

AudioCodes

300HD Series of High Definition IP Phones

HD VoIP

310HD IP Phone

# Administrator's Manual

310HD IP Phone

Version 1.4.0



 **AudioCodes**  
Connecting Networks

 **HD** VoIP  
*Sounds Better*



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## Notice

This manual provides the system administrator a description for setting up and configuring the 310HD IP Phone.

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Date Published: July-27-2010

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## Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used, and only Industry standard terms are used throughout this manual. The symbol 0x indicates hexadecimal notation.

## Related Documentation

Document Name
300HD IP Phone Release Notes
310HD IP Phone Quick Guide
310HD IP Phone User's Manual

## Regulatory Information

The Compliancy and Regulatory Information can be viewed at <http://www.audiocodes.com/library>.

# 1 Introduction

This manual is intended for the system administrator who is responsible for setting up and configuring the 310HD IP phone.

AudioCodes 310HD IP Phone is based on AudioCodes proprietary High Definition (HD) voice technology, providing clarity and a rich audio experience in Voice-over-IP (VoIP) calls. The 310HD IP Phone is a fully-featured telephone that provides voice communication over an IP network, allowing you to place and receive phone calls, put calls on hold, transfer calls, make conference calls, and so on.

The 310HD IP Phone offers a wide variety of management and configuration tools:

- **Phone's LCD display user interface** - easy-to-use, menu-driven display screen, providing basic phone configuration and status capabilities
- **Embedded Web server** - provides a user-friendly Web interface that runs on a Web browser (Microsoft® Internet Explorer is the recommended browser).
- **Configuration file** - text-based file (created using any plain text editor such as Microsoft's Notepad) containing configuration parameters and which is loaded to the phone using the Web interface or a TFTP, FTP, HTTP or HTTPS server.
- **TR-069** for remote configuration and management
- **CLI over Telnet**

For a detailed description on hardware installation and for operating the phone's call features, refer to the *User's Manual*.

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**Reader's Notes**

## 2 LCD-Based Management

The IP phone provides a liquid crystal display (LCD) based screen, offering an intuitive, menu-driven interface for configuring the phone.

This chapter provides step-by-step procedures for configuring the following administration settings using the phone's LCD interface:

- Network settings (refer to Section 2.2 on page 12)
- SIP accounts (refer to Section 2.3 on page 13)
- Restoring factory defaults (refer to Section 2.4 on page 14)
- Restarting the phone (refer to Section 2.5 on page 15)

These administration tasks are performed in the phone's **Administration** menu (refer to Section 2.1).

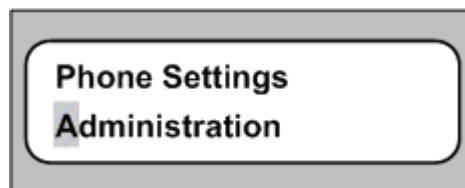
This chapter also describes how to view the phone's current network settings (refer to Section 2.6 on page 15) and firmware version (refer to Section 2.7 on page 16).



### 2.1 Accessing the Administration Submenus

The submenus of the **Administration** menu are password protected. When attempting to access a submenu, the phone prompts you to enter the access password.

➤ **To access the Administration menu:**

1. In idle display, press the MENU key; the menu list is displayed:



2. Select the required **Administration** submenu (using the  and  Navigation keys), and then press the ENTER key; the Administration submenus are displayed.
3. Select a submenu; you are prompted for a password:



4. Enter the password, and then press the ENTER key.



**Notes:**

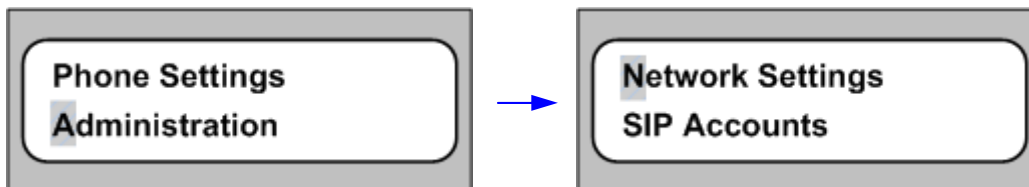
- The default password is "1234". To change the login password, use the phone's Web interface (refer to Section 3.11 on page 42).
- Once you have entered the password, the access session is applied to all the submenus.

## 2.2 Configuring Network Settings

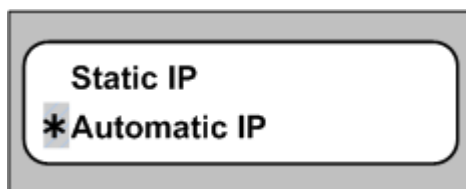
The phone's LAN connection interface can be manually defined (static IP address) or automatically configured using a DHCP server from where the LAN IP address is obtained.



➤ **To configure the phone's LAN connection type:**

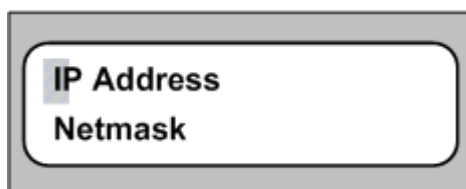
1. Access the Network Settings screen (MENU key > **Administration** menu > **Network Settings**).



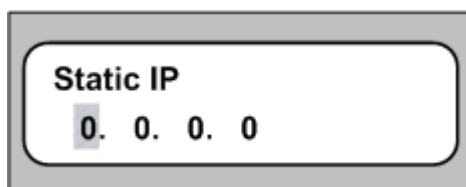
The following screen appears:




2. Choose one of the following IP addressing schemes (using the  and  Navigation keys):
  - Static IP
  - Automatic IP
3. Press the ENTER key. If you selected **Static IP**, then continue with Step 4; otherwise, skip to Step 5.
4. If you selected the **Static IP** option, the following screen appears:

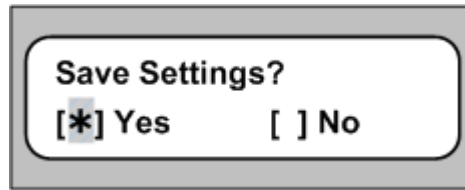




- a. Select the required network parameter (**IP Address**, **Netmask**, **Gateway**, **Primary DNS**, and/or **Secondary DNS**).
- b. Press the ENTER key; the screen prompts you to enter the IP address of the selected network parameter (e.g., static IP address below):



- c. Press the ENTER key; the IP address is applied
- d. Repeat steps 4.a through 4.c for each required network parameter.

5. Press the left  Navigation key until you are prompted to save settings, as shown below:



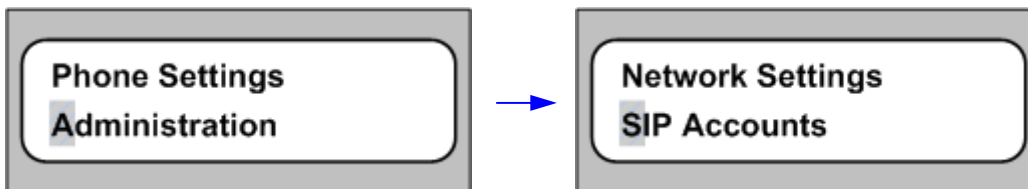
6. Select the **Yes** option (using the  and  Navigation keys), and then press the ENTER key; the network settings are saved.

## 2.3 Configuring SIP Accounts

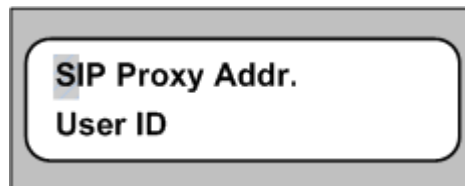
The SIP Accounts submenu allows you to configure parameters related to the phone's SIP account.



- **To configure the phone's SIP account:**

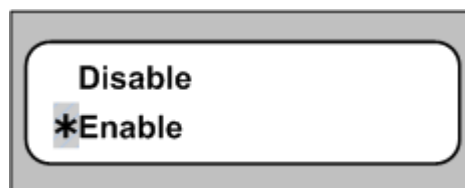
1. Access the SIP Accounts screen (MENU key > **Administration** menu > **SIP Accounts**).





The following screen appears:

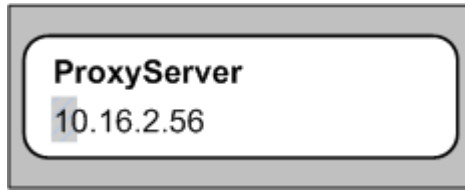




2. Define the SIP proxy IP address:
- a. Select the **SIP Proxy Addr.** option (using the  and  Navigation keys), and then press the ENTER key: the following screen appears:





- b. Select one of the following options (using the  and  Navigation keys):
  - ◆ **Disable:** disables SIP proxy
  - ◆ **Enable:** enables SIP proxy
 An asterisk appears alongside the selected option.

- c. Press the ENTER key; you are prompted to enter the IP address:



- d. Enter the IP address, and then press the ENTER key; you are returned to the SIP Accounts options.
3. Define the remaining SIP proxy parameters, by selecting (using the  and  Navigation keys) the required parameter:
    - **User ID**
    - **Display Name:** caller ID
    - **Auth Name:** authentication name
    - **Auth Password:** authentication password

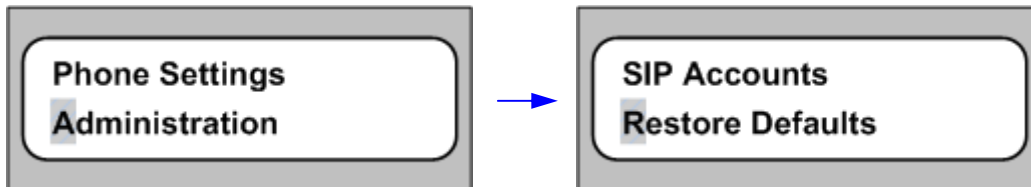
For each parameter, once you have entered the required value, press the ENTER key.
  4. Select the **Save** option (using the  and  Navigation keys), and then press the ENTER key; your Sip account settings are saved.

## 2.4 Restoring Defaults

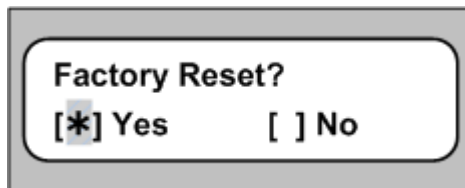
The procedure below describes how to restore the phone to factory defaults.

- **To restore the phone to default settings:**

1. Select the **Restore Defaults** submenu (**MENU** key > **Administration** menu > **Restore Defaults**):



A confirmation message appears, prompting you to confirm factory reset:



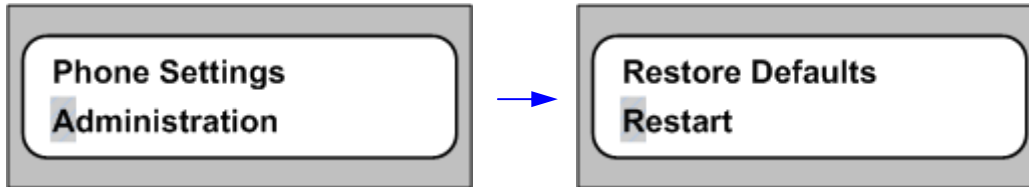
2. Select the **Yes** option (using the  and  Navigation keys), and then press the ENTER key; the phone is restored to factory defaults.

## 2.5 Restarting Phone

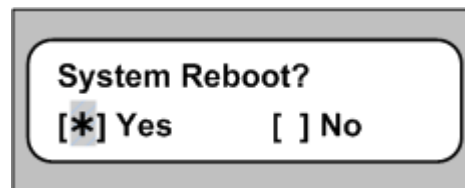
The procedure below describes how to restart the phone.



➤ **To restart the phone:**

1. Select the **Restart** submenu (**MENU** key > **Administration** menu > **Restart**):



A confirmation message appears, prompting you to confirm phone reset:



2. Select the **Yes** option (using the  and  Navigation keys), and then press the ENTER key; the phone resets.

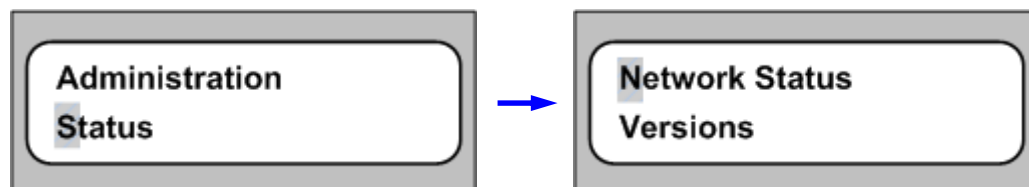
## 2.6 Viewing Network Status

You can view the phone's network parameters. These include the following:

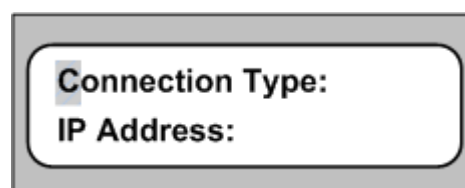
- IP address
- Subnet mask
- Default gateway
- Primary and secondary DNS addresses
- MAC address

➤ **To view the phone's network status:**

1. From the idle screen, press the MENU key, select the **Status** menu, and then the **Network Status** submenu.



The following screen appears:



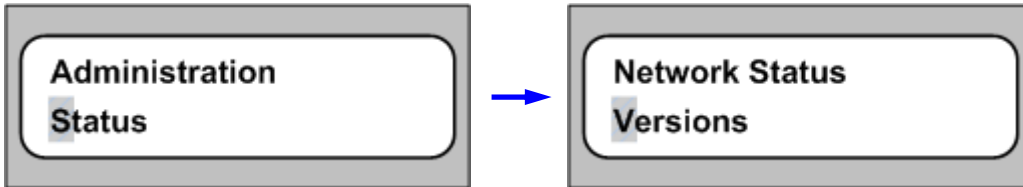
2. Scroll through the list (using the  and  Navigation keys) to view the required network parameters.

## 2.7 Viewing Phone Model and Firmware Version

You can view the phone's model number and software (firmware) version.

➤ **To view the phone's model and firmware version:**

1. From the idle screen, press the MENU key, select the **Status** menu, and then the **Network Versions** submenu.



The following screen appears:



2. Press the ENTER key; the phone's model and firmware version is displayed.



## 3 Web-based Management

This chapter describes the phone's embedded Web server interface for configuring the phone.



**Note:** Where Web parameters can also be configured using the Configuration file, a table is provided listing the corresponding Configuration file parameter. In such cases, the description of the parameter (Web and Configuration file parameters) appears only in the Chapter on Configuration file parameters (see Chapter 4).

### 3.1 Accessing the Web Interface

You can use any standard Web browser (such as Microsoft Internet Explorer) to access the phone's Web interface. The IP address used for accessing the Web interface is the phone's IP address, received from a DHCP server or manually configured (static IP address).

➤ **To access the phone's Web interface:**

1. Connect the LAN port of your phone to the IP network (using the Cable or ADSL modem from your Internet Service Provider).
2. Determine the phone's IP address obtained from the DHCP server, using the phone's LCD screen as described in Section 2.1 on page 11 (in the 'IP Address' field).
3. Open a Web browser, and then in the URL address field, enter the phone's IP address (for example, `http://192.168.1.2:8000` or `https://192.168.1.2`), as displayed below:



The Web login window appears:

**Figure 3-1: Web Login Window**



4. Alternatively, if your DHCP and DNS servers are synchronized you can access the phone Web browser by using the following method:

**`http://310hd-<MAC Address>.<Domain Name>`**  
 (e.g. `http://310hd-001122334455.corp.YourCompany.com`)

5. Enter the **User name** and **Password**, and then click **OK**.

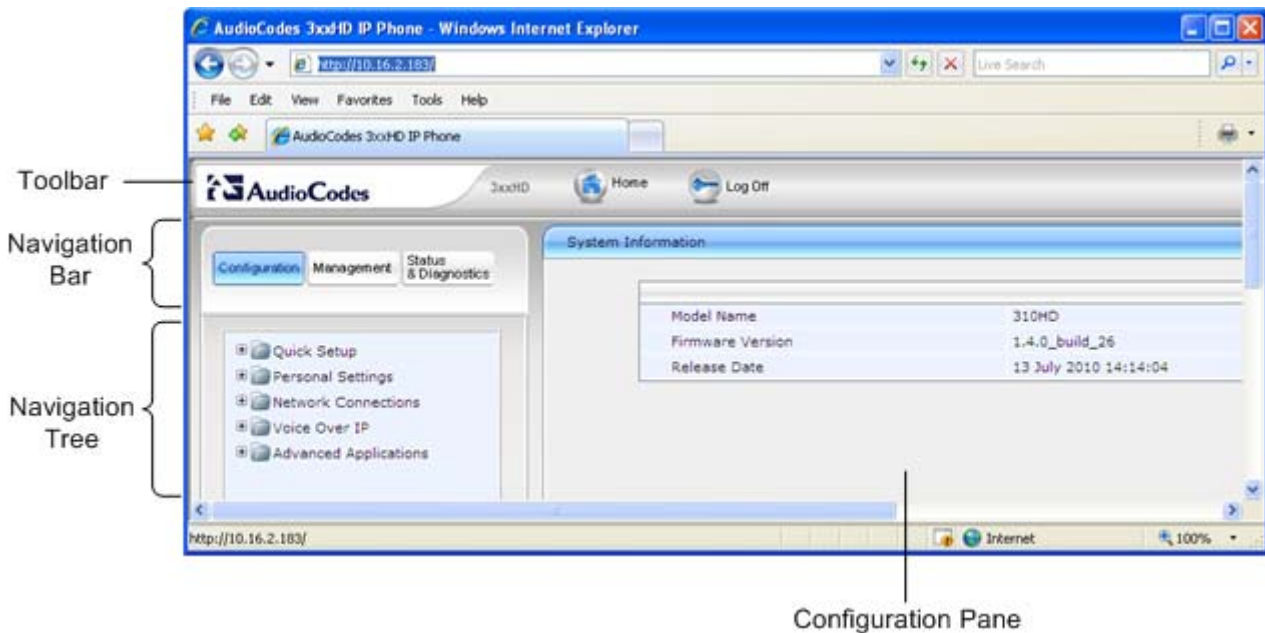


**Note:** The administrator's default login user name and password are "admin" and "1234" respectively. To change the login credentials, refer to Section 3.11 on page 42.



## 3.2 Getting Started with the Web

The areas of the Web interface are shown below:

**Figure 3-2: Web Interface Areas**



The Web interface is composed of the following main areas:

- **Toolbar:** displays AudioCodes logo and provides the following buttons:
  -  **Home:** opens the Home page
  -  **Log off:** closes the Web interface
- **Navigation bar:** provides tabs for accessing the configuration menus:
  - **Configuration:** provides menus for configuring the phone.
  - **Management:** provides menus for various management tasks such as firmware upgrade and changing the login username and password.
  - **Status & Diagnostics:** provides menus for displaying information on the status of the phone, such as call history.
- **Navigation tree:** tree-like, hierarchical structure of menus pertaining to the selected tab on the Navigation bar.
- **Configuration pane:** displays the configuration parameters pertaining to a selected menu in the Navigation tree.

