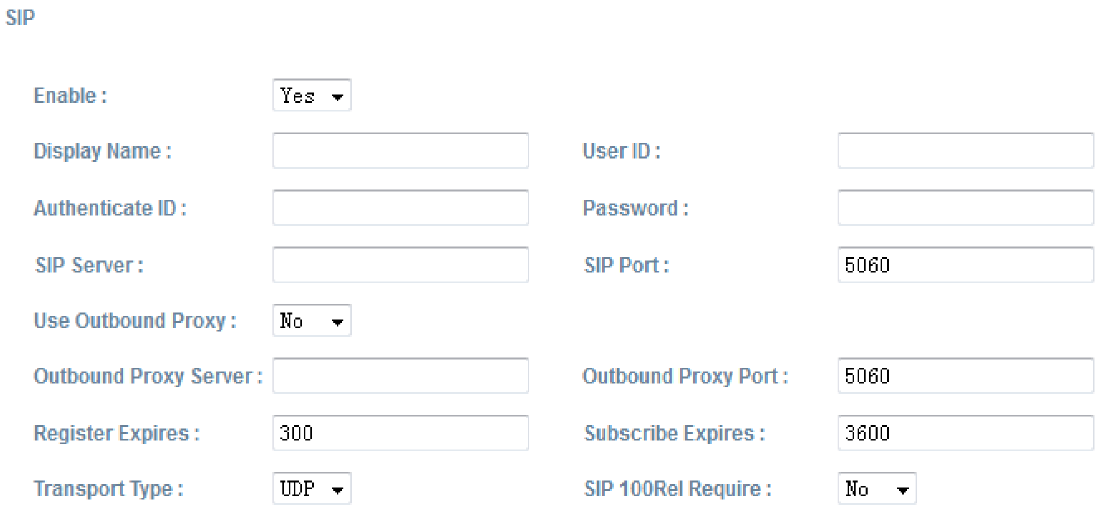
Enable Stun :	No	Stun Server :	

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

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

5.4 Account

5.4.1 SIP



A. Enable: Whether to enable this sip register or not.

B. Display Name: This name will display on the screen. When making outgoing calls, the callee will show the display name on its screen.

C. User ID: Set user ID for sip account.

D. Authenticate ID: Normally is the same with User ID, can also leave it as blank. If the server has requirements on it, set it as required!

E. Password: Set password for sip account.

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

G. SIP Port: Set SIP server port. Default to 5060.

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

I. 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.

J. Outbound Proxy Port: Set outbound proxy server port.

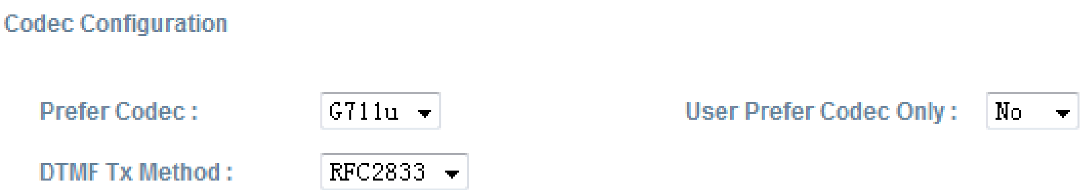
K. Register Expires: Set register expires time. Default to 300s. AT820 will auto configure this expires time to the server recommended setting if it is different from the SIP server.

