A21 IP Phone User Manual

Version: 1.0

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Content

Contact ATCOM

Overview 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 oveA210 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 have been available in 100+ countries and used by millions of end users.

Contact Sales

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Tel	+ (86) 755-83018618-8806
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Download Center： http://www.atcom.cn/download.html

1. Overview of A21

A21

Type	POE	Power adapter	Wi-Fi
A21	Yes	Optional accessory	Yes

1.1 Interfaces
  ➢ Power input: DC 5V, 1000mA or POE
  ➢ LAN: RJ45 port
  ➢ PC: RJ45 port
  ➢ Headset jack 1 : RJ9 port
  ➢ Handset jack 1 : RJ9 port
1.2 Hardware
  ➢ LCD: 132×58
  ➢ CPU: 400MHz Dual Core
  ➢ LED indicator: 1 Status Light, 2 line indicators
1.3 Software
  ➢ Sip 2.0 (RFC3261) and other related SIP RFCs

- ➢ 2 SIP lines
- ➢ STUN
- ➢ Jitter Buffer, VAD,CNG
- ➢ G.711A/u, G722, G.723, G.726-16, G.726-24, G.726-32, G.726-40, G.729 , Lin16-16, iLBC
- ➢ Echo Cancellation
- ➢ SIP Domain name, Authentication
- ➢ DTMF（inband, RFC2833, info）
- ➢ Call transfer, Call forward, 3-way conference, Call hold, Call back
- ➢ DND(Do Not Disturb), Auto answer, Blacklists, Block Call-ID, Block Anonymous call, Dial plan, IP call
- ➢ Phone book with 1000 white records and 100 black records, 200 answered calls, 200 missed calls, 200 dialed calls
- ➢ Update via HTTP, FTP, TFTP, PNP
- ➢ Syslog
- ➢ SNTP
- ➢ WEB access with different login level
- ➢ Multi-language: English, Chinese, Farsi, French, German, Hebrew, Italian, Portuguese, Russian, Spanish, Turkish
- ➢ Soft button: soft button * 3
- ➢ Redundancy SIP server
- ➢ BLF

1.4 Network
- ➢ LAN/PC：Support bridge mode
- ➢ Support VLAN
- ➢ Support L2TP VPN
- ➢ LAN support Primary and Secondary DNS
- ➢ LAN support DHCP Client
- ➢ Support QoS

1.5 Management and Maintenance
- ➢ Support safe mode and firmware updating under safe mode
- ➢ Support different level user management
- ➢ Configuration via web , keyboard
- ➢ Support multi-language
- ➢ Auto provision (Firmware and configuration file)
- ➢ Support system log and call log

1.6 Protocol
- ➢ IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- ➢ 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
- ➢ DNS: Domain Name Server
- ➢ TFTP: Trivial File Transfer Protocol
- ➢ HTTP:Hypertext Transfer Protocol
- ➢ FTP:File Transfer Protocol

1.7 Compliant and Authenticated Standard
- ➢ CE: AGC01180140201E2，AGC01180140202E2

➢ Comply with ROHS in EU
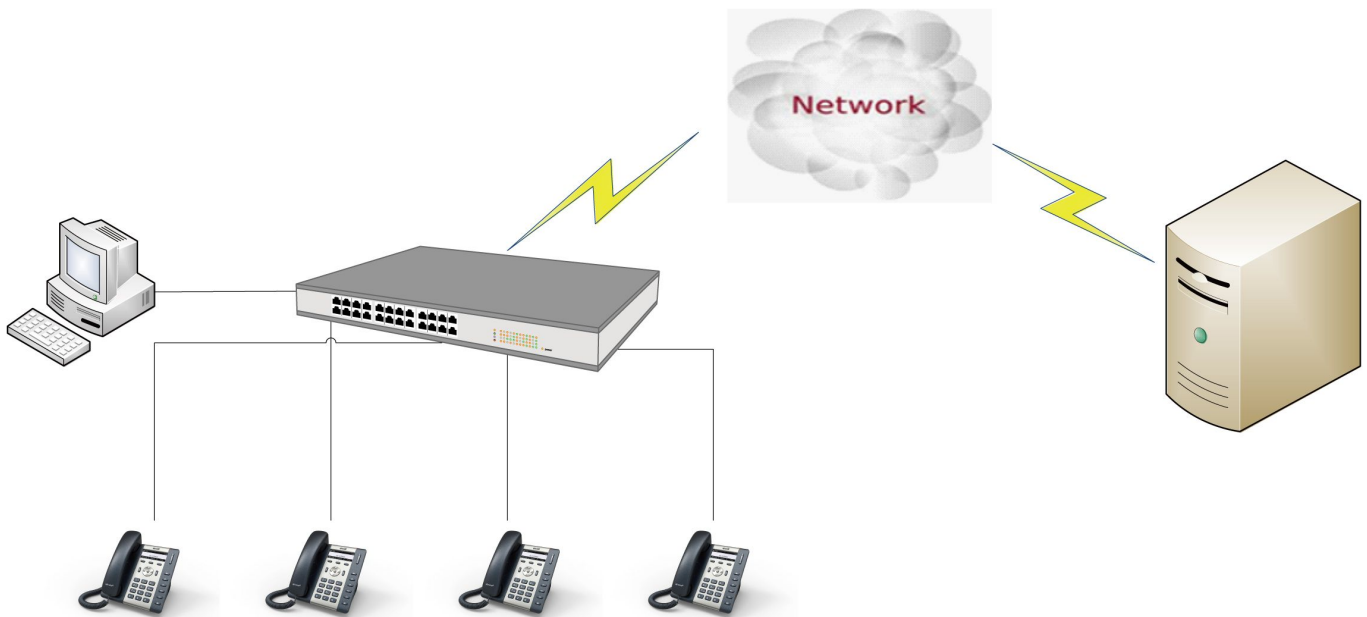➢ Comply with ROHS in China

➢

$$CE \quad FC \quad \cancel{\boxtimes} \quad RoHS$$

➢ Operation temperature: lower than 60° C
➢ Storage temperature: lower than 60° C
➢ Humidity: 10 to 90% no dew

1.8 Packing List

Model	Phone	Handset	Handset line	Stand bracket	Ethernet Cable	User Manual CD
A21	1	1	1	1	1pcs 1.5m	1

Note: Power adapter (Input: AC 100~240V, 50/60Hz; Output: 5V, 1000mA;)

1.9 Installation

Connect LAN port to PC with Ethernet cable, or connect A21 to a switch/router which is in the same network as your PC. It uses DHCP mode by default, and you can review its current IP address by pressing ✓ key on idle state. To access the web interface, you can input the IP address in IE browser. E.G. The IP address of your A21 is 192.168.1.100, you can input 192.168.1.100 and press enter key on your browser to access its webpage. There are two login level:
User
Admin

No password is set for those two accounts with factory settings. You can click admin button on the right corner of the webpage to switch from the user to admin mode. To set the password for user and admin login you can firstly login as admin and enter the Network--->Advance page as following.

Installation instruction
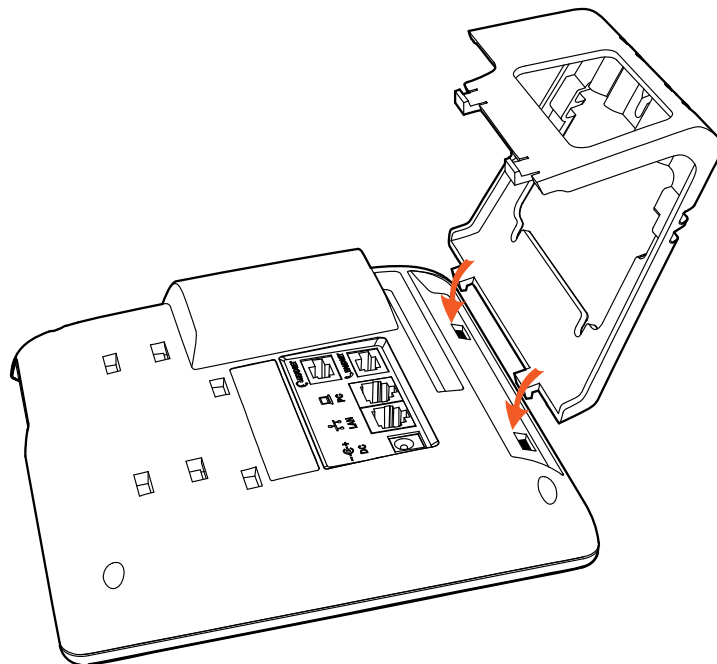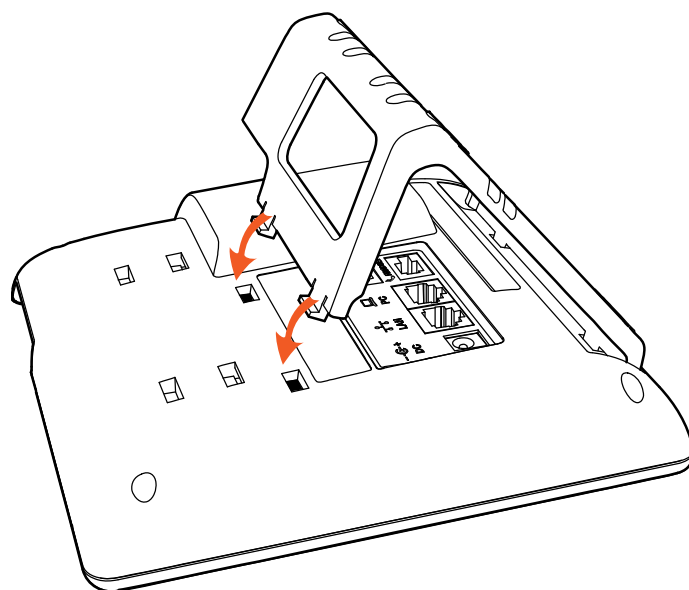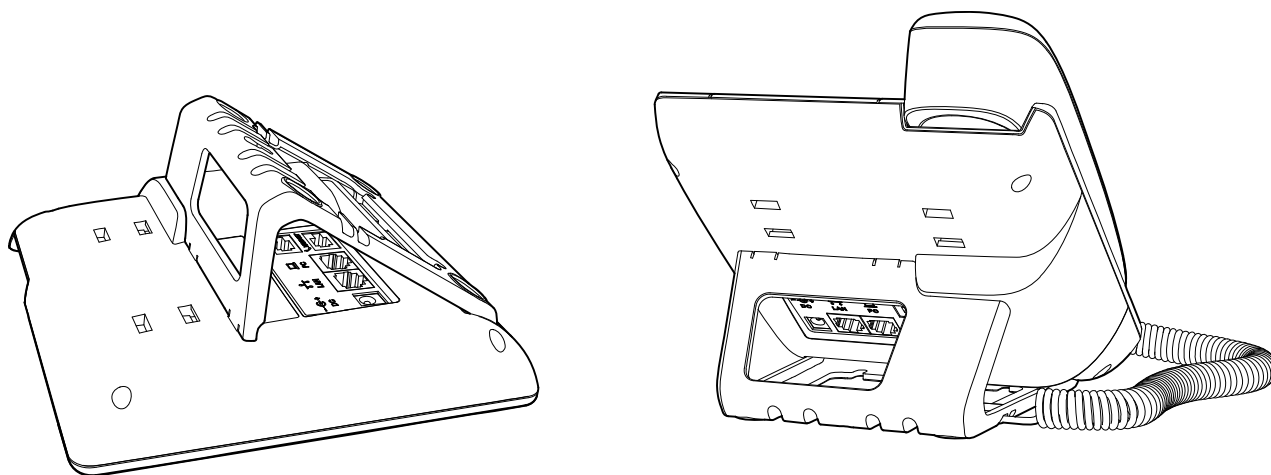
1. Desktop installation

