AudioCodes 300HD Series of High Definition IP Phones

HD VoIP 320HD IP Phone

Administrator's Manual

320HD 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 320HD IP Phone.

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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
320HD IP Phone Quick Guide
320HD IP Phone User's Manual

1 Introduction

AudioCodes 320HD IP Phone is based on AudioCodes proprietary High Definition (HD) voice technology, providing clarity and a better audio experience in Voice-over-IP (VoIP) calls.

The 320HD 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.

This manual is intended for the system Administrator whose responsibility is to setup and configure the phone.

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

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

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

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:

- Defining LAN connection type (refer to Section 2.2 on page 14)
- Defining SIP accounts (refer to Section 2.3 on page 16)
- Restoring factory defaults (refer to Section 2.6 on page 19)
- Restarting the phone (refer to Section 2.7 on page 20)

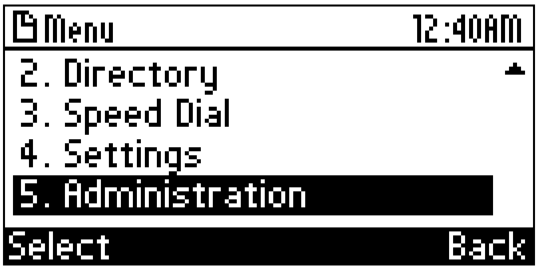
These administration settings are performed in the phone's **Administration** menu (refer to Section 2.1).

2.1 Accessing the Administration Menu

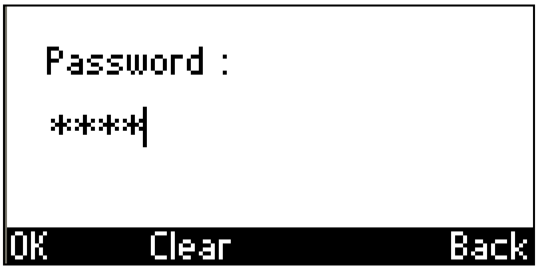
The **Administration** menu is password protected. The default password is "1234".

➢ **To access the Administration menu:**

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



2. Using the ▲ and ▼ Navigation keys, choose the **Administration** submenu, and then press the **Select** softkey; you are prompted for a password:



3. Enter your password, and then choose **OK**.

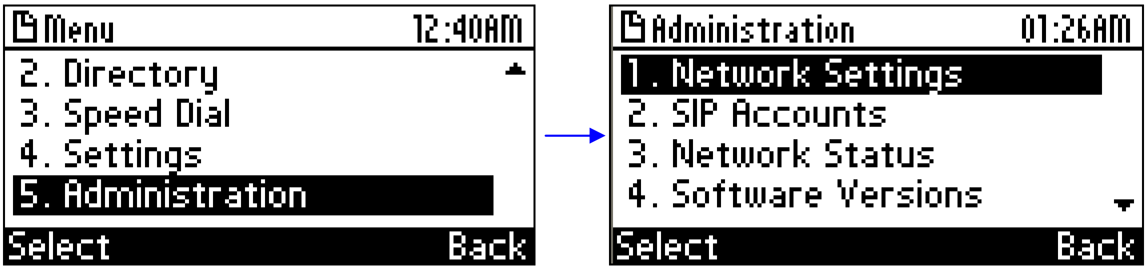
Note: To change the **Administration** menu's login password, use the phone's Web interface (refer to Section 3.11).

2.2 Defining LAN Network

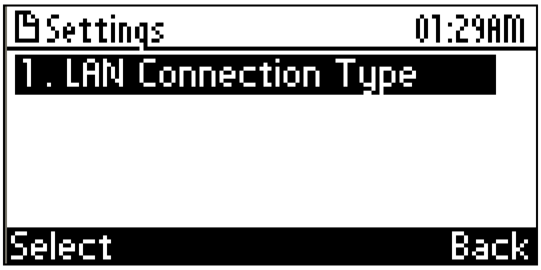
The phone's LAN connection interface can be manually defined (static IP address) or automatically provisioned using a DHCP server from where the LAN IP address is obtained.

