ATCOM

ATCOM®

AT800 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.

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AT800 Series Overview

AT800D/AT800DP   AT800/AT800P

Model	LCD Screen	PoE
AT800DP	Yes	Yes
AT800D	Yes	No
AT800P	No	Yes
AT800	No	No

Interfaces

- Power input: DC 6~15V, 5W
- LAN: RJ45 port
- PC: RJ45 port
- Headset jack 1 : RJ9 port

Hardware

- LCD: 128×32   dot matrix

- FLASH: 8MB
- SDRAM: 16MB
- CPU: 262MHz Dual Core
- LED indicator：Login indicator , MWI, Mute indicator, Call indicator
- Support HAC(Hearing Aid Compatible)

Software

- SIP 2.0 (RFC3261)
- 1 SIP account
- STUN
- Jitter Buffer,VAD,CNG
- G.711A/u, G.722, G.723, G.726-16, G.726-24, G.726-32, G.726-40, G.729, iLBC, Lin-16 Codec
- G.168 compliant 96ms echo cancellation
- Support SIP domain，SIP authentication(none,basic,MD5),NDS.
- Support inband , RFC2833 and SIP info DTMF
- Call Forward, Call transfer, Call hold, 3-way conference, Redial
- DND(Do Not Disturb), Auto answer, Blacklist, Block Caller ID, Block Anonymous Call
- Dial rule, IP call
- Phone book with 100 records, 50 answered records, 50 missed records, 50 dialed records
- Support HTTP, FTP, TFTP updating the configuration
- Syslog
- Support SNTP client
- WEB access
- Support different level user management

Network

- LAN: Static IP, DHCP, PPPoE（ADSL)
- PC: Bridge and route mode
- Support NAT
- Support VLAN（DATA VLAN and VOICE VLAN）
- LAN support DHCP Server
- Qos support Diffserv

Management and Maintenance

- Support firmware updating in safe mode
- Support different level user management
- Configuration via web , keypad

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- Support system log and call log
- Auto provision

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
- VAD/CNG
- DNS: Domain Name Server
- TFTP: Trivial File Transfer Protocol
- HTTP: Hypertext Transfer Protocol
- FTP: File Transfer Protocol
- STUN: Simple traversal of UDP over Nats
- SNTP: Simple network time protocol
- ICMP: Internet Control Message Protocol

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 arrow, stands for recycle. The number 20 stands for the years of environment protection. Please note the years of environment protection is not discarding year nor usage life.

Operating Requirement

- Operation temperature: 0~45°C (32~113°F)
- Storage temperature: -5~55°C (23~ 131°F)
- Humidity: 10 to 90% no dew

Packing List

- AT800x IP phone
- Power adapter (Input: AC 100~240V, 50/60Hz; Output: DC 12V ,500mA)
- Manual CD

Note: There is no power adapter in AT800DP and AT800P packaging by default. Please contact ATCOM if user wants to use with power adapter.
The default packing is exclusive of headset. User can buy it from ATCOM additionally, or use with any RJ9 headset. Refer to how to connect headset .

Getting started

Keypad

MWI：Message Waiting Indicator.
Ringing Indicator

Up/Down key：Adjusts the volume
of the headset and ringer.
Scroll through the displayed
information.

Mute key：Toggles the Mute
feature on of off.

Keypad：Provides the digits,
letters,and special characters
in context-sensitive applications.

Soft keys：Labels automatically to
identity their context-sensitive features.

Headset key：Toggles and indicates
the headset mode.

Login key：Toggles the Login feature
on of off.

Multi-function keys：：Function of this
buttons can be set according to
the requirements.

(AT800D/AT800DP)

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(AT800/AT800P)

Description of the keys and Screen:

Soft buttons	Press to select an feature shown in the soft button features
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 button features.
Navigation button	1. Move cursor up and down when pressed in menu reviewing. 2. Adjust output volume when pressed in talking state 3. Adjust ringer volume when pressed in idle state.
1~9,*#	For entering numbers, letters or characters
Programmable Key	Function key can be defined as Speed dial key, CFWD, Hold, Xfer, ConF, DND, Redial, Voicemail.
Menu	Enter Keypad menu settings
Mute button	Mute/un-mute mic-phone in talking state
Headset button	Pick up phone to make outgoing calls or answer incoming calls.
MWI	Message waiting indicator/ Ringing indicator
Login button	1. Press to enter login interface to input SIP account username and password to register to sip server. 2. Press in login-ed state to input SIP account password to logout.

Indicator status

1. Login indicator

LED Status	Frequency	Description
Red /Flash	500ms on/500ms off	LAN Disconnected
Red	-	SIP Unregistered
OFF	-	SIP Registered

2. Mute indicator

LED Status	Frequency	Description
Red	-	Mute
OFF	-	Unmute

3. Voicemail indicator

LED Status	Frequency	Description
Red/Flash	500ms on/500ms off	New message
Red/Flash	250ms on/250ms off	Call ringing
Red	-	Normal working(AT800/AT800P)
OFF	-	No new message

4. Call indicator

LED Status	Frequency	Description
Red	-	In a conversation
Red/Flash	500ms on/500ms off	On-hold
OFF	-	Idle state

When boot up,all indicators will be lighted once.

When phone is stand by,all the indicators are off. For AT800/AT800P, WMI indicator will be lighted when the phone is working.

When updating firmware, all indicators will be flash in frequence 1s on/ 1s off.

Installation

Power on

There are two options for network and power source connections. The administrator will advise which one to use.

- DC Power

- Power over Ethernet (PoE)
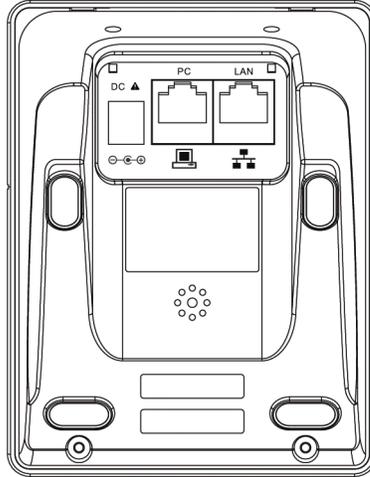
DC Power

1. Plug in network cable at LAN port.

2. Plug in the power jack with DC 12V/500mA.

3. The phone will show a loading bar and ATCOM logo, and then come into idle display after boot up absolutely.

PoE

Power over Ethernet. Using a regular CAT5 cable, the phone can be powered from a PoE (IEEE 802.3af) compliant switch or hub.
To connect the PoE:

- Plug in network cable at LAN port which is connected to a PoE switch.

