



IP 250

Thank you for choosing your Atl IP250 telephone. You have selected a carefully designed telephone that incorporates the very latest technology, offering a stylish instrument, and providing many years of excellent service.

This booklet will ensure you obtain the best use of your IP telephone.



User Guide

IP250 IP Phone

Web Configuration Manual

To access the web configuration and status pages for your phone simply open up your preferred web browser and enter the IP address of your IP250 into the address URL field. The web interface has two privilege levels: Administrator and User. Both privilege levels are password protected by default. The user names for the Administrator and User are “admin” and “user” respectively. And the default password for both privilege levels is “voip”. The major difference between an Administrator and a User is that the Administrator has full control over the IP250 device while the User mode only allows limited changes.

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1 HOME

This following page is the first page displayed when the device is logged in to. Figure 1 and 2 show the variations in pages dependent on an Administrator or a User logging in. As the screenshots show below, the user level privilege (Figure 1) doesn't have access to the configuration page of CODECS, Download and Configuration.

It displays how long the device has been running since its last reboot, the IP address the device is currently using, whether or not the device is password protected, and also displays the main application and downloader application firmware versions. In addition, the MAC address of the WAN port, and serial number of the device, if it has one, are also displayed in this page.

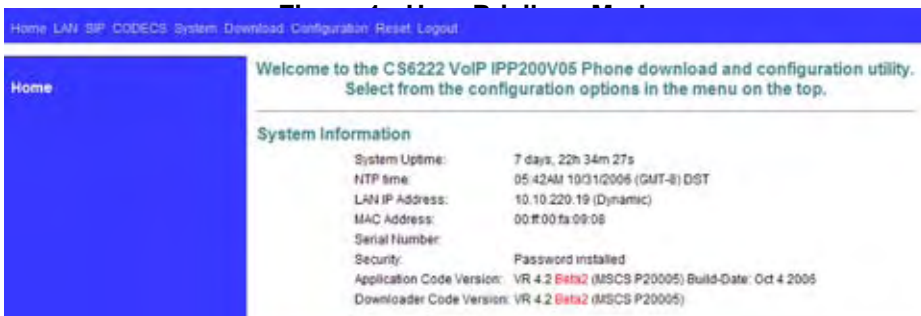
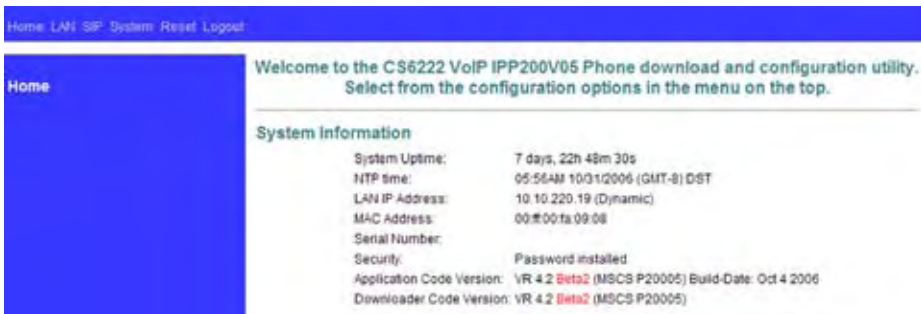


Figure 2 - Administrator Privilege

2 LAN

2.1 LAN Status

The following page allows configuration of the phone's local network settings. Sub-pages are available for the following on the left of page: the LAN interface status, configuration of the LAN interface settings, and configuration of DHCP server settings, and Port Forwarding (NAPT) Settings.

Home **LAN** SIP CODECS System Download Configuration Reset Logout

LAN Status
LAN Settings
PPFcE
IPSEC

LAN Status

Interface Status

Enabled:	Yes
Protocol:	Ethernet
Interface Status:	Up
Link Status:	100M bps, Full Duplex

Network Settings

IP Address:	10.10.220.19
MAC Address:	00:ff:00:fa:09:08
Subnet Mask:	255.255.0.0
Default Gateway:	10.10.1.2
Host Name:	
Domain Name:	
Priority Tag:	Not set

Update

2.2 LAN Settings

The following page allows the user to configure the private LAN interface settings. Assign an IP address to the LAN Ethernet port. This IP address is also the default router address for the devices on the private LAN. The default LAN interface IP address is set to 192.168.1.1. Enter the subnet mask for the private LAN. If you wish to set the broadcast and multicast limits for the bridge/router, enter these values as percentages of the LAN interface Ethernet bit rate. Leaving these values blank will imply values of 100%.

Press "Save LAN Settings" to save and apply the LAN interface settings. Any new settings will take effect immediately.

The screenshot shows a web interface for LAN configuration. At the top, a blue navigation bar contains the text: Home LAN SIP CODECS System Download Configuration Reset Logout. On the left, a blue sidebar menu lists: LAN Status, LAN Settings (highlighted), PPPoE, and IPSEC. The main content area is titled "LAN Configuration" and contains two sections. The first section has two radio buttons: "Use DHCP to obtain LAN configuration" (selected) and "Specify fixed LAN configuration". Below the second radio button are three input fields for "IP Address:", "IP Netmask:", and "IP Gateway:". The second section has two radio buttons: "Automatically obtain DNS server settings" (selected) and "Manual DNS server settings". Below the second radio button are four input fields for "IP DNS Server:", "IP DNS Server2:", "Host Name:", and "Domain Name:". At the bottom of the form is a "Save LAN Settings" button.

2.3 PPPoE

This page allows you to configure the phone to authenticate to your network using PPPoE. If you wish to use this authentication method please fill out the appropriate fields and select Yes in the Enable PPPoE's selection Box. Please note that you will need to obtain your username and password from who ever provides your network service.