A. Put the bottom side of the IP phone upside and press one-side joints of stand bracket into the slot, please refer the picture as below:

B. Press the other side joints into the slot according to the direction of the arrow
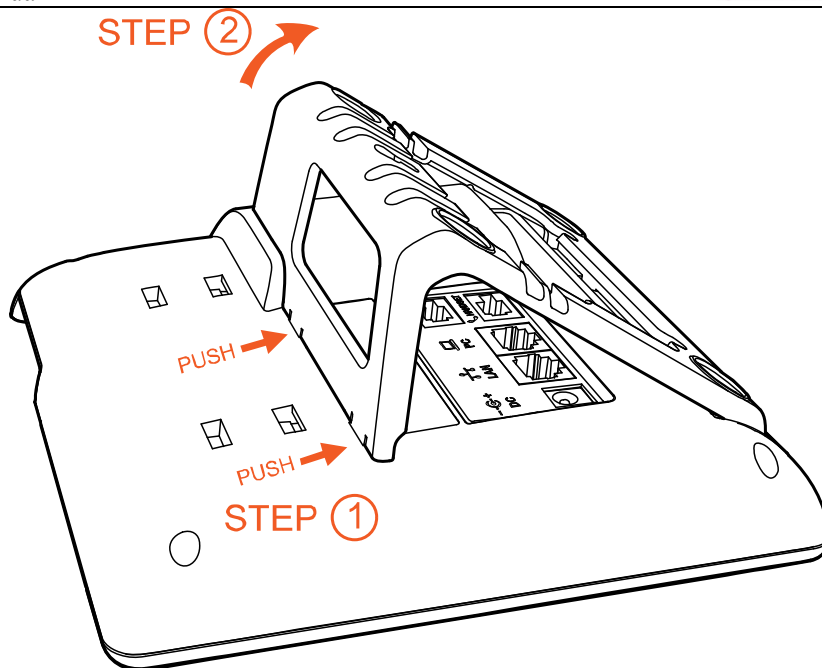
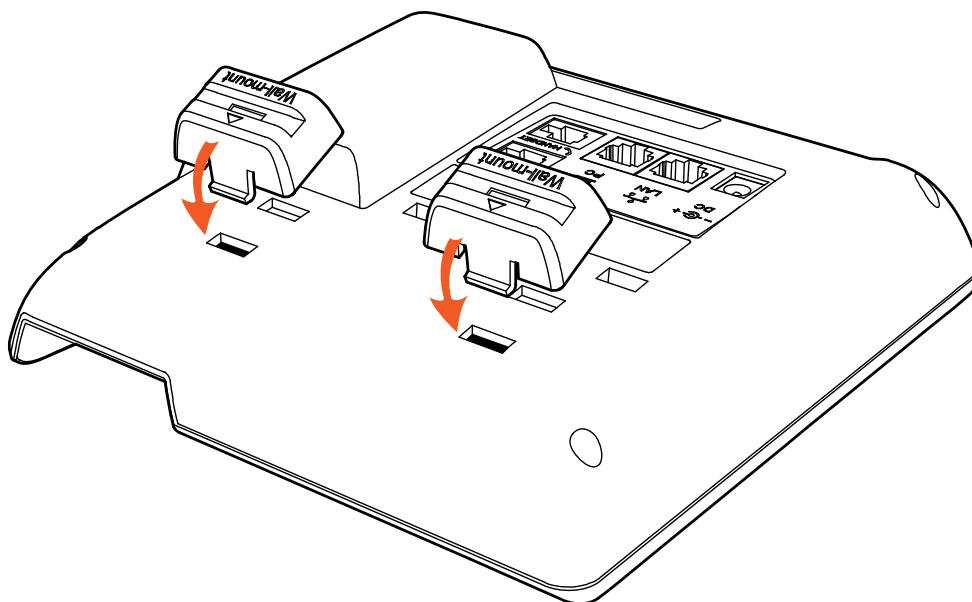C. It is the right picture after fixing the stand bracket below:

D. Disassemble the stand bracket:
Push the spring joint of stand and pull the stand according to the direction of the arrow. When the joints are pulled out of the slot, you can take off the stand bracket.
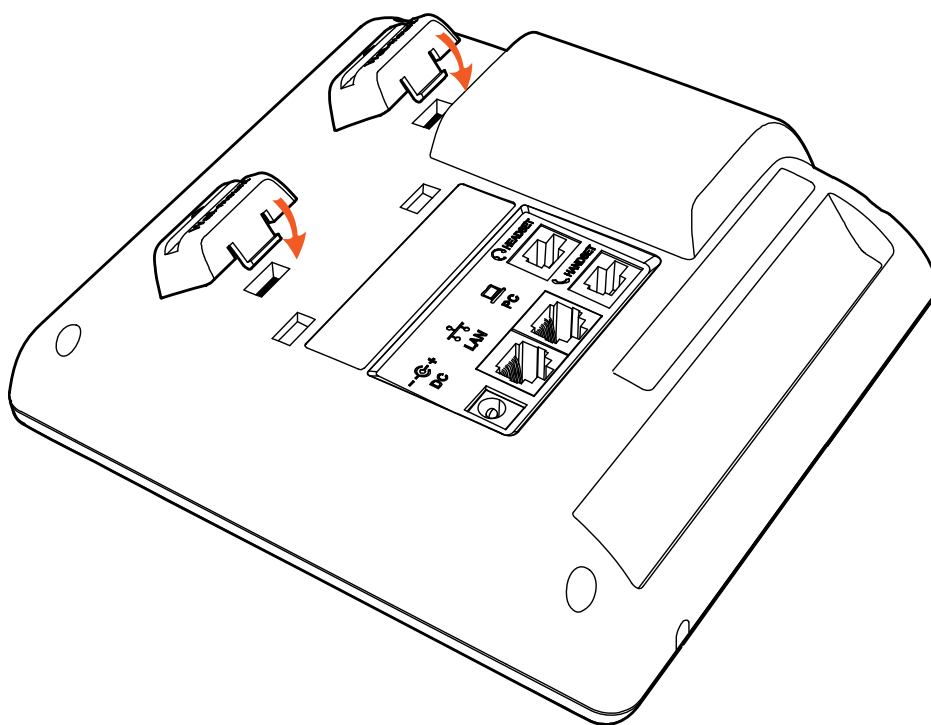
STEP ②

PUSH

PUSH

STEP ①

2. Wall-hung Installation

A. Put the bottom side of the IP phone upside and press one-side joints of wall-hung stand bracket into the slot, please refer the picture as below:

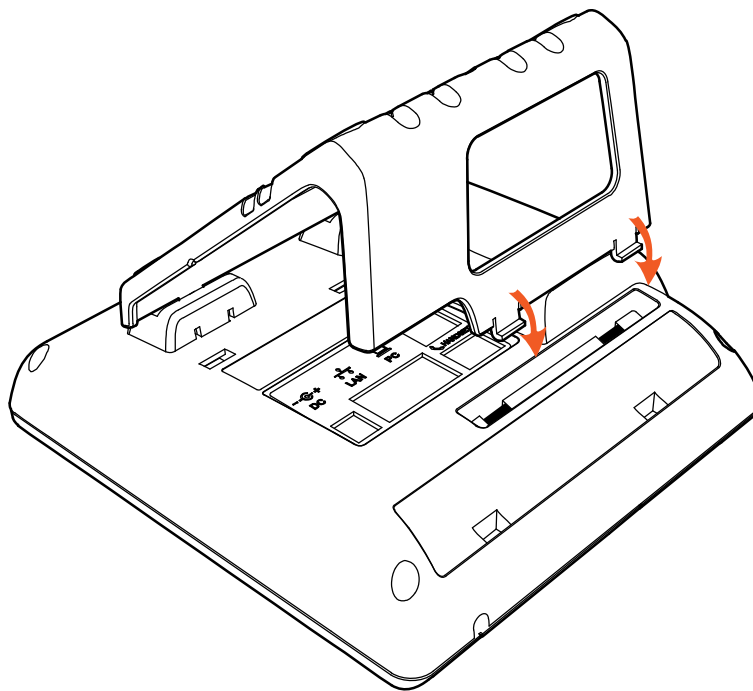B. Press the other side joints into the slot according to the direction of the arrow

C. After install the wall-hung stand bracket, press one-side joints of stand bracket into the slot, please refer the picture as below:
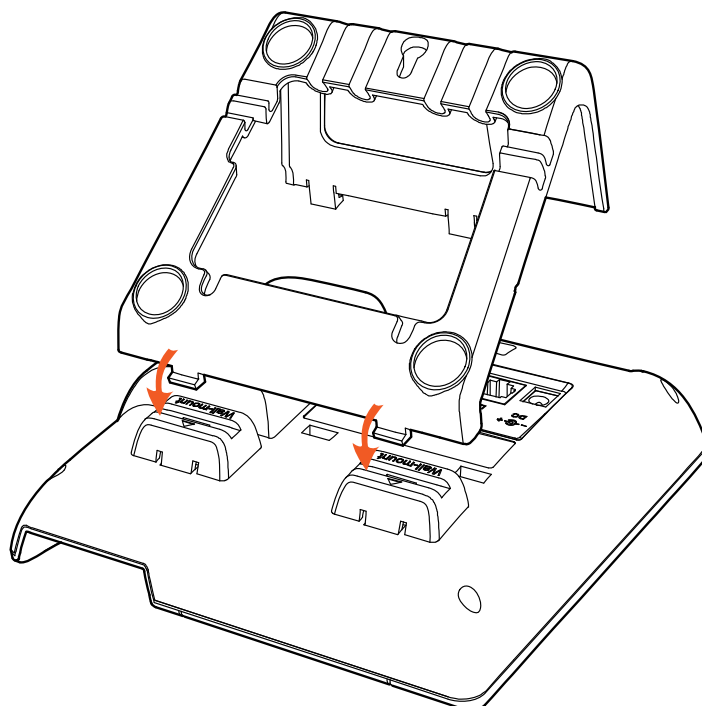
D. Press the other side joints into the slot according to the direction of the arrow