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

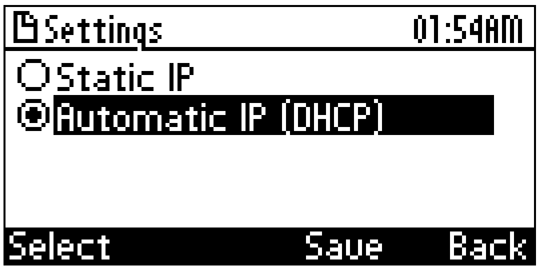
1. Access the Settings screen (MENU key > **Administration** menu > **Network Settings** submenu).



The following screen appears:



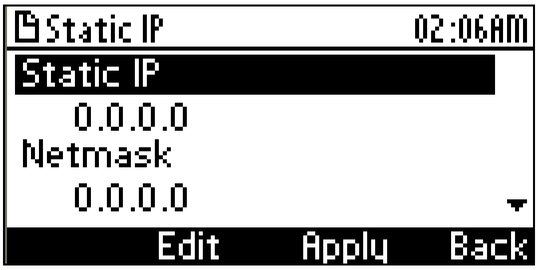
2. Press the **Select** softkey to choose **LAN Connection Type**; the following screen appears:



3. Using the ▲ and ▼ Navigation keys, choose one of the following IP addressing schemes:
   - "Static IP"
   - "Automatic IP (DHCP)"
4. Press the **Select** softkey. If you selected "Static IP" in Step 3, continue with Step 5; otherwise, skip to Step 6.

5. Define a static IP addressing scheme:

   a. Press the **Edit** softkey; the Static IP screen appears:

   b. To configure each required network parameter (i.e., “Static IP”, “Netmask”, “Gateway”, “Primary DNS”, and “Secondary DNS”), choose the parameter using the ▲ and ▼ Navigation keys, and then press the **Edit** softkey. Enter the new address in dotted-decimal notation, using the following keys:

      - **Navigation keys:** moves the cursor left or right in the IP address
      - **Clear softkey:** deletes the digit to the left of the cursor.

   c. Press the **Save** and then **Apply** softkey.

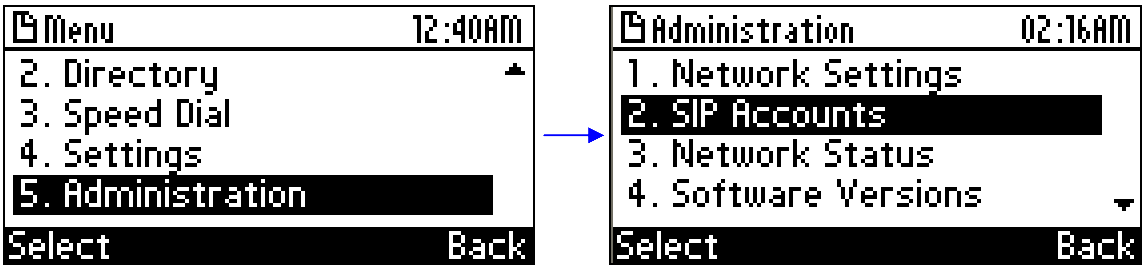
6. Press the **Save** softkey.

2.3 Defining SIP Accounts

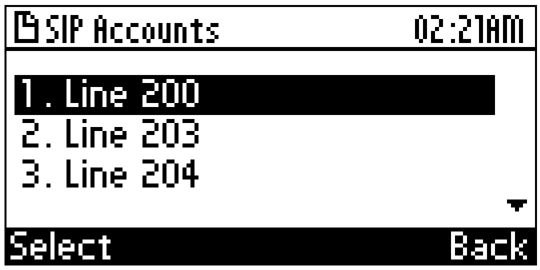
You need to configure parameters related to the phone's SIP account. This also allows you to set up the phone with up to four lines, each with its own extension phone number. These lines correspond to the LINE keys on your phone. For each line you need to configure a SIP account.