Headset

To connect the headset:

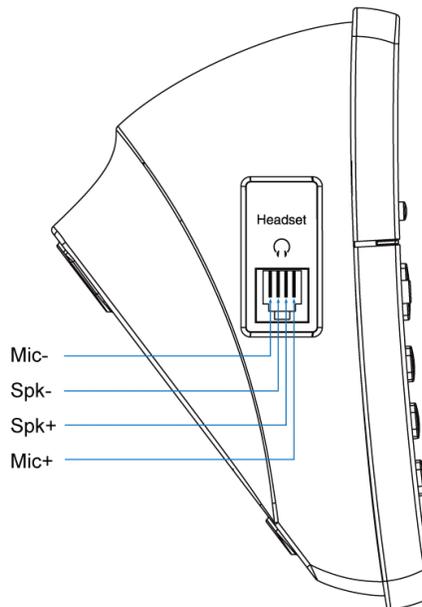- Connect the RJ-9 headset to the headset jack on the left side of the phone. The pin sequence of RJ9 port on the phone is like following picture:

Sip Register

Before using the phone, user have to follow below steps to register it.

To register the phone:
1. Viewing the Network Information:
    - Press Menu softkey
    - Enter login password (default: 123)
    - Scroll to [9.Network] and press select softkey
    - Select [1. LAN] to view the network information:
        Current IP : 172.16.0.110, etc.
        Current Net mask:255.255.255.0
2. Access the IP Phone administrator web user interface:
    - Launch a web browser on a computer that can reach the phone on the network.
    - Enter the IP address of the phone in web browser address bar. For example:
        http://172.16.0.110

3. Configuring the SIP parameters:
   - Click admin on the right corner of the webpage to login administrator interface
   - Click Account tab
   - Under SIP parameters, make the necessary configuration changes to the fields shown in the following table and click submit:

Enable: yes/no	Enable/disable the SIP registration
User ID:	The username, provided by ISP.
Password:	The password, provided by ISP.
SIP Server:	The registrar and proxy server (we don't distinguish between the registrar server and the proxy server right now).
SIP Port:	The default SIP port is: 5060, user can modify it to fit the SIP server's port.
Transport Type:	Optional transport type, can be set as UDP/TCP/TLS

After that, click submit button to save the configuration. The phone will be restart automatically to make the changes effect.

For AT800/AT800P, it's not able to view the current IP by man-machine interface because there's no LCD screen. User should login via PC port. Please refer to Web Interface part, and log in the web interface through PC port IP address.

Warning: If the parameter **remember account** is set to **yes**, the account information will be saved automatically, user don't have to make a manual login after a restart; if it is set to **no**, user will have to enter the **User ID** and **password** manually every time restating the phone.

Basic Phone Functions

Place or Answer a call

- Press headset button to place or answer a call
- If the auto answer parameter is set to yes, the incoming call will be answered automatically.
- If dial assistance is enabled, when placing calls, user will see a list of closely-matched phone numbers on the screen when dialing. To choose one of them: using the navigation button to scroll down and highlight a phone number on the LCD screen, press dial to complete the call.
- Enter the personal directory, use navigation button to scroll down to highlight an entry then press Dial to place a call
- Enter the Redial list menu, use navigation button to scroll down to highlight an history number, and press Dial to complete the call.
- Use the programmable key to place a call to the pre-configured number.

For AT800/AT800P, user can place a call as followings:

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1.   Press headset button, enter number and # (send key) to dial
2.   Press headset button, enter a speed dial number, the pre-configured number will be dialed
3.   To hang up the active call, press headset button again.

Note: If setting * as the send key, use * instead of #, If send key is set to none, wait for about 10 seconds, the number will be dialed automatically.

Volume and Mute

Volume

To adjust the output volume in a conversation, press right navigation button to increase the volume, or press left navigation button to decrease the volume.

Pressing the right/left navigation button on idle state can adjust the volume of the phone's ringer.

Mute

To mute the call in a conversation, press the Mute button, the button lights in red. Press the Mute button again to unmute.

Hold

•   To put a call on hold in a conversation, press the hold softkey to hold the call
•   When setting a programmable key to hold function, press the specified key will hold the call.

For AT800/AT800P, using programmable key to put a call on hold, refer to Programmable key for more details.

Resume a call

•   To resume a held call, press Resume softkey.
•   When setting a programmable key to hold function, pressing the specified key again in hold mode can resume the call.
•   For AT800/AT800P, using programmable hold key can resume the call, refer to Programmable key for details.

Ending a call

Press headset key to hangup.

Transfer

There are two types of call transfer:
- Attended   Talking to the third party before transfer the call.
- Blind   Transfer the call directly without talking to the third party.
- For AT800/AT800P, use programmable Xfer key to make attended transfer, refer to Programmable key for more details.

Attended transfer

- When in a conversation, press Xfer key to enter attended transfer mode and enter transferred number.
- Press Dial key to call the third party.
- Do one of the following:
  1. To transfer the call before the third party answers, please press Xfer button directly in Called party ringing state. And when the third party answers, the other two parties can talk to each other.
  2. To transfer after the third party answers, please wait for the transferred call is answered and then press Xfer key to transfer the call. After that ,the other two parties can talk to each other.

Blind transfer

- When in a conversation, press Bxfer key to enter blind transfer mode and enter the transferred number.
- Press Dial key to call the third party. The phone will hang up automatically after dial the third party. And the other two party can talk to each other.

3-Way conference Call

To initiate a conference call:
- Press the **Conf** softkey button in a conversation to hear a dial tone.
- Enter the telephone number of the third party and press dial key to send it out.
- 1. Press the **Conf** softkey in Called party ringing state. When the third party answers the call, it will enter 3-way conference automatically.

  2. After the third party answers the call, pressing Conf key to enter 3-way conference.

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- Press Join key to leave the conference and make the two other parties still in a conversation.
- Press headset key to hang up the conference, and the two other parties will all hang up.