L. Subscribe Expires: Set subscribe expires time. Default to 3600s.

M. Transport Type: UDP/TCP/TLS.

N. 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.

5.4.2 Codec Configuration

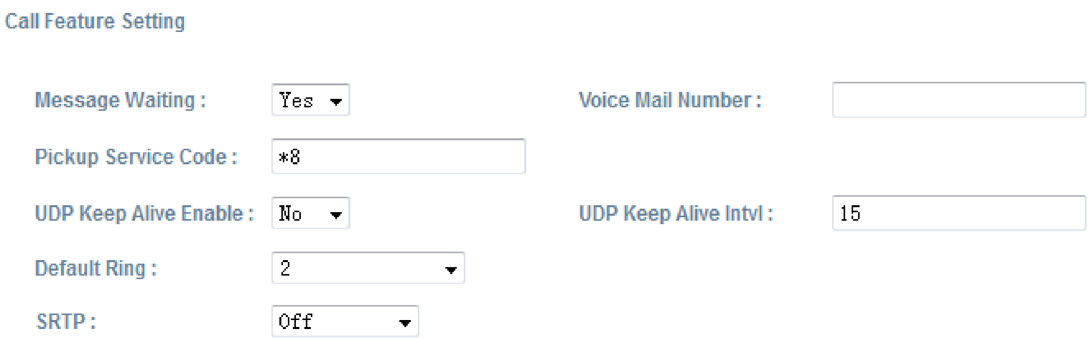


A. Prefer Codec: Select a preferred codec for all calls, it will use the prefer codec to create calls if the far end supports this codec. However, the actual codec used in a call still depends on the outcome of the codec negotiation protocol.

B. User Prefer Codec Only: Whether to use the preferred codec only for all calls or not. If select "Yes", the call may fail if the far end does not support this codec.

C. DTMF Tx Method: Select the method to transmit DTMF signals to the far end: InBand, RFC2833, and INFO. Default to RFC2833.

5.4.3 Call Feature Setting



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

B. Voice Mail Number: If user sets a voicemail number, then user can just dial it to enter the voicemail box.

C. Pickup Service Code: Set the code to pick up calls. Default to "*8".

D. 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.

E. UDP Keep Alive Intvl: Set the interval time to send UDP packet.

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

G. SRTP: Choose the mode of Secure Real Time Control Protocol. Supports from the server will be needed.

5.4.4 Dial Plan

Dial Plan

Dial Plan : (xxxxxxxxxxxx.)

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.

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

   1. Interdigit Long Timer---the details please refer to Interdigit Long Timer.
   2. Interdigit Short Timer---the details please refer to Interdigit Short Timer.
   3. 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.)".

B. 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 user input characters or symbols, such dials will be rejected.
   2. It will wait user to go on inputting 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 "#".

C. Syntax

   1. "x": Stands for a digit which ranges 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 xxxxxx.)

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>)

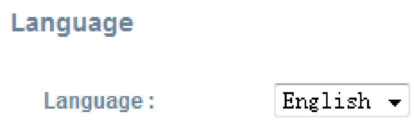
The following provides a Warm Line to a local office operator (1000) after 5 seconds, unless a 4 digit extension is dialed by the user.

(P5 <:1000> | xxxx)

5.5 Phone Setting

5.5.1 Preference

5.5.1.1 Language



Select the language shown on the LCD.

5.5.1.2 Output Volume（1~8）

A. Handset Volume: Specify handset volume grade, range from 1 to 8.

B. SpeakerPhone Volume: Specify speakerphone volume grade, range from 1 to 8.

C. Headset Volume: Specify headset volume grade, range from 1 to 8.

5.5.1.3 Input Gain

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

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

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

5.5.1.4 LCD

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

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

C. LCD Contrast: Specify the LCD contrast,range from 1 to 8.

5.5.1.5 Control Timer Values (Seconds)

A. 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. Range from 1 to 99, the default value is 20.

B. 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. Range from 1 to 99, the default value is 8.

C. Reorder Delay: The phone will wait for the reorder delay after far end hangs up to play the reorder tone. 0 means playing it immediately. Range from 0 to 99, the default value is 5.

5.5.1.6 Data And Time

A. 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.

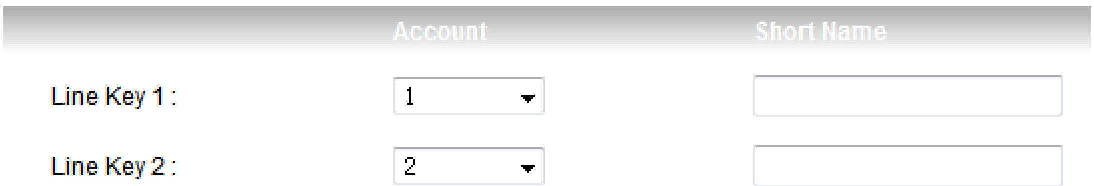
The display time will be 1 hour early than the standard time.