E. Knock in nails or screws on the wall according to the proportion of the distance between the hanging holes as below:
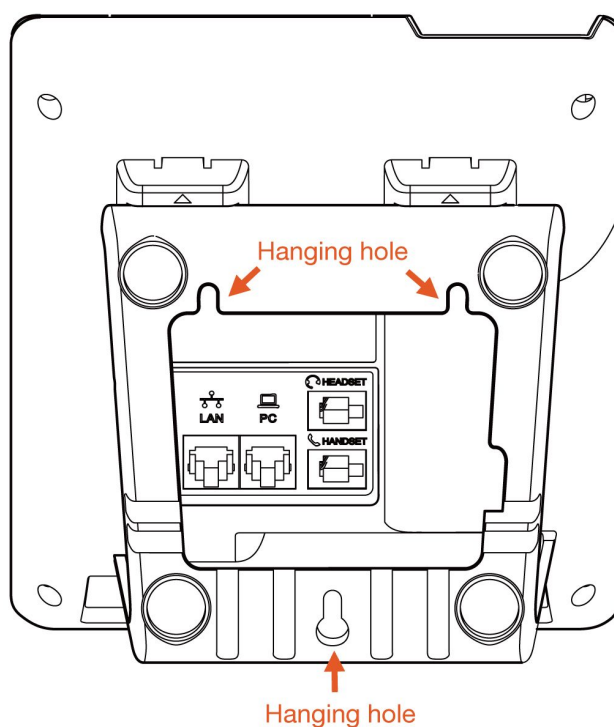
F. It is the right picture after fixing the stand bracket below:
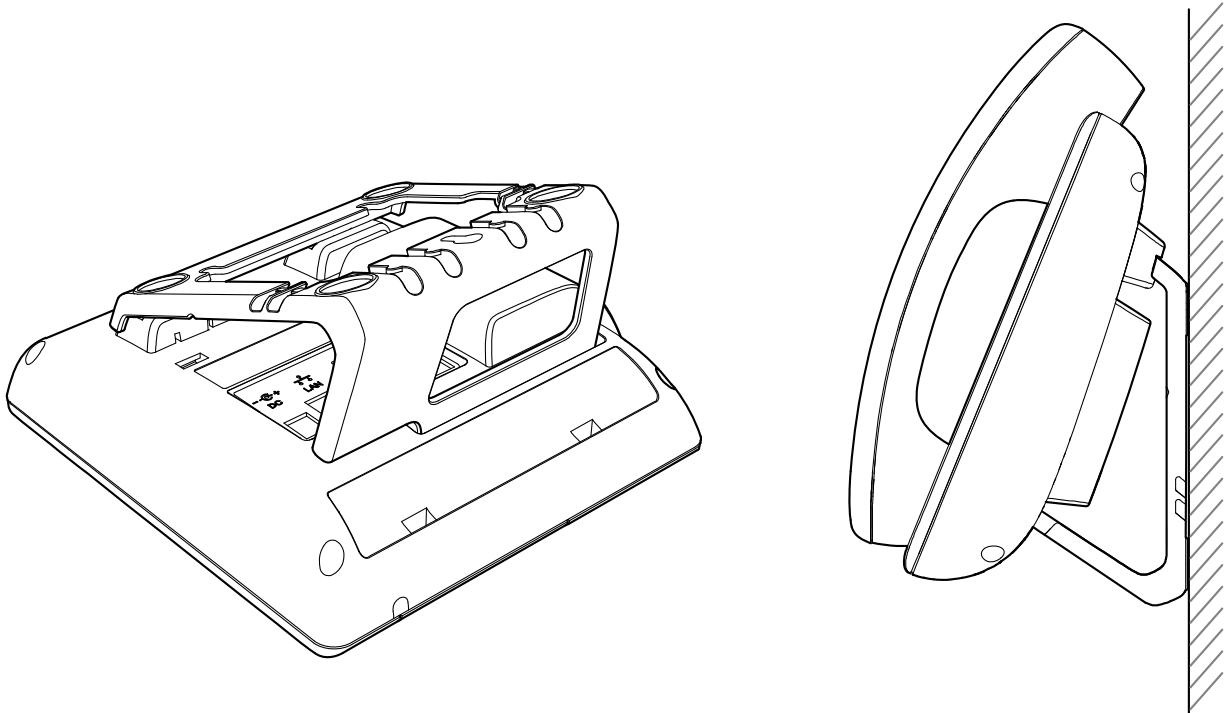
G. Disassemble the stand bracket:
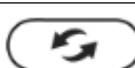
Push the spring joint of stand and pull the stand according to the direction of the arrow. According to the direction of the arrow:

2. Keypad of IP Phone

LCD   Status Indicator LED

Line key

Soft Key

Volume

Navigation

Menu

OK

Cancel

Voice Mail

Mute

Headset

Dial pad

Redial

Speaker

2.1 Describe of the buttons and Screen
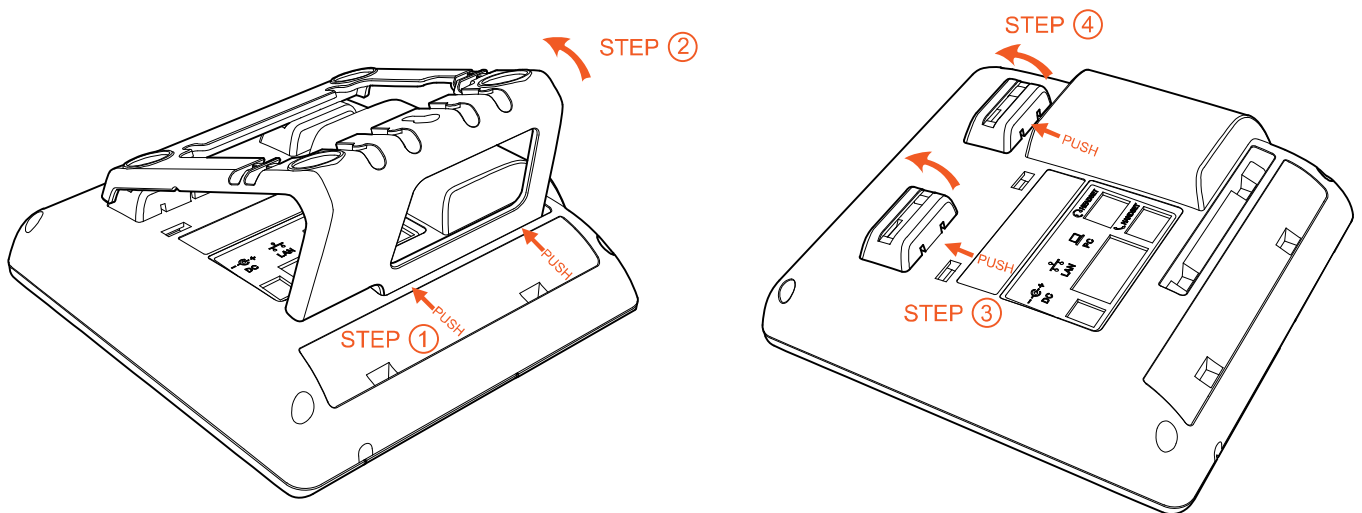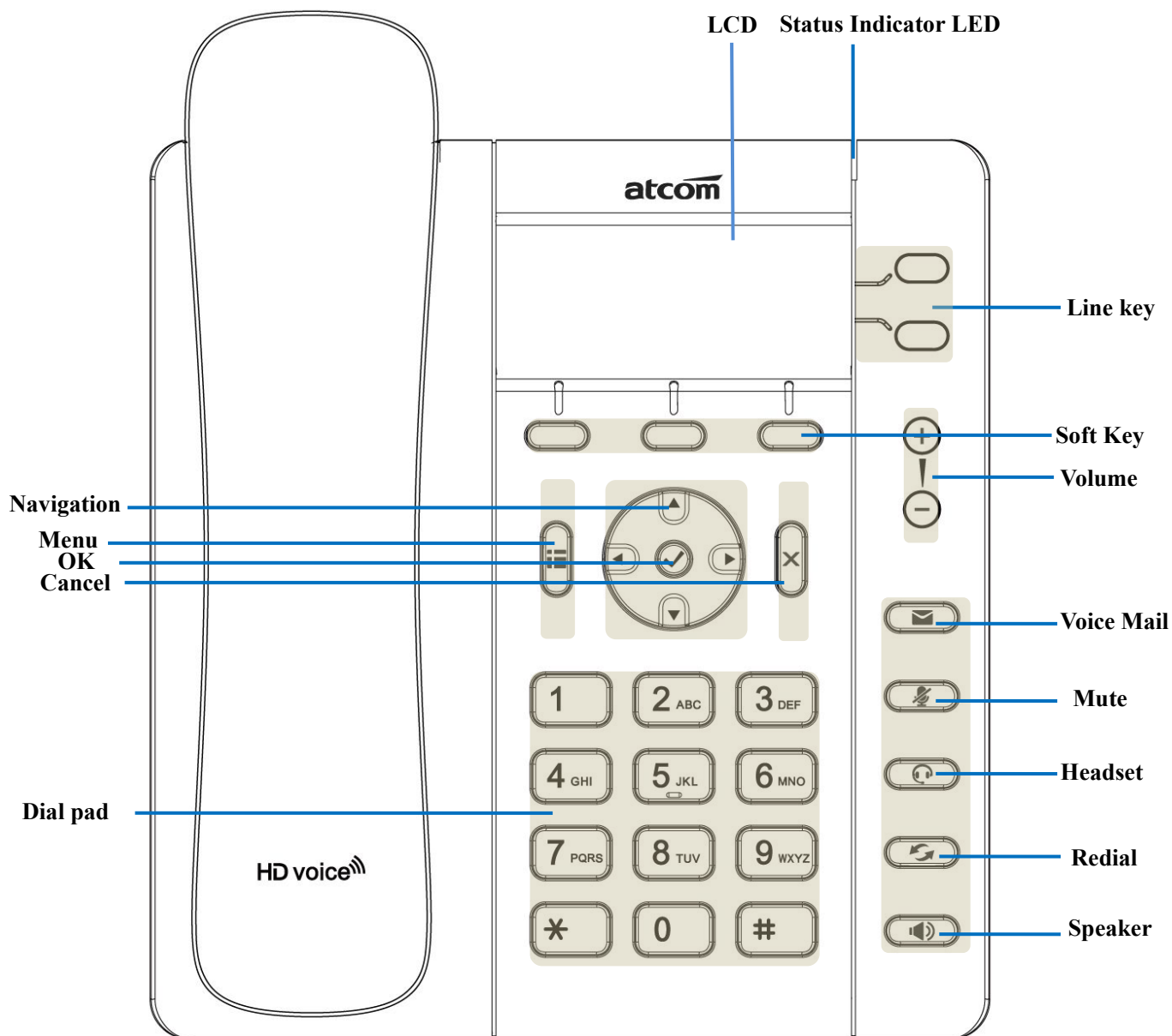
Soft Keys	Press to select a function which displayed at the soft key zone of screen Display all optional functions at the last line of LCD screen.
Status Indicator LED	Indicates the phone's status, ➢ If the phone is being started ,the LED is on ➢ If the phone is standby, the LED is off ➢ If there is income calling, the LED will blink at 120ms on, 120ms off. ➢ If there is a new message, the LED will blink at 600ms on, 600ms off.
LCD Screen	All information such as date, time, phone number, incoming caller's ID(if available),line/call status, extension numbers and the soft key features are displayed on it.
Line Key	Shows extension number and status.
Dial pad	Entering numbers or characters.
Navigation keys	Allows users to navigate (left, right, up, down). Press ▼ can enter "Directory". Press ▲ can enter "Hitory".
+ / −	Adjust the volume (speaker/handset/headset/ring). Volume level will be displayed on LCD when pressing volume key.
✓	Confirm users' operation and show status when A21 is idle.
✕	Cancel users' operation.
☰	Enter menu settings.
🔊	Pick up and hung up under the speaker mode.
🔇	Mute the mic in a conversation by pressing the Mute button; this prevents the person on the active call from hearing what you or someone else in the room is saying. To un-mute, press the Mute button again.
🎧	Pick up and hung up under headset mode.
✉	Check the Voicemail status.
↻	Redial the last dialed number