Home LAN SIP CODECS System Download Configuration Reset Logout

LAN Status
LAN Settings
PPPoE
IPSEC

LAN PPPoE Configuration

Enable PPPoE:

Authentication

Username:

Password:

Settings

Idle Timeout: minutes

Service Name:

AC Name:

2.4 IPSEC UPDATE TO FOLLOW

IPSec Configuration

Select Tunnel to view/modify: Tunnel 1

Enable tunnel 1:

Remote IP Address range: -

Remote security gateway:

Security mode: Tunnel1

Outbound AH SPI (DEC):

Outbound AH Authentication Algorithm: HMAC-SHA1

Outbound AH Authentication Key (HEX):

Outbound ESP SPI (DEC):

Outbound ESP Encryption Algorithm: NULL

Outbound ESP Authentication Algorithm: NULL

Outbound ESP Encryption Key (HEX):

Outbound ESP Authentication Key (HEX):

Inbound AH SPI (DEC):

Inbound AH Authentication Algorithm: HMAC-SHA1

Inbound AH Authentication Key (HEX):

Inbound ESP SPI (DEC):

Inbound ESP Encryption Algorithm: NULL

Inbound ESP Authentication Algorithm: NULL

Inbound ESP Encryption Key (HEX):

Inbound ESP Authentication Key (HEX):

Save Tunnel Settings

3 SIP

3.1 Server

The following page allows configuration of the SIP server and endpoint settings.

Enter the address and port value of the SIP server. The address may be an IP address or the name of the server. If no SIP server address is entered, the device will attempt to self provision a SIP server using a DNS query. For this to be successful, ensure that the DNS settings on the device include a DNS server address which is configured with the SIP server address and will respond to the query, and the appropriate domain name of the network.

If you wish to specify a special SIP domain name, you may enter the domain name here. If no domain name is entered, the SIP domain name will be set to that of the network (i.e. that which is obtained via DHCP, or specified on the LAN settings page).

The currently provisioned SIP Server and Domain are displayed beside "SIP Server Settings" for informational purposes. Select whether or not to send a Registration Request to the SIP server by checking the box next to "Send Registration Request".

For the endpoint, set the dial plan to be used by all lines, and select the transport method to be used for SIP signaling (either UDP or TCP). For each line on the endpoint (NOTE: The IP Phone has a single line), enter the Line Phone Number, Caller-ID Name, signaling port value, authentication Username and Password, and select if AEC is to be performed on this line.

Press "Save SIP Settings" to save the new values.

Home LAN SIP (DDCC) System Overview Configuration Manual List

SIP Server Configuration

Primary Server Settings	Secondary Server Settings
(Current Server: 5000, Domain:)	(Current Server: 0, Domain:)
* Address: <input type="text"/> (IP or FQDN)	* Address: <input type="text"/> (IP or FQDN)
* Port: <input type="text"/>	* Port: <input type="text"/>
Domain Name: <input type="text"/>	Domain Name: <input type="text"/>
<input checked="" type="checkbox"/> Send Registration Request with Expires Time: 3600	<input checked="" type="checkbox"/> Send Registration Request with Expires Time: <input type="text"/>
Outbound Proxy IP: <input type="text"/> (IP or FQDN)	Outbound Proxy IP: <input type="text"/> (IP or FQDN)
Outbound Proxy Port: <input type="text"/>	Outbound Proxy Port: <input type="text"/>

RTP Port Number Setting: 5000-60535) -

NAT Traversal Settings

NONE

UFWP Control Point

STUN Server IP: (IP or FQDN) STUN Server Port:

* Leaving a setting blank will force the unit to use the information obtained via DHCP and/or DNS.
** Leaving a setting blank will disable server redundancy function.

3.2 Extension

The following page allows specification of the SIP signaling stack behavior under certain scenarios.

Home LAN SIP CODECS System Download Configuration Reset Logout

Server
Extensions
Digit Map
User 1
OOB Signalling
ToS/DiffServ
Tone
Ring
Service Code
Phone Book

SIP Extensions

- Support PRACK method with provisional response reliability
- Encode SIP URI with user parameter
- Session Timer use UPDATE method
- Call Hold using c=0.0.0.0 (RFC 2543) in SDP
- enable Global Number support (E.164)
- send NOTIFY for REFER request
- send Message Waiting Indicator (MWI) SUBSCRIBE command
- No Authorization Header in re-REGISTER
- Check existence of To Tag in INVITE 2xx response

SIP Timers

- Send INVITE with Timer header value: Seconds
- SIP Session Timer value: Seconds
- SIP Keep Alive Timer value: Seconds
- Conditional Call Forwarding Timer: Seconds

Inter Digit Timer: Seconds.

SIP T1 Timer: Milliseconds

SIP T2 Timer: Milliseconds

SIP T4 Timer: Milliseconds

Save SIP Extension Settings

- ▶ Support PRACK method: If you wish for the SIP stack to implement reliable transmission of provisional responses according to RFC 3262 (using the PRACK method) ,check the option.
- ▶ Encode SIP URI with user parameter: include the user parameter “user=phone” in the SIP URI headers.
- ▶ Session Timer use UPDATE method: Session timer use update instead of reinvite.
- ▶ Call Hold using C=0.0.0.0: using the call hold method described in RFC2543. If unchecked, the call hold would follow RFC3263 method.

- ▶ Enable Global Number support (E.164): add prefix “+” for dialed numbers in sip invitation.
- ▶ Send NOTIFY for REFER request: send out NOTIFY request to transferer for unattended and attended call transfer.
- ▶ Send Message Waiting Indicator (MWI) SUBSCRIBE command: send SUBSCRIBE after registered to server to check if there are any messages to be read.
- ▶ Send INVITE with Timer header: encode Timer header in all INVITE requests for ringing timeout.
- ▶ Enable SIP session timer: If you wish for the SIP stack to implement a session timer according to “draft-sip-session-timer”, select the option.
- ▶ SIP T1 Timer, SIP T2 Timer, and SIP T4 Timer: please refer to RFC3261.

Press “Save SIP Extension Settings” to save the new values.