### 3.3 Quick Setup

The Web interface allows you to quickly configure the main parameters required for basic phone functioning. This is provided by the 'Quick Setup' page, as described below.

➤ **To quickly setup your phone:**

1. Access the 'Quick Setup' page (**Configuration** tab > **Quick Setup** menu > **Quick Setup**).

**Figure 3-3: Quick Setup Page**

▼ LAN Setup	
IP Type:	<input type="radio"/> Static IP <input checked="" type="radio"/> Automatic IP (DHCP)
IP Address:	<input type="text" value="0.0.0.0"/>
Subnet Mask:	<input type="text" value="0.0.0.0"/>
Default Gateway Address:	<input type="text" value="0.0.0.0"/>
Primary DNS:	<input type="text" value="0.0.0.0"/>
Secondary DNS:	<input type="text" value="0.0.0.0"/>
▼ SIP Proxy and Registrar	
Use SIP Proxy:	<input type="text" value="Disable"/>
Use SIP Registrar:	<input type="text" value="Disable"/>
▼ Line Settings	
Line Activate:	<input type="text" value="Enable"/>
User ID:	<input type="text" value="0"/>
Authentication User Name:	<input type="text" value="0"/>
Authentication Password:	<input type="text" value="0"/>

2. For a description of the parameters on this page, refer to the following:
  - Parameters under the **LAN Setup** group, refer to Section 3.7 on page 24.
  - Parameters under the **SIP Proxy and Registrar** group, refer to Section 3.8.1 on page 25
  - Parameters under the **Line Settings** group, refer to Section 3.8.5 on page 32

### 3.4 Changing LCD Display Language

You can choose your preferred language for the LCD display.

➤ **To choose a language:**

1. Access the 'Language' page (**Configuration** tab > **Personal Settings** menu > **Language**).

**Figure 3-4: Language Page**

The screenshot shows a web interface for the 'Language' page. At the top, there is a dropdown menu labeled 'Language'. Below it, the 'Phone Display Language' is set to 'English' with a small dropdown arrow next to the text.

2. Select the language according to the parameter in the table below, and then click **Submit**; the phone reboots and changes the LCD display language accordingly.

**Table 3-1: Language Parameter Description**

Web Parameter	Configuration File Parameter	Description
Phone Display Language	personal_settings/language	Refer to Section 4.2.8

### 3.5 Defining Speed Dials

You can add up to 10 speed dials to your phone.



**Note:** The phone's speed dials can also be defined in a simple text-based editor, placed on a server (e.g., HTTP or FTP/TFTP), and then uploaded to the phone using a configuration file. For a detailed description, refer to Section 4.3.

➤ **To add a speed dial:**

1. Access the 'Speed Dial' page (**Configuration** tab > **Personal Settings** menu > **Speed Dial**).

**Figure 3-5: Speed Dial Page**

Button	Type	Number	Delete
1	Speed Dial	4001	<input type="checkbox"/>
2	Speed Dial	4006	<input type="checkbox"/>
3	Speed Dial	4010	<input type="checkbox"/>
4	Speed Dial	90527766554	<input type="checkbox"/>
5	Speed Dial		<input type="checkbox"/>
6	Speed Dial		<input type="checkbox"/>
7	Speed Dial		<input type="checkbox"/>
8	Speed Dial		<input type="checkbox"/>
9	Speed Dial		<input type="checkbox"/>
10	Speed Dial		<input type="checkbox"/>

2. In the 'Number' field corresponding to the phone's Speed Dial key (in the 'Button' column), enter the speed dial number to which you want to assign the Speed Dial key.
3. Click **Submit**.

➤ **To delete speed dials:**

- Deleting selected speed dial entries: select the 'Delete' check boxes corresponding to the speed dials that you want to delete, and then click **Submit**.
- All speed dials: Click **Delete All**, and then at the prompt, click **OK**.
- To clear (unselect) all your selected 'Delete' check boxes, and then click **Reset**.

## 3.6 Regional Call Progress Tones and Ring Tones

You can suite your phone’s call progress tones to the location in which the phone is located. In addition, instead of using the default ring tone, you can upload a different ring tone file and use that ring tone for indicating incoming calls. This ring tone can only be selected using the phones LCD interface.

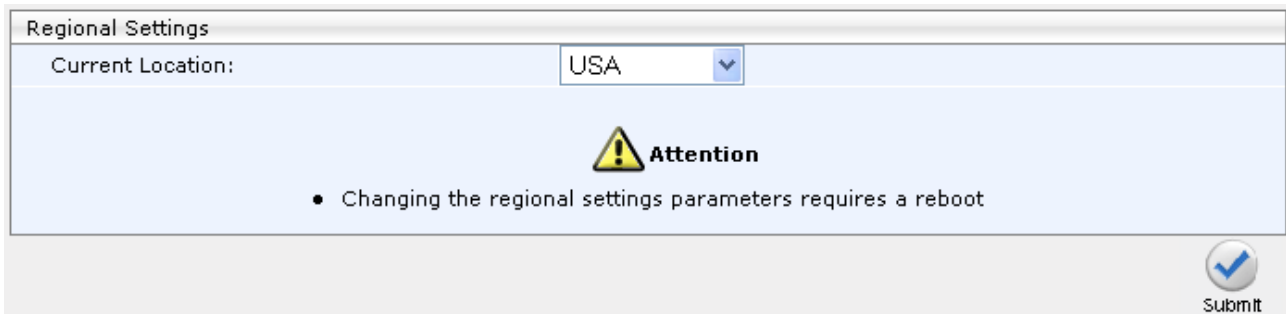
### 3.6.1 Selecting the Region for Call Progress Tones

Follow the procedure below for selecting the region in which your phone is located. This is important for suiting your phone for the call progress tones of the country in which the phone is located.

➤ **To select the geographical location of your phone:**

1. Access the ‘Tones’ page (**Configuration** tab > **Personal Settings** menu > **Tones**).

**Figure 3-6: Tones Page**



2. From the ‘Current Location’ drop-down list, select the country in which your phone is located.
3. Click **Submit**.

**Table 3-2: Region CPT Parameter Description**

Web Parameter	Configuration File Parameter	Description
Current Location	voip/regional_settings/selected_country	Refer to Section 4.2.6 (includes additional configuration file parameters for CPT)

### 3.6.2 Uploading Ring Tones

Follow the procedure below for uploading new ring tones.



**Notes:**

- The ring tone file must be in WAV file format (A/Mu-Law, 8-kHz audio sample rate and 8-bit audio sample size or PCM 16-kHz audio sample rate and 16-bit audio sample size, Intel PCM encoding).
- If you want the phone to use an uploaded ring tone, you need to use the phone's LCD screen to select it (refer to the phone's *User's Manual*).

➤ **To upload a ring tone:**

1. Access the 'Tones' page (**Configuration** tab > **Personal Settings** menu > **Tones**).

**Figure 3-7: Tones Page**

ID	Ringling Tone Name	Delete
1	Become insane 1	<input type="checkbox"/>
2	Become insane 2	<input type="checkbox"/>
3	Skittle	<input type="checkbox"/>
4	IPP_ring1	<input type="checkbox"/>
5	Soda pop	<input type="checkbox"/>
6	Rihanna	<input type="checkbox"/>
7	Colosseum	<input type="checkbox"/>
8	Beautiful world	<input type="checkbox"/>

2. For uploading a ring tone, perform the following:
  - a. In the 'Ringling Tone Name' field, enter the name of the ring tone file that you want to upload.
    - Click the **Browse** button, navigate to the folder in which the ring tone file is located, select the file, and then click Open; the file name and path is displayed in the 'File Location' field.
    - Click **Submit**; the file is loaded to the phone and displayed in the Ring Tone list.

➤ **To delete Ring Tones:**

- Deleting selected ring tones: select the 'Delete' check boxes corresponding to the ring tone that you want to delete, and then click **Submit**.

**Table 3-3: Ring Tone Parameter Description**

Web Parameter	Configuration File Parameter	Description
Ringling Tone Name File Location	provisioning/ring_tone_uri	Refer to Section 4.2.3

### 3.7 Configuring LAN Interface

The phone's LAN configuration includes defining the method for obtaining an IP address as well as VLAN settings. The phone's IP address can be *static* whereby the IP address is manually entered, or *automatic* whereby the IP address is acquired from a DHCP server.

➤ **To define the phone's LAN settings:**

1. Access the 'LAN Settings' page (**Configuration** tab > **Network Connections** menu > **LAN Settings**).

**Figure 3-8: LAN Settings Page**

The screenshot shows the LAN Settings page with the following fields:

- IP Type:** Radio buttons for Static IP and Automatic IP (DHCP). Automatic IP (DHCP) is selected.
- IP Address:** Text input field containing 0.0.0.0
- Subnet Mask:** Text input field containing 0.0.0.0
- Default Gateway Address:** Text input field containing 0.0.0.0
- Primary DNS:** Text input field containing 0.0.0.0
- Secondary DNS:** Text input field containing 0.0.0.0
- MAC Address:** Text input field containing 00:90:8F:1E:D2:F0
- VLAN Settings:**
  - VLAN Discovery Mode:** Dropdown menu set to Automatic Configuration of VLAN (CDP)

2. Configure the LAN settings according to the parameters in the table below, and then click **Submit**.

**Table 3-4: LAN Parameters Description**

Web Parameter	Configuration File Parameter	Description
IP Type	network/lan_type	Refer to Section <a href="#">4.2.2</a>
IP Address	network/lan/fixed_ip/ip_address	
Subnet Mask	network/lan/fixed_ip/netmask	
Default Gateway Address	network/lan/fixed_ip/gateway	
Primary DNS	network/lan/fixed_ip/primary_dns	
Secondary DNS	network/lan/fixed_ip/secondary_dns	
<b>VLAN Settings</b>		
VLAN Discovery Mode	network/lan/vlan/mode	Refer to Section <a href="#">4.2.2</a>
VLAN ID	network/lan/vlan/id	
VLAN Priority	network/lan/vlan/priority	

## 3.8 VoIP Settings

The **Voice Over IP** menu allows you to configure the following VoIP settings:

- Signaling protocol (refer to Section 3.8.1 on page 25)
- Dialing (refer to Section 3.8.2 on page 28)
- Media streaming (refer to Section 3.8.3 on page 30)
- Voice (refer to Section 3.8.4 on page 30)
- Line (refer to Section 3.8.5 on page 32)
- Services (refer to Section 3.8.6 on page 33)
- Volume (refer to Section 3.8.7 on page 35)

### 3.8.1 Configuring Signaling Protocol

The 'Signaling Protocol' page allows you to define various SIP signaling parameters.

➤ **To define the phone's SIP settings:**

1. Access the 'Signaling Protocol' page (**Configuration** tab > **Voice Over IP** menu > **Signaling Protocols**).

**Figure 3-9: Signaling Protocol Page**

▼ SIP General	
SIP Transport Protocol:	UDP ▼
SIP Local Port:	5060
Gateway Name:	<input type="text"/>
PRACK Mode:	Enable ▼
Enable RPORT:	Enable ▼
Include PTIME in SDP:	Enable ▼
Enable Keep Alive using OPTIONS:	Disable ▼
Connect Media on 180 Response:	Disable ▼
Block Caller ID on Outgoing Calls:	Disable ▼
Incoming Anonymous Call Blocking:	Disable ▼
▼ SIP Proxy and Registrar	
Use SIP Proxy:	Disable ▼
Use SIP Registrar:	Disable ▼
Use SIP Outbound Proxy:	Disable ▼
Use Redundant Proxy:	Disable ▼
▼ SIP Timers	
Retransmission Timer T1:	500
Retransmission Timer T2:	4000
Retransmission Timer T4:	5000
INVITE Timer:	32000
Session-Expires:	1800
Min-SE:	90
▼ Quality of Service Parameters	
Type of Service (ToS):	0x60 Hex

2. Configure SIP according to the parameters in the table below, and then click **Submit**.

**Table 3-5: SIP Parameters Description**

Web Parameter	Configuration File Parameter	Description
<b>SIP General Parameters</b>		
SIP Transport Protocol	voip/signalling/sip/transport_protocol	Refer to Section <a href="#">4.2.5.6</a>
SIP Local Port	voip/signalling/sip/port	
Gateway Name	voip/signalling/sip/proxy_gateway	
PRACK Mode	voip/signalling/sip/prack/enabled	
Enable RPORT	voip/signalling/sip/rport/enabled	
Include PTIME in SDP	voip/signalling/sip/sdp_include_ptime	
Enable Keep Alive using OPTIONS	voip/signalling/sip/keepalive_options/enabled	
Keep Alive Period	voip/signalling/sip/keepalive_options/timout	
Connect Media on 180 Response	voip/signalling/sip/connect_media_on_180	
Block Caller ID on Outgoing Calls	voip/signalling/sip/block_callerid_on_outgoing_calls	
Incoming Anonymous Call Blocking	voip/signalling/sip/anonymous_calls_blocking	
<b>SIP Proxy and Registrar Parameters</b>		
Use SIP Proxy	voip/signalling/sip/use_proxy	Refer to Section <a href="#">4.2.5.6</a>
Proxy IP Address or Host Name	voip/signalling/sip/proxy_address	
Proxy Port	voip/signalling/sip/proxy_port	
Maximum Number of Authentication Retries	voip/signalling/sip/proxy_timeout	
Use SIP Proxy IP and Port for Registration	voip/signalling/sip/use_proxy_ip_port_for_registrar	
Use SIP Registrar	voip/signalling/sip/sip_registrar/enabled	
Registrar IP Address or Host Name	voip/signalling/sip/sip_registrar/addr	
Registrar Port	voip/signalling/sip/sip_registrar/port	
Registration Expires	voip/signalling/sip/proxy_timeout	
Use SIP Outbound Proxy	voip/signalling/sip/sip_outbound_proxy/enabled	
Outbound Proxy IP Address or Host Name	voip/signalling/sip/sip_outbound_proxy/addr	
Outbound Proxy Port	voip/signalling/sip/sip_outbound_proxy/port	
Use Redundant Proxy	voip/signalling/sip/redundant_proxy/enabled	
Redundant Proxy Address	voip/signalling/sip/redundant_proxy/address	

Web Parameter	Configuration File Parameter	Description
Redundant Proxy Port	voip/signalling/sip/redundant_proxy/port	
Redundant Proxy Keep Alive Period	voip/signalling/sip/redundant_proxy/keepalive_period	
Switch back to Primary SIP proxy when available	voip/signalling/sip/redundant_proxy/symmetric_mode	
<b>SIP Timers</b>		
Retransmission Timer T1	voip/signalling/sip/sip_t1	Refer to Section <a href="#">4.2.5.6</a>
Retransmission Timer T2	voip/signalling/sip/sip_t2	
Retransmission Timer T4	voip/signalling/sip/sip_t4	
INVITE Timer	voip/signalling/sip/sip_invite_timer	
Session-Expires	voip/signalling/sip/session_timer	
Min-SE	voip/signalling/sip/min_session_interval	
<b>SIP QoS</b>		
Type of Service (ToS)	voip/signalling/sip/tos	Refer to Section <a href="#">4.2.5.6</a>

### 3.8.2 Configuring Dialing

The dialing parameters can be configured in the 'Dialing' page, as described below.

➤ **To define the dialing parameters:**

1. Access the 'Dialing' page (**Configuration** tab > **Voice Over IP** menu > **Dialing**).

**Figure 3-10: Dialing Page**

▼Dialing Parameters		
Dialing Timeout:	<input type="text" value="5"/>	Seconds
Phone Number Length:	<input type="text" value="19"/>	Digits
Enable Dialing Complete Key:	<input type="text" value="Enable"/>	
Dialing Complete Key:	<input type="text" value="#"/>	
Dial Tone Timeout:	<input type="text" value="30"/>	Seconds
Reorder Tone Timeout:	<input type="text" value="40"/>	Seconds
No Answer Call Timeout:	<input type="text" value="60"/>	Seconds
Howler Tone Timeout:	<input type="text" value="120"/>	Seconds
Secondary Dial Tone:	<input type="text" value="Enable"/>	
Secondary Dial Tone Key:	<input type="text" value="9"/>	
DTMF Transport Mode:	<input type="text" value="RFC 2833"/>	
Digit Map:	<input type="text"/>	
Dial Plan:	<input type="text"/>	
▼Automatic Dialing		
Activate:	<input type="text" value="Disable"/>	

2. Configure dialing options according to the parameters in the table below, and then click **Submit**.

**Table 3-6: Dialing Parameters Description**

Web Parameter	Configuration File Parameter	Description
Dialing Timeout	voip/dialing/timeout	Refer to Section <a href="#">4.2.5.5</a>
Phone Number Length	voip/dialing/phone_number_max_size	
Enable Dialing Complete Key	voip/dialing/dial_complete_key/enabled	
Dialing Complete Key	voip/dialing/dial_complete_key/key	
Dial Tone Timeout	voip/dialing/dialtone_timeout	
Reorder Tone Timeout	voip/dialing/warning_tone_timeout	
No Answer Call Timeout	voip/dialing/unanswered_call_timeout	
Howler Tone Timeout	voip/dialing/offhook_tone_timeout	
Secondary Dial Tone	voip/dialing/secondary_dial_tone/enabled	
Secondary Dial Tone Key	voip/dialing/secondary_dial_tone/key_sequence	

Web Parameter	Configuration File Parameter	Description
DTMF Transport Mode	voip/media/out_of_band_dtmf	
Digit Map	voip/signalling/sip/digit_map	
Dial Plan	voip/signalling/sip /number_rules	
<b>Automatic Dialing</b>		
Activate	voip/dialing/auto_dialing/enabled	Refer to Section <a href="#">4.2.5.5</a>
Timeout	voip/dialing/auto_dialing/timeout	
Destination Phone Number	voip/dialing/auto_dialing/destination	

### 3.8.3 Configuring Media Streaming

The media streaming parameters are configured in the 'Media Streaming' page, as described below.