2.2 Line keys LED

LED Status	Description
Off	The line is idle or disabled for register
Blink quickly in green	The line has incoming call
Green	The line is talking
Blink quickly in red	The line is held

2.3 Status Indicator

LED Status	Description
On	Rainbow 2LW is being started
Off	Rainbow 2LW is standby
Blink in red at 120ms on, 120ms off	There is incoming call

2.4 Programmable keys LED

LED Status	Description
Off	The monitored account is unregistered
Green	The monitored account is idle
Red	The monitored account is talking
Fast blink in red	The monitored account is ringing

When powering on, all LEDs will be lighted once.

When there is an incoming call, the corresponding line LED and status LED will be blinking.

2.3 Describe of the LCD Screen Icons

Icon	Description
☎	The extension is registered
📞	The extension is unregistered
✉	There is a new voice mail
🔔	There is an incoming call
⏸	The call is held
🔊	A21 is in speaker mode
📞	A21 is in handset mode
🎧	A21 is in headset mode
🔇	The call is muted

	Connect to Wi-Fi successfully, ×means failed，and ? means connecting

3. Basic functions and operations

3.1 Answer the calls

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

A. Answer by handset

Pick up the handset and talk with the caller. If you want to hang up, just put back the handset. When you are talking with the handset and want to switch to speaker mode or headset mode, please press ⊲» key or ⌒ key, then put down the handset.

B. Answer by speaker

Press ⊲» key and talk with callers by built-in Micro-phone and Speaker. If you want to hang up, please press ⊲» key again. Switch calling or talking into handset mode by lifting the handset under speaker mode. Press ⌒ key will switch calling or talking into headset mode.

C. Answer by headset

Keep your microphone connected with the RJ9 headset jack, when there is an incoming call, press ⌒ key and talk with the caller. If you want to hang up, please press ⌒ key again. Pressing ⊲» key can change calling or talking into speaker mode, and lifting the handset switches to handset mode.

3.2 Make Call

A.   Use the handset

Pickup the handset and input a phone number. Press soft key "Send" to dial the number. When you hear the tones of "du~~du~~" and the phone number your dialed is being displayed on the LCD, the phone at the side of being called should be ringing. If the called party answers this calling, the call is established and the calling timer is started immediately.

B.   Use the speaker

Press ⊲» key and input a phone number. Press soft key "Send" to dial the number. When caller hear the tones of "du~~du~~" and the phone number your dialed is being displayed on the LCD, the phone at the side of being called should be ringing. If the called party answers this calling, the call is established and the calling timer is started immediately.

C.   Use the headset

Press ⌒ Key and input a phone number. Press soft key "Send" to dial the number. When caller hear the tones of "du~~du~~" and the phone number your dialed is being displayed on the

LCD, the phone at the side of being called should be ringing. If the called party answers this calling, the call is established and the calling timer is started immediately.

D.　Dial from phone book

1.　Press ⊞ key and input the keypad password 123 to enter the menu and choose "Directory" option. Press "Select" soft key and then find the contact person by navigation keys. When the certain contact person is highlighted, press "Dial" or just pick up the handset to call this number.
2.　Pick up the handset, press "Directory" soft key, then select the contact person and press "Dial" soft key.

3.　Pick up the handset, press ▼ and enter "Directory", then select the contact person and press "Dial" soft key.

E.　Dial from call history

1.　Press ⊞ key and input the keypad password 123 to enter the menu and choose "History" option, then enter sub-directory "Dialed Calls", "Received Calls" or " Missed Calls" to select one of call history entry, and press "Dial" soft key or pick up handset to call this number.
2.　Pickup the handset, press "History" soft key, then select one of call history entry, and press "Dial" soft key to call this number.

3.　Pickup the handset, press ▲ and enter "Call History" ,then select one of call history entry, and press "Dial" soft key to call this number.

3.3 Pre- dial

It's a method to dial a phone number immediacy at standby mode.
A、Dial-up the phone number at standby mode
B、Press soft key "Dial" to send out the number

Multiple line dial-up

A21 supports 2 SIP lines. That means user can register on2 different sip accounts simultaneity. User is able to choose line1 or line2 by pressing corresponding line key. Line1 is the priority line.

A21 maximum supports 4 concurrent calls. When there is an incoming call to another line, LCD will show the incoming telephone number and line LED will be blinking in green. User is able to press the corresponding line key on which you're talking then press "Answer" to receive the incoming call from another line.

3.4 Hang up the phone

1.　Handset hang up
Put back the handset at handset mode, the current calling will be hung up.
2.　Speaker hang up

Press 🔊 key at speaker model, the current calling will be hung up.

3. Headset Hang up

Press ⌢◉⌣ key at headset model, the current calling will be hung up.

4. Hang up one line call

Press the hook to hang up the current calling when 2 calls happened simultaneously.

3.5 Call Transfer

1. Attended call transfer

The attended transfer allows user to call a third-party before transferring the calling.

While on calling, press the "Transfer" soft key to hold the current call and phone the third party. Then dial the target number you want to transfer to on the activated line and press "Send" soft key to call that number. After the target party answers the call, press "Transfer" soft key again to complete the transfer.

2. Blind call transfer

The blind transfer allows user to transfer a call without speaking to the third party. On the user side, the call will be ended as soon as the target phone number is dialed.

Operating steps: Press "More" soft key to get more option, then press "Bxfer" soft key, input the transfer target number and press "Send" soft key.

3.6 Voicemail

A21 has a ⌢✉⌣ key for entering voicemail box. Press ⌢✉⌣ key to enter the menu to configure voicemail number if never configure it previously. Otherwise, the voicemail number will be called after press it. If you want to modify it after configured it, please go to the Account webpage to modify the voicemail number.

3.7 Mute calls

The input audio will be not transmitted to peer phone after pressing ⌢🎤̸⌣ key, and the phone will be muted even switched among different modes of speaker, handset and headset. To un-mute, just press ⌢🎤̸⌣ key again.

3.8 3-Way Conference Call

To initiate a conference call:

Press "More" soft key on calling to get more options, then press "Conf" soft key to start a conference call. Enter the phone number of the third party and press "Send" soft key to send it out.

1. After the third party answers the call, pressing "Conf" key again to establish the 3-way conference.
2. 3-way conference initiator can press "Exit" soft key to quit from the conference and leave the other two parties still in the conversation.
3. If the initiator hangs up the call or press the "End Conf" soft key, the conference will be ended and the calling between the other two parties will be hung up.

3.9 Call History

A21 supports 200 missed calls list, 200 incoming calls list and 200 dialed calls list. When the storage is full, the old record will be erased by the new one.

Press "History" soft key or ⬆ key when A21 is standby, all the incoming(->), outgoing(<-) and missed calls(!) will be listed. There is other ways to check them:

A. Missed call

1. Press ☰ key.

2. Press ▲ key and ▼ key to select "Call History" then press "Select" soft key.

3. Press ▲ key and ▼ key to select "Missed Calls" then press "Select" soft key.

4. Press ▲ key and ▼ key to browse the missed call record. If there is no record, the LCD display will be indicated "List is Empty".

B. Answered call

1. Press ☰ key .

2. Press ▲ key and ▼ key to choose "Call History" and then press "Select" soft key.

3. Press ▲ key and ▼ key to choose "Answered Calls" and then press "Select" soft key.

4. Press ▲ key and ▼ key to browse the answered call records. If there is no record, the LCD display will be indicated "List is Empty".

C. Dialed call

1) Press ☰ key.

2) Press ▲ key and ▼ key to select "Call History" and then press "Select" soft key.

3) Press ▲ key and ▼ key to select "Dialed Calls" and then press "OK" soft key.

4) Press ▲ key and ▼ key to browse the dialed call records. If there is no record, the LCD display will be indicated "List is Empty".

3.10 Default Password

Password is needed to access menu and web.

The default password for accessing menu is 123.

There are 2 modes to access web: User mode and Admin mode. All the parameters can be visible at Admin mode while part of them can be visible at User mode.

➢ User Mode
  Username：user   no default Password
➢ Admin Mode：
  Username：admin   no default Password

3.11 Check the Phone's IP address

Press key, then the status of the phone will be displayed on the screen and you will see the current IP address of the phone.

3.12 Dial Plan

Dial plan defines the rule of dial. The syntax of dial plan for A21 is closely similar as the corresponding syntax specified by MGCP and MEGACO.

Dial plan is stipulated by the below configurable parameters:

- Interdigit Long Timer——refer to Interdigit long timer for more detail description

- Interdigit Short Timer——refer to Interdigit short timer for more detail description

- Dial Plan

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

Default: (*xx.|xxxxxxxxxxxx.)