For AT800/AT800P, it can only place a Three-way conference call using the programmable key. Please refer to Programmable key part for more details.

Redial

To redial a number:
- Press Redial softkey and select the desired number.
- Press the programmable key which is set to Redial function, it will dial out the last call in redial list immediately.

Missed calls

For AT800D/AT800DP, there will be a prompt display on LCD screen to indicate the missed calls.
To view all missed calls:
- If the LCD screen shows a missed call, press the Missed softkey to view a list of missed calls.
- If the LCD screen does not show a missed call:
  1. Press **Menu** softkey
  2. Scroll to **Call history** and press **select**
  3. Scroll to **Missed calls** and press **select**.
     a) Press dial to **dial** the number
     b) Press edit to **edit** the number
     c) Press delete to **delete** the number

Last Call Return

For AT800D/AT800DP, to return a missed call:
- If the LCD screen shows a missed call,press the **LCR** softkey to return the last missed call.
- If the LCD screen shows more then one missed call, press the **Missed** softkey button and select a record then press **dial** key to return the call.

Call Forward

It supports all forward, busy forward and no answer forward. When Call Forward is enabled, LCD will display "Call forwarded" on AT800D/AT800DP.

To forward calls:
- Press Menu
- Scroll and select **Call forward**:
  1. Forwarding calls to a single number. select **CFWD All Number,** enter the forward number;

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    2. Forwarding calls when phone is busy. Select **CFWD Busy Number**, enter the forward number;

    3. Forwarding calls when no one answer the call. Select **CFWD No Ans Number,** enter the forward number.

- Press **OK** to save the forward number.
- Press **save**.

    For AT800/AT800P, use the web interface to configure the call forward function. Please refer to Call Forward part for more details.

Do Not Disturb

Do Not Disturb can reject all the incoming calls. If there is an incoming call, the phone will not ring and only display 1 missed call. When Do Not Disturb is enabled, LCD will display "Do Not Disturb" on AT800D/AT800DP.

To set Do Not Disturb, choose one of the following options:
- Set Do Not Disturb via Web Interface
  1. Login to the IP Phone Web Interface
  2. Click **Phone Setting**, then **Feature**
  3. Set Do Not Disturb to yes or no
- Set Do Not Disturb via keypad
  1. Press **menu** and scroll to select **Preference**
  2. Scroll to select **Do Not Disturb,** then press **edit**
  3. Press **Y/N** to set Do Not Disturb, then **OK**
  4. Press **Save**

For AT800/AT800P,
1. Using the programmable key to activate or inactivate Do Not disturb function, refer to Programmable key for more details.
2. Using the web interface, refer to Feature part for more details.

Personal directory

Using personal directory to view, add, edit and delete the contacts. It supports up to 100 phonebook entires. To access the Personal Directory, press the **Dir** softkey in idle state, or press **menu** softkey then select **Directory**.
Choose one of the following:
- Press **add** to add an entry
- Press **delete** to delete the selected entry
- Press **edit** to edit the selected entry
- Press **cancel** to exit the directory

Call History

To view the Call History Lists:
- Press **menu** softkey
- Scroll to **Call History** and press select
- There are three options：
  1. Redial List
  2. Answered Calls
  3. Missed Calls
- Press select to view the list

Choose one of the following options:
- Press d**ial** to dial the selected number
- Press **delete** to delete the selected history
- Press **edi**t to edit the selected number
- Press **cancel** to exit

Advance Phone Features

Privacy and Security Features

Block Caller ID

Block Caller ID from the phone to place an anonymous call.

To block Caller ID:
- Press **menu** softkey
- Scroll to [**5. Preference**] and press select
- Select [**Block Caller ID**] and press edit
- Change setting by press **Y/N**, then **OK** to save setting, **Cancel** to cancel setting and exit
- Press **save** again.

Block Anonymous Calls

It will block the anonymous incoming calls.

To block anonymous calls:
- Press **Menu** softkey
- Scroll to **Preference** and press **select**
- Scroll to **Block Anonymous Call** and press **edit**

- Change setting by press **Y/N**, then **OK** to save setting, **Cancel** to cancel setting and exit
- Press **save** again.

Auto Answer

The phone will answer the call automatically in headset.
To auto answer calls:

- Press **Menu** softkey
- Scroll to **Preference** and press **select**
- Scroll to **Auto Answer** and press **edit**
- Change setting by press **Y/N**, then **OK** to save setting, **Cancel** to cancel setting and exit
- Press save again.

Speed Dial

Configure up to 8 speed dials. Each speed dial can be a phone number or IP address.

To create or edit a speed dial:

- Press **Menu** softkey
- Scroll to **Speed dia**l and press **select**
- Press **edit** to input the speed dial numbers
- Enter a phone number or IP Address for speed dial.

To call speed dial numbers, choose one of the following options:

- Press headset key and enter the number for the speed dial,E.G. Set number 123456789 in speed dial2, then dial 2, the phone will call the number 123456789.
- Press **Menu**, select **speed dial**, choose the number, and press **dial**.

LCD screen

Customize LCD screen contrast and backlight.

LCD Contrast

To adjust the contrast of the LCD screen:

- Press the **Menu** softkey
- Scroll to **LCD Contrast** and press **select**
- Use the **right navigation key** to increase and **left navigation key** to decrease the LCD contrast level.
- Press **save** to save the change, **cancel** to cancel.

Backlight

To adjust the screen Backlight level:

- Press Menu softkey
- Scroll to Back Light and press select
- Use the **right navigation key** to increase and **left navigation key** to decrease the LCD Backlight level.
- Press **save** to save the change, **cancel** to cancel.

Ring Tones

To change a ring tone:

- Press **Menu** softkey
- Scroll to **Ring Tone** and press **selec**t
- Press change
- Scroll through the ring tones and press Play to play the highlighted ring tone. Press select to choose it as incoming call ringtone.