➤ **To define the media streaming parameters:**

1. Access the 'Media Streaming' page (**Configuration** tab > **Voice Over IP** menu > **Media Streaming**).

**Figure 3-11: Media Streaming Page**

▼Media Streaming Parameters		
RTP Port Range - Contiguous Series of 4 Ports Starting From:	<input type="text" value="4000"/>	
DTMF Relay RFC 2833 Payload Type:	<input type="text" value="101"/>	
▼Quality of Service Parameters		
Type of Service (ToS):	<input type="text" value="0xb8"/> Hex	
▼Codecs		
Codec Priority	Codec Type	Packetization Time (milliseconds)
1st Codec	<input type="text" value="G.722"/>	<input type="text" value="20"/>
2nd Codec	<input type="text" value="G.711, 64 Kbps, μ-Law"/>	<input type="text" value="20"/>
3rd Codec	<input type="text" value="G.711, 64 Kbps, A-Law"/>	<input type="text" value="20"/>
4th Codec	<input type="text" value="G.729, 8 Kbps"/>	<input type="text" value="20"/>
5th Codec	<input type="text" value="G.723, 5.3/6.3 Kbps"/>	<input type="text" value="30"/>
▼G.723 Bitrate		
G.723 Bitrate:	<input type="text" value="High"/>	

2. Configure media streaming according to the parameters in the table below, and then click **Submit**.

**Table 3-7: Media Streaming Parameters Description**

Web Parameter	Configuration File Parameter	Description
RTP Port Range	voip/media/media_port	Refer to Section <a href="#">4.2.5.4</a>
DTMF Relay RFC 2833 Payload Type	voip/media/dtmf_payload	
<b>Quality of Service (QoS)</b>		
Type of Service (ToS)	voip/media/media_tos	Refer to Section <a href="#">4.2.5.4</a>
<b>Codecs</b>		
Codec Type	voip/codec/codec_info /%d/name	Refer to Section <a href="#">4.2.5.3</a>
Packetization Time	voip/codec/codec_info /%d/ptime	
<b>G.723 Bitrate</b>		
G.723 Bitrate	voip/codec/g723_bitrate	Refer to Section <a href="#">4.2.5.3</a>

### 3.8.4 Configuring Voice

The voice parameters are configured in the 'Voice' page, as described below.

➤ **To define the voice parameters:**

1. Access the 'Voice' page (**Configuration** tab > **Voice Over IP** menu > **Voice**).

**Figure 3-12: Voice Page**

▼Gain Control	
Enable Automatic Gain Control:	Disable ▾
▼Jitter Buffer	
Minimum Delay (10 to 150 milliseconds):	35 msec
Optimization Factor (1 to 13):	07 ▾
▼Silence Compression	
Enable Silence Compression:	Disable ▾

2. Configure voice options according to the parameters in the table below, and then click **Submit**.

**Table 3-8: Voice Parameters Description**

Web Parameter	Configuration File Parameter	Description
<b>Gain Control</b>		
Enable Automatic Gain Control	<b>voip/audio/gain/automatic_gain_control/enabled</b>	Refer to Section <a href="#">4.2.5.2</a>
Automatic Gain Control Direction	<b>voip/audio/gain/automatic_gain_control/direction</b>	
Target Energy	<b>voip/audio/gain/automatic_gain_control/target_energy</b>	
<b>Jitter Buffer</b>		
Minimum Delay	<b>voip/audio/jitter_buffer/min_delay</b>	Refer to Section <a href="#">4.2.5.7</a>
Optimization Factor	<b>voip/audio/jitter_buffer/optimization_factor</b>	
<b>Silence Compression</b>		
Enable Silence Compression	<b>voip/audio/silence_compression/enabled</b>	Refer to Section <a href="#">4.2.5.7</a>

### 3.8.5 Configuring Line

Before you can make a call on your phone, you must configure a phone line.

➤ **To define the lines:**

1. Access the 'Line Settings' page (**Configuration** tab > **Voice Over IP** menu > **Line Settings**).

**Figure 3-13: Voice Page**

▼Line Settings	
Line Activate:	Enable ▾
User ID:	0
Display Name:	310HD
Authentication User Name:	0
Authentication Password:	0

2. Configure the line according to the parameters in the table below, and then click **Submit**.

**Table 3-9: Line Parameters Description**

Web Parameter	Configuration File Parameter	Description
Line Activate	<b>voip/line/%d/enabled</b>	Refer to Section <a href="#">4.2.5.1</a>
User ID	<b>voip/line/%d/id</b>	
Display Name	<b>voip/line/%d/description</b>	
Authentication User Name	<b>voip/line/%d/auth_name</b>	
Authentication Password	<b>voip/line/%d/auth_password</b>	

### 3.8.6 Configuring Services

You can configure various supplementary services supported by your phone such as call waiting, call forwarding, three-way conferencing, and message waiting indication (MWI).



**Note:** The Busy Lamp Field (BLF) feature will be supported in the next applicable release.

➤ **To define services:**

1. Access the 'Services' page (**Configuration** tab > **Voice Over IP** menu > **Services**).

**Figure 3-14: Services Page**

<b>▼Application Server</b>	
Type:	Generic ▾
<b>▼Call Waiting</b>	
Activate:	Enable ▾
Call Waiting SIP Reply:	Queued ▾
<b>▼Call Forward</b>	
Activate:	Enable ▾
Call Forward Type:	No Reply ▾
Forward on No Reply Timeout:	6 <input type="text"/> Seconds
<b>▼DND (Do Not Disturb)</b>	
Activate:	Enable ▾
<b>▼Message Waiting Indication (MWI)</b>	
Voice Mail Number:	<input type="text"/>
Activate:	Enable ▾
Subscribe To MWI:	Disable ▾
<b>▼BLF Support</b>	
Activate:	Disable ▾
<b>▼General Parameters</b>	
Stutter Tone Duration:	2500 <input type="text"/> msec
Out of Service Behavior:	Reorder Tone ▾
Automatic Disconnect:	Enable ▾

2. Configure the services according to the parameters in the table below, and then click **Submit**.

Table 3-10: Supplementary Services Parameters Description

Web Parameter	Configuration File Parameter	Description
Application Server Type	voip/services/application_server_type	Refer to Section <a href="#">4.2.5.8</a>
<b>Call Waiting</b>		
Activate	voip/services/call_waiting/enabled	Refer to Section <a href="#">4.2.5.8</a>
Call Waiting SIP Reply	voip/services/call_waiting/sip_reply	
<b>Call Forward</b>		
Activate	voip/services/call_forward/line/0/enabled	Refer to Section <a href="#">4.2.5.8</a>
Call Forward Type	voip/services/call_forward/line/0/type	
Forward on No Reply Timeout	voip/services/call_forward/line/0/timeout	
<b>DND (Do Not Disturb)</b>		
Activate	voip/services/do_not_disturb/enabled	Refer to Section <a href="#">4.2.5.8</a>
<b>Message Waiting Indication</b>		
Voice Mail Number	voip/services/msg_waiting_ind/voice_mail_number	Refer to Section <a href="#">4.2.5.8</a>
Activate	voip/services/msg_waiting_ind/enabled	
Subscribe To MWI	voip/services/msg_waiting_ind/subscribe	
MWI Server IP Address or Host Name	voip/services/msg_waiting_ind/subscribe_address	
MWI Serve Port	voip/services/msg_waiting_ind/subscribe_port	
MWI Subscribe Expiry Time	voip/services/msg_waiting_ind/expiration_timeout	
<b>General Parameters</b>		
Stutter Tone Duration	voip/services/msg_waiting/stutter_tone_duration	Refer to Section <a href="#">4.2.5.8</a>
Out of Service Behavior	voip/services/out_of_service_behavior	
Automatic Disconnect	voip/dialing/automatic_disconnect	

### 3.8.7 Configuring Volume

You can configure various volume parameters such as speaker, tones, ringer, analog/digital input and output gain.

➤ **To define volume settings:**

1. Access the 'Services' page (**Configuration** tab > **Voice Over IP** menu > **Volume Settings**).

**Figure 3-15: Volume Settings Page**

Volume Setting	
Additional Speaker Volume:	3dB
Tone Volume:	-10dB
Ringer Volume:	0dB
Hands Free Digital Output:	7dB
Hands Free Digital Input:	0dB
Hands Free Analog Output:	0dB
Hands Free Analog Input:	33.0dB
Handset Digital Output:	0dB
Handset Digital Input:	0dB
Handset Analog Output:	-9.0dB
Handset Analog Input:	19.5dB
Handset Analog Sidetone Gain:	-12dB
Headset Digital Output:	0dB
Headset Digital Input:	0dB
Headset Analog Output:	-12.0dB
Headset Analog Input:	33.0dB
Headset Analog Sidetone Gain:	-12dB

2. Configure the volume settings according to the parameters in the table below, and then click **Submit**.

**Table 3-11: Volume Parameters Description**

Web Parameter	Configuration File Parameter	Description
<b>Additional Speaker Gain</b>	<code>voip/audio/gain/additional_speaker_gain</code>	Refer to Section <a href="#">4.2.5.2</a>
<b>Tone Volume</b>	<code>voip/audio/gain/tone_signal_level</code>	
<b>Ringer Volume</b>	<code>voip/audio/gain/ringer_signal_level</code>	
<b>Hands-free Gain Parameters</b>		
<b>Hands Free Digital Output</b>	<code>voip/audio/gain/handsfree_digital_output_gain</code>	Refer to Section <a href="#">4.2.5.2</a>
<b>Hands Free Digital Input</b>	<code>voip/audio/gain/handsfree_digital_input_gain</code>	
<b>Hands Free Analog Output</b>	<code>voip/audio/gain/handsfree_analog_output_gain</code>	

Web Parameter	Configuration File Parameter	Description
Hands Free Analog Input	voip/audio/gain/handsfree_analog_input_gain	
<b>Handset Gain Parameters</b>		
Handset Digital Output	voip/audio/gain/handset_digital_output_gain	Refer to Section <a href="#">4.2.5.2</a>
Handset Digital Input	voip/audio/gain/handset_digital_input_gain	
Handset Analog Output	voip/audio/gain/handset_analog_output_gain	
Handset Analog Input	voip/audio/gain/handset_analog_input_gain	
Handset Analog Sidetone Gain	voip/audio/gain/handset_analog_sidetone_gain	
<b>Headset Gain Parameters</b>		
Headset Digital Output	voip/audio/gain/headset_digital_output_gain	Refer to Section <a href="#">4.2.5.2</a>
Headset Digital Input	voip/audio/gain/headset_digital_input_gain	
Headset Analog Output	voip/audio/gain/headset_analog_output_gain	
Headset Analog Input	voip/audio/gain/headset_analog_input_gain	
Headset Analog Sidetone Gain	voip/audio/gain/headset_analog_sidetone_gain	

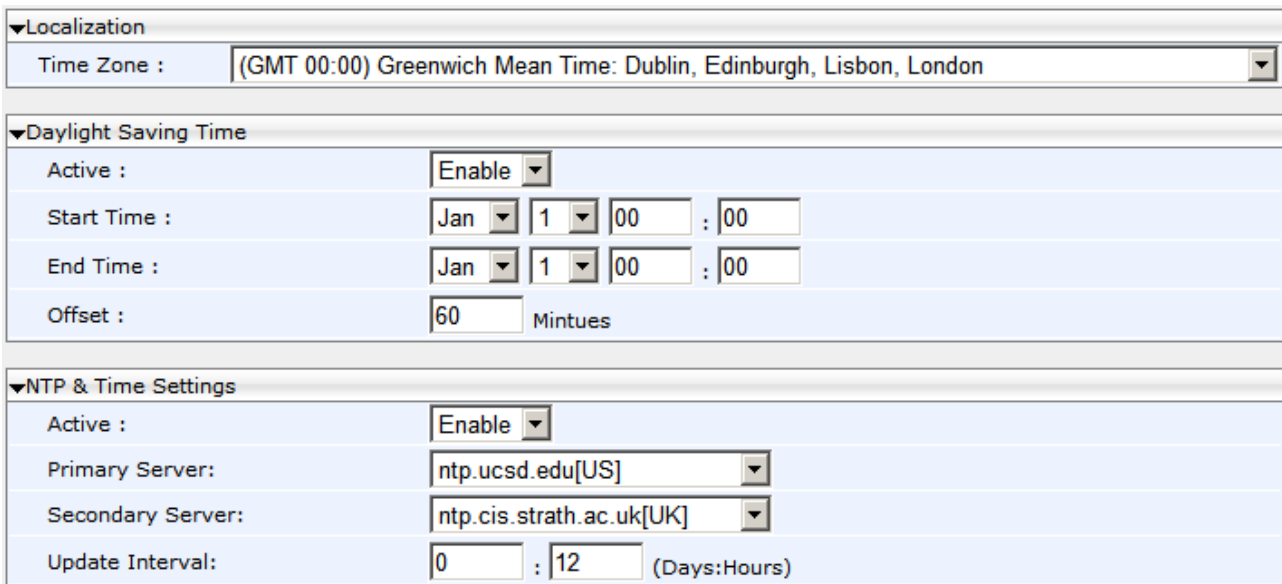
### 3.9 Configuring Date and Time

Generally, a phone retrieves the date and time from a Network Time Protocol (NTP) server when it connects to the Internet. Alternatively, the date and time can be configured manually. NTP is a protocol for distributing the Coordinated Universal Time (UTC) by means of synchronizing the clocks of computer systems over packet-switched, variable-latency data networks.

➤ **To define the date and time:**

1. Access the 'Date and Time' page (**Configuration** tab > **Advanced Applications** menu > **Date and Time**).

**Figure 3-16: Date and Time Page**



2. Configure the date and time according to the parameters in the table below, and then click **Submit**.

**Table 3-12: Date and Time Parameters Description**

Web Parameter	Configuration File Parameter	Description
<b>Localization</b>		
Time Zone	system/ntp/gmt_offset	Refer to Section 4.2.1
<b>Daylight Saving Time</b>		
Active	system/ntp/daylight_saving/activate	Refer to Section 4.2.1
Start Time	system/ntp/daylight_saving/start_date	
End Time	system/ntp/daylight_saving/end_date	
Offset	system/ntp/daylight_saving/offset	
<b>NTP &amp; Time Settings</b>		
Active	system/ntp/enabled	Refer to Section 4.2.1
Primary Server	system/ntp/primary_server_address	
Secondary Server	system/ntp/secondary_server_addresses	
Update Interval	system/ntp/sync_time	

## 3.10 Firmware and Configuration Management

The Web interface allows you to perform the following:

- Automatic update of firmware and configuration files
- Manual update of firmware and configuration files

### 3.10.1 Configuring Automatic Update of Firmware and Configuration File

The IP phone offers a built-in mechanism for automatically upgrading its software image and updating its configuration. This method is used to upgrade the phone firmware and update its configuration, by remotely downloading an updated software image and configuration file.

The automatic update mechanism helps you keep your software image and configuration up-to-date, by performing routine checks for newer software versions and configuration files, as well as allowing you to perform manual checks.

The automatic update mechanism is as follows:

1. Before connecting the phone, verify that the provisioning server is running and that the firmware and configuration files are located in the correct location.
2. Connect your phone to the IP network, and then connect the phone to the power outlet.
3. During DHCP negotiation, the phone requests for DHCP options 66/67/160 to receive provisioning information. The DHCP server should respond with Option 160 providing the provisioning URL or options 66 and 67 providing the TFTP IP address and firmware file name respectively.
4. The phone then checks whether new firmware is available by checking the firmware file header. If the version is different from the one currently running on the phone, the phone downloads the complete image and burns it to its flash memory.
5. If a new firmware is unavailable, the phone then checks whether a new configuration is available. If a configuration file is available on the server, the phone downloads it and updates the phone's configuration after verifying that the configuration file is related to the phone model and the version is later than the current one. When configuration update is needed, the phone reboots.



#### Notes:

- In the DHCP Discover message, the phone publishes its model name in Option fields 60 and 77 (e.g. 310HD). If the administrator wants to provide different provisioning information to different phone models, the administrator can set up a policy in the DHCP server according to the phone model name.
- If the phone for some reason is powered off during the firmware upgrade process, the phone is unusable and the recovery process must be performed (refer to Section 3.10.3 on page 41).
- You can only use firmware files with an *img* extension and configuration files with a *.cfg* extension.
- To “force” the firmware or configuration file to be retrieved immediately regardless of the “Check Period” value, click the **Check Now** button.
- An additional auto-provisioning mechanism is supported if the provisioning environment does not provide all the required information (e.g. DHCP options). For more information, refer to the document *3xxHD Advanced Support for Auto-provisioning Application Note*.

➤ **To define automatic update:**

1. Access the 'Automatic Update' page (**Management** tab > **Automatic Update** menu > **Automatic Update**).

**Figure 3-17: Automatic Update Page**

Firmware Version :	1.4.0_build_26	
Provisioning Method :	Static URL	
Firmware URL :	ftp://10.16.2.130/310HD_1.4.0_build_26.img	Check Now
Configuration URL :		Check Now
Check Period :	Daily	
Every day at :	00:00	
Random Provisioning Time :	120	minutes

2. Configure automatic update of firmware and configuration files according to the parameters in the table below, and then click **Submit**.

**Table 3-13: Automatic Provisioning Parameters Description**

Web Parameter	Configuration File Parameter	Description
Provisioning Method	provisioning/method	Refer to Section <a href="#">4.2.3</a>
Firmware URL	provisioning/firmware/url	
Configuration URL	provisioning/configuration/url	
DHCP Option Value	provisioning/url_option_value	
Check Period	provisioning/period/type	
Every (Check Period = Hourly)	provisioning/period/hourly/hours_interval	
Every day at	provisioning/period/daily/time	
Every (Check Period = Weekly)	provisioning/period/weekly/day provisioning/period/weekly/time	
Random Provisioning Time	provisioning/random_provisioning_time	

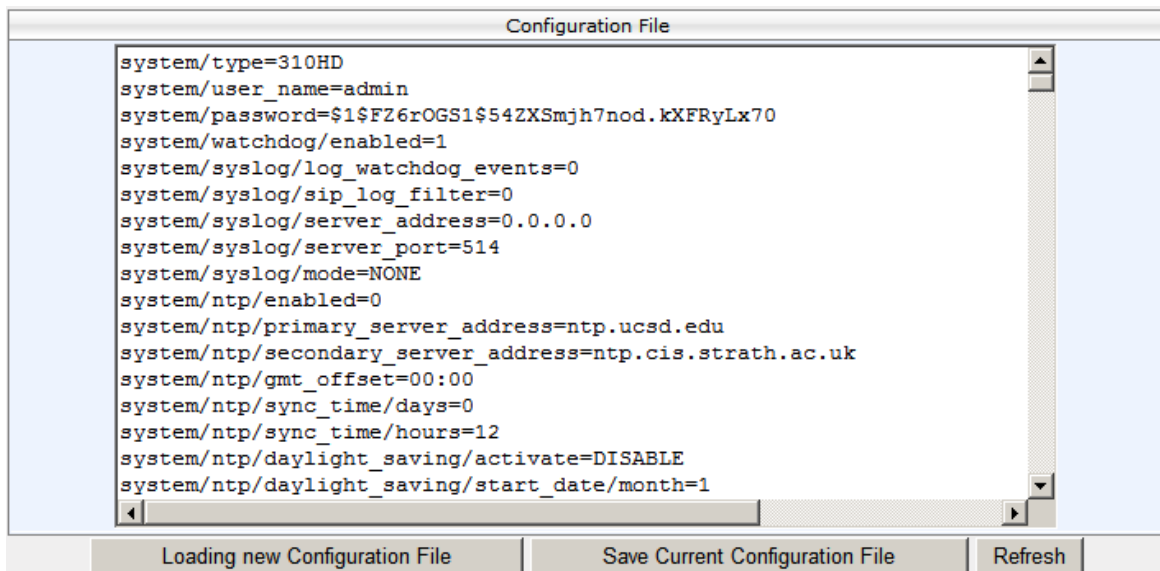
### 3.10.2 Updating (Manually) Configuration File

The phone enables you to view, save, and load its configuration file to backup and restore the current configuration.