When user dials a series of digits, A21 will response in below way:

• No candidate sequences matched, the number will be rejected and "call ended" will be displayed on the screen. For instance, if the default dial plan only supports digits, any '*' character or letters input will be rejected.

• More than one candidate sequences matched, A21 will wait for more digits input.

• When input timeout occurs, A21 will dial the digits input already.

• When input '#' character, A21 will dial the input digits immediately.

Digit Sequence Syntax:

'x': Matches any one numeric digit ('0' ... '9')

'[]': Numeric ranges are allowed within '[]'. For example, [389] means '3' or '8' or '9', [3-6] means '3' or '4' or '5' or '6' , [235-8*] means '2' or '3' or '5' or '6' or '7' or '8' or '*'.

'.' ：Any element can be repeated zero or more times by appending a '.' character. For example, 01. matches 01,011,0111,01111,...,011111111...etc.

'<>': subsequence substitution. For example, '<8:1650>xxxxxxx' would match '85551212' and the first digit '8' will be replaced by '1650' and '16505551212' will be dialed.

',': 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.

'!': A sequence can be barred (rejected) by placing a '!' character at the end of the sequence. For example, "137xxxxxxxx!" will forbid numbers which have 11 digits and start by 137 to be dialed.

Example:

(xxxxxxx|[*#]xxxx|9,1xxxxxxxxxx|00xxx!) contains 4 subsequences:

1. Allow to dial numbers with 7 digits
2. Allow to dial numbers with 4 digits and start by '*' or '#'
3. Allow to play an "outside line" dial tone after pressing '9' and dial numbers with 11 digits and start by 1
4. Forbid to dial numbers with 5 digits and start by 00

4. Web settings

Input the IP address in the web browser and press 'Enter' key to access A21's user webpage. Click "admin" which is on the right top corner to enter administrator webpage.
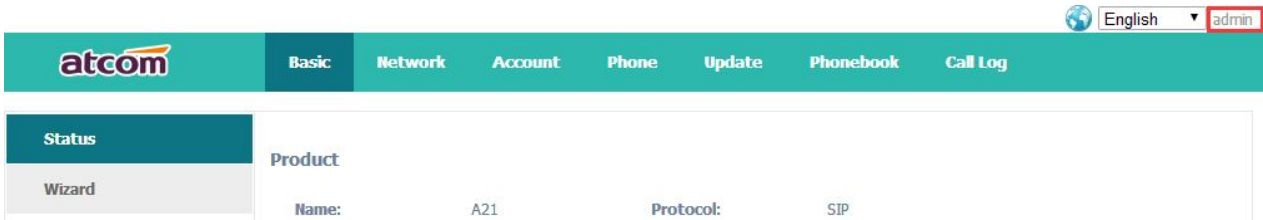
4.1 System status
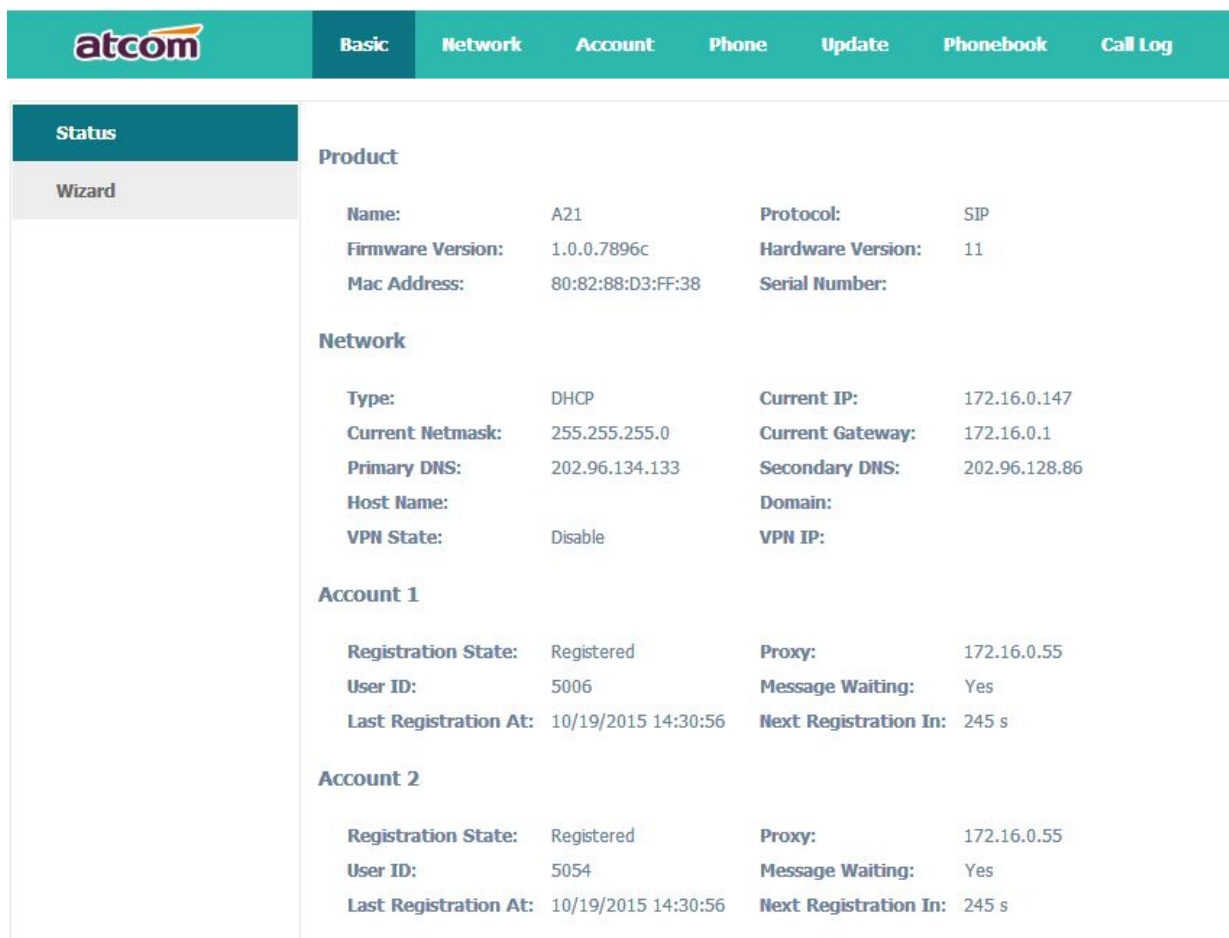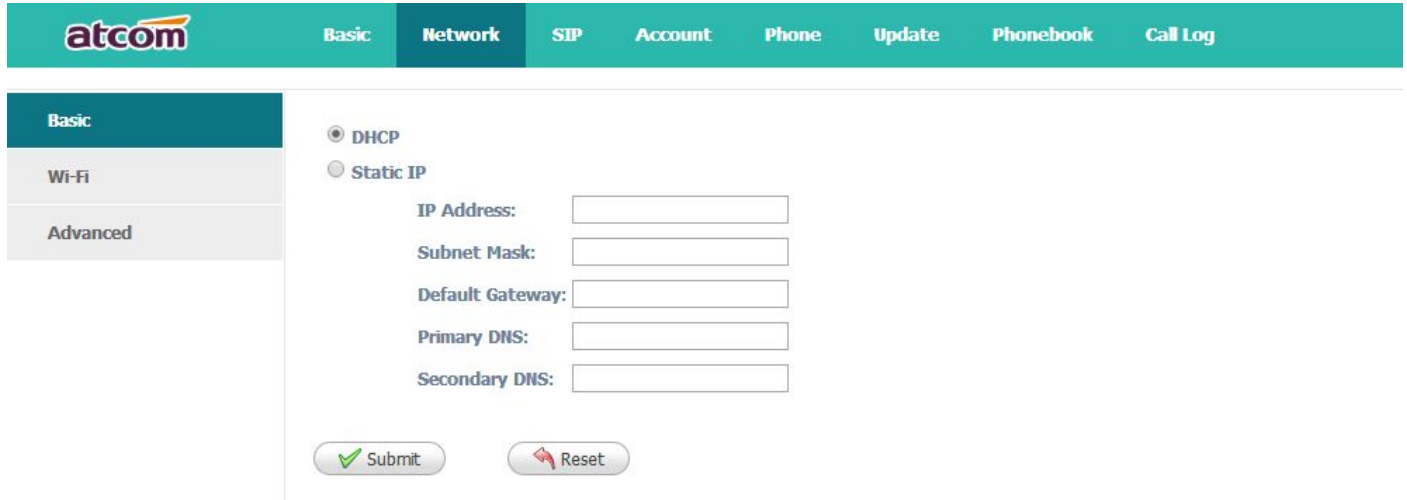
This page shows the IP phone's system status.

1. **Product Information** shows the product name, protocol, software and hardware version, Mac address and serial number.

2. **Network Information** shows the network connection type, IP address, netmask, gateway, DNS, host name, domain and VPN information.

3. **Account Information** account 1/2 shows registration state, proxy, user ID, message waiting on or off , the last and next registration time.

4.2 Network

4.2.1 Basic

There are 3 ways to connect to the internet: DHCP, Static and PPPoE, please choose one according to your own situation.

Network mode: DHCP ,Static IP

1. DHCP

   Obtain automatically dynamic IP from DHCP server.

2. Static IP

   a) IP Address: set IP address

   b) Subnet Mask：set netmask

   c) Default Gateway：set gateway

   d) Primary DNS：set primary DNS server

   e) Secondary DNS：set secondary DNS server

Press 'Submit' button after finishing setting and all the settings info will be saved and taken effect after A21 reboots.
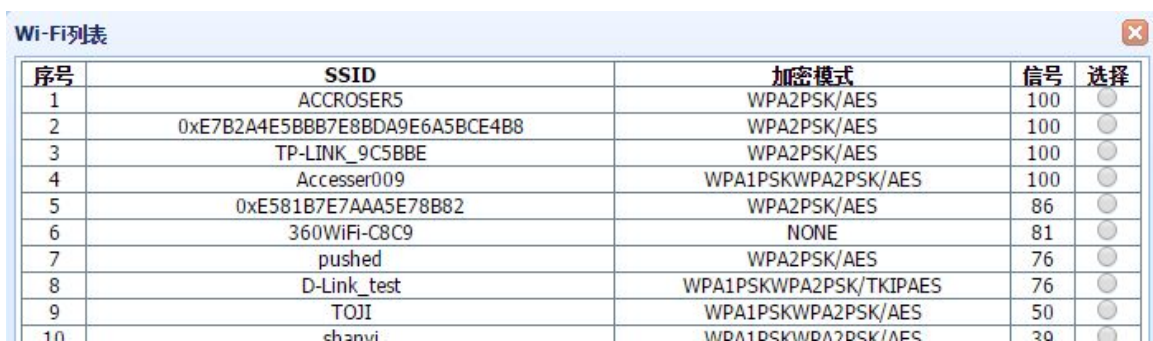
4.2.2 Wi-Fi

1、Configure the wifi on the WEB

A21 connect to network via Wi-Fi acquiescently, click "Search Wi-Fi" button to search the available Wi-Fi.