3.3 Digit Map

UPDATE TO FOLLOW

Home LAN SIP CODECS System Download Configuration Reset Logout

Server
Extensions
Digit Map
User 1
DOB Signaling
ToS/DiffServ
Tone
Ring
Service Code
Phone Book

Gateway Settings

Dial Plan:

Name	Digits for matching	Operation	Digits for operation
Digit Map1	<input type="text"/>	dropped	<input type="text"/>
Digit Map2	<input type="text"/>	dropped	<input type="text"/>
Digit Map3	<input type="text"/>	dropped	<input type="text"/>
Digit Map4	<input type="text"/>	dropped	<input type="text"/>
Digit Map5	<input type="text"/>	dropped	<input type="text"/>
Digit Map6	<input type="text"/>	dropped	<input type="text"/>
Digit Map7	<input type="text"/>	dropped	<input type="text"/>
Digit Map8	<input type="text"/>	dropped	<input type="text"/>

*The fields must be set to 'null' if this field will do nothing.

use as a quick dial function * use as a quick dial function
 To enable # to be recognized as dial number To enable * to be recognized as dial

Save SIP Settings

3.4 User 1

The following page allows the user to configure the phone to connect to an IPPBX or a SIP service. Enter the phone number, caller ID, username and password specified by your ITSP or the administrator of your IPPBX.

Home LAN SIP CODECS System Download Configuration Reset Logout

Server
Extensions
DigIP Map
User 1
OOB Signaling
Tools/Outputs
Tone
Ring
Service Code
Phone Book

User 1 Configuration

Line 1	Phone Number	CallerID Name	Port	User Name	Password
Primary Server	<input type="text"/>	<input type="text"/>	\$060	<input type="text"/>	<input type="text"/>
Secondary Server	<input type="text"/>	<input type="text"/>	\$060	<input type="text"/>	<input type="text"/>

Line1 AEC Control

Line1 Gain Control

Input Gain Control (-12 ~ 18)db db

Output Gain Control (-12 ~ 18)db db

Supplementary Service Subscription

Enable Call Waiting (Reject second incoming call)

Enable Caller-ID Display

Reject anonymous call

Block Caller-ID in outgoing call

Distinctive Ring Settings

Ring1 Caller:	<input type="text"/>	Ring2 Caller:	<input type="text"/>
Ring3 Caller:	<input type="text"/>	Ring4 Caller:	<input type="text"/>
Ring5 Caller:	<input type="text"/>	Ring6 Caller:	<input type="text"/>
Ring7 Caller:	<input type="text"/>	Ring8 Caller:	<input type="text"/>

Speed Dial Settings

Speed Dial 1:	<input type="text"/>	Speed Dial 2:	<input type="text"/>
Speed Dial 3:	<input type="text"/>	Speed Dial 4:	<input type="text"/>
Speed Dial 5:	<input type="text"/>	Speed Dial 6:	<input type="text"/>
Speed Dial 7:	<input type="text"/>	Speed Dial 8:	<input type="text"/>

Line1 Polarity

Idle

Caller connected

Callee connected

3.5 OOB Signaling

The following page allows configuration of the out-of-band signaling options for SIP. Select whether OOB telephone event signaling is to be done using the SIP INFO message, or to be done via RFC2833 RTP signaling.

The screenshot shows a web interface with a blue navigation bar at the top containing links: Home, LAN, SIP, CODECS, System, Download, Configuration, Reset, Logout. On the left is a blue sidebar menu with the following items: Server, Extensions, Digit Map, User 1, **OOB Signalling**, ToS/DiffServ, Tone, Ring, Service Code, Phone Book. The main content area is titled "RTP Telephone Event Configuration" and contains the following settings: "Send DTMF Events" is set to "Out-of-Band (RFC2833)" via a dropdown menu; "RFC2833 signalling using payload value:" is set to "96" in a text box; there is an unchecked checkbox for "Regenerate OOB DTMF tone"; and a "Save OOB Settings" button is located at the bottom of the configuration area.

3.6 Tos/Diffserv

The following page is used to configure the Type-of-Service/Diffserv byte values which are to be used in the IP header of all transmitted SIP signaling packets and RTP packets. The ToS/DiffServ byte values are entered as two-digit hexadecimal values. If no special ToS/DiffServ value is to be used for a particular traffic type, enter "00" or leave the setting empty. Press "Save ToS/DiffServ Settings" to save these new settings.

The screenshot shows a web interface with a blue navigation bar at the top containing links: Home, LAN, SIP, CODECS, System, Download, Configuration, Reset, Logout. On the left is a blue sidebar menu with the following items: Server, Extensions, Digit Map, User 1, OOB Signalling, **ToS/DiffServ**, Tone, Ring, Service Code, Phone Book. The main content area is titled "ToS/DiffServ" and contains the following settings: "Call Signalling Packets:" is set to "10" (2 Hex digit byte value) in a text box; "RTP Packets:" is set to "14" (2 Hex digit byte value) in a text box; and a "Save ToS/DiffServ Settings" button is located at the bottom of the configuration area.

3.7 Tone

The following page is used to configure Tones which applies in order to acknowledge users.

- ▶ **Dial Tone:** The tone you hear when you pick up handset
- ▶ **Recall Dial Tone:** The tone when you hold callee and prepare to make another call.
- ▶ **Confirm Tone:** The tone after you've set up some service, like DND (Do Not Disturb), Call Forwarding, etc.
- ▶ **Ring Back Tone:** The audible ringing you hear before callee picks up and answers your call.
- ▶ **Busy Tone:** The tone indicates the number you dialed is in busy now.
- ▶ **Reorder Tone:** The tone you hear if you dial an invalid number or the call is not available.
- ▶ **Receiver-Off-Hook Tone:** The tone to alert you to place the handset on-hook.
- ▶ **Message-Waiting-Indicator Tone:** The tone to notify you to call for message box.
- ▶ **Call-Waiting-Indicator Tone:** The tone to make you aware of the second incoming call while you're in conversations.