➤ **To manually update the configuration file:**

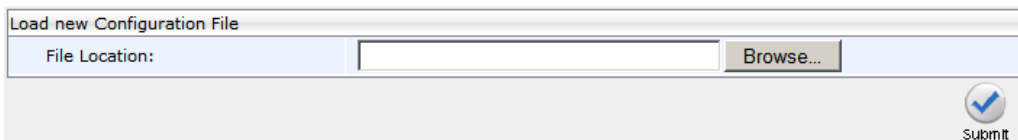
1. Access the 'Configuration File' page (**Management** tab > **Manual Update** menu > **Configuration File**). The current configuration file settings are displayed in the text pane.

**Figure 3-18: Configuration File Page**



2. Click the **Loading new Configuration File** button; the following page appears:

**Figure 3-19: Load new Configuration File**



3. Click the **Browse** button and then select the required configuration file located on your local PC; the phone verifies that the configuration file is related to the phone model. The configuration file is then loaded to the phone. Once loaded, the phone reboots (indicated by a message displayed on the phone's screen). The phone is now updated with the new configuration.



**Note:** The configuration file name must have the extension *\*.cfg*.

➤ **To save the configuration file:**

- In the 'Configuration File' page, click the **Saving Current Configuration File** button, and then save the current phone configuration file to a folder on your local PC.

### 3.10.3 Recovering Firmware

If the phone is powered off for some reason during the firmware upgrade process, the phone becomes unusable.

➤ **To recover the phone firmware:**

1. Ensure that your DHCP server supports Options 66 (TFTP server address) and 67 (firmware file), and that these are configurable.
2. Before connecting the phone, verify that the TFTP server is running and the firmware file for recovery is located in the correct location.
3. Connect your phone to the IP network, and then connect the phone to the power outlet;
  - b. The phone sends a TFTP request to the IP address indicated in the DHCP Option 66 field to retrieve the firmware file indicated in the DHCP Option 67 field.
  - c. The phone, in the DHCP Discover message sends its model name in the DHCP Option 77 field. The DHCP server, according to the phone model, sets the appropriate firmware file name in the DHCP Option 67 field sent to the phone (e.g. 310HD\_1.2.2\_build\_5.img).
  - d. The phone then upgrades to the recovery firmware.
  - e. After the firmware upgrade process completes, the phone boots up successfully.



**Note:** In Recovery Mode, the phone is unable to communicate with a TFTP server located on a Linux machine or any other operating system (OS) that is not Microsoft Windows XP.

### 3.11 Changing Login Username and Password

You can change the phone’s login user name and password. This is the login required for accessing the Web interface and the **Administration** menu of the phone’s LCD screen. The default user name and password is “admin” and “1234” respectively.

➤ **To change the login username and password:**

1. Access the ‘System Authorization’ page (**Management** tab > **Administration** menu > **Users**).

**Figure 3-20: System Authorization Page**

Username	<input type="text" value="admin"/>
Password	<input type="password" value="••••"/>
Confirm Password	<input type="password" value="••••"/>

2. In the ‘Username’ field, enter a user name.
3. In the ‘Password’ field, enter a new password, and then in the ‘Confirm Password’ field, re-enter this new password.
4. Click **Submit**; a confirmation box appears.
5. Click **OK**.

**Table 3-14: Login Username and Password Parameters Description**

Web Parameter	Configuration File Parameter	Description
Username	system/user_name	Refer to Section <a href="#">4.2.1</a>
Password	system/password	

## 3.12 Restoring Phone Defaults

You can restore all your phone's settings to factory default settings, as described in the procedure below.

➤ **To restore the phone to factory defaults:**

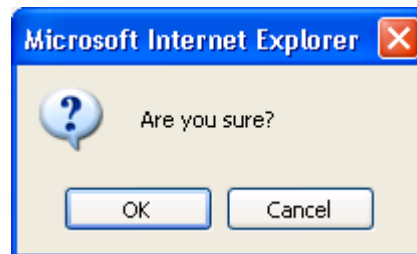
1. Access the 'Restore Defaults' page (**Management** tab > **Administration** menu > **Restore Defaults**).

**Figure 3-21: Restore Defaults Page**



2. Click the **Submit** button; a confirmation box appears prompting you to confirm.

**Figure 3-22: Submit Confirmation Box**



3. Click **OK**.

## 3.13 Restarting Phone

You can use the Web interface to restart your phone.

➤ **To restart the phone:**

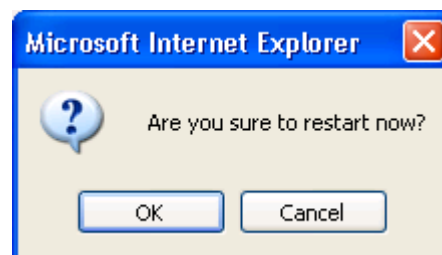
1. Access the 'Restart System' page (**Management** tab > **Administration** menu > **Restart System**).

**Figure 3-23: Restart System Page**



2. Click the **Restart** button; a confirmation box appears prompting you to confirm.

**Figure 3-24: Confirmation Box**



3. Click **OK**.

### 3.14 Configuring TR-069 Remote Management

To connect to the remote management server the TR-069 client on the phone should be configured as following:

➤ **To configure the TR-69 parameters:**

1. Access the 'TR-069' page (**Management** tab > **Remote Management** menu > **TR-069**).

Figure 3-25: TR-069 Page

▼TR-069	
Activate:	Enable ▾
TR-069 Key:	<input type="text"/>
TR-069 ACS URL:	<input type="text"/>
TR-069 User:	<input type="text"/>
TR-069 Password:	<input type="text"/>
TR-069 Request Connection User:	<input type="text"/>
TR-069 Request Connection Password:	<input type="text"/>
Informed:	Enable ▾
Informed Interval:	3600

2. Configure TR-069 according to the parameters in the table below.
3. Click **Submit**, and then **OK** at the confirmation box prompt.

Table 3-15: TR-069 Parameters Description

Web Parameter	Configuration File Parameter	Description
Activate	management/tr069/enabled	Refer to Section 4.2.4
TR-069 Key	management/tr069/feature_key	
TR-069 ACS URL	management/tr069/acs_url	
TR-069 User	management/tr069/user_name	
TR-069 Password	management/tr069/password	
TR-069 Request Connection User	management/tr069/connection_request/user_name	
TR-069 Request Connection Password	management/tr069/connection_request/password	
Informed	management/tr069/inform/enabled	
Informed Interval	management/tr069/inform/interval	

## 3.15 Viewing Status Information

The Web interface allows you to view a variety of status information about your phone.



**Note:** Currently, the 'VoIP Status' page (**Status & Diagnostics** tab > **System Status** menu > **VoIP Status**) does not exist.

### 3.15.1 Viewing LAN Information

You can view various LAN details such as IP address and default gateway.

➤ **To view LAN status information:**

- Access the 'Network Status' page (**Status & Diagnostics** tab > **System Status** menu > **Network Status**).

**Figure 3-26: Network Status Page**

LAN Information	
Type:	DHCP Client
IP Address:	10.13.22.15
Subnet Mask:	255.255.0.0
Default Gateway Address:	10.13.0.1
Primary DNS:	10.1.1.11
Secondary DNS:	10.1.1.10
MAC Address:	00:90:8F:0C:E3:92

### 3.15.2 Viewing Call History

You can view a list of received calls, missed calls, and dialed numbers.

➤ **To view call history log:**

1. Access the 'Call History' page (**Status & Diagnostics** tab > **History** menu > **Call History**).

**Figure 3-27: Call History Page**

Type: Missed Calls ▾ Page: 1 ▾				
No.	Number	Time	Duration	Delete
1	anonymous	2000/01/02 Sunday 21:41:27	00:00:00	<input type="checkbox"/>
2	222	2000/01/02 Sunday 19:34:23	00:00:00	<input type="checkbox"/>
3	anonymous	2000/01/01 Saturday 23:06:32	00:00:00	<input type="checkbox"/>
4	anonymous	2000/01/01 Saturday 20:30:19	00:00:00	<input type="checkbox"/>

2. From the 'Type' drop-down list, select the type of call history (i.e., missed calls, received calls, and dialed numbers) that you want to view; the table lists the call history according to the chosen call history type.

You can delete a logged call history entry, by selecting the 'Delete' check box corresponding to the entry that you want to delete, and then clicking the **Delete** button.

### 3.15.3 Viewing Phone's Version Number

You can view the phone's model name as well as the firmware version currently running on the phone.

➤ **To view the phone's model and version number:**

- Access the 'System Information' page (**Status & Diagnostics** tab > **System Information** menu > **Versions**).

**Figure 3-28: System Information Page**

Model Name	310HD
Firmware Version	1.4.0_build_26
Release Date	13 July 2010 14:14:04

## 3.16 Diagnostics

### 3.16.1 Configuring System Logging

System logging allows you to debug the different software components of the phone.

➤ **To define the logging parameters:**

1. Access the 'System logging' page (**Status & Diagnostics** tab > **Diagnostics** menu > **Logging**).

**Figure 3-29: System Logging Page**

System Logging	
Activate:	Enable
Server IP Address or Host Name:	0.0.0.0
Server Port:	514
Category:	0

2. Configure system logging according to the parameters in the table below, and then click **Submit**.

**Table 3-16: System Logging Parameters Description**

Web Parameter	Configuration File Parameter	Description
Activate	<b>system/watchdog/enabled</b>	Refer to Section <a href="#">4.2.1</a>
Server IP Address or Host Name	<b>system/syslog/server_address</b>	
Server Port	<b>system/syslog/server_port</b>	
Category	<b>system/syslog/sip_log_filter</b>	

### 3.16.2 Configuring Packet Recording

Recording parameters allows you to debug the voice activity of the phone.

➤ **To define the recording parameters:**

1. Access the 'Recording' page (**Status & Diagnostics** tab > **Diagnostics** menu > **Recording**).

**Figure 3-30: Recording Page**

Recording	
Remote IP Address or Host Name:	<input type="text" value="0.0.0.0"/>
Remote Port:	<input type="text" value="50000"/>
Enable DSP Recording:	<input type="button" value="Enable"/>
Enable RTP Recording:	<input type="button" value="Disable"/>
Enable EC Debug Recording:	<input type="button" value="Disable"/>
Enable Network Recording:	<input type="button" value="Disable"/>
Enable TDM Recording:	<input type="button" value="Disable"/>

2. Configure packet recording according to the parameters in the table below, and then click **Submit**.

**Table 3-17: Packet Recording Parameters Description**

Web Parameter	Configuration File Parameter	Description
Remote IP Address or Host Name	<b>voip/packet_recording/remote_ip</b>	Refer to Section <a href="#">4.2.7</a>
Remote Port	<b>voip/packet_recording/remote_port</b>	
Enable DSP Recording	<b>/voip/packet_recording/enabled</b>	
Enable RTP Recording	<b>voip/packet_recording/rtp_recording/enabled</b>	
Enable EC Debug Recording	<b>voip/packet_recording/ec_debug_recording/enabled</b>	
Enable Network Recording	<b>voip/packet_recording/network_recording/enabled</b>	
Enable TDM Recording	<b>voip/packet_recording/tdm_recording/enabled</b>	

## 4 Configuration File Parameters

This section describes the Configuration file and the parameters that you can set in the Configuration file.

### 4.1 Overview

The Configuration file can be loaded to the phone using the automatic provisioning mechanism, or manually from your local PC using the Web interface (refer to Section 3.10.2). The subsections below describe the Configuration file syntax and linking additional Configuration files to a Configuration file.

#### 4.1.1 File Syntax

The Configuration file can be created using a standard ASCII, text-based program such as Notepad. The Configuration file is a \*.cfg file with the file name being the phone's MAC address: **<phone's MAC address>.cfg**.

The syntax of the Configuration file is as follows:

```
<parameter name>=<value>
```

Ensure that the Configuration file adheres to the following guidelines:

- No spaces on either side of the equals (=) sign.
- Each parameter must be on a new line.

Below is an example of part of a Configuration file:

```
system/type=310HD
voip/line/0/enabled=1
voip/line/0/id=1234
voip/line/0/description=320HD
voip/line/0/auth name=1234
voip/line/0/auth password=4321
```

#### 4.1.2 Linking Additional Files using “Include”

The Configuration file allows you to include links (URL and/or file name) to other Configuration files that provide additional parameter settings. This is especially useful in deployments with multiple phones, where the phones share common configuration but where each phone has some unique settings. In such a scenario, a phone's Configuration file can include unique parameter settings as well as links to additional Configuration files with settings common to all phones.

Linking additional files is achieved by using the **include** function in the phone's Configuration file. For example, the below Configuration file provides links to additional Configuration files (shown in bold font):

```
system/type=310HD
include 310HD <MAC> voip.cfg
include vlan conf.cfg
include network conf.cfg
include provisioning conf.cfg
```

In addition, the Configuration file can provide URL paths (FTP, TFTP, HTTP, or HTTPS) to where the additional files are located, as shown in the example below (shown in bolded font):

```

system/type=310HD
include http://10.10.10.10/310HD <MAC> voip.cfg
include https://remote-pc/vlan_conf.cfg
include tftp://10.10.10.10/310HD <MAC> network.cfg
include ftp://remote-pc/provisining conf.cfg
  
```



**Note:** If no URL is provided in the Configuration file, the files are retrieved according to the provisioning information (e.g. DHCP Options 66/67 or 160).

## 4.2 Configuration File Parameters

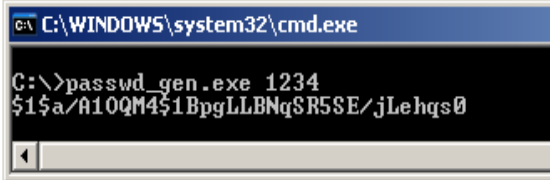
The configuration file parameters are described in this subsection.



**Note:** The optional values of the Configuration file parameters are enclosed in square brackets while its corresponding Web values are rewritten outside the square brackets, for example, [1] Enable.

### 4.2.1 System Parameters

Table 4-1: System Configuration File Parameters

Parameter	Description
<p><b>Note:</b> To add a value to these parameters, enter <b>system/</b> followed by the parameter name, equal sign and then the value (e.g. <code>system/type=310HD</code>).</p>	
<b>system/type</b>	<p>The phone model. The default value <b>310HD</b>.</p>
<b>system/user_name</b>	<p>The phone user name. The default value is admin. <b>Note:</b> This parameter is applicable only to the Web and Telnet interfaces.</p>
<b>system/password</b>	<p>The encrypted phone password. The default value is 1234. To generate an encrypted password for the <b>system/password</b> parameter, run the password generator (<b>passwd_gen</b>) with the required password, as shown below. This application is available on Windows and Linux operating systems. The resultant string is the encrypted password for <b>system/password</b>.</p>  <p><b>Note:</b> This parameter is applicable only to the Web and Telnet interfaces, and LCD display.</p>
<b>system/watchdog/enable d</b>	<p>Enables the system watch dog.</p> <ul style="list-style-type: none"> <li>▪ [0] Disable</li> <li>▪ [1] Enable (default)</li> </ul> <p><b>Note:</b> It is recommended to leave this parameter at its default value.</p>
<b>system/syslog/log_watch dog_events</b>	<p>Enables the watchdog logs to be sent to the address specified in <code>system/syslog/server_address</code>.</p> <ul style="list-style-type: none"> <li>▪ [0] (default) - Disable</li> <li>▪ [1] - Enable</li> </ul>

Parameter	Description
<b>system/syslog/sip_log_filter</b>	Filters the type of the VoIP application logging. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> (default) = Logging is disabled</li> <li>▪ <b>[1]</b> - SIP Call Control</li> <li>▪ <b>[2]</b> - SIP Stack</li> <li>▪ <b>[3]</b> - SIP Call Control and SIP Stack</li> <li>▪ <b>[4]</b> - User Application</li> <li>▪ <b>[5]</b> - SIP Call Control and User Application</li> <li>▪ <b>[6]</b> - SIP Stack and User Application</li> <li>▪ <b>[7]</b> - SIP Call Control, SIP Stack and User Application</li> </ul>
<b>system/syslog/server_address</b>	The IP address (in dotted-decimal notation) of the computer you are using to run the Syslog server (e.g. WireShark). The Syslog server is an application designed to collect the logs and error messages generated by the phone. The default IP address is 0.0.0.0.
<b>system/syslog/server_port</b>	Defines the UDP port of the Syslog server. The valid range is 0 to 65,535. The default port is 514.
<b>system/syslog/mode</b>	Defines the output direction of the Syslog information. <ul style="list-style-type: none"> <li>▪ <b>[NONE]</b> (default) - Disable</li> <li>▪ <b>[UDP]</b> - Toward the network</li> </ul>
<b>Network Time Protocol (NTP) Server</b>	
<b>system/ntp/enabled</b>	Enables the NTP server from which the phone retrieves the date and time. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable – obtains the time information from a configured NTP server</li> </ul>
<b>system/ntp/primary_server_address</b>	Defines the address of the main NTP server (this can be a domain name, for example, tick.nap.com.ar).
<b>system/ntp/secondary_server_address</b>	Defines the address of the secondary NTP server.
<b>system/ntp/gmt_offset</b>	The GMT offset. The format of this value is + or - xx:yy (e.g., +02:00). The default offset is 00:00.
<b>system/ntp/sync_time</b>	This sub-section defines how often the phone must perform an update with the NTP server. <ul style="list-style-type: none"> <li>▪ <b>[days]</b> - defines the number of days</li> <li>▪ <b>[hours]</b> - defines the number of hours</li> </ul> For example: To configure the phone to perform an update with an NTP server every 1 day and 6 hours, set the following: <b>system/ntp/sync_time/days=1</b> <b>system/ntp/sync_time/hours=6</b>
<b>system/ntp/sync_time/days</b>	The number of days. The valid range is 0 to 7. The default of days is 0.
<b>system/ntp/sync_time/hours</b>	The number of hours. The valid range is 0 to 24. The default is 12.