Date and Time

Set NTP or manual date and time:

- Press **Menu**
- Scroll to **Date and Time**, and press **select**
  1. Time Mode. press **chang** to change time mode between **NTP** and **Manual**, then press **OK** to submit change.
  2. Manual Settings. Change day and time manually
  3. NTP Settings. Get the date and time information from a NTP Sever. Modify **NTP server** and **Time Zone**.
  4. Date Format. Change date format between **day/month** and **month/day.**
  5. Time Format. Change time format between **12hr** and **24hr.**
- Press **save**

Voicemail Number

Set voicemail number:

- Press **menu**
- Scroll to **Voice Mail**, and press **select**
- Enter voicemail number then **save**

Reboot

Reboot phone:
- Press **menu**
- Scroll to **reboot**, then press **select**
- Press **OK** to reboot the phone, **Cancel** to exit without rebooting the phone.

Phone Status

To view information about the phone:
- Press **menu** softkey
- Scroll to **Status** and press **select**
- Review the following information:
    1. Phone: the phone date and time, and phone up time
    2. Ext1:   the extension register status, and register proxy
    3. LineKey1: the LineKey1 status

Product Info

To view information about the Product:
- Press **menu** softkey
- Scroll to **Product Info**, then press **select**
- Review the following information:
    1. Product Name: AT800
    2. Serial Number: Product Serial Number
    3. Software Version: Software version of the product
    4. Hardware Version: Hardware Version of the product
    5. MAC Address: MAC address of LAN port
    6. Protocol: SIP

Network

Set mode on LAN port and PC port.AT800x supports static IP, DHCP client and PPPoE on LAN port. It can work as bridge and router mode on PC port.

LAN port

To view and edit LAN port Network parameters:

- Press **menu** softkey
- Scroll to **Network**, and press **select**
- Select **LAN**,
    1. Connect Type: DHCP, PPPoE, Static
    2. Current IP: IP address
    3. Hostname: Hostname
    4. Domain: The domain name
    5. Current Netmask: The current netmask
    6. Current Gateway: The gateway address
    7. Enable Web Server: Access by web interface. Default is set to yes. If disabled, user will not able to access webpage.
    8. Non-DHCP IP Address: Static IP address
    9. Non-DHCP Subnet Mask: Static netmask address
    10. Non-DHCP default route: Static gateway address
    11. Non-DHCP DNS1/DNS2:   Primary DNS and Secondary DNS working in static IP.
    12. PPPoE Login Name: username in PPPoE mode
    13. PPPOE login Password: password in PPPoE mode

PC port

To view or edit the PC port network parameters:

- Press menu soft key
- Scroll to **Network**, and press **select**
- Select **PC**,
    1. LAN IP: the PC port IP address when performed as a router
    2. Netmask: the subnet netmask
    3. NAT: make the device performing as an router which can forward IP packet
    4. Bridge Mode: set to yes will make the LAN port and PC port working in bridge mode. And DHCP service will stop in this mode.
    5. DHCP Service: set to yes, the PC port will work as a DHCP Server.
    6. Lease Start IP: the start IP for DHCP Server.
    7. Lease End IP: the end IP for DHCP Server.
- Press **save**

Web Interface

AT800x is working at route mode by default, and provide DHCPD service on PC port. Connect computer on PC port, and make PC work in DHCP mode. After getting an IP address as 192.168.10.20 by default, access WEB interface by following steps:

- Enter the IP address (default: 192.168.10.1) in the browser
- It enters user webpage by default, and user can click **admin** button on the right corner to switch to **admin** login.

System status

The system status display basic information about the system:

System Status	System Status / System Status			
System Status				user
Network	**Product Information**			
SIP Setting				
Account	Product Name :	AT800	Protocol :	SIP
Phone Setting	Software Version :	2.1.2.10786	Hardware Version :	1.4
Update	Mac Address :	00:09:45:00:02:02	Serial Number :	
Phone Book	**Network Information**			
Call Log				
	Type :	DHCP	Current IP :	172.16.0.146
	Current Netmask :	255.255.255.0	Current Gateway :	172.16.0.1
	Primary DNS :	202.96.134.133	Secondary DNS :	202.96.128.86
	Host Name :		Domain :	
	Account Information			
	Registration State :	Registered	Proxy :	172.16.0.55
	User ID :	5025	Message Waiting :	Yes
	Last Registration At :	09/03/2012 11:54:07	Next Registration In :	78 s

Network

There are three sub-menus under network option: LAN Port, PC Port, Advance.

LAN Port

Set IP mode on LAN port. It supports Static IP,DHCP and PPPoE mode.

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1. DHCP   Obtain IP address from Router/ DHCP server
2. Static IP   Set Static IP:
   a) IP Address
   b) Subnet Mask
   c) Default Gateway
   d) Primary DNS
   e) Secondary DNS
3. PPPoE   Connect LAN port to a modem. Set PPPoE username and password provided by ISP.
   a) User name
   b) Password

PC Port

It supports Bridge and router mode on PC port.

1. As Bridge   LAN port and PC port work in bridge mode
2. As Router   LAN port and PC port work in router/NAT mode
   a) IP Address—PC port IP address
   b) Submit Mask—PC Port submit mask
   c) Enable DHCP server: enable/disable DHCP service

d) Start IP address—start lease IP address for DHCP service

e) End IP address—end lease IP address for DHCP service

Warning: start/end IP address and PC port IP address MUST be in the same subnet.

Advance

1. **Web Server**
   a) **Enable Web Server: yes/no**, enable/disable web server; system will reboot after submit, if disable web server, the web interface will not be available anymore.
      If web server is disabled on AT800/AT800P, it needs restore to factory settings by long press * and # key when powering on. Please refer to How to restore to factory settings.
   b) **Admin password**: set admin password here
      Tips: input http://ip-address/index in the browser to enter admin web interface.
   c) **User password**: set user password here
      Tips: input http://ip-address in the browser to enter user web interface.
   d) **HTTP Port**—the web server port, default is 80. If user has changed the http port, e.g. 8080; to log in web interface by URI: http://ip-address:port in the browser.
2. **VLAN**
   a) **Use VLAN tagging: yes/no**, enable/disable VLAN tagging, default to no.
   b) **Voice VLAN Identifier**—set voice VLAN identifier, range from 1 to 4096
   c) **Voice VLAN Priority**—set voice VLAN priority, range from 0 to 7
   d) **Data VLAN Identifier**—set data VLAN identifier, range from 1 to 4096
   e) **Data VLAN priority**—set data VLAN priority, range from 0 to 7
      TIPS: The RTP/RTCP packet is voice packet, the others is data packet.
      Notes: VLAN can only works in bridge mode. If you enable VLAN in router mode, it will not work with VLAN.
3. **Port Link**

- a) LAN port link—LAN port duplex mode
- b) PC port link—PC port duplex mode
4. **Qos**
   - a) **SIP Qos**—Quality of service for SIP (Diffserv)
   - b) **Voice Qos**—Quality of service for RTP (Diffserv)
5. **Syslog**   use syslog to output the log of the the device
   - a) **Enable syslog:** yes/no, enable/disable syslog service
   - b) **Log level:** none, alert, critical, Error, Warning, Notice, Info, Debug
   - c) **Syslog server**—syslog server address, e.g. 172.16.0.55
   - d) **Port**—syslog server port to receive message, default to 514

Click submit to save changes, and the phone will reboot automatically.