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

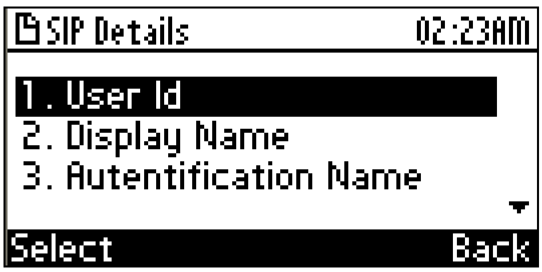
1. Access the SIP Accounts screen (MENU key > **Administration** menu > **SIP Accounts** submenu).



The following screen appears:



2. Choose (using the ▲ and ▼ Navigation keys) the Line that you want to set up with a SIP account, and then press the **Select** softkey; the SIP Details screen appears:



3. Choose the required SIP parameter, and then press the **Select** softkey to define it:
   - "User Id"
   - "Display Name"
   - "Authentication Name"
   - "Authentication Password"
   - "SIP Proxy Address": SIP proxy server's address
4. Press the **Save** softkey to save the parameter setting.

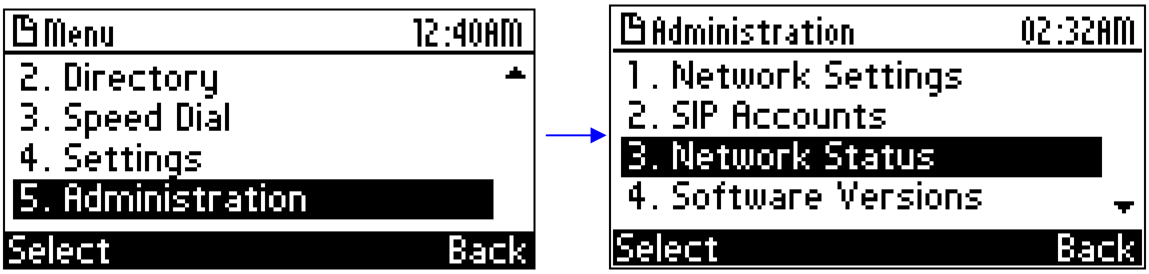
2.4 Viewing Network Status

You can view the following network status information:

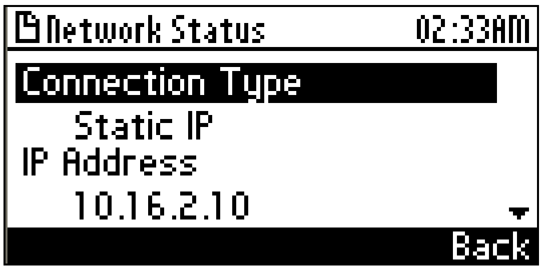
- Connection Type (static IP or automatic IP/DHCP).
- IP address
- Netmask
- Gateway
- Primary and Secondary DNS
- MAC address

➢ To view the phone's network status:

1. Access the Network Status screen (MENU key > **Status** menu > **Network Status** submenu).



The following screen appears:



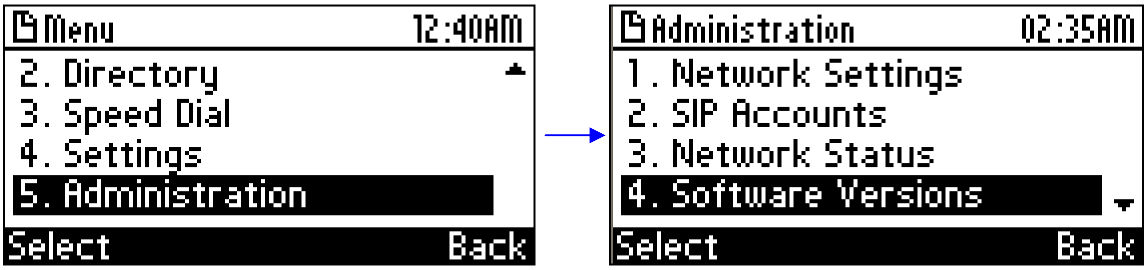
2. Scroll through the list using the ▲ and ▼ Navigation keys to view the desired network parameter.