Parameter	Description
<b>Daylight Saving</b>	
<b>system/ntp/daylight_saving/activate</b>	Determines whether the phone automatically detects the Daylight Saving setting for selected time zones. <ul style="list-style-type: none"> <li>▪ <b>[Disable]</b> Disable (default)</li> <li>▪ <b>[Enable]</b> Enable</li> </ul>
<b>system/ntp/daylight_saving/start_date</b>	This sub-section defines the starting day for the daylight saving offset. <ul style="list-style-type: none"> <li>▪ <b>[month]</b> - defines the specific month in a year</li> <li>▪ <b>[day]</b> - defines the specific day in a month</li> <li>▪ <b>[hour]</b> - defines the specific hour in a day</li> <li>▪ <b>[minute]</b> - defines the specific minute in an hour</li> </ul> For example: To configure the phone to start daylight savings with a specific offset on February 22 <sup>nd</sup> at 14:30, set the following: <pre>system/ntp/daylight_saving/start_date/month=2 system/ntp/daylight_saving/start_date/day=22 system/ntp/daylight_saving/start_date/hour=14 system/ntp/daylight_saving/start_date/minute=30</pre>
<b>system/ntp/daylight_saving/start_date/month</b>	The month in a year. The valid range is 1 to 12.
<b>system/ntp/daylight_saving/start_date/day</b>	The day in a month. The valid range is 1 to 31.
<b>system/ntp/daylight_saving/start_date/hour</b>	The hour in the day. The valid range is 0 to 23.
<b>system/ntp/daylight_saving/start_date/minute</b>	The minute in an hour. The valid range is 0 to 59.
<b>system/ntp/daylight_saving/end_date</b>	This sub-section defines the ending day for the daylight saving offset. <ul style="list-style-type: none"> <li>▪ <b>[month]</b> - defines the specific month in a year</li> <li>▪ <b>[day]</b> - defines the specific day in a month</li> <li>▪ <b>[hour]</b> - defines the specific hour in a day</li> <li>▪ <b>[minute]</b> - defines the specific minute in an hour</li> </ul> For example: To configure the phone to end the daylight savings on July 16 <sup>th</sup> at 22:15, set the following: <pre>system/ntp/daylight_saving/end_date/month=7 system/ntp/daylight_saving/end_date/day=16 system/ntp/daylight_saving/end_date/hour=22 system/ntp/daylight_saving/end_date/minute=15</pre>
<b>system/ntp/daylight_saving/end_date/month</b>	The month in a year. The valid range is 1 to 12.
<b>system/ntp/daylight_saving/end_date/day</b>	The day in a month. The valid range is 1 to 31.
<b>system/ntp/daylight_saving/end_date/hour</b>	The hour in the day The valid range is 0 to 23.

Parameter	Description
<b>system/ntp/daylight_saving/end_date/minute</b>	The minute in an hour. The valid range is 0 to 59.
<b>system/ntp/daylight_saving/offset</b>	The offset value for the daylight saving. The valid range is 0 to 180. The default offset is 60.

## 4.2.2 Network Parameters

**Table 4-2: Network Configuration File Parameters**

Parameter	Description
<b>Note:</b> To add a value to these parameters, enter <b>network/</b> followed by the parameter name, equal sign and then the value (e.g. <code>network/lan_type=DHCP</code> ).	
<b>network/lan_type</b>	Defines the IP addressing method: <ul style="list-style-type: none"> <li>▪ <b>[DHCP]</b> Automatic IP DHCP (default) - Phone's IP address is acquired automatically from a DHCP server</li> <li>▪ <b>[STATIC]</b> Static IP - Phone's IP address is defined manually</li> </ul>
<b>network/lan/fixed_ip</b>	This sub-section defines the parameters which are relevant in case <b>lan_type</b> is configured to "STATIC".
<b>network/lan/fixed_ip/ip_address</b>	The LAN IP address. <b>Note:</b> This parameter is applicable only when the phone is assigned a static IP address.
<b>network/lan/fixed_ip/netmask</b>	The subnet mask address. <b>Note:</b> This parameter is applicable only when the phone is assigned a static IP address.
<b>network/lan/fixed_ip/gateway</b>	The IP address of the default gateway. <b>Note:</b> This parameter is applicable only when the phone is assigned a static IP address.
<b>Domain Name Server (DNS)</b>	
<b>network/lan/fixed_ip/primary_dns</b>	The primary DNS server address. <b>Note:</b> This parameter is applicable only when the phone is assigned a static IP address.
<b>network/lan/fixed_ip/secondary_dns</b>	The secondary DNS server address. The phone connects to this server if the primary DNS server is unavailable. <b>Note:</b> This parameter is applicable only when the phone is assigned a static IP address.

Parameter	Description
<b>VLAN</b>	
<b>network/lan/vlan/mode</b>	<p>Determines the VLAN mode of operation.</p> <ul style="list-style-type: none"> <li>▪ <b>[Disable]</b> Disable</li> <li>▪ <b>[Manual]</b> Manual Configuration of LAN - Static configuration of VLAN ID and priority</li> <li>▪ <b>[Automatic]</b> Automatic Configuration of VLAN/CDP (default) - VLAN discovery mechanism based on Cisco Discovery Protocol (CDP)</li> </ul>
<b>network/lan/vlan/id</b>	<p>The VLAN ID.</p> <p>The valid range is 0 to 4096. The default VLAN ID is 0.</p>
<b>network/lan/vlan/priority</b>	<p>The priority of traffic pertaining to this VLAN.</p> <p>The valid range is 0 to 7 (where 7 is the highest priority). The default VLAN priority is 0.</p>

## 4.2.3 Provisioning Parameters

Table 4-3: Provisioning Configuration File Parameters

Parameter	Description
<p><b>Note:</b> To add a value to these parameters, enter <code>provisioning/</code> followed by the parameter name equals the value (e.g. <code>provisioning/method=dynamic</code>).</p>	
<code>provisioning/method</code>	<p>Defines the provisioning method:</p> <ul style="list-style-type: none"> <li>▪ <b>[Disable]</b> Disable - Automatic update is disabled. The phone attempts to upgrade its firmware and configuration</li> <li>▪ <b>[Dynamic]</b> DHCP Options (Dynamic URL) (default) - Using DHCP options 160 or 66/67 for provisioning</li> <li>▪ <b>[Static]</b> Static URL - Using Static URL for provisioning</li> </ul>
<code>provisioning/firmware/url</code>	<p>The static URL for checking the firmware file. The URL must be entered using one of the following syntax options:</p> <ul style="list-style-type: none"> <li>▪ <code>&lt;protocol&gt;://&lt;server IP address or host name&gt;</code></li> <li>▪ <code>&lt;protocol&gt;://&lt;server IP address or host name&gt;/&lt;firmware file name&gt;</code></li> </ul> <p>Where <code>&lt;protocol&gt;</code> can be one of the following protocols: "ftp", "tftp", "http" or "https". For example:</p> <ul style="list-style-type: none"> <li>▪ <code>tftp://192.168.2.1</code> – retrieved firmware file is 310HD.img</li> <li>▪ <code>ftp://192.168.2.1/Different_Firmware_Name.img</code> - retrieved firmware file is Different_Firmware_Name.img</li> </ul> <p><b>Note:</b> This parameter is applicable only when <b>method</b> is configured to "Static".</p>
<code>provisioning/configuration/url</code>	<p>The static URL for checking the configuration file. The URL must be entered using one of the following syntax options:</p> <ul style="list-style-type: none"> <li>▪ <code>&lt;protocol&gt;://&lt;server IP address or host name&gt;</code></li> <li>▪ <code>&lt;protocol&gt;://&lt;server IP address or host name&gt;/&lt;configuration file name&gt;</code></li> </ul> <p>Where <code>&lt;protocol&gt;</code> can be one of the following protocols: "ftp", "tftp", "http" or "https". For example:</p> <ul style="list-style-type: none"> <li>▪ <code>http://192.168.2.1</code> - configuration file name is <code>&lt;MAC Address&gt;.cfg</code>, for example, 001122334455.cfg</li> <li>▪ <code>https://192.168.2.1/310HD_&lt;MAC&gt;_conf.cfg</code> - retrieved configuration file name is 310HD_&lt;MAC Address&gt;_conf.cfg, for example, 310HD_001122334455_conf.cfg</li> </ul> <p><b>Note:</b> This parameter is applicable only when <b>method</b> is configured to "Static".</p>
<code>provisioning/period/type</code>	<p>Defines the period type for automatic provisioning:</p> <ul style="list-style-type: none"> <li>▪ <b>[hourly]</b> Hourly - Sets an interval in hours.</li> <li>▪ <b>[daily]</b> Daily (default) - Sets an hour in the day.</li> <li>▪ <b>[weekly]</b> Weekly - Sets a day in the week and an hour in the day.</li> <li>▪ <b>[powerup]</b> On Power-up Only - The phone tries to upgrade only after power-up.</li> </ul>
<code>provisioning/period/hourly/hours_interval</code>	<p>The interval in hours for automatically checking for new firmware and configuration files.</p> <p>The valid range is 1 to 168. The default is 24.</p> <p><b>Note:</b> This parameter is applicable only when <b>type</b> is configured to "hourly".</p>

Parameter	Description
<b>provisioning/period/daily/time</b>	<p>The hour in the day for automatically checking for new firmware and configuration files.</p> <p>The format of this value is <b>hh:mm</b>, where <b>hh</b> is hour and <b>mm</b> is minutes. For example, 00 : 30 .</p> <p>The default time is <b>00:00</b>.</p> <p><b>Note:</b> This parameter is applicable only when <b>type</b> is configured to “daily”.</p>
<b>provisioning/period/weekly/day</b>	<p>The day in the week for automatically checking for new firmware and configuration files.</p> <ul style="list-style-type: none"> <li>▪ <b>[Sunday]</b> Sunday (default)</li> <li>▪ <b>[Monday]</b> Monday</li> <li>▪ <b>[Tuesday]</b> Tuesday</li> <li>▪ <b>[Wednesday]</b> Wednesday</li> <li>▪ <b>[Thursday]</b> Thursday</li> <li>▪ <b>[Friday]</b> Friday</li> <li>▪ <b>[Saturday]</b> Saturday</li> </ul> <p><b>Note:</b> This parameter is applicable only when <b>type</b> is configured to “weekly”.</p>
<b>provisioning/period/weekly/time</b>	<p>The hour in the day for automatically checking for new firmware and configuration files.</p> <p>The format of this value is: <b>hh:mm</b>, where <b>hh</b> is hour and <b>mm</b> is minutes. For example: 00 : 30</p> <p>The default time is <b>00:00</b>.</p> <p><b>Note:</b> This parameter is applicable only when <b>type</b> is configured to “weekly”.</p>
<b>provisioning/url_option_value</b>	<p>Determines the DHCP option number to be used for receiving the URL for provisioning.</p> <p>The default value is 160.</p> <p>The phone supports DHCP Option 160 for complete URL and Options 66/67 for TFTP usage. Option 160 has the highest priority and if absent, Options 66/67 are used.</p> <p>The following syntax is available for DHCP option 160:</p> <ul style="list-style-type: none"> <li>▪ &lt;protocol&gt;://&lt;server IP address or host name&gt;</li> <li>▪ &lt;protocol&gt;://&lt;server IP address or host name&gt;/&lt;firmware file name&gt;</li> <li>▪ &lt;protocol&gt;://&lt;server IP address or host name&gt;/&lt;firmware file name&gt;;&lt;configuration file name&gt;</li> <li>▪ &lt;protocol&gt;://&lt;server IP address or host name&gt;/&lt;configuration file name&gt;</li> </ul> <p>Where &lt;protocol&gt; can be one of the following: “ftp”, “tftp”, “http” or “https”.</p> <p>For example:</p> <ul style="list-style-type: none"> <li>▪ <b>ftp://192.168.2.1</b> – retrieved firmware file is <i>310HD.img</i> and the configuration file name is &lt;MAC address&gt;.cfg. For example, 001122334455.cfg</li> <li>▪ <b>tftp://192.168.2.1/different_firmware_name.img</b> - retrieved firmware file is <i>Different_Firmware_Name.img</i> and the configuration file name is &lt;MAC address&gt;.cfg. For example, 001122334455.cfg</li> <li>▪ <b>http://192.168.2.1/different_firmware_name.img; 310HD_&lt;MAC&gt;_conf.cfg</b> - retrieved firmware file is <i>different_firmware_name.img</i> and the configuration file name is <i>310HD_&lt;MAC address&gt;_conf.cfg</i>. For example, <b>310HD_001122334455_conf.cfg</b></li> </ul>

Parameter	Description
	<ul style="list-style-type: none"> <li>▪ <b>https://192.168.2.1/310HD_&lt;MAC&gt;_conf.cfg</b> - In this case the retrieved firmware file is 310HD.img and the configuration file name is 310HD_&lt;MAC Address&gt;_conf.cfg. For example, 310HD_001122334455_conf.cfg</li> </ul> <p>The following syntax is available for DHCP Options 66/67:</p> <ul style="list-style-type: none"> <li>▪ Option 66 must be a valid IP address or host name of a TFTP server only.</li> <li>▪ Option 67 must be the firmware name.</li> </ul> <p>If Option 67 is absent, the phone requests for the 310HD.img image file. For example:</p> <ul style="list-style-type: none"> <li>▪ <b>Option 66:</b> 192.168.2.1 or myTFTPServer</li> <li>▪ <b>Option 67:</b> 310HD_1.2.2_build_5.img</li> </ul> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>▪ This parameter is applicable only when <b>method</b> is configured to “Dynamic”.</li> <li>▪ It is recommended to leave the parameter at its default value to avoid conflict with other DHCP options settings.</li> </ul>
<b>provisioning/random_provisioning_time</b>	<p>Defines the maximum random number to start the provisioning process.</p> <p>This is used for periodic checking of firmware and configuration files to avoid multiple devices from starting the upgrade process at the same time. When the device is meant to start the upgrade, the device randomly selects a number between 1 and the value set for <b>random_provisioning_time</b> and performs the check only after the random time.</p> <p>The valid range is 0-65535. The default value is 120.</p>
<b>provisioning/XXX_uri</b>	<p>The URI that allows the administrator to retrieve relevant information from separate files. The URI's can include specific protocol and path of the provisioning server or alternatively, the files can be retrieved from the server whose information was provided during provisioning (e.g. DHCP options 66/67 or 160).</p> <p>For example, in the below setting, <b>YYY.ext</b> is retrieved from the server whose information was provided during provisioning (e.g. DHCP options 66/67 or 160):</p> <p><b>provisioning/XXX_uri=YYY.ext</b></p> <p>Alternatively, the full URL and protocol can be set as follows:</p> <p><b>provisioning/XXX_uri=&lt;protocol&gt;://&lt;full path&gt;/YYY.ext</b></p> <p>For example:</p> <ul style="list-style-type: none"> <li>▪ provisioning/XXX_uri=tftp://10.10.10.10/YYY.ext</li> <li>▪ provisioning/XXX_uri=ftp:// 10.10.10.10/YYY.ext</li> <li>▪ provisioning/XXX_uri=http:// 10.10.10.10/YYY.ext</li> <li>▪ provisioning/XXX_uri=https:// 10.10.10.10/YYY.ext</li> </ul>
<b>provisioning/ring_tone_uri</b>	<p>The URI for retrieving the ring tones file. The ring tones can be compressed to *.zip or *.tgz files and provided to the phone during provisioning.</p> <p>For example: <b>provisioning/ring_tone_uri=tones.tgz</b></p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>▪ The ringtone file is downloaded only after boot up, and not periodically.</li> <li>▪ If the tones file is new, the phone updates the information, but does not reboot.</li> </ul>

Parameter	Description
<b>provisioning/corporate_directory_uri</b>	<p>The URI for retrieving the corporate directory. The corporate directory must be included in a separate file that can be downloaded to the phone during provisioning.</p> <p>For example: <b>provisioning/corporate_directory_uri=corporate_dir.txt</b></p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>▪ The corporate directory file is downloaded after boot up and periodically.</li> <li>▪ If the corporate directory file is new, the phone updates the information, does not reboot.</li> <li>▪ For creating a Corporate Directory file, refer to Section 4.3.</li> </ul>
<b>provisioning/speed_dial_uri</b>	<p>The URI for retrieving the speed dial list. The speed dial list must be included in a separate file that can be downloaded to the phone during provisioning.</p> <p>For example: <b>provisioning/speed_dial_uri=speed_dial_list.txt</b></p> <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>▪ The speed dial file is downloaded after boot up and periodically.</li> <li>▪ If the speed dial file is new, the phone reboots.</li> <li>▪ For creating a Speed Dial file, refer to Section 4.4.</li> </ul>

## 4.2.4 TR-069 Management Parameters

Table 4-4: TR-069 Management Configuration File Parameters

Parameter	Description
<p><b>Note:</b> To add a value to these parameters, enter <b>management/</b> followed by the parameter name, equal sign and then the value (e.g. <code>management/tr069/enabled=0</code>).</p>	
<b>management/tr069/enabled</b>	<p>Enables TR-069.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul> <p><b>Note:</b> This feature can be enabled only if the valid key is provided in the <b>feature_key</b> field.</p>
<b>management/tr069/feature_key</b>	Feature key to enable the TR-069.
<b>management/tr069/acs_url</b>	<p>URL for connecting the device to the ACS.</p> <p><b>Note:</b> This parameter must be in the form of a valid HTTP or HTTPS URL.</p>
<b>management/tr069/user_name</b>	Username used to authenticate the device when making a connection to the ACS.
<b>management/tr069/password</b>	Password used to authenticate the device when making a connection to the ACS.
<b>management/tr069/connection_request/user_name</b>	Username to authenticate an ACS making a Connection Request to the device.
<b>management/tr069/connection_request/password</b>	Password to authenticate an ACS making a Connection Request to the device.

Parameter	Description
<b>management/tr069/inform/enabled</b>	Determines whether or not the device must periodically send device information to the ACS using the <b>Inform method</b> call. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable</li> <li>▪ <b>[1]</b> Enable (default)</li> </ul>
<b>management/tr069/inform/interval</b>	The duration (in seconds) of the interval for which the device must attempt to connect with the ACS and call the <b>Inform method</b> if TR-069 is enabled ( <b>management/tr069/inform/enabled</b> ). The valid range is 0 to 65535. The default is 3600.

## 4.2.5 VoIP Parameters

### 4.2.5.1 Line Parameters

**Table 4-5: Line Configuration File Parameters**

Parameter	Description
<b>voip/line/%d/enabled</b>	Enables the line. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/line/%d/id</b>	Lines VoIP user's ID for identification to initiate and accept calls. The user's ID can be up to 30 characters.
<b>voip/line/%d/description</b>	Arbitrary name to intuitively identify the line and that is displayed to remote parties as your caller ID.
<b>voip/line/%d/auth_name</b>	User name provided to you from the VoIP service provider. This is used when sending a response to Unauthorized or Proxy Authentication Requested (401/407). The authentication name can be up to 35 characters.
<b>voip/line/%d/auth_password</b>	Password provided to you from the VoIP Service Provider. This is used when sending a response to Unauthorized or Proxy Authentication Requested (401/407). The authentication password can be up to 35 characters.
<b>voip/line/%d/do_not_disturb/activated</b>	Activates the Do Not Disturb (DnD) feature, if enabled, using the parameter <b>voip/services/do_not_disturb/enabled</b> . <ul style="list-style-type: none"> <li>▪ <b>[0]</b> (default) - Activate</li> <li>▪ <b>[1]</b> - Deactivate</li> </ul> <b>Notes:</b> DnD can also be activated using the LCD screen interface (more common).