After all available Wi-Fi hotspots are list out, click the round behind the available Wi-Fi which you want to connect.

Input the right "Secret Key" and then click the "Submit" button, the phone will reboot, and you should disconnect the wire that connect to A21 during rebooting, or the phone will connect to network via wire preferentially. The Wi-Fi icon will display on the home screen, it means connect failed when the icon come with ×, you should to check the configuration.

2、Configure the wifi on the phone

1、Press "Menu"-"Network"-"Connect Mode",change for the wifi connection mode,press "OK".

2、Press "Menu"-"Network"-"WIFI setting"-"WLAN",choose the wifi you want to connect,Press "Connect",then input the correct wifi password.

3、Return to the standby interface,and phone successful connected to the wifi.

4.2.3 Advance

➢ **Web Server**

✓ Enable Web Server: Enable or disable web access. If choose "no", you're not able to access A21's webpage.

✓ Admin password: Set password for admin webpage access. Input 'http://ip-address/index.asp' in the web browser to access admin's login webpage after setting the admin password, then input username(admin) and password to access the admin's webpage.

✓ User password: Set password for user webpage access. Input 'http://ip-address/user.asp' in the web browser to access user's login webpage after setting the user password, then input username(user) and password to access the user's webpage.

✓ HTTP port: set port for HTTP access (defaults to 80)

   For example, A21's IP is 192.168.1.223

   HTTP port was set as 100, you have to type "http://192.168.1.223:100" in web browser to enter A21's webpage.

➢ **VPN**

1. VPN Enable: Select "Yes" or "No" to enable or disable VPN
2. VPN Type: Only support L2TP VPN
3. Server address: VPN server address
4. Username: VPN account's username
5. Password: VPN account's password

After apply, the phone will be reboot. The VPN IP address will be shown on the System Status webpage.

➢ **VLAN**

1. VLAN Enable: Select "Yes" or "No" to enable or disable VLAN
2. LAN VLAN Identifier(1..4094) : Assign VLAN ID for voice stream, range from 1 to 4094
3. LAN VLAN Priority: range from 0 to 7, 7 is the highest priority.
4. PC VLAN Identifier(1..4094) : Assign VLAN ID for data stream, range from 1 to 4094
5. PC VLAN Priority: range from 0 to 7, 7 is the highest priority.

> ➢ **Port Link**

Choose the network type and port link of LAN/PC
1. LAN Port Link: Auto negotiate, full duplex 10Mbps, full duplex 100Mbps, half duplex 10Mbps, half duplex 100Mbps.
2. PC Port Link: Auto negotiate, full duplex 10Mbps, full duplex 100Mbps, half duplex 10Mbps, half duplex 100Mbps.

> ➢ **Qos**

QoS: Quality of service
1. SIP Qos: Quality of service for SIP (Diffserv)
2. Voice Qos: Quality of service for RTP (Diffserv)

> ➢ **Syslog**

1. Enable Syslog: Select "Yes" or "No" to enable or disable syslog.
2. Log level: None，Alert，Critical，Error，Warning，Notice，Info，Debug. The debug level is the most detailed.
3. Syslog Server: Syslog Server address.
4. Port: Syslog server port, defaults to 514.

4.3 SIP Setting

> ➢ **Sip Timer Values**

1. Sip T1: RFC 3261 T1 value (RTT). Range: 0 – 64 sec, defaults to 0.5
2. Sip T2: RFC 3261 T2 value (Maximum retransmit interval for non-INVITE requests and INVITE responses). Range: 0 – 64 sec, defaults to 4
3. Sip T4: RFC 3261 T4 value (Maximum duration a message will remain in the network). Range: 0 – 64 sec, defaults to 5

4. Reg Retry Intvl: Interval to wait before the phone retries registration again after encountering a failure condition during last registration. Range: 0 –65535, defaults to 8
5. Sub Retry Intvl: Interval to wait before the phone retries subscriber again after encountering a failure condition during last subscriber. Range: 0 –65535, defaults to 10

➢ **RTP Parameters**

RTP Parameters

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

1. RTP Port Min: Minimum port number for RTP transmission and reception. Range: 1–65535, defaults to 16384
2. RTP Port Max: Maximum port number for RTP transmission and reception. <RTP Port Max> should be at least 2 larger than <RTP port Min>.Range: 1–65535, defaults to 16482
3. RTP Packet Size(ms): Packet size in milliseconds, which can be 10ms, 20ms, 30ms, 40ms, 60ms

➢ **SDP Payload Types**

SDP Payload Types

G711a Codec Name:	PCMA	G711u Codec Name:	PCMU
G723 Codec Name:	G723	G722 Codec Name:	G722
G729ab Codec Name:	G729		
AVT Dynamic Payload:	101	AVT Codec Name:	telephone-event
iLBC Dynamic Payload:	98	iLBC Codec Name:	iLBC
G726r16 Dynamic Payload:	108	G726r16 Codec Name:	G726-16
G726r24 Dynamic Payload:	109	G726r24 Codec Name:	G726-24
G726r32 Dynamic Payload:	110	G726r32 Codec Name:	G726-32
G726r40 Dynamic Payload:	111	G726r40 Codec Name:	G726-40
Lin16 Dynamic Payload:	107	Lin16 Codec Name:	L16

1. G711a Codec Name---G711a codec name used in SDP, defaults to PCMA
2. G711u Codec Name---G711u codec name used in SDP, defaults to PCMU
3. G723 Codec Name---G723 codec name used in SDP, defaults to G723
4. G722 Codec Name---G722 codec name used in SDP, defaults to G722
5. G729ab Codec Name---G729ab codec name used in SDP, defaults to G729
6. AVT Codec Name---AVT codec name used in SDP, defaults to telephone-event
7. iLBC Dynamic Payload---iLBC dynamic payload type. Defaults to 98
8. iLBC Mode---iLBC codec rate, which can be 13.3kBit/s, 15.2kBit/s. Defaults to 13.3kBit/s
9. G726r16 Dynamic Payload ---G726 dynamic payload type. Defaults to 108

10. G726r16 Codec Name---G726 codec name used in SDP, defaults to G726-16

11. G726r24 Dynamic Payload ---G726 dynamic payload type. Defaults to 109

12. G726r24 Codec Name---G726 codec name used in SDP, defaults to G726-24

13. G726r32 Dynamic Payload ---G726 dynamic payload type. Defaults to 110

14. G726r32 Codec Name---G726 codec name used in SDP, defaults to G726r32

15. G726r40 Dynamic Payload ---G726 dynamic payload type. Defaults to 111

16. G726r40 Codec Name---G726 codec name used in SDP, defaults to G726r40

17. Lin16 Dynamic Payload ---Lin16 dynamic payload type, defaults to 102

18. Lin16 Codec Name--- Lin16 codec name used in SDP, defaults to Lin16

➢ **NAT Support Parameters**

NAT Support Parameters

Enable Stun :   No ▼   Stun Server :   [   ]

1. Enable Stun: Select "Yes" or "No" to enable or disable using stun to discover NAT mapping.
2. Stun Server: Set stun server, which can be IP address or domain name.

4.4 Account

A21 has 2 line which is enabled to register by default.

Account

Account: Account 1   ▼

➢ **SIP**

1. Display Name: This name will be displayed on the LCD. It will show the User ID instead if leave Display Name as blank.

2. User ID: Username of sip account.

3. Authenticate ID: Normally is the same as User ID, but blank is acceptable.

4. Password: Password of SIP account.

5. SIP Server: SIP server address, support both IP address and domain name.

6. SIP Port: SIP server port, defaults to 5060.

7. SIP Redundancy Server: SIP redundancy server address.

8. Use Outbound Proxy: Select "Yes" or "No" to enable or disable outbound proxy.

9. Outbound Proxy Server：Set address of Outbound proxy server. All signaling requests will be sent to outbound proxy server firstly.

10. Outbound Proxy Port: Outbound proxy server port.

11. Register Expires: Register expiration time, defaults to 300 seconds.

12. Subscribe Expires: Subscriber expiration time, defaults to 3600 seconds.

13. Transport Type: UDP/TCP/TLS. Defaults to UDP.

14. SIP 100Rel Require: Select "Yes" or "No" to enable or disable 100Rel. If enabled, 100rel parameters will be added to the SIP request to support PRACK.

15. Session Timer Enable：Select "Yes" or "No" to enable or disable Session Timer.

16. Early Update Enable：Select "Yes" or "No" to enable or disable Early Update.

17. Caller ID Display: Select "Yes" or "No" to enable or disable Caller ID display

18. AutoSubscribeMWIEnable：Select "Yes" or "No" to enable or disable SubscribeMWI

19. Server List：Choose the server type.

20. DNS Mode: Choose the DNS mode

21. BLF List URI: Set BLF list URI when A21 cooperates with Broadworks (Broadsoft).

➢ **Codec Configuration**

1. Disabled Codecs: The codecs in this frame are disabled, choose one and then click [>>] to enable the specified codec.

2. Enabled Codecs: The codecs in this frame are enabled, the uppermost codec has the maximum priority. Choose one and then click [<<] to disable the specified codec, click [∧] to promote priority and click [∨] to reduce priority.

➢ **Call Feature Setting**

1. Message Waiting: Select "Yes" or "No" to enable or disable indication of new voicemail existed.

2. Voice Mail Number: Set voicemail number.

3. Pickup Service Code: Set Pickup Service Code.

4. UDP Keep Alive Enable: Select "Yes" or "No" to enable or disable UDP keep alive. If enabled, A21 sends UDP packets periodically to keep the server port alive.

5. UDP Keep Alive Intvl: Set interval to send UDP packets.

6. Default Ring: Set default ring tone.

7. SRTP: Select "Yes" or "No" to enable or disable SRTP (Secure Real Time Control Protocol). This feature will be available only when the server supports SRTP.

➢ **Dial Plan**

Dial Plan

Dial Plan : (*xx.|xxxxxxxxxxxx.)

Dial Plan: Configure dial rule for SIP account, please refer to <u>dial plan</u>.