2.5 Viewing Software Version

You can view the phone's model type, firmware version, and configuration file version.

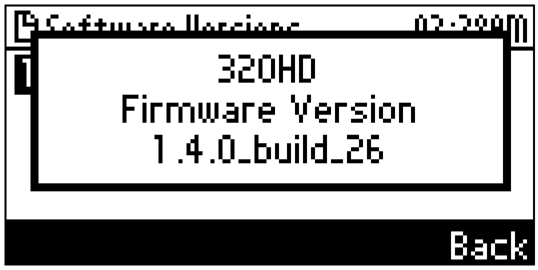
➢ **To view the phone's firmware version, model type, and configuration version:**

1. Access the Software Versions screen (MENU key > **Status** menu > **Versions**).



The following screen appears:

2. Press the **Select** softkey; the phone model and the its firmware version are displayed:

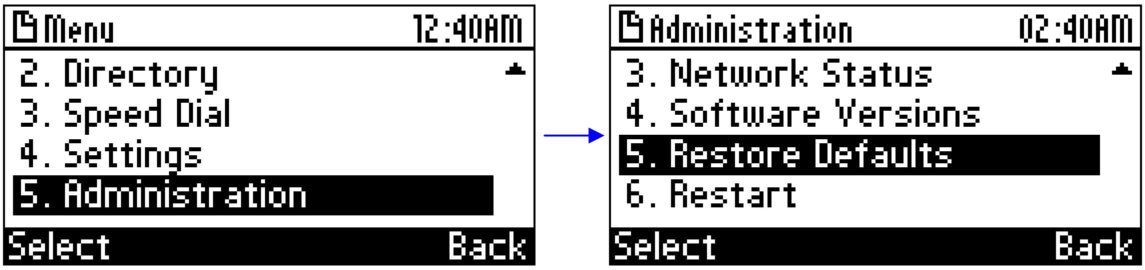


2.6 Restoring Default Settings

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

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

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



A warning message appears, requesting you to confirm:

2. Press the **Yes** softkey to confirm reset to defaults or **No** if you want to cancel.

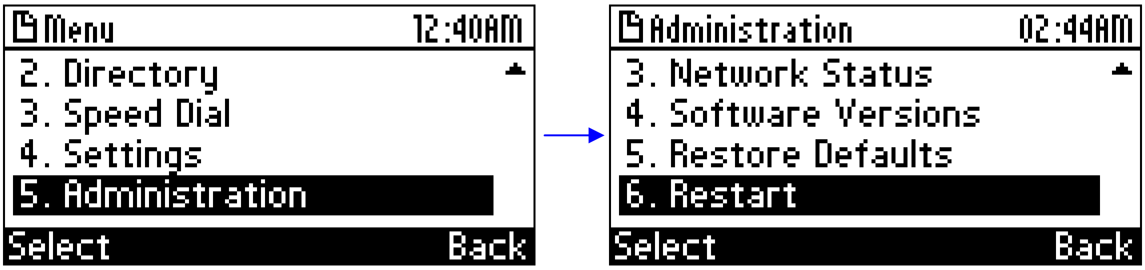
2.7 Restarting Phone

The procedure below describes how to restart the phone.

Note: The phone can also be remotely reset by sending a SIP NOTIFY message to the phone with the "check-sync" event. This reset allows for graceful shutdown, whereby the phone resets only when there are no calls, i.e., when the phone is in idle state.

➢ **To reset the phone:**

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



A warning message appears, requesting you to confirm:

2. Press the **Yes** softkey to confirm phone restart or **No** if you want to cancel.

3 Web-based Management

This chapter describes the phone's embedded Web server (*Web interface*).

> **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).