Home LAN SIP CODECS System Download Configuration Reset Logout

Tone Configuration

Dial Tone:	<input type="text" value="350@-13+440@-13#ON(1000),R"/>
Recall Dial Tone:	<input type="text" value="350@-13+440@-13#[ON(100),OFF(100)]3,ON(1000),R"/>
Confirm Tone:	<input type="text" value="350@-13+440@-13#[ON(100),OFF(100)]3,OFF(1000),R"/>
Ring Back Tone:	<input type="text" value="440@-19+480@-19#ON(2000),OFF(4000),R"/>
Busy Tone:	<input type="text" value="480@-24+620@-24#ON(500),OFF(500),R"/>
Reorder Tone:	<input type="text" value="480@-24+620@-24#ON(250),OFF(250),R"/>
Receiver-Off-Hook Tone:	<input type="text" value="1400@-3+2060@-3+2450@-3+2600@-3#ON(100),OFF(100),R"/>
Message-Waiting Indicator Tone:	<input type="text" value="350@-13+440@-13#[ON(100),OFF(100)]10"/>
Call-Waiting Indicator Tone:	<input type="text" value="400@-14#ON(150)"/>

3.8 Ring

The following page is used to configure Ring Cadences required by Rings, Call-Waiting-Indicator, and Distinctive Ring features.

- ▶ **Default Ring:** Default ring cadence when the phone rings.
- ▶ **Call-Waiting Reminder Ring:** Ring cadence of Call-Waiting Reminder Ring.
- ▶ **Distinctive Ring 1-8:** Ring cadences provided for distinctive ring function.

You may customise them according to the fixed format.

For example,

ON(500),OFF(500),R

Will cause 500 milliseconds ring on, then 500 milliseconds off, and repeat steadily.

Home LAN SIP CODECS System Download Configuration Reset Logout

Server
Extensions
Digit Map
User 1
OOB Signaling
ToSIPdSrv
Tone
Ring
Service Code
Phone Book

Ring Configuration

Default Ring:	ON(1000),OFF(2000),R
Call-Waiting Reminder Ring:	ON(125),OFF(425),ON(2000),OFF(2875),R

Distinctive Ring Configuration

Distinct Ring 1:	ON(500),OFF(1500),R
Distinct Ring 2:	ON(400),OFF(200),ON(400),OFF(2000),R
Distinct Ring 3:	ON(200),OFF(100),ON(200),OFF(100),ON(400),OFF(2000)
Distinct Ring 4:	ON(400),OFF(500),ON(200),OFF(25),ON(200),OFF(1500)
Distinct Ring 5:	ON(250),OFF(50),R
Distinct Ring 6:	ON(500),OFF(500),R
Distinct Ring 7:	ON(150),OFF(1000),ON(150),OFF(1000),ON(500),OFF(10)
Distinct Ring 8:	ON(500),OFF(10),R

Save Ring Settings

3.9 Service Code

The following page is used to configure the service codes available on the phone, you can setup a service by setting the code you wish to dial through the keypad.

Service Code Configuration:

Conditional Call Forwarding: *70#

Call Forward On: *72#

Call Forward Off: #72#

Do Not Disturb On: *74#

Do Not Disturb Off: #74#

Call Transfer: *98#

Call Return: *69#

Speed Dial: *68

Note: Do NOT change the service code values unless there is a conflict between the settings of your VoIP device and the settings provided by your service provider.

Home LAN **SIP** CODECS System Download Configuration Reset Logout

Service Code Configuration

Conditional Call Forwarding:	<input type="text" value="*70#"/>
Call Forwarding On Busy:	<input type="text" value="*71#"/>
Call Forwarding On:	<input type="text" value="*72#"/>
Call Forwarding Off:	<input type="text" value="#72#"/>
Do Not Disturb On:	<input type="text" value="*74#"/>
Do Not Disturb Off:	<input type="text" value="#74#"/>
Call Transfer:	<input type="text" value="*98#"/>
Call Return:	<input type="text" value="*69#"/>
Speed Dial:	<input type="text" value="*68"/>

- use *XX# or #xx# format , xx=01-99

Save Service Code Settings

- Server
- Extensions
- Digit Map
- User 1
- OOB Signalling
- ToS/DiffServ
- Tone
- Ring
- Service Code**
- Phone Book

3.10 Phonebook

This page allows you to setup 10 numbers you wish to store on your phone. To add an entry into your phone book fill out the number section with the phone number of the person you wish to add. Once the number has been entered then you will need to enter the sip domain that the phone number is registered to in the IP Address field along with a port number in the Port field. The default port number for sip is 5060. When you have finished filling out the contact details you will need to click the Save SIP Phone Book Settings button at the bottom of the page to save your new entries.

Home LAN SIP CODECS System Download Configuration Reset Logout

[Server](#)
[Extensions](#)
[Digit Map](#)
[User 1](#)
[OOB Signalling](#)
[ToSiDiffServ](#)
[Tone](#)
[Ring](#)
[Service Code](#)
[Phone Book](#)

SIP Phone Book Configuration

	Number	IP Address	Port
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>
Phone Number:	<input type="text"/>	<input type="text"/>	<input type="text"/>

4 CODECS

If the device is running one of the four VoIP applications, the following page is available for configuring the audio CODEC parameters, as well as the Jitter Buffer settings for the CODEC decoders.

Enter which CODECs are to be supported. For some protocols (e.g. H.323 and SIP), the G711U and G711A protocols are always supported by default. For MGCP and H.248, it is possible to remove these CODECs from the devices list of supported “capabilities”.

Select which complex codec is to be supported. Due to memory limitations, it is not possible to select more than one complex codec.

Select the packetization period to be used for each selected CODEC. For MGCP, a range of packetizations may be provided for each CODEC (to be advertised in the device’s “capabilities” set).

Select whether Silence Suppression is to be supported for each CODEC.

The Jitter Buffer settings apply to all active CODEC decoders. You may choose between an adaptive jitter buffer and a fixed jitter buffer. For an adaptive jitter buffer, choose the maximum allowable playout delay (in milliseconds). For a fixed jitter buffer, choose the fixed playout delay (in milliseconds).