SIP settings

To view and edit SIP and RTP parameters:
- Log in to the web interface.
- Choose **SIP settings**

SIP Parameters

Sip Parameters

Sip User Agent Name :   AT800 2.1.2.10786   Sip Reg User Agent Name : AT800 2.1.2.10786

1. **SIP User Agent Name**---This name will appear in the header field User-agent in all SIP method which is not REGISTER
2. **SIP Reg User Agent Name**---This name will appear in the header field User-agent in the SIP method REGISTER

SIP Timer Values

Sip Timer Values

Sip T1 :   0.5   Sip T2 :   4
Sip T4 :   5
Reg Retry Intvl : 8   Sub Retry Intvl : 10

1. **SIP T1**---RFC 3261 T1 value (RTT estimate). Range from 0 to 64 seconds. Defaults to 0.5 seconds
2. **SIP T2**---RFC 3261 T2 value, which is the maximum retransmit interval for non-INVITE requests and INVITE responses. Ranges from 0 to 64 seconds. Defaults to 4 seconds

3. **SIP T4**---RFC 3261 T4 value, which is the maximum duration a message remains in the network. Ranges from 0 to 64 seconds. Defaults to 5 seconds
4. **Reg Retry Intvl**---interval to wait before the phone retries registration after failing during the previous registration. Defaults to 8 seconds.
5. **Sub Retry Intvl**---The retry interval when the last SUBSCRIBE request fails. Defaults to 10 seconds.

RTP Parameters

RTP Parameters

RTP Port Min :	16384	RTP Port Max :	16482
RTP Packet Size(ms) :	10		

1. **RTP Port**---Minimum port number for RTP transmission and reception. <RTP Port Min> and <RTP Port Max> should define a range that contains at least 2 even number ports . Defaults to 16384
2. **RTP Port Max**---Maximum port number for RTP transmission and reception. <RTP port Min> and <RTP Port Max> should define a range that contains at least 2 even number ports. Defaults to 16482.
3. **RTP Packet Size (ms)**---Packet size in milliseconds, which can be 10, 20, 30, 40, 60

Warning: Lin16-16 codec only supports RTP packet size = 10 ms because of the voice chip limitation. G.723 only allows 30ms/60ms RTP packet size.

SDP Payload Types

SDP Payload Types

AVT Dynamic Payload :	101	G729b Dynamic Payload :	18
G726r16 Dynamic Payload :	108	G726r24 Dynamic Payload :	97
G726r32 Dynamic Payload :	2	G726r40 Dynamic Payload :	96
iLBC Dynamic Payload :	98	iLBC Mode :	13.3 kBit/s
Lin16-16 Dynamic Payload :	102		
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
G729ab Codec Name :	G729		
G726-16 Codec Name :	G72616	G726-24 Codec Name :	G72624
G726-32 Codec Name :	G72632	G726-40 Codec Name :	G72640
iLBC Codec Name :	iLBC	Lin16-16 Codec Name :	L16

1. **AVT Dynamic Payload**---AVT dynamic payload type. Defaults to 101
2. **G729b Dynamic Payload**---G729b dynamic payload type.   Defaults to 18
3. **G726r 16 Dynamic Payload**---G726r 16 dynamic payload type. Defaults to 108
4. **G726r 24 Dynamic Payload**---G726r 24 dynamic payload type. Defaults to 97
5. **G726r 32 Dynamic Payload**---G726r 32 dynamic payload type. Defaults to 2
6. **G726r 40 Dynamic Payload**---G726r 40 dynamic payload type. Defaults to 96
7. **iLBC Dynamic Payload**---iLBC dynamic payload type. Defaults to 98
8. **iLBC Mode**---iLBC codec code rate, which can be 13.3kbit/s, 15.2kbit/s
9. **Lin16-16 Dynamic Payload**---Lin16-16 dynamic payload type, defaults to 102
10. **G711a Codec Name**---G711a codec name used in SDP, defaults to PCMA
11. **G711u Codec Name**---G711u codec name used in SDP, defaults to PCMU
12. **AVT Codec Name**---AVT codec name used in SDP, defaults to telephone-event
13. **G722 Codec Name**---G722 codec name used in SDP, defaults to G722
14. **G723 Codec Name**---G723 codec name used in SDP, defaults to G723
15. **G723 Mode**---G723 codec code rate, which can be 6.3kbit/s, 5.3kbit/s
16. **G729ab Codec Name**---G729ab codec name used in SDP, defaults to G729
17. **G726-16 Codec Name**---G726-16 codec name used in SDP, defaults to G72616
18. **G726-24 Codec Name**---G726-24 codec name used in SDP, defaults to G72624
19. **G726-32 Codec Name**---G726-32 codec name used in SDP, defaults to G72632
20. **G726-40 Codec Name**---G726-40 codec name used in SDP, defaults to G72640
21. **iLBC Codec Name**---iLBC codec name used in SDP, defaults to iLBC
22. **Lin16-16 Codec Name**---Lin16-16 codec name used in SDP, defaults to L16

NAT Support Parameters

1. **Enable Stun: yes/no**---enable or disable STUN
2. **Stun Server**---the stun server, which can be URI or IP address.

After finishing the configuration in this page, click submit to save changes, and the phone will reboot automatically.