B. 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.

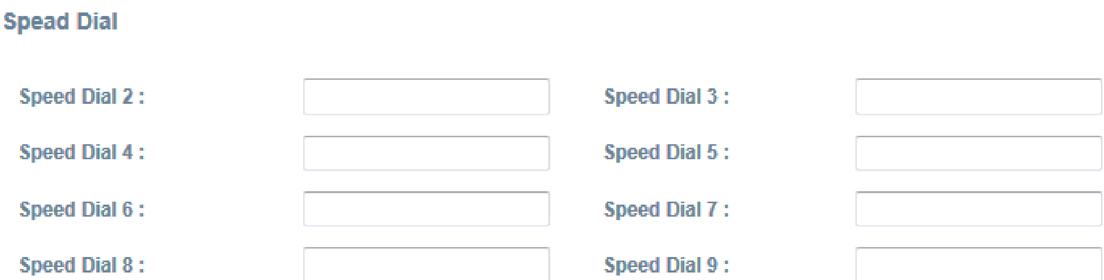
5.5.2 Line



Set name for each line showed on LCD, if keep it as blank, it will use phone SIP account user ID as short name. If set the two lines' account as the same account, both line can make calls by using one account, but the other account is equal to be forbidden, and it can answer incoming calls only with the first line.

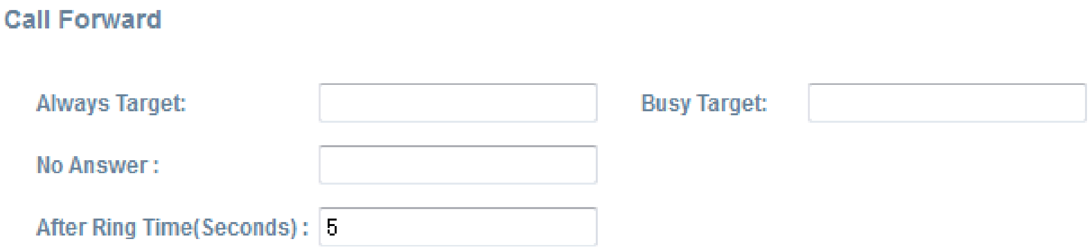
5.5.3 Features

5.5.3.1 Speed Dial



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

5.5.3.2 Call Forward



A. Always Target: Set a target number that forward all the incoming call to.

B. Busy Target：Set a target number that forward the incoming call to when user's phone is busy.

C. No Answer: Set a target number that forward the incoming call to when no one answer it.

D. 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 seconds.

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.

5.5.3.3 Call Settings

A. 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".

B. 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".

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

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

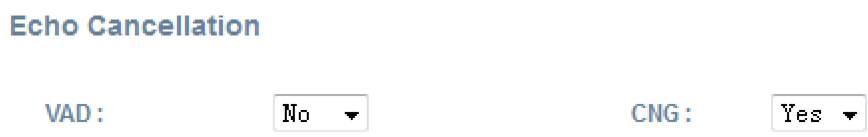
E. Auto Answer: Whether to enable auto answer function or not. The phone will answer the incoming call automatically through headset or hand-free while enable it.