Finally, select whether or not a decoder should automatically switch from an adaptive jitter buffer to a fixed jitterbuffer upon fax/modem tone detection. Adaptive jitter buffers are sometimes detrimental to fax transmission over G711 CODECs if they have to adapt too rapidly or too extensively due to inconsistent and widespread packet delays. In these adverse network conditions, a fixed jitter buffer provides superior performance when handling incoming fax transmissions over G711 CODECs. Press “Save CODEC Settings” to save the new CODEC parameters.



5 SYSTEM

5.1 Security

The password needed to access the VoIP device via web interface can be set in the following page. First enter the old password, then enter the new password and confirm the new password. Click on the Change Password button to save the change. The VoIP device will log the user out and redirect the user to the login page.

Home LAN SIP CODECS **System** Download Configuration Reset Logout

Security
Timeout
Localization
Handset
Port Number
Ringer Tone
SNMP

Set Security Password

Password is currently installed

Account: admin

Old password:

New password:

Confirm new password:

5.2 Timeout

This field will only be active if the admin password is set. To change the Authentication time out open the System menu from the top headings. This will bring up the system sub menus on the left hand side of the page. Select Timeout from the menu. In HTTP Authentication Timeout field, input the timeout value you wish to use, then press the Change Time button. The phone's browser authentication will now timeout according to the time you have set in this field. Once the timeout expires the user will be prompted to log into the phone again before being allowed to change anymore settings.

Home LAN SIP CODECS **System** Download Configuration Reset Logout

Security
Timeout
Localization
Handset
Port Number
Ringer Tone
SNMP

Set Web System Timeout

HTTP Authentication Timeout (Seconds)

Change Time

5.3 Localization

Timezone:

Find the current time from a list of cities.

Country Caller ID:

The caller ID can find out who's calling you and keep track of how often they call.

Users should set the country field according to their geographical location, otherwise the Caller ID function might not work properly.

Timezone setting:

Click "System" on the top menu.

Click "Localization" on the left menu.

In NTP Server field, enter a NTP server IP address.

If you want to use the default NTP server, this field should be blank.

In Time Zone drop down menu, select one time zone.

In Adjust clock for daylight savings checkbox, if your country has daylight savings time, you can enable it. Press Save Localization Settings button, then system will redirect to the web page of reset.

Country Caller ID setting:

Click "System" on the top menu.

Click "Localization" on the left menu.

In Country drop down menu, select one country.

Press Save Localization Settings button, then system will redirect to the web page of reset.

The screenshot shows a web interface with a blue header bar containing navigation links: Home, LAN, SIP, CODECS, **System**, Download, Configuration, Reset, Logout. On the left, a blue sidebar menu lists: Security, Timeout, **Localization**, Handset, Port Number, Ringer Tone, SNMP. The main content area is titled "Localization" and contains the following fields:

- Country: United Kingdom (dropdown menu)
- NTP Server: (text input field)
- Time Zone: (GMT) Greenwich Mean Time: Dublin, Edinburgh, Lisbon, London (dropdown menu)
- Adjust clock for daylight savings

At the bottom of the form is a button labeled "Save Localization Settings".

5.4 Handset

The following page allows user to configure the flash hook time interval.

Home LAN SIP CODECS **System** Download Configuration Reset Logout

Security
Timeout
Localization
Handset
Port Number
Ringer Tone
SNMP

IP Phone Handset Configuration

Display String:

Display Number:

Program Key1:

Program Key2:

Program Key3:

Program Key4:

Program Key5:

Program Key6:

Program Key7:

Program Key8:

Program Key9:

Program Key10:

Program Key11:

Allow Network Configuration on LCD Menu

Control Timer Values

Hook Flash Timer Min: Milliseconds

Hook Flash Timer Max: Milliseconds

***Please enter a multiple of 10.(ex:10,20,30...)**

5.5 Port Number

This allows the device to use different port number for the http server. The default port number is 80.



5.6 Ringer Tone

The following page allows users to select desired ring tone from the drop down list, click Save Ring Tone Setting once selected.



5.7 SNMP

The following page is used for configuring the device's SNMP manager. Configure the SNMP Trap Host IP address and community, the SNMP read and write community parameters, and the SNMP System Description and System Object ID parameters.

Press "Save SNMP Settings" to apply the new values. These settings will only take effect when the device is rebooted.

The screenshot shows a web interface for configuring SNMP. At the top, a blue navigation bar contains the text: Home LAN BIP CODECS **System** Download Configuration Reset Logout. On the left, a blue sidebar menu lists: Security, Timeout, Localization, Handset, Port Number, Ringer Tone, and **SNMP**. The main content area is titled "SNMP Configuration" and is divided into three sections:

- SNMP Trap Configuration:** Contains two input fields: "IP address:" (empty) and "Trap Community:" (empty).
- SNMP Community Configuration:** Contains two input fields: "Read Community:" with the value "public" and "Write Community:" with the value "private".
- SNMP System Configuration:** Contains two input fields: "System Description:" (empty) and "System Objectid:" with the value "4528".

At the bottom of the configuration area is a button labeled "Save SNMP Settings".

6 DOWNLOAD

The following page provides two options for downloading a new firmware application image to the device. If you wish to download the new firmware image using TFTP, enter the filename of the ROM image and enter the IP address of the TFTP server on which this file resides.

To initiate the TFTP download process, press “Start TFTP Download.” If the ROM image is stored on the same local machine you are using to access the device’s web pages, you can choose to download the ROM file to the device using an HTTP post. Enter the filename of the ROM image or press “Browse” to help locate the file.

To initiate the HTTP download process, press “Start HTTP Download.”

If the main application is executing at the time, the device will automatically reboot itself into the downloader mode and begin the download process. If the downloader application is executing at the time, the download process will begin. The download status will be displayed when the image download process is complete. Please refer to Section A “The Downloader Application” for more details on the download process.