Account Settings

To configure the account parameters:

- Log in to the web administrator interface
- Click admin button
- Click account menu

SIP

1. **Enable: yes/no**---enable or disable the account
2. **User ID**---extension number provided from ISP
3. **Password**---Password from provided from ISP
4. **SIP Server**---SIP registration and proxy server provided from ISP
5. **SIP port**---SIP server port provided from ISP, default is 5060
6. **Use Outbound Proxy**---Enable an outbound proxy. If set to no, the outbound proxy field is

ignored. Defaults to no.

7. **Outbound proxy Server**---SIP outbound proxy server where all outbound requests are sent to.

8. **Outbound proxy port**---the port of the outbound proxy server

   Register expires--- SIP Register interval time. Default is 300s.

9. **Subscribe expires**---The interval time of the sip subscription. Default is 3600s.

10. **Transport type**---select from UDP, TCP, or TLS. Default is UDP.

11. **SIP 100rel require**---To enable the support of 100REL SIP extension for reliable transmission of provisional response and use of PRACK requests, select yes, Otherwise, select no. Default is set to no.

Codec Configuration

Codec Configuration

| Prefer Codec : | G711u | User Prefer Codec Only : | no |
| DTMF Tx Method : | RFC2833 | | |

1. **Prefer codec**---preferred codec for all calls. (The actual codec used in a call still depends on the outcome of the codec negotiation protocol.). Which can be one of: G711u, G711a, G729, G722, G723, G726-16, G726-24, G726-32, G726-40, Lin16-16, iLBC. Defaults to G711u.

2. **User Prefer codec Only**---Choose yes to use only the prefer codec for all calls. (The call fails if the far end does not support this codec.) Select no, the phone will contains all the codec G.711A/u, G.722, G.723, G.726-16, G.726-24, G.726-32, G.726-40, G.729, iLBC, Lin-16 in SDP. Defaults to no.

3. **DTMF Tx Method**---select the method to transmit DTMF signals to the far end: Inband, rfc2833, INFO. Defaults to RFC2833.

Call feature setting

Call Feature Setting

Message Waiting :	yes	Voice Mail Number :	
UDP Keep Alive Enable :	no	UDP Keep Alive Intvl :	15
Default Ring :	1		
SRTP :	Off		

1. **Message waiting---**Choose yes to enable Message waiting indicator.

2. **Voicemail Number---**Set voicemail number.

3. **UDP Keep alive enable---**Choose yes to send UDP message to sip server port, so as to ensure the server port is still available.

4. **UDP keep alive intvl---**Set expire time of the keep alive UDP.

5. **Default Ring---**Set default ring tone for incoming calls.

6. **SRTP---**Enable SRTP(Secure Real-time Transport Protocol), and it should work with sip server.

Dial Plan

Dial Plan

Dial Plan :   (xxxxxxxxxxxx.)

✔ Submit   ↩ Reset

Dial plans specifies how to interpret digit sequences dialed by the user, and how to convert those sequences into an outbound dial string. The syntax for dial plan closely resembles the corresponding syntax specified by MGCP and MEGACO.

The dial plan functionality is regulated by the following configurable parameters:
- Interdigit_Long_Timer---refer to Interdigit_long_timer part for details
- Interdigit_Short_Timer---refer to interdigit long timer part for details
- Dial_plan

Dial plan contains a series of digit sequences, separated by "|" character. The collection of sequences are enclosed in parentheses, "(" and ")".

Defaults to (xxxxxxxxxxxx.)

When a user dials a series of digits, each sequence in the dial plan is tested as a possible match.   The matching sequences form a set of candidate digit sequences.   As more digits are entered by the user, the set of candidates diminishes until only one or none are valid.

Any one of a set of terminating events triggers the AT00 to both accept the user-dialed sequence, and transmit it to initiate a call, or else reject it as invalid. The terminating events are:
- No candidate sequences remain: the number is rejected.
- Only one candidate sequence remains, and it has been matched completely: the number is accepted and transmitted after any transformations indicated by the dial plan, unless the sequence is barred by the dial plan (barring is discussed later), in which case the number is rejected.
- A timeout occurs: the digit sequence is accepted and transmitted as dialed if incomplete, or transformed as per the dial plan if complete.
- An explicit 'send' (user presses the '#' key): The digit sequence is accepted and transmitted as dialed if incomplete, or transformed as per the dial plan if complete.

The timeout duration depends on the matching state. White space is ignored, and may be used for readability.

Digit Sequence Syntax:
Each digit sequence within the dial plan consists of a series of elements, which are individually matched to the keys pressed by the user. Elements can be one of the following:

Individual keys '0', '1', '2' . . . '9', '*', '#'.
The letter 'x' matches any one numeric digit ('0'   ... '9')

A subset of keys within brackets (allows ranges): '['set ']' (e.g. [389] means '3' or '8' or '9')

Numeric ranges are allowed within the brackets: digit '-'digit (e.g. [2-9] means '2' or '3' or … or '9')

Ranges can be combined with other keys: e.g. [235-8*] means '2' or '3' or '5' or '6' or '7' or '8' or '*'.

Element repetition:

Any element can be repeated zero or more times by appending a period ('.' character) to the element. Hence, "01." matches "0", "01", "011", "0111" … etc.

Subsequence Substitution:

A subsequence of keys (possibly empty) can be automatically replaced with a different subsequence using an angle bracket notation: '<' dialed-subsequence ':' transmitted-subsequence '>'. So, for example, "<8:1650>xxxxxxx" would match "85551212" and transmit "16505551212".

Intersequence Tones:

An "outside line" dial tone can be generated within a sequence by appending a ',' character between digits. Thus, the sequence "9, 1xxxxxxxxxx" sounds an "outside line" dial tone after the user presses '9', until the '1' is pressed.

Number Barring:

A sequence can be barred (rejected) by placing a '!' character at the end of the sequence.   Thus, "1900xxxxxxx!"   automatically rejects all 900 area code numbers from being dialed.

Interdigit Timer Master Override:

The long and short interdigit timers can be changed in the dial plan (affecting a specific line) by preceding the entire plan with the following syntax:

Long interdigit timer: 'L' ':' delay-value ','

Short interdigit timer: 'S' ':' delay-value ','

Thus, "L=8, (. . .)" would set the interdigit long timeout to 8 seconds for the line associated with this dial plan.   And, "L: 8, S: 4, (. . .)" would override both the long and the short timeout values.