**Note:** The variable `%d` refers to the line number, however, only line 0 is relevant since the phone supports only one extension.

## 4.2.5.2 Gain Parameters

Table 4-6: Gain Configuration File Parameters

Parameter	Description
<b>voip/audio/gain/additional_speaker_gain</b>	Additional parameter for speaker gain configuration. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> 0dB</li> <li>▪ <b>[1]</b> 1dB</li> <li>▪ <b>[2]</b> 2dB</li> <li>▪ <b>[3]</b> 3dB (default)</li> </ul>
<b>voip/audio/gain/tone_signal_level</b>	Call progress tone volume. This volume can be modified on-the-fly by pressing the phone's VOLUME key in certain scenarios. The valid range is 1 to 31 (-dB). The default value is 10 (-10dB).
<b>voip/audio/gain/ringer_signal_level</b>	Ringing tone volume. This volume can be modified on-the-fly by pressing the phone's VOLUME key when the phone is in idle state. The valid range is 0 to 63: <ul style="list-style-type: none"> <li>▪ <b>[0]</b> - Mute</li> <li>▪ <b>[1]</b> - -31 dB</li> <li>▪ <b>[32]</b> - 0 dB</li> <li>▪ <b>[63]</b> - 31 dB</li> </ul>
<b>Hands-free Gain Parameters</b>	
<b>Note:</b> Values are in decibels (dB) and represented as follows: <ul style="list-style-type: none"> <li>▪ Negative values: use the word "minus" (e.g. =minus9db).</li> <li>▪ Positive values: use the word "plus" (e.g. =plus9db).</li> </ul> Decimal places: use underscore instead of period (e.g. plus19_5db).	
<b>voip/audio/gain/handsfree_digital_output_gain</b>	Digital output gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 7 (7dB).
<b>voip/audio/gain/handsfree_digital_input_gain</b>	Digital input gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).
<b>voip/audio/gain/handsfree_analog_output_gain</b>	Analog output gain (in dB). Valid values: <b>[0db]</b> , <b>[minus1_5db]</b> , <b>[minus3db]</b> , <b>[minus4_5db]</b> , <b>[minus6db]</b> , <b>[minus7_5db]</b> , <b>[minus9db]</b> , <b>[minus10_5db]</b> , <b>[minus12db]</b> , <b>[minus13_5db]</b> , <b>[minus15db]</b> , <b>[minus16_5db]</b> , <b>[minus18db]</b> , <b>[minus19_5db]</b> , <b>[minus21db]</b> , <b>[minus22_5db]</b> , <b>[minus24db]</b> , <b>[minus25_5db]</b> , <b>[minus27db]</b> , <b>[minus28_5db]</b> , <b>[minus30db]</b> , <b>[minus31_5db]</b> , <b>[minus33db]</b> , <b>[minus34_5db]</b> , <b>[minus36db]</b> , <b>[minus37_5db]</b> , <b>[minus39db]</b> , <b>[minus39db]</b> , <b>[minus42db]</b> , <b>[minus48db]</b> , <b>[minus54db]</b> , <b>[MUTE]</b> The default value is <b>0db</b> .

Parameter	Description
<b>voip/audio/gain/handsfree_analog_input_gain</b>	Analog input gain (in dB). Valid values: [0db], [plus1_5db], [plus3db], [plus4_5db], [plus6db], [plus7_5db], [plus9db], [plus10_5db], [plus12db], [plus13_5db], [plus15db], [plus16_5db], [plus18db], [plus19_5db], [plus21db], [plus22_5db], [plus24db], [plus25_5db], [plus27db], [plus28_5db], [plus30db], [plus31_5db], [plus33db], [plus34_5db], [plus36db], [plus37_5db], [plus39db], [plus40_5db], [plus42db], [plus48db], [plus54db], [MUTE] The default value is <b>plus33db</b> .
<b>Handset Gain Parameters</b> <b>Note:</b> Values are in decibels (dB) and represented as follows: <ul style="list-style-type: none"> <li>▪ Negative values: use the word “minus” (e.g. =minus9db).</li> <li>▪ Positive values: use the word “plus” (e.g. =plus9db).</li> </ul> Decimal places: use underscore instead of period (e.g. plus19_5db).	
<b>voip/audio/gain/handset_digital_output_gain</b>	Digital output gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).
<b>voip/audio/gain/handset_digital_input_gain</b>	Digital input gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).
<b>voip/audio/gain/handset_analog_output_gain</b>	Analog output gain (in dB). Valid values: [0db], [minus1_5db], [minus3db], [minus4_5db], [minus6db], [minus7_5db], [minus9db], [minus10_5db], [minus12db], [minus13_5db], [minus15db], [minus16_5db], [minus18db], [minus19_5db], [minus21db], [minus22_5db], [minus24db], [minus25_5db], [minus27db], [minus28_5db], [minus30db], [minus31_5db], [minus33db], [minus34_5db], [minus36db], [minus37_5db], [minus39db], [minus39db], [minus42db], [minus48db], [minus54db], [MUTE] The default value is <b>minus9db</b> .
<b>voip/audio/gain/handset_analog_input_gain</b>	Analog input gain (in dB). Valid values: [0db], [plus1_5db], [plus3db], [plus4_5db], [plus6db], [plus7_5db], [plus9db], [plus10_5db], [plus12db], [plus13_5db], [plus15db], [plus16_5db], [plus18db], [plus19_5db], [plus21db], [plus22_5db], [plus24db], [plus25_5db], [plus27db], [plus28_5db], [plus30db], [plus31_5db], [plus33db], [plus34_5db], [plus36db], [plus37_5db], [plus39db], [plus40_5db], [plus42db], [plus48db], [plus54db], [MUTE] The default value is <b>plus19_5db</b> .
<b>voip/audio/gain/handset_analog_sidetone_gain</b>	Analog side tone gain (in db). Valid values: [minus9db], [minus12db], [minus15db], [minus18db], [minus21db], [minus24db], [minus27db], [MUTE] The default value is <b>minus12db</b> .

Parameter	Description
<b>Headset Gain Parameters</b>	
<p><b>Note:</b> Values are in decibels (dB) and represented as follows:</p> <ul style="list-style-type: none"> <li>Negative values: use the word "minus" (e.g. =minus9db).</li> <li>Positive values: use the word "plus" (e.g. =plus9db).</li> </ul> <p>Decimal places: use underscore instead of period (e.g. plus19_5db).</p>	
<b>voip/audio/gain/headset_digital_output_gain</b>	<p>Digital output gain (in dB).</p> <p>The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).</p>
<b>voip/audio/gain/headset_digital_input_gain</b>	<p>Digital input gain (in dB).</p> <p>The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).</p>
<b>voip/audio/gain/headset_analog_output_gain</b>	<p>Analog output gain (in dB).</p> <p>Valid values: [0db], [minus1_5db], [minus3db], [minus4_5db], [minus6db], [minus7_5db], [minus9db], [minus10_5db], [minus12db], [minus13_5db], [minus15db], [minus16_5db], [minus18db], [minus19_5db], [minus21db], [minus22_5db], [minus24db], [minus25_5db], [minus27db], [minus28_5db], [minus30db], [minus31_5db], [minus33db], [minus34_5db], [minus36db], [minus37_5db], [minus39db], [minus39db], [minus42db], [minus48db], [minus54db], [MUTE]</p> <p>The default value is <b>minus12db</b>.</p>
<b>voip/audio/gain/headset_analog_input_gain</b>	<p>Analog input gain (in dB).</p> <p>Valid values: [0db], [plus1_5db], [plus3db], [plus4_5db], [plus6db], [plus7_5db], [plus9db], [plus10_5db], [plus12db], [plus13_5db], [plus15db], [plus16_5db], [plus18db], [plus19_5db], [plus21db], [plus22_5db], [plus24db], [plus25_5db], [plus27db], [plus28_5db], [plus30db], [plus31_5db], [plus33db], [plus34_5db], [plus36db], [plus37_5db], [plus39db], [plus40_5db], [plus42db], [plus48db], [plus54db], [MUTE]</p> <p>The default value is <b>plus33db</b>.</p>
<b>voip/audio/gain/headset_analog_sidetone_gain</b>	<p>Analog side tone gain (in db).</p> <p>Valid values: [minus9db], [minus12db], [minus15db], [minus18db], [minus21db], [minus24db], [minus27db], [MUTE]</p> <p>The default value is <b>minus12db</b>.</p>
<b>Automatic Gain Control (AGC)</b>	
<b>voip/audio/gain/automatic_gain_control/enabled</b>	<p>Enables the AGC. AGC automatically adjusts the phone's voice volume to compensate for weak or loud signals.</p> <ul style="list-style-type: none"> <li>[0] Disable (default)</li> <li>[1] Enable</li> </ul>
<b>voip/audio/gain/automatic_gain_control/direction</b>	<p>Determines whether AGC is located before the encoder input (CTL_LOCAL) or after the decoder output (CTL_REMOTE).</p> <ul style="list-style-type: none"> <li>[CTL_REMOTE] For Remote User (default) - AGC is located after the Decoder output</li> <li>[CTL_LOCAL] For Local User - AGC is located before the Encoder input</li> </ul>
<b>voip/audio/gain/automatic_gain_control/target_energy</b>	<p>The required output energy (in dBm) of the AGC.</p> <p>The valid range is -31 to 0 (dB). The default value is -19.</p>

### 4.2.5.3 Codec Parameters

**Table 4-7: Codec Configuration File Parameters**

Parameter	Description
<b>voip/codec/codec_info/%d/enabled</b>	<p>Determines the codecs that you want to implement and their priority. Up to five codecs can be configured, where the first codec (i.e., <b>voip/codec/0/...</b>) has the highest priority. To make a call, at least one codec must be configured. In addition, for best performance it is recommended to select as many codecs as possible.</p> <p>When you start a call to a remote party, your available codecs are compared with the remote party's to determine the codec to use. If there is no codec that both parties have made available, the call attempt fails. Note that if more than one codec is common to both parties, you cannot force which of the common codecs are used by the remote party's client. If you do wish to force the use of a specific codec, configure the list with only that specific codec.</p> <p>The <i>%d</i> variable stands for the priority:</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> - Disabled</li> <li>▪ <b>[1]</b> (default) - Enabled</li> </ul>
<b>voip/codec/codec_info /%d/name</b>	<p>Name of the codec. The variable <i>%d</i> depicts the index number of the codec entry and its priority, where the first codec (i.e. <b>voip/codec/codec_info/0/name=...</b>) has the highest priority. The default order of codec priority from highest to lowest is G.722, G.711 Mu-Law, G.711 A-Law, G.729, and then G.723.</p> <ul style="list-style-type: none"> <li>▪ <b>[G722]</b> G.722</li> <li>▪ <b>[PCMA]</b> G.711 A-Law</li> <li>▪ <b>[PCMU]</b> G.711 Mu-Law</li> <li>▪ <b>[G729]</b> G.729</li> <li>▪ <b>[G723]</b> G.723</li> </ul> <p>For example, <b>voip/codec/codec_info/0/name=G722</b>.</p>
<b>voip/codec/codec_info /%d/ptime</b>	<p>Packetization time - length of the digital voice segment that each packet holds.</p> <p>The default is 20 millisecond packets, excluding G.723 which is 30 millisecond packets.</p>
<b>voip/codec/g723_bitrate</b>	<p>Low or high bit rate for G.723.</p> <ul style="list-style-type: none"> <li>▪ <b>[LOW]</b> Low</li> <li>▪ <b>[HIGH]</b> High (default)</li> </ul>
<b>voip/codec/g722_bitrate</b>	<p>G.722 bit rate.</p> <ul style="list-style-type: none"> <li>▪ <b>[G722_64K]</b> (default)</li> <li>▪ <b>[G722_56K]</b></li> <li>▪ <b>[G722_48K]</b></li> </ul> <p><b>Note:</b> Currently, only 64bps is supported.</p>

#### 4.2.5.4 Media Streaming Parameters

**Table 4-8: Media Streaming Configuration File Parameters**

Parameter	Description
<b>voip/media/dtmf_payload</b>	Defines the RTP payload type used for RFC 2833 DTMF relay packets. The valid range is 96 to 127. The default value is 101.
<b>voip/media/media_port</b>	Defines the starting port range for Real Time Protocol (RTP) voice transport. The valid range is 1024 to 65535. The default value is 4000.
<b>voip/media/rtp_mute_on_hold</b>	Mute sending RTP packets to remote in HOLD state. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> - Disabled. RTP packets are sent to remote end when in HOLD state.</li> <li>▪ <b>[1]</b> (default) - Enabled. RTP packets are not sent to remote end when in HOLD state.</li> </ul>
<b>voip/media/out_of_band_dtmf</b>	DTMF transport mode. <ul style="list-style-type: none"> <li>▪ <b>[INBAND]</b> Inband</li> <li>▪ <b>[RFC2833]</b> RFC 2833 (default)</li> <li>▪ <b>[VIA_SIP]</b> Via SIP</li> </ul>
<b>Quality of Service (QoS)</b>	
<b>voip/media/media_tos</b>	QoS in hexadecimal format. This is a part of the IP header that defines the type of routing service to tag outgoing voice packets originated from the phone. It informs routers that this packet must receive a specific QoS. The default value is <b>0xb8</b> .  Values can be set in decimal (e.g. 184) or hexadecimal (e.g. 0xb8).

### 4.2.5.5 Dialing Parameters

**Table 4-9: Dialing Configuration File Parameters**

Parameter	Description
<b>voip/dialing/timeout</b>	<p>The duration (in seconds) of allowed inactivity between dialed digits. When you work with a proxy, the number you have dialed before the dialing process has timed out is sent to the proxy as the user ID to be called. This is useful for calling a remote party without creating a speed dial entry (assuming the remote party is registered with the proxy).</p> <p>The valid range is 0 to 10. The default value is 5.</p>
<b>voip/dialing/phone_number_max_size</b>	<p>The maximum length of shortcut numbers that you can enter and the maximum number of digits that you can dial</p> <p>The valid range is 0 to 32. The default value is 19.</p>
<b>voip/dialing/dial_complete_key/enabled</b>	<p>Enables the feature for defining a key to indicate that dialing has completed. Pressing the Dialing Complete key (defined below) forces the phone to make a call to the dialed digits even if there is no match in the dial plan or digit map.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable</li> <li>▪ <b>[1]</b> Enable (default)</li> </ul> <p><b>Note:</b> This parameter is available only if the parameter <b>voip/dialing/dial_complete_key/enabled</b> is set to 1.</p>
<b>voip/dialing/dial_complete_key/key</b>	<p>Defines the Dialing Complete key.</p> <p>The valid value is a single character. The default value is the pound (#) key.</p>
<b>voip/dialing/dialtone_timeout</b>	<p>The maximum duration of the dial tone (in seconds) after which the dial tone stops and a reorder tone is played.</p> <p>The valid range is 0 to 300. The default value is 30.</p>
<b>voip/dialing/warning_tone_timeout</b>	<p>The maximum duration of the reorder tone (in seconds) after which the reorder tone stops and a howler tone is played.</p> <p>The valid range is 0 to 300. The default value is 40.</p>
<b>voip/dialing/unanswered_call_timeout</b>	<p>Timeout before the phone automatically sends a Cancel message. When the phone makes a call and the other side doesn't answer, the phone sends a Cancel after this timeout</p> <p>The valid range is 0 to 300. The default value is 60.</p>
<b>voip/dialing/offhook_tone_timeout</b>	<p>The duration (in seconds) of the howler tone. If the limit is exceeded, the howler tone stops. The howler tone indicates that the phone has been left in an off-hook state.</p> <p>The valid range is 0 to 300. The default value is 120.</p>
<b>voip/dialing/secondary_dial_tone/enabled</b>	<p>Enables the secondary dial tone.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable - Phone doesn't use secondary dial tone.</li> <li>▪ <b>[1]</b> Enable (default) - Phone plays secondary dial tone if the secondary dial tone key is pressed (first digit).</li> </ul>
<b>voip/dialing/secondary_dial_tone/key_sequence</b>	<p>Secondary dial tone is played if this is the first key pressed.</p> <p>The valid range is 0 to 9. The default value is 9.</p> <p><b>Note:</b> This parameter is available only if the parameter <b>voip/dialing/secondary_dial_tone/enabled</b> is set to 1.</p>

Parameter	Description
<b>voip/dialing/automatic_disconnect</b>	Determines whether the phone automatically goes idle (i.e. on-hook) when the last remaining call is disconnected. This is only relevant when the speaker or headset is used. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable</li> <li>▪ <b>[1]</b> Enable(default)</li> </ul>
<b>Automatic Dialling</b>	
<b>voip/dialing/auto_dialing/enabled</b>	Determines whether automatic dialing is enabled (i.e., phone number is automatically dialed when you off-hook the phone). <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/dialing/auto_dialing/timeout</b>	Timeout (in seconds) before automatic dialing occurs after the phone is off-hooked. When set to 0, automatic dialing is performed immediately. The valid range is 0 to 120. The default value is 15.
<b>voip/dialing/auto_dialing/destination</b>	The number that is automatically dialed when the phone is off-hooked. The valid value can be up to 32 characters.