6.1 HTTP Download method

When using http to upgrade firmware, it will check firmware version before starting download process.

In Filename field, press Browsing Button.

Press Start HTTP Download button to start downloading file.

If firmware version doesn't fit in with old version, it won't allow updating.

6.2 AutoUpdate

AutoUpdate: This feature is an auto-installation system. It can update device configuration values or firmware through TFTP, HTTP, or HTTPS.

Click "Download" item on the top menu.

Click "AutoUpdate" item on the left menu.

Select "YES" from Enable AutoUpdate drop down menu.

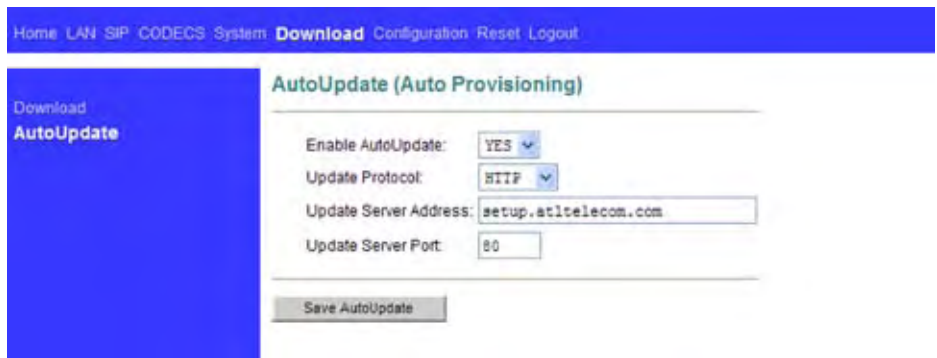
Select item form Update Protocol drop down menu.

In Update Server Address field, key in IP address by connected server.

In Update Server Port field, key in port number by connected server. (e.g.: 23/80/443)

Press "Save AutoUpdate" button.

The web system will redirect to reset web page.



The screenshot shows a web interface for configuring AutoUpdate. At the top, a blue navigation bar contains the links: Home LAN SIP CODECS System **Download** Configuration Reset Logout. On the left, a blue sidebar menu has "Download" and "AutoUpdate" (highlighted). The main content area is titled "AutoUpdate (Auto Provisioning)" and contains the following form fields:

Enable AutoUpdate:	YES
Update Protocol:	HTTP
Update Server Address:	setup.atltelecom.com
Update Server Port:	80

Below the form is a "Save AutoUpdate" button.

7 CONFIGURATION

7.1 Backup

Backup configuration values of system settings to a file from the device:

Click "Configuration" item on the top menu.

Click "Backup" item on the left menu.

Press Backup Configure File button to save configuration file.



7.2 Restore

The following page is used to restore configuration values of system settings from a previously saved configuration file, or default factory values that stored inside the device.

Restore configuration from a file:

Click "Configuration" item on the top menu.

Click "Restore" item on the left menu.

Press Browsing button to select file by backup from local machine.

Press Start Download button to process downloading file.

After downloading file is finished, the web system will redirect to restart device.

Restore default factory values form device:

Click "Configuration" item on the top menu.

Click "Restore" item on the left menu.

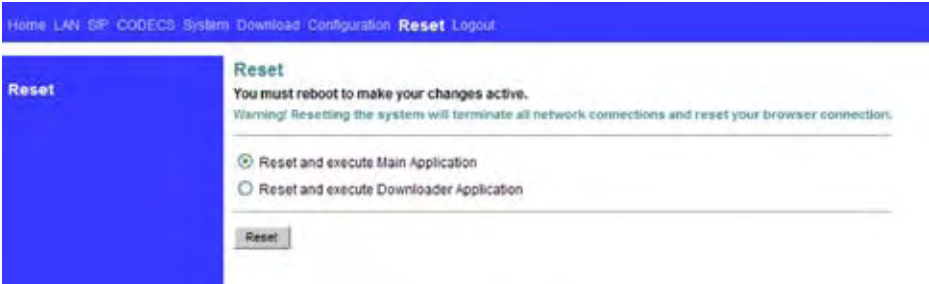
Press "Start Restore Default Factory" button.

After restoring default factory, the web system will redirect to restart device.

The screenshot shows a web interface with a blue header and a blue sidebar. The header contains navigation links: Home, LAN, SIP, CODECS, System, Download, Configuration, Reset, and Logout. The sidebar has two options: Backup and Restore. The main content area is titled "Configure Restore" and contains a section for "Configure Restore method (Select filename on local browser machine)". This section includes a "Filename:" input field, a "Browse..." button, and a "Start Download" button. Below this is a section titled "Restore Factory Default" with a "Start Restore Default Factory" button.

8 RESET

The following page provides options for resetting the device. Select whether you wish to reset the device and start executing the main (default) application, or whether you wish to reset the device and start executing the internal downloader application. Press "Reset" to reset the device



Phone Functions

MESSAGE

Press to link to voicemail server to retrieve voice messages

UP and DOWN arrows

Navigation keys

ENTER

Press to move to sub-menus

DELETE

1. Delete digits or characters under the circumstances of editing
 - a. Press to edit the number while initiating a call or a call conference or pre-dial
 - b. Press to edit the number when setting forwarding number
 - c. Press to edit data when add new contact information to phone book
 - d. Press to edit the phone number when programming speed dial
2. Press to delete contact information
 - a. Delete contact information from phone book
 - b. Delete contact information from received calls
 - c. Delete contact information from missed calls
 - d. Delete contact information from outgoing calls

STORE

Confirm settings

- a. Confirm forwarding phone number
- b. Add new phone number to phone book
- c. Store speed-dial memory
- d. Confirm ring melody
- e. Confirm phone settings

FORWARD

- a. Press to activate forwarding function to forward incoming call to an assigned telephone number
- b. Press to deactivate forwarding function if it is previously activated