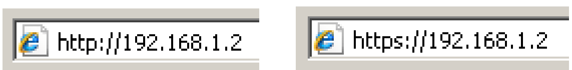
3.1 Accessing Web Interface

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 2.4.
3. Start your Web browser, and then in the URL address field, enter the phone's IP address, for example, http://192.168.1.2 or https://192.168.1.2, as displayed below.

Figure 3-1: Phone's URL in Web Browser



The Web login window appears:

Figure 3-2: Web Login Window

> **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 47.

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

   http://320hd-<MAC address>.<domain name>

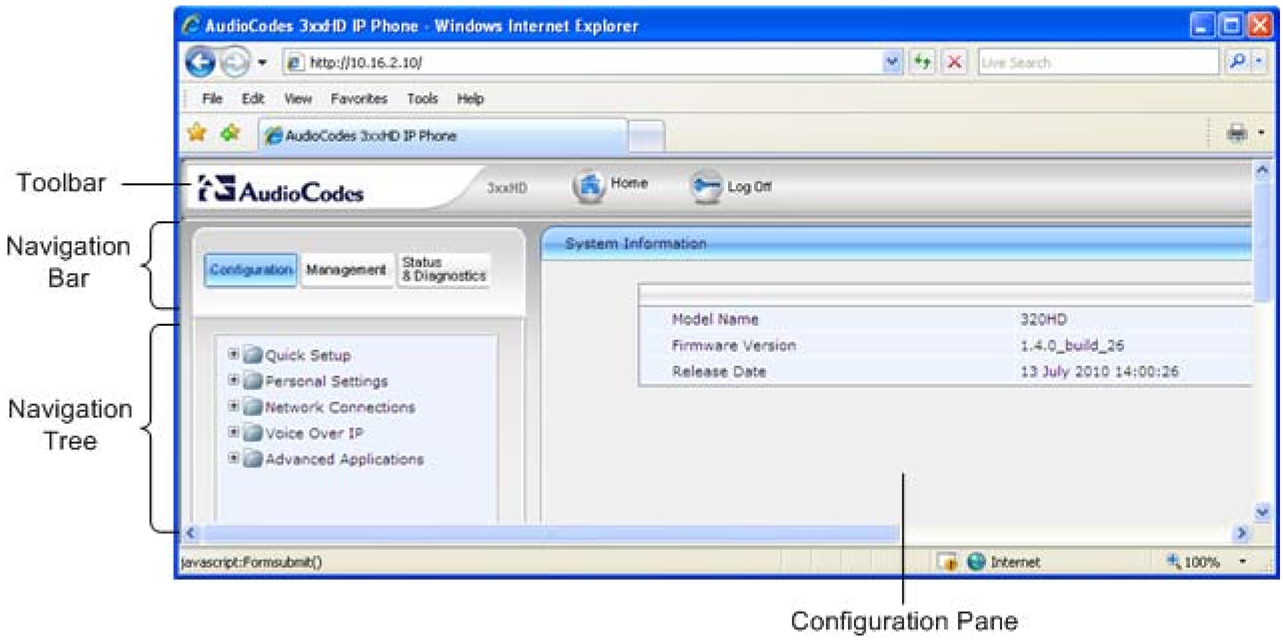
   For example, **http://320hd-001122334455.corp.YourCompany.com**

5. Enter the user name and the password, and then click **OK**.

3.2 Getting Started with the Web

The areas of the Web interface are shown below:

Figure 3-3: Main Areas of Web Interface



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.