#### 4.2.5.6 SIP Signaling Parameters

**Table 4-10: SIP Configuration File Parameters**

Parameter	Description
<b>voip/signalling/sip/transport_protocol</b>	Determines the transport layer for outgoing SIP calls initiated by the phone. <ul style="list-style-type: none"> <li>▪ <b>[UDP]</b> UDP (default)</li> <li>▪ <b>[TCP]</b> TCP</li> <li>▪ <b>[TLS]</b> TLS</li> </ul>
<b>voip/signalling/sip/port</b>	Defines the local SIP port (UDP, TCP or TLS) for SIP messages. The valid range is 1024 to 65535. The default value is 5060.
<b>voip/signalling/sip/proxy_gateway</b>	Assigns a name to the phone. The name is used as the host part of the SIP URI in the From header. <p><b>Notes:</b></p> <ul style="list-style-type: none"> <li>▪ Ensure that the name you choose is the one with which the Proxy is configured to identify the phone.</li> <li>▪ If not specified, the phone's IP address is used (default).</li> </ul>
<b>voip/signalling/sip/prack/enabled</b>	Determines whether the phone sends PRACK (Provisional Acknowledgment) messages upon receipt of 1xx SIP reliable responses. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable</li> <li>▪ <b>[1]</b> Enable (default)</li> </ul>
<b>voip/signalling/sip/rport/enabled</b>	Determines whether the phone adds the 'rport' parameter to the relevant SIP message (in the SIP Via header). <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>

Parameter	Description
<b>voip/signalling/sip/sdp_include_ptime</b>	Determines whether the phone adds the PTIME parameter to the SDP message body. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/signalling/sip/keepalive_options/enabled</b>	Determines whether keep-alive is performed using SIP OPTIONS messages sent to the Proxy. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/signalling/sip/keepalive_options/timeout</b>	Defines the Proxy keep-alive time interval (in seconds) between Keep-Alive messages. The valid range is 0 to 86400. The default value is 300.
<b>voip/signalling/sip/connect_media_on_180</b>	Determines whether the media is connected upon receipt of SIP 180, 183, or 200 messages. When the parameter is disabled, media is connected upon receipt of 183 and 200 messages only. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/signalling/sip/block_callerid_on_outgoing_calls</b>	When enabled, the outgoing INVITE message is sent with an anonymous From header and P-Asserted-Identity header. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul> For example: <ul style="list-style-type: none"> <li>▪ FROM header contains anonymous URI: <i>From: "Anonymous" sip:anonymous@anonymous.invalid</i></li> <li>▪ P-Asserted-Identity header: <i>P-Asserted-Identity: "1001" 1115551001@proxy.net</i></li> </ul>
<b>voip/signalling/sip/anonymous_calls_blocking</b>	When enabled, incoming INVITE messages with anonymous From header are rejected. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul> For example: <i>From: "Anonymous" &lt;sip:anonymous@anonymous.invalid&gt;</i> The phone responds with a SIP 403 "Forbidden" response.
<b>voip/signalling/sip/authentication_retries</b>	Defines the number of times authenticated register messages are re-sent if 401 or 407 SIP responses with a different "nonce" are received. The valid range is 0 to 100. The default value is 4.
<b>voip/signalling/sip/display_name_in_registration_msg/enabled</b>	Sets the Display Name in the "To" and "From" fields of the SIP REGISTER message. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>

Parameter	Description
<b>voip/signalling/sip/digit_map</b>	<p>Enables the administrator to predefine possible formats (or patterns) for the dialed number. A match to one of the defined patterns terminates the dialed number.</p> <p>The valid value can be up to 256 characters.</p> <p>There are two main formats for the digit map configuration. The formats are distinguished by the separator ';' or ' '.</p> <ul style="list-style-type: none"> <li>▪ <b>Using ' ' separator:</b> The following constructs can be used in each numbering scheme: <ul style="list-style-type: none"> <li>✓ <b>Digit:</b> A digit from "0" to "9".</li> <li>✓ <b>DTMF:</b> A digit, or one of the symbols "A", "B", "C", "D", "#", or "*". Extensions may be defined.</li> <li>✓ <b>Wildcard:</b> The symbol "x" which matches any digit ("0" to "9").</li> <li>✓ <b>* Range:</b> One or more DTMF symbols enclosed between square brackets "[" and "]").</li> <li>✓ <b>Sub range:</b> Two digits separated by hyphen ("-") which matches any digit between and including the two. The subrange construct can only be used inside a range construct, i.e., between "[" and "]".</li> <li>✓ <b>Position:</b> A period (".") which matches an arbitrary number, including zero, of occurrences of the preceding construct.</li> </ul> <p><b>For example:</b> [2-9]11 0 100 101 011xxx. 9011xxx. 1[2-9]xxxxxxxx 91[2-9]xxxxxxxx 9[2-9]xxxxx *xx [8]xxxx [2-7]xxx</p> <p>This example includes the following rules:</p> <ul style="list-style-type: none"> <li>✓ <b>[2-9]11:</b> 911 rule: 211, 311, 411, 511, 611, 711, 811, 911 are dialed immediately</li> <li>✓ <b>0:</b> Local operator rule: After dialing "0" the phone waits T seconds and then completes the call automatically</li> <li>✓ <b>100:</b> Auto-attendant default extension</li> <li>✓ <b>101:</b> Voicemail default extension</li> <li>✓ <b>011xxx.:</b> International rule without prefix</li> <li>✓ <b>9011xxx.:</b> International rule with prefix</li> <li>✓ <b>1[2-9]xxxxxxxx:</b> LD rule without prefix</li> <li>✓ <b>91[2-9]xxxxxxxx:</b> LD rule with prefix</li> <li>✓ <b>9[2-9]xxxxx:</b> Local call with prefix</li> <li>✓ <b>*xx:</b> 2-digit star codes</li> <li>✓ <b>[1-7]xx:</b> A regular 3 digit extension that does not start with 9 or 8 is dialed immediately</li> <li>✓ <b>[2-7]xx:</b> A regular 3 digit extension that does not start with 9 or 8 or 1 is dialed immediately</li> <li>✓ <b>[2-7]xxx:</b> A regular 4 digit extension that does not start with 9 or 8 or 1 is dialed immediately</li> <li>✓ <b>[8]xxx:</b> A 3 digit extension prefixed with an 8 (routes calls directly to voicemail of extension xxx)</li> <li>✓ <b>[8]xxxx:</b> A 4 digit extension prefixed with an 8 (routes calls directly to voicemail of extension xxxx)</li> <li>✓ <b>T:</b> Refers to the <b>Dialing Timeout</b>.</li> </ul> </li> <li>▪ <b>Using ';' separator:</b> An 'x' in the pattern indicates any digit. ';' separates between patterns.  For example: '10x;05xxxxxxxx;4xxx'.  In this example, three patterns are defined. A number that starts with 10 is terminated after the third digit, and so on. If the user dials a number that does not match any pattern, the number is terminated using the timeout or when the user presses the pound ('#') key.</li> </ul>

Parameter	Description
<b>voip/signalling/sip/number_rules</b>	<p>This parameter works in conjunction with the parameter <b>voip/signalling/sip/digit_map</b> and enables translation of specific patterns to specific SIP destination addresses. An 'x' represents any dialed digit. Each backslash at the right side of the '=' represents one of the dialed digits. Rules are separated by the character ';'.</p> <p>The valid value can be up to 256 characters.</p> <p>For example: '4xxx=Line_\\@10.1.2.3'</p> <p>This rule issues a call to 10.1.2.3 with the SIP ID of Line_ followed by the last three digits of the dialed number.</p>
<b>SIP Proxy and Registrar</b>	
<b>voip/signalling/sip/use_proxy</b>	<p>Determines whether to use a SIP Proxy server.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/signalling/sip/proxy_address</b>	<p>The IP address or host name of the SIP proxy server.</p> <p>The default value is 0.0.0.0.</p>
<b>voip/signalling/sip/proxy_port</b>	<p>The UDP or TCP port of the SIP proxy server.</p> <p>The valid range is 1024 to 65535. The default value is 5060.</p>
<b>voip/signalling/sip/proxy_timeout</b>	<p>The SIP proxy server registration timeout (in seconds).</p> <p>The valid range is 0 to 86400. The default value is 3600.</p>
<b>voip/signalling/sip/sip_registrar/enabled</b>	<p>Determines whether the phone registers to a separate SIP Registrar server.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/signalling/sip/use_proxy_ip_port_for_registrar</b>	<p>Determines whether to use the SIP proxy's IP address and port for registration. When enabled, there is no need to configure the address of the registrar separately.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/signalling/sip/sip_registrar/addr</b>	<p>The IP address or host name of the Registrar server.</p> <p>The default value is 0.0.0.0.</p>
<b>voip/signalling/sip/sip_registrar/port</b>	<p>The UDP or TCP port of the Registrar server.</p> <p>The valid range is 1024 to 65535. The default value is 5060.</p>
<b>voip/signalling/sip/sip_outbound_proxy/enabled</b>	<p>Determines whether an outbound SIP proxy server is used (all SIP messages are sent to this server as the first hop).</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/signalling/sip/sip_outbound_proxy/addr</b>	<p>The IP address of the outbound proxy. If this parameter is set, all outgoing messages (including Registration messages) are sent to this Proxy according to the Stack behavior.</p> <p>The default value is 0.0.0.0.</p>
<b>voip/signalling/sip/sip_outbound_proxy/port</b>	<p>The port on which the outbound proxy listens.</p> <p>The valid range is 1024 to 65535. The default value is 5060.</p>

Parameter	Description
<b>voip/signalling/sip/redundant_proxy/enabled</b>	<p>Enables the redundant proxy mechanism.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default) - Phone doesn't use redundant proxy (default).</li> <li>▪ <b>[1] Enable</b> - Phone registered to redundant proxy if the primary proxy does not respond to SIP signaling messages.</li> </ul>
<b>voip/signalling/sip/redundant_proxy/address</b>	<p>The IP address or host name of the redundant proxy. The default value is 0.0.0.0.</p> <p><b>Note:</b> This parameter is applicable only if the parameter <b>use_redundant_proxy</b> is set to 1.</p>
<b>voip/signalling/sip/redundant_proxy/port</b>	<p>The UDP or TCP port of the redundant proxy server. The valid range is 1024 to 65535. The default value is 5060.</p> <p><b>Note:</b> This parameter is applicable only if the parameter <b>use_redundant_proxy</b> is set to 1.</p>
<b>voip/signalling/sip/redundant_proxy/keepalive_period</b>	<p>Defines the interval in seconds for sending keep-alive messages to the proxy. The valid range is 0 to 300. The default value is 60.</p> <p><b>Note:</b> This parameter is applicable only if the parameter <b>use_redundant_proxy</b> is set to 1.</p>

Parameter	Description
<b>voip/signalling/sip/redundant_proxy/symmetric_mode</b>	<p>Determines the proxy redundancy mode.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> (default) - Asymmetric mode.</li> <li>▪ <b>[1]</b> - Symmetric mode.</li> </ul> <p>The Redundant Proxy feature allows the configuration of a backup SIP proxy server to increase QoS stability. Once this feature is enabled, the phone identifies cases where the primary proxy does not respond to SIP signaling messages. In these scenarios, the phone registers to the redundant proxy and the phone seamlessly continues normal functionality, without the user noticing any connectivity failure or malfunction with the primary proxy.</p> <p>The Redundant Proxy feature includes two operational modes:</p> <ul style="list-style-type: none"> <li>▪ <b>Asymmetric mode:</b> the primary proxy is assigned a higher priority for registration than the redundant proxy. In the Asymmetric mode, once the phone is registered to the primary proxy, it sends keep-alive messages (using SIP OPTIONS messages) to the primary proxy. If the primary proxy does not respond, the phone registers to the redundant proxy, but continues sending keep-alive messages to the primary proxy. If the primary proxy responds to these keep-alive messages, the phone re-registers to the primary proxy. Therefore, the phone assigns the primary proxy a higher priority for registration.</li> <li>▪ <b>Symmetric mode:</b> both proxies are assigned the same priority for registration. In the Symmetric mode, once the phone is registered to a proxy, it sends keep-alive messages to this proxy. The phone switches proxies only once the proxy to which it has registered does not respond. Therefore, the phone assigns both proxies the same priority for registration</li> </ul> <p>In both modes, the following applies:</p> <ul style="list-style-type: none"> <li>▪ If the phone is not registered (i.e., if the proxy server—redundant or primary—to which the phone currently tries to register does not respond), the phone attempts to register to an alternative proxy. These attempts continue until the phone successfully registers.</li> <li>▪ If this feature is enabled and the user reboots the phone, the phone registers to the last proxy to which it was trying to register, and not necessarily to the primary proxy.</li> </ul> <p><b>Note:</b> This parameter is applicable only if the parameter <b>use_redundant_proxy</b> is set to 1.</p>

Parameter	Description
<b>SIP Timers</b>	
<b>voip/signalling/sip/sip_t1</b>	<p>The time interval (in msec) between the first transmission of a SIP message and the first retransmission of the same message (according to RFC 3261).</p> <p>The valid range is 100 to 60000. The default value is 500.</p> <p><b>Note:</b> The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx. For example (assuming that SipT1Rtx = 500 and SipT2Rtx = 4000):</p> <ul style="list-style-type: none"> <li>▪ The first retransmission is sent after 500 msec.</li> <li>▪ The second retransmission is sent after 1000 (2*500) msec.</li> <li>▪ The third retransmission is sent after 2000 (2*1000) msec.</li> <li>▪ The fourth retransmission and subsequent retransmissions until SIPMaxRtx are sent after 4000 (2*2000) msec..</li> </ul>
<b>voip/signalling/sip/sip_t2</b>	<p>The maximum interval (in msec) between retransmissions of SIP messages (according to RFC 3261).</p> <p>The valid range is 4000 to 60000. The default value is 200.</p> <p><b>Note:</b> The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx.</p>
<b>voip/signalling/sip/sip_t4</b>	<p>The SIP T4 retransmission timer according to RFC 3261.</p> <p>The valid range is 5000 to 60000. The default value is 1000.</p>
<b>voip/signalling/sip/sip_invite_timer</b>	<p>The SIP INVITE timer according to RFC 3261.</p> <p>The valid range is 0 to 65535. The default value is 32000.</p>
<b>voip/signalling/sip/session_timer</b>	<p>The time (in seconds) at which an element considers the call timed out if no successful INVITE transaction occurs beforehand. This value is inserted into every INVITE in the Session-Expires header unless it is configured to 0. If the timer option tag is not part of the supported list, the sessionExpires value is ignored.</p> <p>The valid range is 0 to 65535. The default value is 1800.</p>
<b>voip/signalling/sip/min_session_interval</b>	<p>The minimum value for the session interval that the application is willing to accept.</p> <p>The valid range is 0 to 65535. The default value is 900.</p>
<b>SIP QoS</b>	
<b>voip/signalling/sip/tos</b>	<p>QoS in hexadecimal format. This is a part of the IP header that defines the type of routing service to tag outgoing signalling packets originated from the phone. It informs routers that this packet must receive a specific QoS.</p> <p>The default value is 0x60.</p> <p>Values can be set in decimal (e.g. 96) or hexadecimal (e.g. 0x60).</p>

### 4.2.5.7 Voice

**Table 4-11: Voice Configuration File Parameters**

Parameter	Description
<b>voip/audio/jitter_buffer/min_delay</b>	The initial and minimal delay of the adaptive jitter buffer mechanism, which compensates for network problems. The value should be set according to the expected average jitter in the network (in milliseconds). The valid range is 0 to 300. The default value is 35.
<b>voip/audio/jitter_buffer/optimization_factor</b>	The adaptation rate of the jitter buffer mechanism. Higher values cause the jitter buffer to respond faster to increased network jitter. The valid range is 0 to 13. The default value is 7.
<b>voip/audio/echo_cancellation/enabled</b>	Enables echo cancellation. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> - Disable</li> <li>▪ <b>[1]</b> (default) – Enable</li> </ul> <b>Note:</b> Disabling echo cancellation should be done for testing purposes only.
<b>voip/audio/silence_compression/enabled</b>	Enables silence compression for reducing network bandwidth consumption. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>

### 4.2.5.8 Supplementary Services Parameters

**Table 4-12: Supplementary Services Configuration File Parameters**

Parameter	Description
<b>voip/services/application_server_type</b>	Defines the type of the Application server to which the device is registered. <ul style="list-style-type: none"> <li>▪ <b>[Generic]</b> Generic (default)</li> <li>▪ <b>[Asterisk]</b> Asterisk</li> <li>▪ <b>[BSFT]</b> Broadsoft</li> </ul> <b>Note:</b> Currently, this parameter is not applicable (only Generic is supported).
<b>voip/services/out_of_service_behavior</b>	Determines whether a reorder tone is played instead of a dial tone if you configured a Registrar IP address and the registration failed. <ul style="list-style-type: none"> <li>▪ <b>[NONE]</b> No Tone</li> <li>▪ <b>[REORDER_TONE]</b> Reorder Tone (default)</li> </ul>
<b>Call Waiting</b>	
<b>voip/services/call_waiting/enabled</b>	Enables the Call Waiting feature. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable</li> <li>▪ <b>[1]</b> Enable (default)</li> </ul>
<b>voip/services/call_waiting/sip_reply</b>	Determines the SIP response that is sent when another call arrives while a call is in progress: <ul style="list-style-type: none"> <li>▪ <b>[RINGING]</b> Ringing - 180 Ringing</li> <li>▪ <b>[QUEUED]</b> Queued (default) - 182 Queued</li> </ul>
<b>Call Forward</b>	
<b>voip/services/call_forward/line/0/enabled</b>	Enables the Call Forward feature. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable</li> <li>▪ <b>[1]</b> Enable (default)</li> </ul>

Parameter	Description
<b>voip/services/call_forward/line/0/type</b>	Determines the condition upon which incoming calls are forwarded to another destination: <ul style="list-style-type: none"> <li>▪ <b>[Unconditional]</b> Unconditional - incoming calls are forwarded independently of the status of the line.</li> <li>▪ <b>[Busy]</b> Busy - incoming calls are forwarded only if the phone is busy.</li> <li>▪ <b>[No_Reply]</b> No Reply (default) - incoming calls are forwarded only if the phone does not answer before a user-defined timeout.</li> </ul>
<b>voip/services/call_forward/line/0/timeout</b>	If calls are forwarded when the condition is No-Reply, then this parameter defines the time (in seconds) after which incoming calls are forwarded when this is no reply. The valid range is 0 to 7200. The default value is 6.
<b>voip/services/call_forward/line/0/destination</b>	The destination to which the call is directed when call forward is activated.
<b>voip/services/call_forward/line/0/active</b>	Activates call forwarding, if it has been enabled (using the parameter <b>voip/services/call_forward/line/0/enabled</b> ). <ul style="list-style-type: none"> <li>▪ <b>[0]</b> (default) - Disable</li> <li>▪ <b>[1]</b> - Enable</li> </ul> <b>Note:</b> Call forwarding can also be activated using the LCD screen (common).
<b>Message Waiting Indication (MWI)</b>	
<b>voip/services/msg_waiting_ind/enabled</b>	Enables the MWI feature. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable</li> <li>▪ <b>[1]</b> Enable (default)</li> </ul>
<b>voip/services/msg_waiting_ind/subscribe</b>	Determines whether the phone registers to an MWI server. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<b>voip/services/msg_waiting_ind/subscribe_port</b>	The port number of the MWI server. The valid range is 1024 to 65535. The default value is 5060.
<b>voip/services/msg_waiting_ind/subscribe_address</b>	The IP address or host name of the MWI server. The default value is 0.0.0.0.
<b>voip/services/msg_waiting_ind/expiration_timeout</b>	The interval between the MWI Subscribe messages. The valid range is 0 to 86400. The default value is 3600.
<b>voip/services/msg_waiting_ind/stutter_tone_duration</b>	Defines the duration for which a stutter tone is played when you have unheard messages. The valid range is 0 to 7200. The default value is 2500.
<b>voip/services/msg_waiting_ind/voice_mail_number</b>	Defines the extension number for accessing your voice mail messages. The valid value is up to 64 characters.
<b>Do Not Disturb (DND)</b>	
<b>voip/services/do_not_disturb/enabled</b>	Enables the Do not Disturb feature. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable</li> <li>▪ <b>[1]</b> Enable (default)</li> </ul>