5.5.3.4 Others

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

5.5.4 Voice

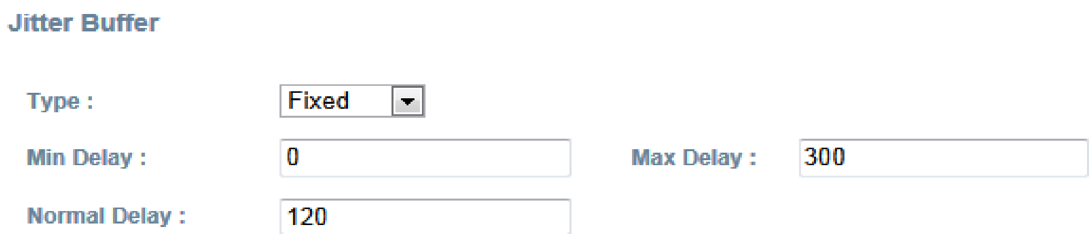
5.5.4.1 Echo Cancellation



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

B. 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.

5.5.4.2 Jitter Buffer



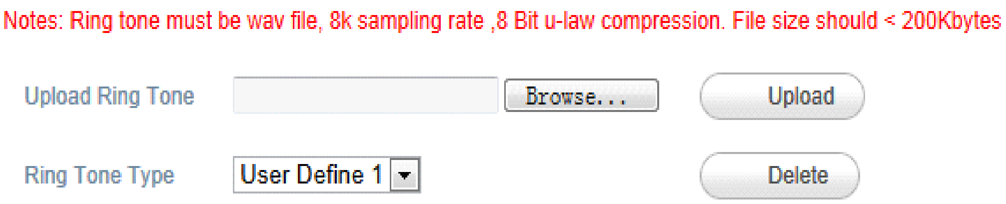
A. Type: Specify type for jitter buffer: Adaptive or Fixed. The SPA can buffer incoming voice packets to minimize out-of-order packet arrival. This process is known as jitter buffer.

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

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

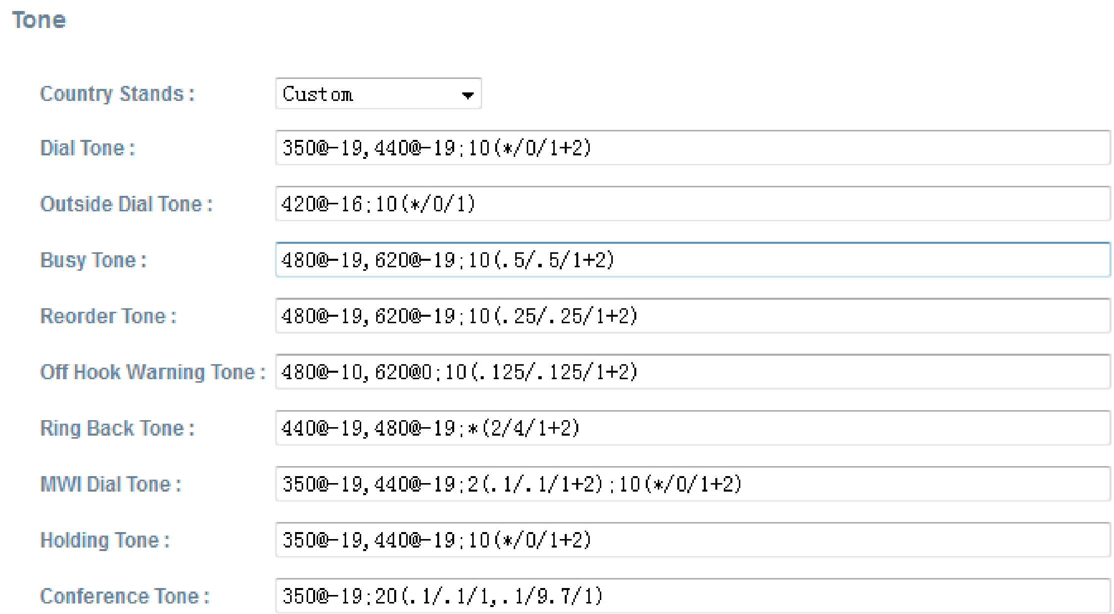
D. Normal Delay: It is used as fixed delay which should be between min delay and max delay. Ranges from 1 to 998. The default value is 120.

5.5.5 Ring



Administrator can upload 2 user define rings for AT820. The ring file should be wav (8k, 8bit, u-law) and no larger than 200kb.

5.5.6 Tone



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:

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

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

C. 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.

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

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

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

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

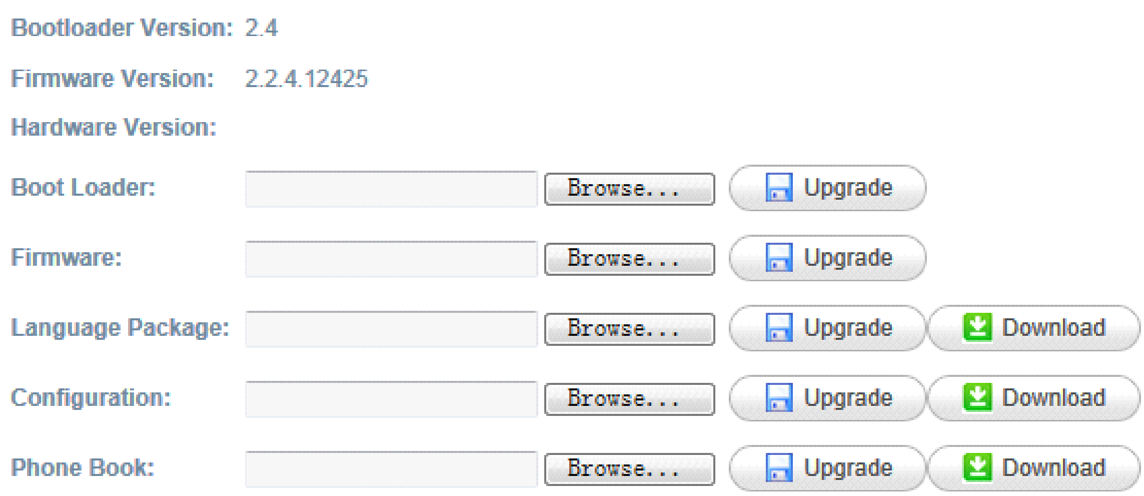
H. MWI Dial Tone：This tone is played instead of <Dial Tone> when there are unheard messages in the subscriber’s mail box.

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

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

5.6. Update

5.6.1 Manual Update



A. Boot Loader: Download Boot file on ATCOM's webpage, then find it from user’s PC to upgrade.

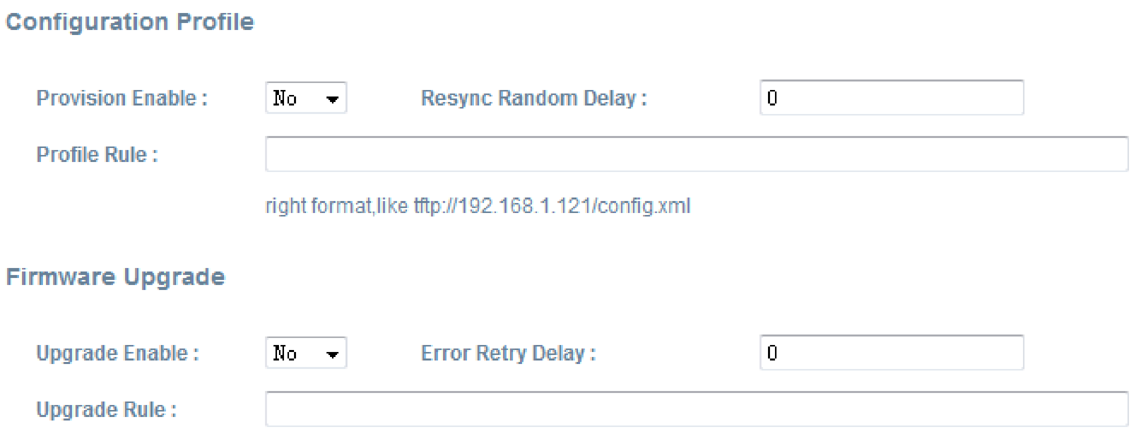
B. Firmware: Download firmware on ATCOM's webpage, then find it from user’s PC to upgrade.

C. Language Package: Upgrade or download language.xml file.

D. Configuration: Upgrade or download config.xml file.

E. Phone Book: Upgrade or download phonebook.xml.

5.6.2 Auto Provision



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

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

C. 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:

1. TFTP rule: tftp://ip-address/config.xml.
2. FTP rule: ftp://ip-address/config.xml.
3. 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.

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

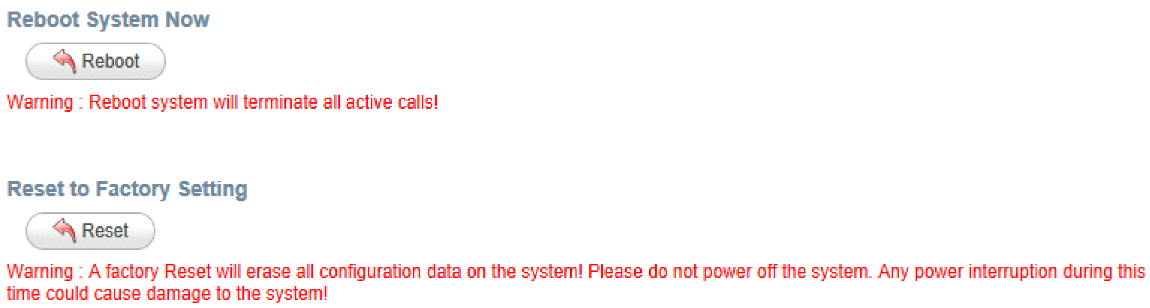
E. Error Retry Delay: Set a retry delay when error occurs.