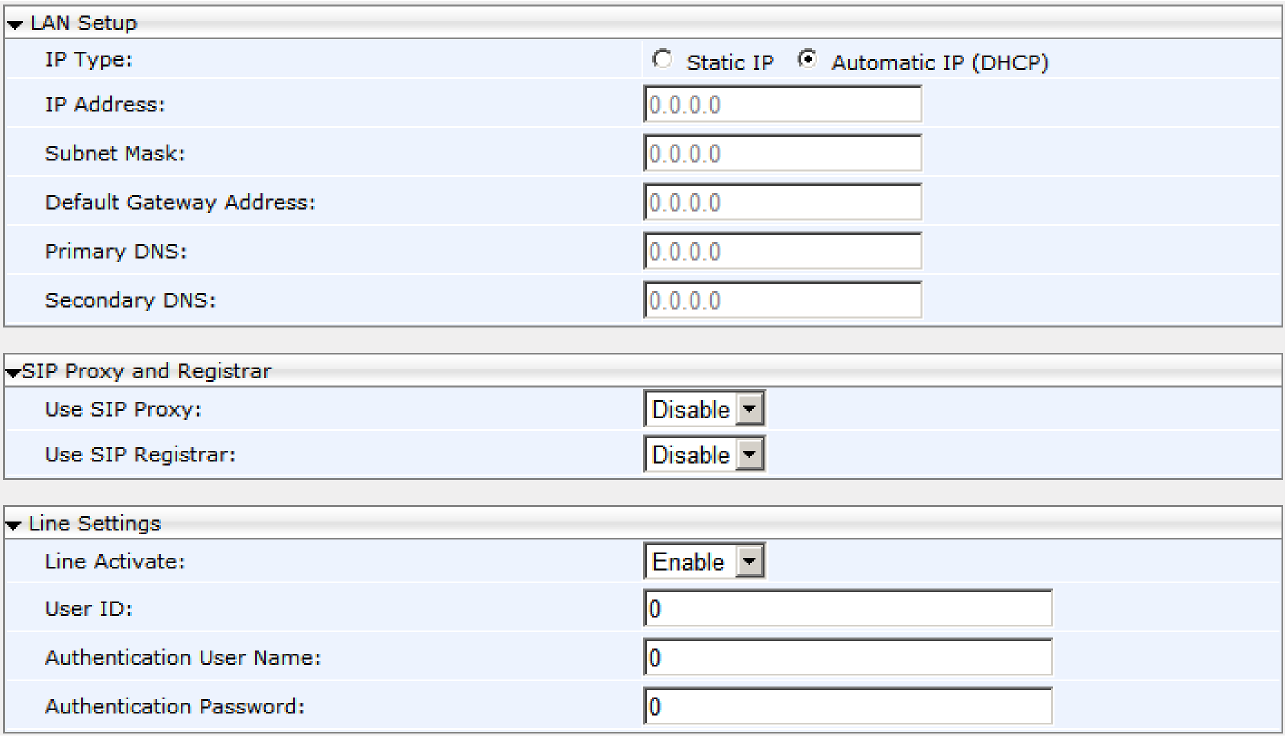
3.3 Quick Setup

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 3-4: Quick Setup Page



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 28.
   - Parameters under the **SIP Proxy and Registrar** group, refer to Section 3.8.1 on page 29
   - Parameters under the **Line Settings** group, refer to Section 3.8.5 on page 36

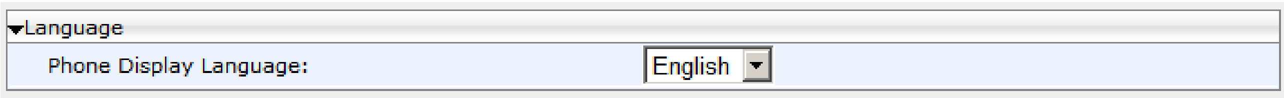
3.4 Changing LCD Display Language

You can choose the language of the LCD display.

➢ **To choose a language:**

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

Figure 3-5: Language Page



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 Configuring Speed Dials

You can add up to 12 speed dials to your phone. In addition, you can enable busy lamp field (BLF) per speed dial if the network includes a third-party Application server supporting BLF functionality. For configuring BLF for BroadWorks and Asterisk application servers, refer to Section 5.1 on page 93, and Section 5.2 on page 95 respectively.

Notes:

- 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 4.4.
- For BLF configuration file parameters, refer to Section 4.2.5.8.6.