SET

Press to enter sub-menus for phone settings

CLID

Press to enter received and missed calls selection page

MUTE

- a. Press to mute all microphones
- b. Press to resume conversation from mute

HOLD

- a. Press to put an active call on hold
- b. Press to resume conversation from hold

CONF

- a. Press to join a third party into a conference call
- b. Press to remove the last joined party from conference call

AUTO RD

On-hook, press to dial out the last dialed number 15 times automatically if phone line of second party is occupied

REDIAL/OUT

- a. On-hook, press to show last dialed number on the LCD
- b. Off-hook, press to dial last dialed number automatically

SPEAKER/HEADSET

- a. Press to activate speakerphone or headset
- b. Press to deactivate speakerphone or headset

VOLUME up and down

- a. On handset, press to adjust handset volume
- b. On speakerphone, press to adjust speaker volume
- c. On headset, press to adjust headset volume

TRANSFER

Press to transfer a call

OPERATION

Make a call

- a. Lift up the handset and then dial the number, speak through handset.
- b. Press SPEAKER/HEADSET (Headset plugged in and handset down) and then dial the number, speak through headset
- c. Press SPEAKER/HEADSET (Headset not plugged in) and then dial the number, speak through speakerphone.
- d. On-hook, dial the number, and then pick up the handset or press SPEAKER/HEADSET (Headset plugged in and handset down) or Press SPEAKER/ HEADSET (Headset not plugged in)

Receive a call

- a. Lift up handset to conduct conversation through handset
- b. Press SPEAKER/HEADSET (Headset plugged in and handset down) to conduct conversation through headset.
- c. Press SPEAKER/HEADSET (Headset not plugged in) to conduct conversation through speakerphone

End a call

- a. On handset: place handset back on-hook
- b. Headset: press SPEAKER/HEADSET or lift up handset and put it back on cradle to terminate a conversation
- c. Speakerphone: press SPEAKER/HEADSET button or lift up handset and put it back on cradle to terminate a conversation

Call transfer

- a. During an active call, press TRF to place the current call on hold, call the third party to which the call is to be transferred, and then hang up the phone.
- b. During an active call, press HOLD to place the current call on hold, call the third party to which the call is to be transferred, press TRF once the call is connected and then hang up the phone

Conference

- a. During a call, either an incoming call or outgoing call, press HOLD or CONF to place the current call on hold, and then dial the number of third party that is to be joined into the conference, press CONF again once the call is connected to initiate a 3 way conference.
- b. Press CONF or hook switch anytime during a conference call to remove the last joined party from the conference.

Call forwarding

- a. On-hook: press FWD and enter the telephone number to which the calls are to be forwarded, press STORE (PROGRAM) to confirm
- b. Press FWD to deactivate forwarding function if it is previously activated

Redial

- a. On-hook, press REDIAL to view the last dialed number, and then review other last 19 dialed numbers by using navigation keys; select a desired number and lift up the handset or press SPEAKER/HEADSET to dial out automatically.
- b. Off hook, press REDIAL to dial out the last dialed number automatically.

Switch Talk Mode from Handset to Speakerphone or Headset

On handset, press SPEAKER/HEADSET and speak through speakerphone or headset

Switch Talk Mode from Speakerphone or Headset to Handset

On speaker or headset, lift up handset and speak through handset

Pre-dial

On-hook, dial the phone number and then lift up the handset or press SPEAKER/HEADSET to dial out automatically.

CALLER ID

Caller ID review

On-hook, press CLID and the LCD displays 1.RECEIVED 2.MISSED; select 1 or 2 by using navigation keys and press ENTER to access 1.received call page or 2.missed call page.

a. Received: Use navigation keys to view 100 most recent received calls in corresponding order, select a number and press ENTER to show the details of the call including name of calling party, phone number, call in date and time.

b. Missed: Use navigation keys to view 100 most recent missed calls in corresponding order, select a number and press ENTER to show the details of the call including name of calling party, phone number, call in date and time.

Call back from caller ID

Select a number from received or missed calls, and then lift up the handset or press SPEAKER/HEADSET to dial out automatically.

Delete caller ID

a. Delete one: Select a number from received or missed calls, press DELETE to bring up sub-menu and select DELETE ONE by using navigation keys, press DELETE again to confirm

b. Delete all: Press DELETE under received or missed call page to bring up sub-menu, select DELETE ALL by using navigation keys and press DELETE again to confirm

PHONE BOOK

On-hook, press PHONE BOOK and the LCD displays 1.ADD 2.SEARCH; select 1 or 2 by using navigation keys and press ENTER to access 1.name input page or 2.view contact page

View phone book

a. SEARCH: in view contact page, use navigation keys to browse between stored contact information

b. SEARCH: in view contact page, enter the initial of the name to bring up a list of names with the same initial and then browse contact information by using navigation keys

Store contact information to phone book

- a. **ADD:** in name input page, key in the name and press ENTER to access number input page, key in number and then press STORE (PROGRAM) to confirm
- b. Select a number from received or missed calls or outgoing (redial) page, press PHONE BOOK, select ADD and press ENTER to access name input page, key in the name and press ENTER to access number input page, press STORE (PROGRAM) to confirm.
- c. Repeat the same approach to add more names and numbers to the phone book, if maximum capacity of 100 is reached, MEMORY FULL will display on LCD

Remove contact information from phone book

- a. **DELETE ONE:** select a number from view contact page, press DELETE to bring up sub-menu and select DELETE ONE by using navigation keys, press DELETE again to confirm
- b. **DELETE ALL:** press DELETE under view contact page to bring up sub-menu, select DELETE ALL by using navigation keys, press DELETE again to confirm

Edit contact information in phone book

Select a contact information to be edited in view contact page, press DELETE to bring up sub-menu and select EDIT, press DELETE to enter name input page and use DELETE button to edit name, press ENTER and then edit number in number input page, press STORE to confirm.

Call from phone book

On-hook, select a desired number in view contact page and lift up the handset or press SPEAKER/HEADSET button (headset plugged in) or press SPEAKER/HEADSET button (headset not plugged in) to dial out automatically

Auto-redial

On-hook, press AUTO RD to dial out the last dialed number repeatedly and activate speakerphone.