F. 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:

4. TFTP rule: tftp://ip-address/AT820.xml.
5. FTP rule: ftp://ip-address/AT820.xml.
6. HTTP rule: http://ip-address/AT820.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.

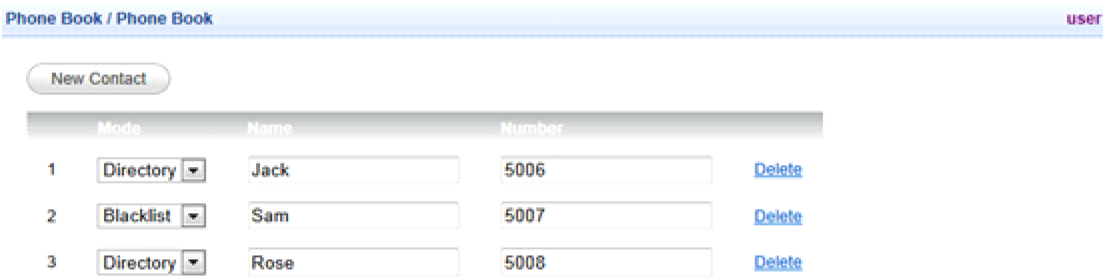
5.6.3 Reset & Reboot



A. Reboot: Press to reboot the phone. Such action will terminate all active calls!

B. Reset: Press to set AT820 to factory default, please backup the config.xml, phonebook.xml and attendant_keypad.xml before reset.

5.7 Phone Book



A. New Contact：Click on it to create a new contact.

B. Mode:Directory mode and blacklist mode are optional. Maximum 100 directory recorders, and 50 blacklist recorders.

C. Delete: Delete the current recorder.

5.8 Call Log

A. Dialed: The recorder of the dialed calls, maximum 50 items.

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

Call Log / Received user

	Name	Number	Time
1 :	5057	5057	05/01 11:03
2 :	5055	5055	05/01 11:02
3 :	5058	5058	05/01 11:01

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

Call Log / Missed user

	Name	Number	Time
1 :	5054	5054	05/01 11:06
2 :	5055	5055	04/01 09:07

6. FAQ & Trouble Shooting

6.1. Frequently Asked Questions

6.1.1. How to make the factory reset

There are two ways to make the factory reset:

A. Factory reset from keyboard，steps are:
   1. Press soft key "MENU", input right password;
   2. Find and select "Factory Reset" ;
   3. Press "OK", then the phone will restart automatically and factory reset.

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 AT820.FW;
3. Make sure the TFTP server and the phone are contacting with the same switch and reachable to each other;

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

The way to make direct IP call is similar to digit call, steps are:

1. After hooking off, press right navigation key and press the third soft key "Num" to switch to "IP".
2. Using * key to inputting ".". If user wants to dial 192.168.1.100, then press 192*168*1*100.
3. After inputting the IP address, press dial key to dial it out.

6.2 Trouble Shooting

6.2.1 The phone can’t register successfully

A. Check the IP address, and if the mode of WAN port is DHCP, please make sure the DHCP server is in service.
B. Check the gateway.
C. Check the DNS.
D. Make sure the information of the account is consistent with which offered by the service supplier.
E. Make sure the SIP server is on.
F. Check the port of the SIP server whose default value is 5060.

6.2.2 The phone can’t obtain IP address

A. Make sure the cable has been connected to the WAN port of the phone.
B. Make sure the cable and the switch’s port that connected with the cable is available.
C. Make sure the DHCP server is on and there are some assignable IP addresses in the address pool.
D. Try to change the LAN port mode as Static.

6.2.3 Only one part can hear the voice during the call

A. Make an IP dial-up call to make sure the headset's telephone receiver and Mic are normal.
B. Enable STUN on web page.
C. Set STUN server as stun.3cx.com.
D. Click on "submit" and wait for the phone to restart.
E. Try to make calls again.