4.5 Phone Setting

4.5.1 Preference

Output Volume(1~8)

| Handset Volume: | 5 | SpeakerPhone Volume: | 1 |
| Headset Volume: | 5 | Ring Volume: | 6 |

Input Gain

| Handset Gain: | 0 | SpeakerPhone Mic Gain: | 0 |
| Headset Gain: | 0 | | |

LCD

| Backlight Level: | 5 | Backlight Time(Seconds): | 10 |
| Contrast: | 8 | Keypad Password: | ••• |

Control Timer Values(Seconds)

| Interdigit Long Timer(1~64): | 20 | Interdigit Short Timer(1~64): | 8 |
| Reorder Delay(0~60): | 3 | Reorder Time(0~60): | 5 |

Watch Dog

Watch Dog: Yes

➢ **Language**
Support customizable multi-language.

➢ **Output Volume(1~8)**
1. Handset Volume: Specify handset volume grade
2. SpeakerPhone Volume: Specify speaker volume grade
3. Headset Volume: Specify headset volume grade
4. Ring Volume: Specify ring tone volume grade

➢ **Input Gain**
1. Handset Gain: Specify handset gain, the bigger the gain, the louder the other party heard.
2. SpeakerPhone Mic Volume: Specify speaker gain, the bigger the gain, the louder the other party heard.
3. Headset Volume: Specify headset gain, the bigger the gain, the louder the other party heard.

➢ **LCD**
1. Backlight Level: select the backlight level
2. Backlight Time(Seconds): select the backlight time
3. Contrast: select the contrast level
4. Keypad Password: set keypad access password

➢ **Control Timer Values(Seconds)**
1. Interdigit Long Timer: If the numbers or characters input have not finished and do not full matched the dial plan, it will be not dialed out automatically until time out. Range: 0 – 64 sec
2. Interdigit Short Timer: If the numbers or characters input are full matched the dial plan, it will be dialed out automatically until time out. Range: 0 – 64 sec
3. Reorder Delay: It means the delay between the remote end hangs up and reorder tone is played. "0" means the reorder tone will be played immediately. Range: 0 – 60 sec
4. Reorder Time: Set the duration about how long the page of Calling–end displayed

➢ **Watch Dog**
Monitor the phone if appears any abnormal phenomena, the default button is enable.

➢ **Date And Time**

Date And Time

◉ NTP

NTP Server : pool.ntp.org

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

○ Manual

Set Local Date(YYYY/mm/dd) : [   ] 📅

Set Local Time(HH:mm:ss) : [   ] ⬍

Daylight Saving Time

Daylight Saving Time : No ▼

Daylight Saving Time Rule : [   ]

NTP: Network time protocol
1. NTP Server: Set NTP server address, IP address or domain name are both acceptable.
2. Time Zone: Choose your own time zone.
3. Daylight Saving Time: Select "Yes" or "No" to enable or disable Daylight saving time.
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=-1Start 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.
> **Manual**

1. Set Local Date(YYYY/mm/dd): manually set local date or click to choose local date .Format: year/month/day.

2. Set Local Time(HH:mm:ss) : manually set local time or click to adjust local time. Format: hour/minute/second. e.g. 12:00:00.

4.5.2 Features

➢ **Call Forward**

1. Always Target: Every incoming call will be forwarded to this target.
2. Busy Target: The incoming call will be forwarded to this target when A21 is busy.
3. No Answer: The incoming call will be forwarded to this target when there is no answer.
4. After Ring Time (Seconds): After this time out, the incoming call will be forwarded to this target if no answer. Defaults waiting for 5 seconds.
   The priority of Always Target is highest when Always Target, Busy Target and No Answer Target have been set. This means all the incoming call will be forwarded to the Always Target.

➢ **Call Settings**

1. Do Not Disturb: Select "Yes" or "No" to enable or disable DND (Do Not Disturb).When DND enabled, all the incoming calls will be rejected. At this moment if Always Target or Busy Target was set, all incoming calls will be forwarded to the targets preferentially.

2. Call Waiting：When A21 is on calling, and Call Waiting enabled , any new incoming calls will not be rejected but ringback tone can be heard by the remote end. if Call Waiting disabled, only one conversation will be available even multi-accounts have been registered on A21 .
3. Block Call ID: Select "Yes" here, Caller ID will be blocked, A21 will call others as anonymous.
4. Block Anc Call: Select "Yes" here, all anonymous call will be blocked, A21 will reject all anonymous callers.
5. Auto Answer: Select "Yes" here, A21 will answer all incoming calls by speaker automatically.
6. Send Key: Define the Send key as "#" key or "*"Key. Press "#" Key or "*"Key follow those phone number you want to dial, then number will be dialed out immediately.

4.5.3 Voice

Echo Cancellation
1. VAD：Select "Yes" or "No" to enable or disable VAD (Voice Active Detection). If enable, RTP packets will not be sent when A21 is mute.
2. CNG：Select "Yes" or "No" to enable or disable CNG (Comfort Noise Generator). If enable, comfortable noise will be sent to the remote end to let it perceive the conversation is still active when A21 is mute.

Jitter Buffer
A21 is able to buffer incoming voice packets to minimize out-of-order packet arrival. This process is known as jitter buffer.
1. Type: Choose type of jitter buffer. When choose Fixed, the size of jitter buffer is fixed. When choose Adaptive, the size of jitter buffer is the sum of Minimum Delay and the size of RTP packets.
2. Min Delay: The minimum delay of the jitter buffer.
3. Max Delay: The maximum delay of the jitter buffer.
4. Normal Delay: This is used to set fixed jitter buffer which should be between Min Delay and Max Delay.

4.5.4 Ring

Administer can upload 2 user define ring. The ring file should be wav (8k, 8bit, u-law) and no larger than 200 KBytes.

4.5.5 Tone

Tone

Country Stands:	Custom ▼
Dial Tone:	350@-30,440@-30;60(*/0/1+2)
Outside Dial Tone:	420@-30;60(*/0/1)
Busy Tone:	480@-23,620@-23;10(.5/.5/1+2)
Reorder Tone:	480@-23,620@-23;10(.25/.25/1+2)
Off Hook Warning Tone:	480@-21,620@0;10(.125/.125/1+2)
Ring Back Tone:	440@-23,480@-23;*(2/4/1+2)
MWI Dial Tone:	350@-30,440@-30;2(.1/.1/1+2);10(*/0/1+2)
Holding Tone:	350@-30,440@-30;60(*/0/1+2)
Conference Tone:	350@-23;20(.1/.1/1,.1/9.7/1)

✔ Submit   ↩ Reset

Set the ToneScript for each tone.
For example, MWI Dial Tone: 350@-30,440@-30;2(.1/,1/1=2);10(*/0/1+2)
Frequency1, Frequency2 ;Cadence Section1;Cadence Section2
350@-30: Frequency1 is 350HZ at -19dBm
440@-30: 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.

1. Country stands: Select a country standard tone as default.
2. Dial Tone：The prompting audio to remind user to enter a phone number for dialing
3. Outside Dial Tone：The prompting audio to remind user to enter an external phone number (versus an internal extension). This is triggered by a "," character encountered in the dial plan.
4. Busy Tone：The promoting audio when a 486 RSC is received for an outbound call.
5. Reorder Tone：The promoting audio when an outbound call failed or after the remote end hangs up an established call.
6. Off Hook Warning Tone: The promoting audio when user does not put the handset on the cradle properly.
7. Ring Back Tone：The promoting audio when the remote end is ringing.
8. MWI Dial Tone：The promoting audio when an unread voicemails existed.
9. Holding Tone：Indicate the local end that the current calling is hold by the remote end.
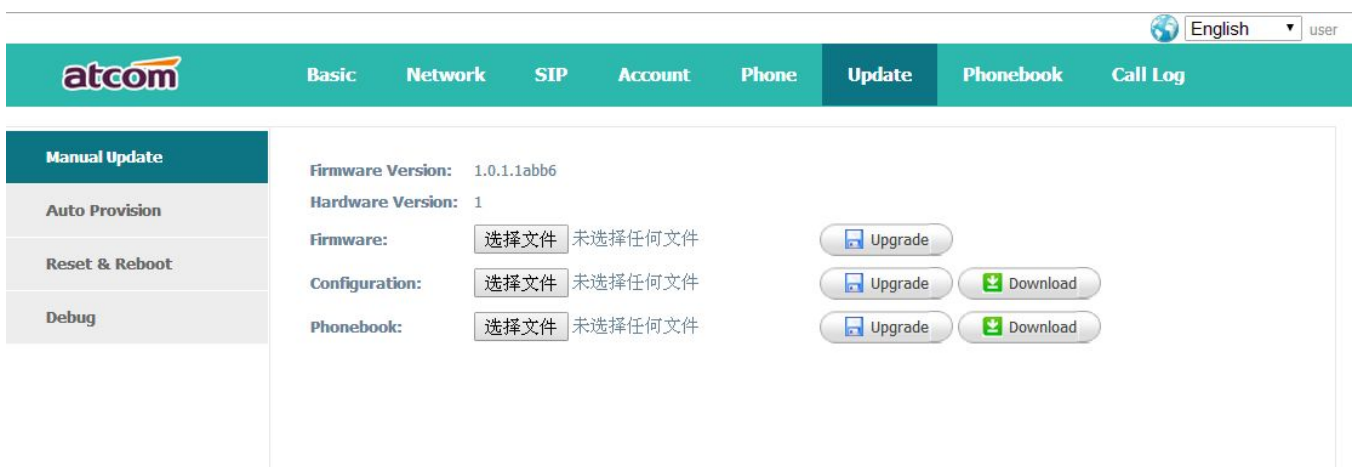10. Conference Tone：The promoting audio to all parties when a 3 way conference is in progress.

4.5.6 Line key

Configure functions for link keys, defaults to Line, BLF and speed dial are available.