Program speed-dial key

Press STORE (PROGRAM) and key in telephone number that is to be set as speed-dial number, assign a memory location on the face plate and press it to confirm.

Speed dialing

- a. On-hook, press desired speed-dial button corresponding to the stored number and then lift up the handset or press SPEAKER/HEADSET to dial out automatically
- b. Off-hook, press desired speed-dial button corresponding to the stored number to dial out automatically

Volume adjustment

- a. On handset, press volume up and down button to adjust handset volume
- b. Speakerphone mode, press volume up and down button to adjust speakerphone volume
- c. Headset mode, press volume up and down button to adjust headset volume

Hold

Mute and resume conversation

When MUTE key is pressed, the transmitter microphone on the handset, speakerphone, and headset will be muted. All microphones will remain muted until the MUTE button is pressed again.

Message storage and retrieval

PHONE SETTING

Date Format

On-hook, press SET to enter submenus, select Phone Set Up and press ENTER, select Date and Time and press ENTER, select Date Format to access date format selection page, select desired date format and press STORE to confirm

- a. DD-MM
- b. MM-DD
- c. DD-MM-YY
- d. MM-DD-YY

Time Format

On-hook, press SET to enter submenus, select Phone Set Up and press ENTER, select Date and Time and press ENTER, select Time Format to access date format selection page, select desired time format and press STORE to confirm

- a. 12HR
- b. 24HR

Ringer Tone

On-hook, press SET to enter submenus, select Phone Set Up and press ENTER, select Ringer and press ENTER, select Ringer Tone to access Ringer Tone selection page, select from 1 to 8 by pressing number key and then press STORE to confirm

Ringer Volume

On-hook, press SET to enter submenus, select Phone Set Up and press ENTER, select Ringer and press ENTER, select Ringer Volume to access Ringer Volume adjustment page, adjust the volume by using volume adjusting keys and then press STORE to confirm.

Handset Volume

On-hook, press SET to enter submenus, select Phone Set Up and press ENTER, select

Ringer and press ENTER, select HS Mic Gain to access Handset Volume adjustment page, adjust the volume by using volume adjusting keys and then press STORE to confirm.

Speaker Volume

On-hook, press SET to enter submenus, select Phone Set Up and press ENTER, select Ringer and press ENTER, select HF Mic Gain to access Speaker Volume adjustment page, adjust the volume by using volume adjusting keys and then press STORE to confirm.

View IP address

On-hook, press SET to enter submenus, select Network Cfg and press ENTER, select Host Config and press ENTER, select IP Config and press ENTER to access IP Address acquiring page, press ENTER again to show IP Address on the LCD

View MAC address

On-hook, press SET to enter submenus, select Network Cfg and press ENTER, select Host Config and press ENTER, select MAC Address and press ENTER to access MAC Address acquiring page where MAC Address is shown on the LCD

View Net Mask address

On-hook, press SET to enter submenus, select Network Cfg and press ENTER, select Host Config and press ENTER, select NetMask Config and press ENTER to access NetMask Address acquiring page, press ENTER again to show NetMask Address on the LCD

View Host name

On-hook, press SET to enter submenus, select Network Cfg and press ENTER, select Host Config and press ENTER, select Host Name and press ENTER to access Host Name acquiring page where Host Name is shown on the LCD

View Domain name

On-hook, press SET to enter submenus, select Network Cfg and press ENTER, select Host Config and press ENTER, select Domain Name and press ENTER to access Domain Name acquiring page where Domain Name is shown on the LCD

View Gate Way address

On-hook, press SET to enter submenus, select Network Cfg and press ENTER, select Router Cfg and press ENTER, select Gateway Addr and press ENTER to access Gateway Address acquiring page where Gateway address is shown on LCD

View DNS Server address

On-hook, press SET to enter submenus, select Network Cfg and press ENTER, select DNS Server and press ENTER to access DNS address acquiring page where DNS Server address is shown on LCD

Safety Instructions

Please read the following instructions carefully to ensure correct use and to prevent unexpected accident and damage caused by incorrect use.

Do not disassemble or modify the IP250 phone or power adaptor if applicable.

The warranty will not cover any defect that occurs due to such mishandling.

Install the unit on a flat, stable surface to ensure safe operation.

Do not install the unit in such a location whereby the unit can be affected by dust or gas.

Do not install the unit in a place subject to direct sunlight or near heat surfaces such as radiators.

Do not install the unit in a humid location to avoid damage, overheating and electric shock.

Make sure to turn off the power supply switch before disconnecting the mains lead.

Disconnecting the mains with the power switch on will cause damage to the unit.

Make sure to only use the power adaptor supplied with the IP250.

Disconnect the power cable when the unit is not in use for an extended period of time.

Guarantee

Your IP250 telephone is designed and manufactured to exacting quality standards. This enables ATL Telecom Limited to offer a 1 year guarantee from the date of purchase. This guarantee protects against faulty material or workmanship, applies to the UK only and is not transferable.

The terms and conditions under which the guarantee will be valid are as set out below.

1. Misuse or any modification carried out to the telephone, or operation other than in accordance with the instructions supplied, will invalidate the guarantee.
2. Damage arising from incorrect installation, accidental damage or consequential loss, are not covered under the guarantee.
3. In the event of a fault developing during the period of the guarantee, the complete telephone should be returned to your supplier, adequately and safely packed, together with proof of date of purchase.
4. The liability of ATL Telecom Limited will be limited to the cost of repair or complete replacement of the same defective instrument, at the discretion of the company. In the event that the same item is not available, a suitable alternate will be offered.
5. The terms of this guarantee do not affect your statutory rights.

Manufacturer's Declaration

ATL Telecom Limited declares that this product is in conformity with the essential requirements of the 'R&TTE directive 1999/5/EC'

Note: A copy of the Declaration of Conformity is available upon request from ATL Telecom Limited.

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