## 4.2.6 Regional Settings Parameters

**Table 4-13: Regional Configuration File Parameters**

Parameter	Description
<code>voip/regional_settings/selected_country</code>	<p>Defines the country in which your phone is located. The behavior and parameters of analog telephones lines vary between countries. The set of Call Progress Tones are all location-specific. The phone automatically selects the correct regional settings according to this setting.</p> <p>Supported countries:</p> <ul style="list-style-type: none"> <li>▪ <b>[Israel]</b> Israel</li> <li>▪ <b>[China]</b> China</li> <li>▪ <b>[France]</b> France</li> <li>▪ <b>[Germany]</b> Germany</li> <li>▪ <b>[Netherlands]</b> Netherlands</li> <li>▪ <b>[UK]</b> UK</li> <li>▪ <b>[Brazil]</b> Brazil</li> <li>▪ <b>[Italy]</b> Italy</li> <li>▪ <b>[Argentina]</b> Argentina</li> <li>▪ <b>[Portugal]</b> Portugal</li> <li>▪ <b>[USA]</b> USA</li> </ul>
<code>voip/regional_settings/user_config_file_values</code>	<p>Enables the user-defined CPT. When this parameter is enabled, the <b>selected_country</b> parameter is not relevant and the below Call Progress Tones values can be determined by the user.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> (default) - Disable</li> <li>▪ <b>[1]</b> - Enable</li> </ul>
<b>Call Progress Tones (CPT)</b>	
<b>Note:</b> Up to 10 CPT's can be configured ( <code>voip/regional_settings/call_progress_tones/0...9</code> ).	
<code>voip/regional_settings/call_progress_tones/%d/enabled</code>	<p>Enables the specific CPT.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> - Disable</li> <li>▪ <b>[1]</b> - Enable</li> </ul>
<code>voip/regional_settings/call_progress_tones/%d/name</code>	<p>Defines the name of the CPT.</p>
<code>voip/regional_settings/call_progress_tones/%d/cadence</code>	<p>Defines the cadence type of the tone.</p> <ul style="list-style-type: none"> <li>▪ <b>[0]</b> - Continuous signal</li> <li>▪ <b>[1]</b> - Cadence signal</li> <li>▪ <b>[2]</b> - Burst signal</li> </ul>
<code>voip/regional_settings/call_progress_tones/%d/frequency_a</code>	<p>Defines the low frequency (in Hz) of the tone.</p> <p>The valid value range is 300 to 1980 Hz, in steps of 1 Hz. Unused frequencies must be set to zero.</p>
<code>voip/regional_settings/call_progress_tones/%d/frequency_b</code>	<p>Defines the high frequency (in Hz) of the tone.</p> <p>The valid value range is 300 to 3000 Hz, in steps of 1 Hz. Unused frequencies must be set to zero.</p>
<code>voip/regional_settings/call_progress_tones/%d/frequency_a_level</code>	<p>Output level of the low frequency tone (in -dBm) in Call Progress generation.</p> <p>The valid range is 0 to 63, where 63 is mute.</p>

Parameter	Description
<b>voip/regional_settings/call_progress_tones/%d/frequency_b_level</b>	Output level of the low frequency tone (in -dBm) in Call Progress generation. The valid range is 0 to 63, where 63 is mute.
<b>voip/regional_settings/call_progress_tones/%d/tone_on_0</b>	tone_on_0 to tone_on_3. If the signal is Cadence or Burst, then this value represents the on duration. If a Continuous tone, then this value represents the minimum detection time. The units are in 10 msec. The valid range is 0 to 10000.
<b>voip/regional_settings/call_progress_tones/%d/tone_off_0</b>	tone_off_0 to tone_on_3. If the signal is Cadence, then this value represents the off duration. The units are in 10 msec. If it is not used, then set it to zero. If the signal is Burst, only tone_off_0 is relevant. It represents the off time that is required from the end of the signal to the detection time. The valid range is 0 to 10000.

## 4.2.7 Packet Recording (Debugging) Parameters

Table 4-14: Packet Recording (Debugging) Configuration File Parameters

Parameter	Description
<code>voip/packet_recording/enabled</code>	Activates the packet recording mechanism. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<code>voip/packet_recording/remote_ip</code>	The IP address (in dotted-decimal notation) of the remote computer to which the recorded packets are sent. The recorded packets should be captured by a network sniffer (such as Wireshark). The default value is 0.0.0.0.
<code>voip/packet_recording/remote_port</code>	Defines the UDP port of the remote computer to which the recorded packets are sent. The valid range is 1024 to 65535. The default value is 50000.
<code>voip/packet_recording/rtp_recording/enabled</code>	Activates the DSP RTP recording. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<code>voip/packet_recording/ec_debug_recording/enabled</code>	Activates the Echo Canceller Debug recording. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<code>voip/packet_recording/network_recording/enabled</code>	Activates the DSP network (TDM Out) recording. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>
<code>voip/packet_recording/tm_recording/enabled</code>	Activates the DSP TDM (TDM In) recording. <ul style="list-style-type: none"> <li>▪ <b>[0]</b> Disable (default)</li> <li>▪ <b>[1]</b> Enable</li> </ul>

## 4.2.8 LCD Display Parameters

Table 4-15: LCD Display Configuration File Parameters

Parameter	Description
<code>personal_settings/language</code>	Determines the LCD display user interface language. <ul style="list-style-type: none"> <li>▪ <b>[English]</b> English (default)</li> <li>▪ <b>[Spanish]</b> Spanish</li> </ul>

## 4.3 Creating a Corporate Directory File

The Configuration file can include a link to a user-defined Corporate Directory file, using the **provisioning/corporate\_directory\_uri** parameter. This allows you to upload a corporate directory to the phone.

The Corporate Directory file includes a list of contacts and their phone numbers. The file must be a simple text file that can be created using an Excel document and saved as a CSV file.

The syntax of the corporate directory file must be as follows:

```
<full name>,<office>,<home>,<mobile>,<user defined 1>  
,<user defined 2>
```

For example:

```
John Smith,1234,98765432,574685746,8888,9999
```

If not all phone numbers are required, the relevant field must be left empty. For example, in the below directory entry, the home and user-defined numbers are absent:

**John Weiss,1234,,574685746**

```
John Smith,1234,,574685746
```

## 4.4 Creating a Speed Dial File

The Configuration file can include a link to a user-defined Speed Dial file, using the **provisioning/speed\_dial\_uri** parameter. This allows you to upload speed dial settings to the phone

The Speed Dial file must include a list of speed dial configurations. The file must be a simple text file that can be created using an Excel document and saved as a CSV file.

The syntax of the speed dials file must be as follows:

```
<memory key serial number>,<memory key phone number>,<type>
```

The *type* variable represents speed dial and **must** be set to "0".

For example:

```
1,4418,0
2,4403,0
3,039764432,0
4,4391,0
.
.
12,1234,0
```

## 5 Built-in Test Plan

The 310HD IP Phone provides a built-in test for verifying its operational integrity. These tests can be performed before deployment and in cases of troubleshooting. These tests are all performed in one continuous sequence, where each test is followed by another consecutively, in the order listed below:

- Keypad and off-hook/on-hook test
- LED test
- Handset microphone (recording/speaking) and handset receiver (playing/listening) test
- Headset microphone (recording/speaking) and headset headphone/receiver (playing/listening) test
- Speaker microphone (recording/speaking) and speaker receiver (playing/listening) test
- MAC address and serial number verification

### 5.1 Test Preparation

Before you can start the built-in test, you need to cable the phone as follows:

1. Connect the phone's LAN port to a switch, using a LAN cable.
2. Ensure that the DHCP server is functioning.
3. Connect a headset to the phone.
4. Power on the phone and wait until initialization is complete.
5. From this stage onwards, follow the procedures described in the subsequent sections and in consecutive order. In other words, continue with the procedure described in Section 5.2 on page 82.

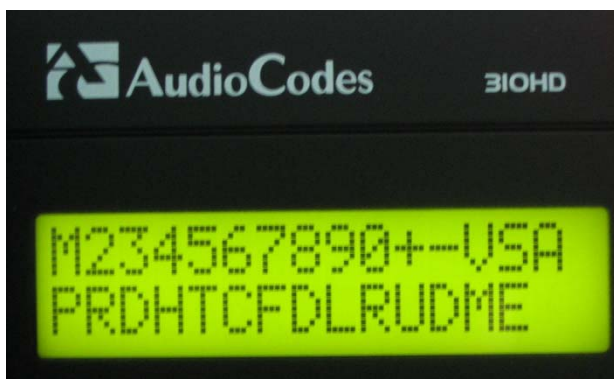
## 5.2 Keypad Test

The Keypad test checks the responsiveness and correct functioning of the keys on the phone's keypad.

➤ **To test the keypad:**

1. Using the keypad, dial the number "0123456789" and then press the star ("\*") key; the phone's LCD screen displays the following:

**Figure 5-1: Testing Keypad**



2. Press all keys except the keys 0-9, \*, and #; for every key pressed, its corresponding character displayed on the LCD disappears.

**Figure 5-2: Testing Keypad – Characters Disappear**



3. Continue with the LED test in Section 5.3 on page 82.

## 5.3 LED Test

Upon the successful completion of the Keypad test (in the previous section), the LCD screen displays the message "All LEDs are on. Press #" and the phone's single LED lights up red, as shown in the figure below:

**Figure 5-3: Successful Completion of Keypad Test**



- Continue with Handset test in Section 5.4 on page 83.

## 5.4 Handset Test

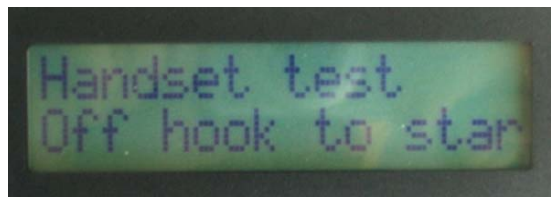
Upon successful completion of the LED test (in the previous section), perform the Handset test. This test verifies the correct functioning of the handset, which includes the following:

- Handset's microphone (transmitter) for speaking
- Handset's receiver (speaker) for listening

➤ **To test the handset:**

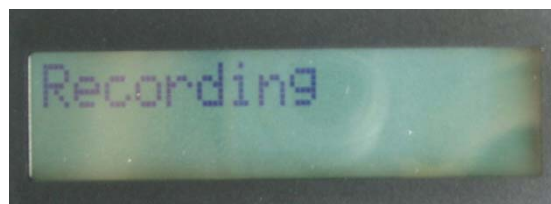
1. Press the pound (#) key; the LCD screen displays the message "Handset test. Off hook to start".

**Figure 5-4: Off hook Handset Test**



2. Off hook the phone; the LCD screen displays the message "Recording".

**Figure 5-5: Recording Test**



3. Speak into the handset microphone; after about five seconds, the LCD screen displays the message "Playing".

**Figure 5-6: Playing Test**



The voice that was recorded when you spoke into the handset microphone is now played from the handset receiver.

4. Continue with the Headset test in Section 5.5 on page 84.

## 5.5 Headset Test

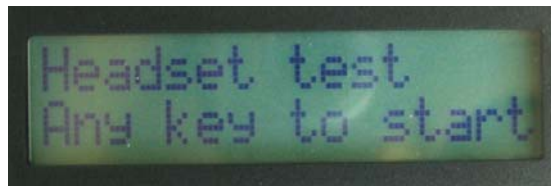
Upon successful completion of the Handset test (in the previous section), perform the Headset test. This test verifies the correct functioning of the headset (for hands-free operation), which includes the following:

- Headset's microphone (transmitter) for speaking
- Headset's headphone (receiver) for listening

➤ **To test the headset:**

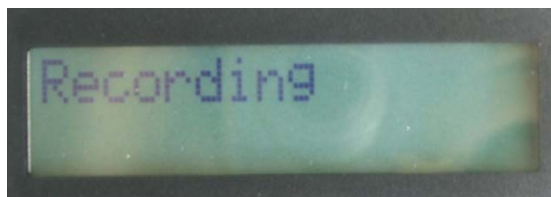
1. After about five seconds after your recorded voice is played in the Handset test (in the previous section), the LCD screen displays the message "Headset test. Any key to start".

**Figure 5-7: Headset Test**



2. Press any key; the LCD screen displays the message "Recording".

**Figure 5-8: Headset Recording Test**



3. Speak into the headset's microphone; after about five seconds, the LCD screen displays "Playing" and the voice that was recorded when you spoke into the headset's microphone is now played from the headset's receiver.

**Figure 5-9: Headset Playing Test**



4. Continue with the Speaker test in Section 5.6 on page 85.

## 5.6 Speaker Test

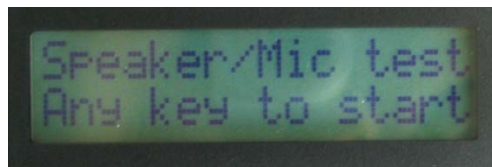
Upon successful completion of the Headset test (in the previous section), perform the Speaker test. This test verifies the correct functioning of the speakers, which includes the following:

- Speaker microphone (transmitter) for speaking (recording)
- Speaker receiver for listening (playing)

➤ **To test the speaker:**

1. After about five seconds after your recorded voice is played in the Headset test (in the previous section), the LCD screen displays the message "Speaker/Mic test. Any key to start".

**Figure 5-10: Speaker Test**



2. Press any key; the LCD screen displays the message "Recording".

**Figure 5-11: Speaker Recording Test**



3. Speak into the phone's speaker microphone; after about five seconds, the LCD screen displays the message "Playing", and the voice that was recorded when you spoke into the speaker's microphone is now played from the speaker's receiver.

**Figure 5-12: Speaker Playing Test**



4. Continue with the MAC Address and Serial Number Verification test in Section 5.7 on page 86.

## 5.7 MAC Address and Serial Number Verification Test

The MAC Address and Serial Number Verification test allows you to verify that the phone's MAC address and serial number are correct.

➤ **To verify the MAC address and serial number:**

1. After about five seconds after your recorded voice is played in the Speaker test (in the previous section), the LCD screen displays a MAC address and serial number:

**Figure 5-13: MAC Address and Serial Number**



2. Check that the LAN MAC address and serial number are correct.
3. Press any key to exit the phone's built-in test.

## 6 Specifications

**Table 6-1: 310HD IP Phone Specifications**

Feature	Details
<b>VoIP Signaling Protocols</b>	<ul style="list-style-type: none"> <li>▪ SIP: RFC 3261, RFC 2327 (SDP)</li> </ul>
<b>Data Protocols</b>	<ul style="list-style-type: none"> <li>▪ IPv4, TCP, UDP, ICMP, ARP, DNS and DNS SRV for SIP Signaling</li> <li>▪ SIP over TLS (SIPS)</li> <li>▪ 802.1p/Q for Traffic Priority and QoS</li> <li>▪ ToS (Type of Service) field, indicating desired QoS DHCP Client</li> <li>▪ DHCP Client</li> <li>▪ NTP Client</li> </ul>
<b>Media Processing</b>	<ul style="list-style-type: none"> <li>▪ Voice Coders: G.711, G.723.1, G.729A/B, G.722</li> <li>▪ Acoustic Echo Cancellation: G.168-2004 compliant, 64-msec tail length</li> <li>▪ Adaptive Jitter Buffer 300 msec</li> <li>▪ Voice Activity Detection</li> <li>▪ Comfort Noise Generation</li> <li>▪ Packet Lost Concealment</li> <li>▪ RTP/RTCP Packetization (RFC 3550, RFC 3551)</li> <li>▪ DTMF Relay (RFC 2833)</li> </ul>
<b>Telephony Features</b>	<ul style="list-style-type: none"> <li>▪ Call Hold / Un Hold</li> <li>▪ Call Transfer</li> <li>▪ 3-way Conference (with local mixing)</li> <li>▪ Redial</li> <li>▪ Caller ID Notification</li> <li>▪ Call Waiting Indication</li> <li>▪ Message Waiting Indication (including MWI LED)</li> <li>▪ Local and Corporate Directories</li> <li>▪ Automatic On-hook Dialing</li> <li>▪ CWRR (Call Waiting Reminder Ring)</li> <li>▪ Secondary Dial Tone</li> <li>▪ Call Logs: Missed/Received Calls and Dialed Numbers</li> <li>▪ Speed Dial</li> <li>▪ Dial Plan</li> <li>▪ Call Forward (Unconditional / Busy / No answer)</li> <li>▪ Redundant SIP Proxy Mechanism</li> </ul>
<b>Configuration/Management</b>	<ul style="list-style-type: none"> <li>▪ TR-069 for remote configuration and management</li> <li>▪ LCD Display User Interface Language Support (Various Languages)</li> <li>▪ Web-based Management (HTTPS)</li> <li>▪ Auto-Provisioning (via TFTP, FTP, HTTP, and HTTPS) for firmware and configuration file upgrade</li> <li>▪ DHCP options (66, 67, and 160) for Auto-provisioning</li> <li>▪ DHCP options (12, 60, and 77) for device information</li> </ul>

Feature	Details
<b>Hardware</b>	<ul style="list-style-type: none"> <li>▪ <b>LCD screen:</b> 2 * 16 characters</li> <li>▪ <b>Connectors interfaces:</b> <ul style="list-style-type: none"> <li>✓ 2 x RJ-45 ports (10/100BaseT Ethernet) for WAN and LAN</li> <li>✓ RJ-9 port (jack) for Headset</li> <li>✓ RJ-9 port (jack) for Handset</li> </ul> </li> <li>▪ <b>Mounting:</b> <ul style="list-style-type: none"> <li>✓ Wall mounting</li> <li>✓ Adjustable angle tilt for desktop mounting</li> </ul> </li> <li>▪ <b>Power:</b> <ul style="list-style-type: none"> <li>✓ DC jack adapter 12V</li> <li>✓ Power supply AC 100 ~ 240V</li> <li>✓ PoE: IEEE802.3af (optional)</li> </ul> </li> <li>▪ <b>Keys:</b> <ul style="list-style-type: none"> <li>✓ 10 x speed dial keys</li> <li>✓ VOICE MAIL message hotkey</li> <li>✓ 4-way navigation keys with ENTER Key</li> <li>✓ MENU</li> <li>✓ DIRECTORY</li> <li>✓ REDIAL</li> <li>✓ HOLD</li> <li>✓ MUTE</li> <li>✓ TRANSFER</li> <li>✓ CONFERENCE</li> <li>✓ FORWARD</li> <li>✓ DND</li> <li>✓ VOLUME control key</li> <li>✓ HEADSET</li> <li>✓ SPEAKER</li> </ul> </li> </ul>
<b>Headset Compatibility</b>	<ul style="list-style-type: none"> <li>▪ GN.com 2100 – both single and dual</li> <li>▪ Jabra BIZ 2400 – both single and dual</li> <li>▪ GN.com 9350</li> <li>▪ Plantronics H261N</li> <li>▪ Plantronics H251N</li> </ul>

## Reader's Notes

# **Administrator's Manual**

## **310HD IP Phone**

**Version 1.4.0**



[www.audiocodes.com](http://www.audiocodes.com)