4.6 Update

4.6.1 Manual Update

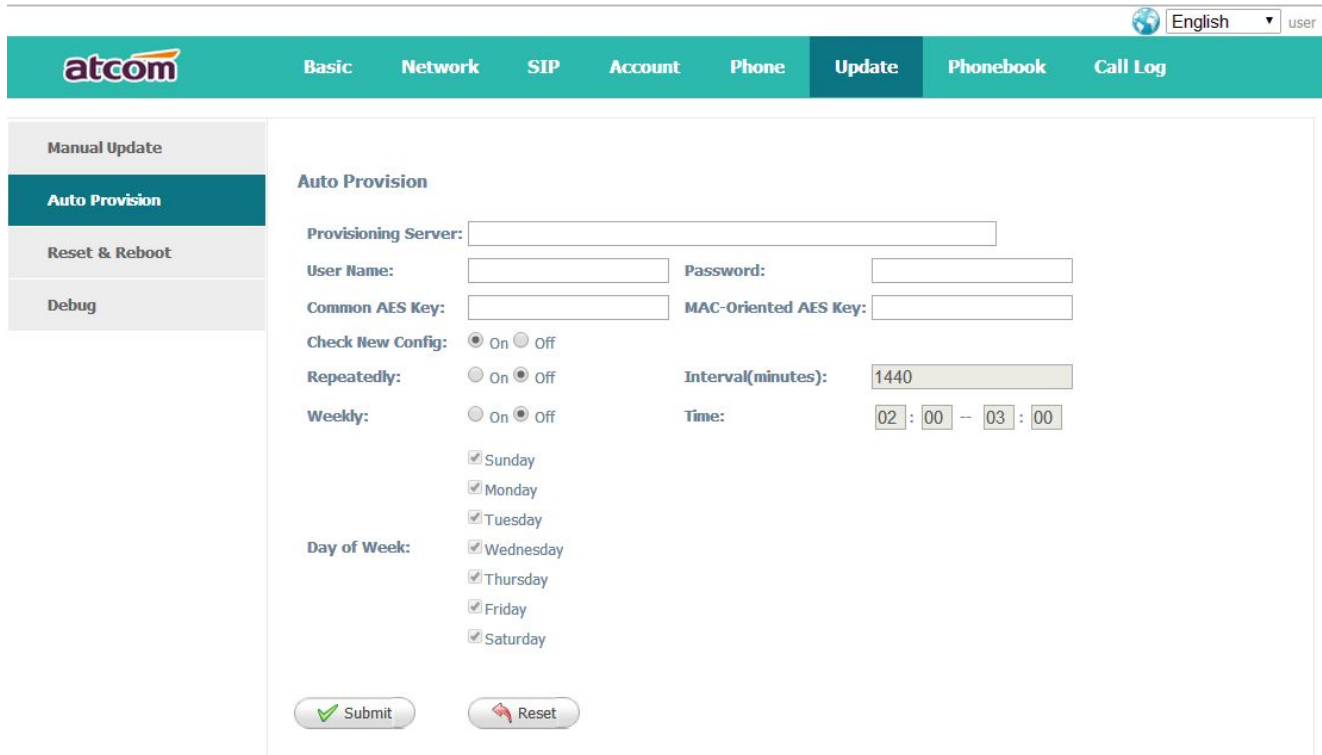1. Firmware: Download firmware from ATCOM's webpage, then select firmware from your PC to upgrade.
2. Configuration: Upload and download config.xml file.
3. Phone Book: Upload and download phonebook.xml file

4.6.2 Update/Auto Provision

Configuration Profile
1. Provisioning Server: The address to save control file for auto upgrading, it can filled by http 、https、tftp server、ftp server.

2. User Name: The user name to access the file server

3. Password: The password to access the file server

4. AES Key: If the configuration file has been encrypted, the AES key will be used to decrypt for auto-upgrading

5. Check New Config: If set this parameter as "On", the phone will do some check for upgrading, and it will upgrade automatically once the conditions are met

6. Repeatedly: If set this parameter as "On", the phone will do some check for upgrading after "Interval(minutes)", and it will upgrade automatically once the conditions are met

7. Interval(minutes) : This parameter is configurable once the Repeatedly is "On"

8. Weekly: If set this parameter as "On", the phone will do the check for upgrading at a certain time in every week

9. Time: The time range that the phone do the check for upgrading every week, and it's configurable once the Repeatedly is "On"

10. Day of week: The day of week that the phone do the check for upgrading every week, and it's configurable once the Repeatedly is "On"

4.6.3 Reset & Reboot
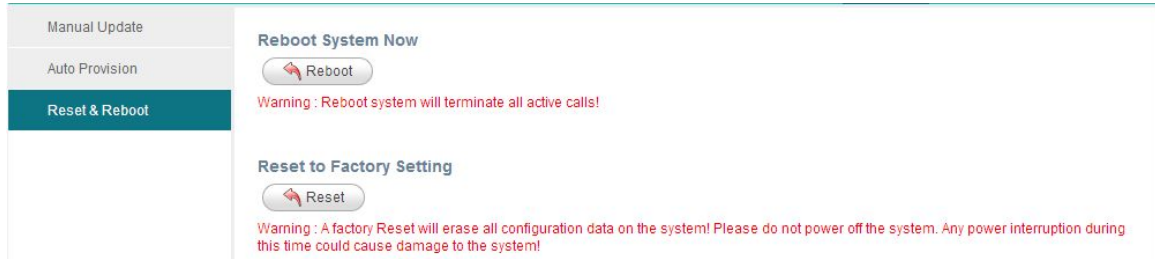
1. Reboot

Reboot will terminate all active calls, and restart the phone in a several seconds.

2. Reset

Click the "Reset" button will set A21 to factory default, please backup the config.xml, phonebook.xml and attendant_keypad.xml before reset.

4.7 Phone Book
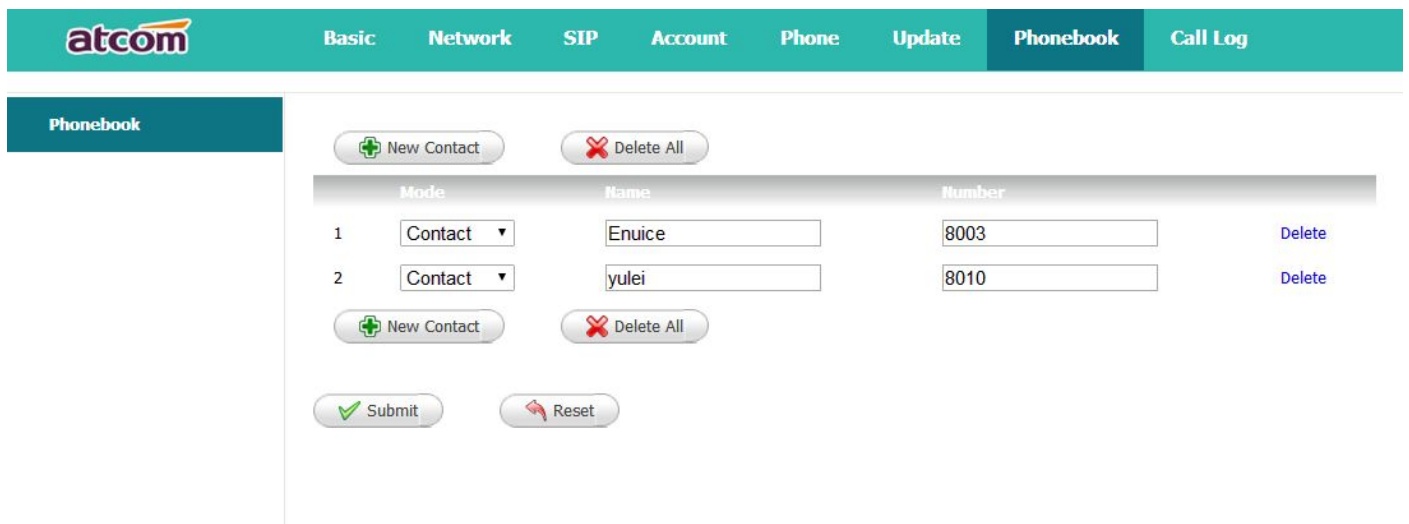
1. New Contact: Add a new phonebook record.
2. Delete All: Delete all the phonebook records.
3. Mode: Contact or Blacklist. All calls from Blacklist will be rejected.
4. Name: Specify a name for each phone number. This name will be displayed on LCD when call to this number or call from this number.
5. Number: Phone number.
6. Submit: Submit to apply the change.
7. Reset: Cancel the inputting.
8. Delete: Delete this record.

Notice：The maximum of records in the phone book is 1000.

4.8 Call Log

	Name	Number	Time
Dialed			
Received			
Missed	Delete All		
1 :	5039	5039	20/02 11:40
2 :	IP Call	172.16.0.116	20/02 11:40
3 :	5069	5069	20/02 11:39

	Name	Number	Time
Dialed			
Received			
Missed	Delete All		
1 :	5039	5039	20/02 11:40
2 :	IP Call	172.16.0.116	20/02 11:40
3 :	5069	5069	20/02 11:39

1. Redial: Record of dialed list, maximum 200 records.
2. Received: Record of received list, maximum 200 records.
3. Missed: Record of Missed list, maximum 200 records.

5. FAQ & Trouble Shooting

Frequently Asked Questions

5.1 How to make a factory reset

There are three ways to make a factory reset:

1.Factory reset from keyboard, steps are:

a) Press ⊡ and input password 123.

b) Find and select '12 Factory Reset'.

c) Press 'OK' key, then the phone will be restarted automatically and factory reset.

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

3. Factory reset during rebooting, steps are:

a) Press '*' and '#' key immediately after plugging in the power adapter.

b) Wait for about 5 seconds, A21 will be reset to factory default setting after rebooted.

5.2 Upgrade firmware under safe mode

If the phone could not start up normally, please 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 A21.FW.

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

4. Keep pressing '#' key and start up the phone until the LCD displays '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 factory configuration.

5.3 How to make direct IP call

When hook off/ pressing speaker or headset key,

1. Press soft key 'Num' until it switch to 'IP'.

2. Using '*' key to input '.'. 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. Trouble Shooting

6.1 The phone can't register successfully

1. Check the IP address, and if the mode of WAN port is DHCP, please make sure the DHCP server is in service.
2. Check the gateway.
3. Check the DNS.
4. Make sure the information of the account is consistent with which offered by the service supplier.
5. Make sure the SIP server is on.
6. Check the port of the SIP server whose default value is 5060.

6.2 The phone can't obtain IP address

1. Make sure the cable has been connected to the LAN port of the phone.
2. Make sure the cable and the switch's port that connected with the cable is available.
3. Make sure the DHCP server is on and there are some assignable IP addresses in the address pool.
4. Try to change the LAN port mode as Static.

6.3 Only one part can hear the voice during the call

1. Make an IP dial-up call to make sure the telephone receiver and microphone are normal.
2. Enable STUN on web page.
3. Set STUN server as stun.sipgate.com.
4. Click 'submit' and wait for the phone to restart.
5. Try to make calls again.

7. Abbreviations

DND ： Do Not Disturb

CFWD ： Call Forward

Bxfer ： Blind Transfer

Conf ： Conference

Num ： Number

DelChr ： Delete Char

Y/N ： Yes/No

SIP：Session Initiate Protocol

RTP：Real-time Transport Protocol

SDP：Session Description Protocol

VPN：Virtual Private Network

VLAN：Virtual Local Area Network

QoS：Quality of Service

Syslog ： System log

UDP：User Data Protocol

TCP：Transmission Control Protocol

TLS：Transport Layer Security Protocol

BLF：Busy Lamp Field

DNS：Domain Name System

SRTP：Secure Real-time Transport Protocol

NTP：Network Time Protocol

VAD：Voice Activity Detection

CNG：Comfort Noise Generator