To add a speed dial:

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

Figure 3-6: Speed Dial Page

Button	Type	Number	Delete
1	Speed Dial+BLF	4391	☐
2	Speed Dial+BLF	4403	☐
3	Speed Dial+BLF	4358	☐
4	Speed Dial+BLF	4071	☐
5	Speed Dial+BLF	90521122333	☐
6	Speed Dial+BLF	90773452345	☐
7	Speed Dial+BLF	4427	☐
8	Speed Dial+BLF	4465	☐
9	Speed Dial+BLF	4312	☐
10	Speed Dial	1111	☐
11	Speed Dial	2222	☐
12	Speed Dial	3333	☐

1. In the 'Type' field corresponding to the phone's Speed Dial key (in the 'Button' column), choose the type of the button ("Speed Dial" or "Speed Dial+BLF") to which you want to assign the Speed Dial key.
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, click **Reset.**

3.6 Configuring Tones

This section describes how to configure the phone's tones.

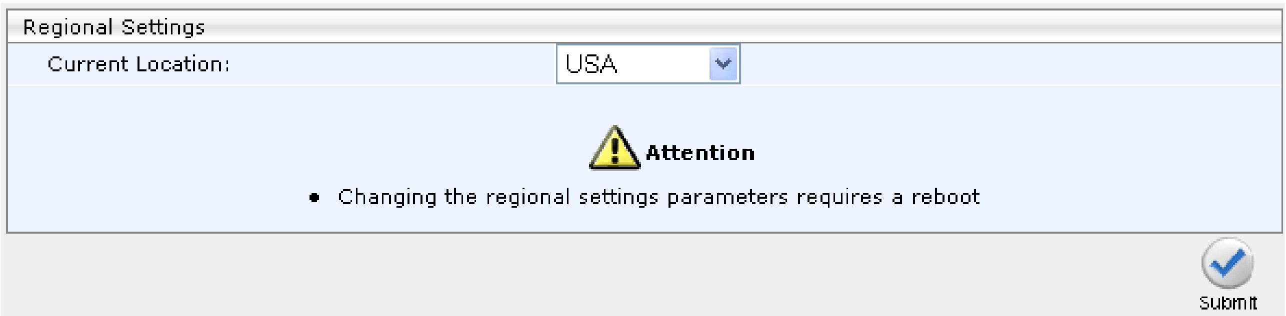
3.6.1 Selecting Region for Call Progress Tones

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 (CPT) 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 3-7: Tones Page



2. Configure the regional settings according to the parameter in the table below, and then 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 Loading Ring Tones

Follow the procedure below for uploading new ring tones. Instead of using the provided ringing tones, you can upload a different ring tone file and use that ring tone for indicating incoming calls.