After finishing, click submit to save changes. The phone will restart automatically.

Phone setting

Preference

Audio Device

Audio Device

Enable Speaker : yes   Headset Volume : 5

Headset Input Gain : 12   Enable Ring Tone : yes

- **Enable Speaker**---Choose yes to allow the ring tone being played in speaker when there is an incoming call. Choose no to play ring tone is headset.
- **Headset Volume**----Set headset receiver volume in a conversation. Default is level 5.
- **Headset Input Gain**---Set headset mic gain volume in a conversation. Default is 12.
- **Enable Ring Tone**---Choose yes to play default ring tone in ringing state. Choose no to disable ring tone. Incoming call will not generate a ring tone then.

LCD

LCD

LCD Contrast : 5   Backlight Time(Seconds) : 10

Backlight Level : high

- **LCD contrast**---Adjust LCD contrast ranging from 1 to 8. Default is level 5.
- **Backlight Time (seconds)**---Set Backlight timeout ranging from 3s to 100s. Default is 10s.
- **Backlight Level---**Set backlight brightness level off, low, middle, high.

Control Timer Values (seconds)

Control Timer Values(Seconds)

Interdigit Long Timer : 10   Interdigit Short Timer : 3

Recorder Delay : 0   Recorder Time : 12

- **Interdigit Long Timer**---If no candidate sequences are as yet complete (as dialed), the Interdigit_Long_Timeout applies. Default is 10s. Range: 1 – 99sec
- **Interdigit Short Timer**---If a candidate sequence is complete, but there exists one or more incomplete candidates, then the Interdigit_Short_Timeout applies. Default is 3s.Range: 1 – 99 sec
- **Recorder Delay**---Delay after far end hangs up before recorder tone is played. 0 = plays immediately, inf = never plays. Range: 0 – 99 sec. Default is 5s.
- **Recorder Time**---The time of displaying call ended. If setting recorder time to 0, it will return to idle state immediately without displaying call ended and playing recorder tone when call is hung up by the other side.Range: 0 – 99 sec. Default is 12s.

Date and Time

- **NTP**
    1. NTP server---Network Time Protocol. Set NTP Server, and both domain and IP address are allowed.
    2. Time zone---Choose proper time zone of the local time.
- **Manual**
    1. Set Local Date (YYYY/MM/DD)---Set a manual date by inputting in YYYY/MM/DD formate. E.G. 2012/10/1. Or click the icon   to choose the date on calendar.
    2. Set Local Time(hh:mm:ss)---Set a manual time by inputting in hh:mm:ss formate. E.G. 12:00:00. Or click up/down button to increase/decrease the value.

    Note: In manual mode, the date and time will be reset after a restart.
- **Daylight Saving Time**
    1. Daylight Saving Time---Enable or disable the daylight saving time.
    2. Daylight Saving Time Rule--- Set daylight saving time rule in format of:

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

    <start-time> and <end-time> are of the form: MM/DD/WW/hh:mm:ss. <start-time> and <end-time> specify the start and end time-date of daylight saving time.

    <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.

    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 one hour early than the standard time.

After finishing, click submit to make the changes effective.

Feature

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. E.G. Set speed dial 2: 123456789, then dialing 2 will call 123456789. Similarly, user can dial 3/4/5.../9 to call numbers in speed dial 3/speed dial 4/ speed dial 5.../speed dial 9.

Call Forward

Call Forward

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

- **Always target**---Set Always forward target. It will forwards all incoming calls to the target number.
- **Busy target**---Set Busy forward target. It will forwards incoming calls to the target number when the phone is busy.
- **No Answer**---Set No answer forward target. It will forwards incoming calls to the target number when there is no one to pick up the call after the no answer timeout.
- **After Ring Time (seconds)**---Set no answer timeout. Default is 5s.

Others

Others

Send Key :	#	Enable Do Not Disturb :	no
Enable Block Call ID :	yes	Enable Block Anc Call:	no
Enable Auto Answer :	yes	Remember Account :	no

- **Send key**---Choose * or # to be the send key to dial out the entered number immediately.
- **Enable Do Not Disturb**---Enable or disable Do not disturb. When enabled, all incoming calls will be rejected. If always call forward or busy call forward is enabled, it will forwards the call to the target number.

- **Enable Block Call ID**---When choosing yes, the phone will call others by an anonymous caller id.
- **Enable Block Anc Call**---When choosing yes, the anonymous incoming call will be rejected
- **Enable Auto Answer**---When choosing yes, the phone will answer the call automatically right away when there is an incoming call.
- **Remember Account**---When choosing yes, the phone will auto login after a restart every time. When choosing no, user should login manually via keypad after a restart. For AT800/AT800P, if choosing remember account to NO, user should login manually on the phone by pressing Login key + *number*password#. E.G. Sip number is 1234, password is 5678, user should press Login key, and then press *1234*5678# to login. To logout on the phone, just press Login key + *password#. E.G. *5678# .

After finishing, click submit button to save changes.

Voice

Echo Cancellation

VAD :   no   CNG :   yes

Jitter Buffer

Type :   Fixed

Min Delay :   0   Max Delay : 300

Normal Delay :   120

Echo Cancellation

- **VAD**---Voice Activity Detection: when choosing yes, the phone will not send RTP packets at the silent moment(for instance, in mute mode) to save bandwidth.
- **CNG-**--Comfort Noise Generator: when choosing yes, the phone will send comfort noise to the other side at the silent moment, so the other side will know the conversation is still valid.

Jitter Buffer

- **Type**---Fixed: the phone will use a fixed jitter buffer size.
  Adaptive: jitter buffer will be the Min delay+ RTP frame size.
- **Min Dela**y---The minimum delay of the jitter buffer.
- **Max Dela**y---The maximal delay of the jitter buffer.
- **Normal Dela**y---It is used as fixed delay which should be between min delay and max delay.

After finishing, click submit button to save changes.

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]

- **Upload Ring Tone**---Choose the desired ring tone file on local PC, and click upload button to upload it.
- **Ring Tone Typ**e---Choose the User Define 1/User Define 2 to upload to.

Please pay attention to the note on webpage: Ring tone must be wav file, 8K, 8Bit ulaw, file size <200KBytes. Otherwise, it may not be uploaded or played on the phone.

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 ToneScript for each tone.

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

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.

- **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

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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 Three way conference is in progress.

After finishing, click submit button to save changes.

Programmable key

- Key
    1. Hold---Set a hold key
    2. Xfer---Set a transfer key bringing attended transfer
    3. Conf---Set a conference key
    4. DND---shortcut key to enable/disable Do Not Disturb
    5. CFWD---shortcut key to enable/disable Call forward.
    6. Speed Dial---speed dial key to dial out the number set in the frame
    7. Redial---redial key to dial out the last number.
    8. Voicemail---Set a voicemail key to dial out the voicemail aceess number which is configured in account settings webpage.

After finishing, click submit button to save changes.

Update and provision

Manual update

Bootloader Version: 2.2
Firmware Version: 2.1.2.10655
Hardware Version: 1.3

Boot Loader: [   ] 浏览...   💾 Upgrade

Firmware: [   ] 浏览...   💾 Upgrade

Configuration: [   ] 浏览...   💾 Upgrade   📥 Download

Phone Book: [   ] 浏览...   💾 Upgrade   📥 Download

- **Bootloader Version**---Show current uboot version.
- **Firmware Version**---Show current firmware version.
- **Hardware Version**---Show current hardware version.
- **Boot Loade**r---Upgrade the uboot file locally. It is a upgrade of the bottom driver of the phone and hardly be used. If user wants to upgrade uboot, please contact ATCOM support.
- **Firmware**---Upgrade the firmware locally.
- **Configuratio**n---Update/Download the configuration file(config.xml) locally.
- **Phone Book**---Update/Download the phonebook configuration file(phonebook.xml) locally.

Auto provision

Configuration Profile

Configuration Profile

Provision Enable :   [no ▾]   Resync On Reset :   [no ▾]
Profile Rule :   [   ]

✔ Submit   🔄 Reset

- **Provision Enabl**e---Enable/disable the auto provision.
- **Resync On Reset**---Choosing yes to upgrade the configuration after each reboot.
- **Profile Rule**---It can support TFTP/FTP/HTTP auto provision. FTP only supports anonymous account login. 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

If there is a sub folder in the root file, e.g. The sub-folder is named "abc", TFTP server address is 192.168.1.100, set the rule like: tftp://192.168.1.100/abc/config.xml

After finishing, please click submit button to effect it. The phone will restart automatically and download the configuration file.

Reset and 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 system now**---Click "Reboot" button to reboot the phone.
- **Reset to Factory setting**---Click "Reset" button to make the phone to factory settings. Please note, all the configurations will be restored in this setting.

Phone book

[New Contact]

	Mode	Name	Number	
1	Directory			Delete
2	Blacklist			Delete

[Submit] [Reset]

- **Phonebook**---View, add, edit, delete phonebook entries.
  Add: Click "New Contact" button to add new entry. Choose Directory mode, and filled in the blank of Name and Number filed.
  Edit: Change the content in Name and Number filed.
  Delete: Click "Delete" button to delete the entry.
After finishing, click submit button to take them effect.

- **Black List-**---View, add, edit, delete black list entires. The numbers in blacklist will not be able to call the phone.
  Add: Click "New Contact" button to add new blacklist entry. Choose Blacklist mode, and filled in the

blank of Name and Number filed.

Edit: Change the content in Name and Number filed.

Delete: Click "Delete" button to delete the entry.

After finishing, click submit button to take them effect.

Call log

- **Redial**---View dialed calls
- **Receive**d---View answered calls
- **Missed-**---View missed calls

FAQ

How to restore to factory settings

There are three ways to perform a factory reset:

a. Reset from the man-machine interface, follow these steps:
   1) Press **menu** softkey
   2) Scroll to **Factory Reset** and press **select**
   3) Press **OK**
b. Reset through the web interface, refer to Reset and Reboot part for more details
c. Reset to factory defaults when power on the device, follow these steps:
   1) Press and hold the * and # key
   2) Plug in the DC adapter
   3) After about 5 seconds, the device will boot up with the default factory settings.

Upgrading in the safe mode

Upgrade firmware in the safe mode whenever the phone is not able to boot up normally.

a) Setting up a TFTP server, and set the IP address of PC to 192.168.1.200
b) Copy the firmware to the TFTP server home directory, and rename it to AT800x.FW
c) Make sure the phone and the TFTP server connect to the same switch
d) Long press # key and plug in the adapter until it displays "Upgrading..." on LCD.
e) The phone will start downloading the AT800.FW firmware from TFTP server
f) After downloading finished, phone will upgrade automatically

g)   Wait till the phone finish upgrading, and it will reboot automatically

How to make direct IP calls

Press headset key to hear dial tone, and then press the third softkey "number" twice to switch to "IP". Using * key to inputting ".". If user wants to dial 192.168.1.100, then press 192*168*1*100. After inputting the IP address, press dial key to dial it out.

How to connect headset

Headset is standard RJ9 jack in following PIN sequence(from left to right):
Mic-
Spk-
Spk+
Mic+
Please keep in mind that the wrong PIN sequence of the headset will bring a voice problem.
1)   Opposite sequence of mic+ and mic-
        Very low volume in headset
2)   Opposite sequence of spk+ and spk-
        No any problem
3)   Opposite sequence of mic+/mic- and spk+/spk-
        Can not use any more

Trouble shooting

Phone can't register to the SIP server.

1)   Check the IP address. If set LAN port in DHCP mode, please make sure that the DHCP server on.
2)   Check the gateway
3)   Check DNS server
4)   Make sure the account information is the same as from ISP
5)   Check the SIP server is on
6)   Check the SIP register port, defaults to 5060

Device can't obtain IP address

1)   Make sure the Ethernet cable is plugged in the LAN port

2) Make sure the cable and connected switch port is OK
3) Make sure the DHCP server is on, and there are available IP addresses in the lease table
4) Try to set LAN port to static IP mode

Conversation with one way speech

1) Make a direct IP call to make sure there is no problem with the headset receiver and microphone.
2) Enable STUN in SIP settings webpage
3) Set STUN server, as stun.xten.com
4) Click submit to restart the phone.
5) Make a call again.