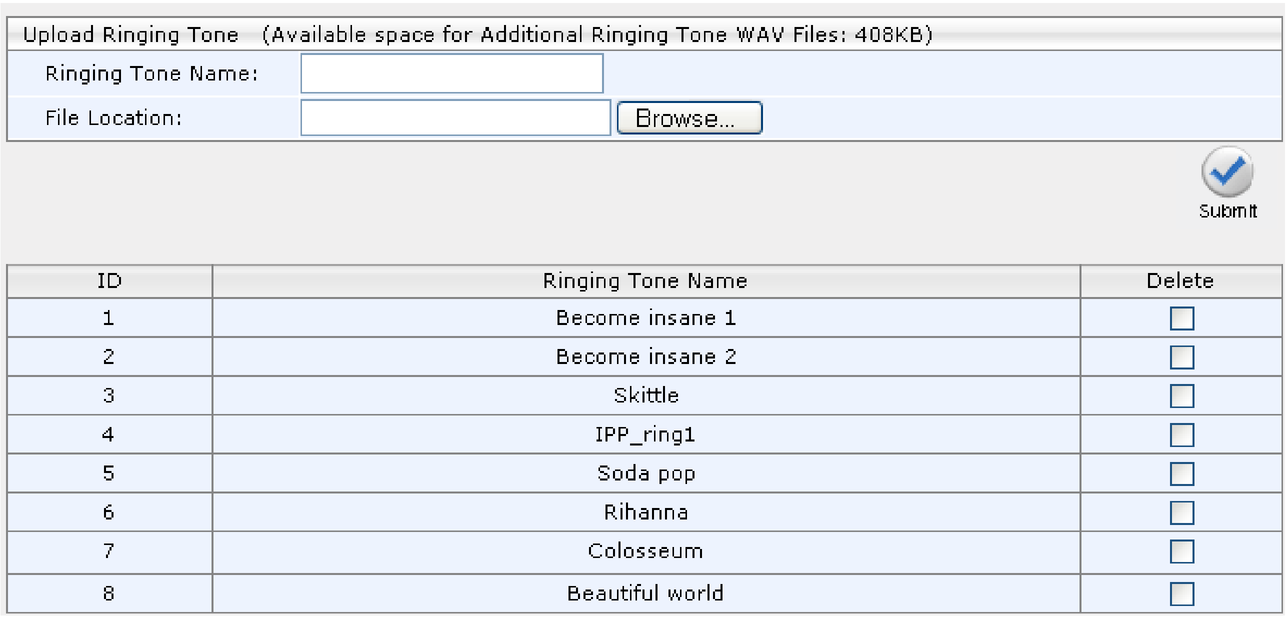
Notes:

- 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 PMC 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 3-8: Tones Page



ID	Ringing Tone Name	Delete
1	Become insane 1	
2	Become insane 2	
3	Skittle	
4	IPP_ring1	
5	Soda pop	
6	Rihanna	
7	Colosseum	
8	Beautiful world	

2. In the 'Ringing Tone Name' field, enter the name of the ring tone file that you want to upload. If you do not enter a name, the phone assigns the tone's file name (without the .wav file extension) as the name of the tone.
3. 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.
4. Click **Submit**; the file is loaded to the phone and displayed in the Ring Tone list.

Table 3-3: Ring Tone Parameter Description

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

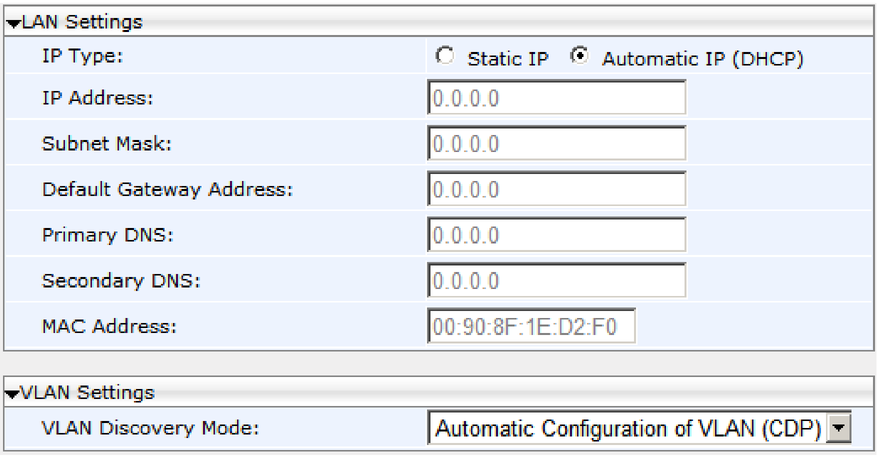
3.7 Configuring LAN Interface

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 3-9: LAN Settings Page



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 4.2.2
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**	
VLAN Settings		
VLAN Discovery Mode	**network/lan/vlan/mode**	Refer to Section 4.2.2
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 29)
- Dialing (refer to Section 3.8.2 on page 32)
- Media streaming (refer to Section 3.8.3 on page 34)
- Voice (refer to Section 3.8.4 on page 35)
- Line (refer to Section 3.8.5 on page 36)
- Services (refer to Section 3.8.6 on page 37)
- Volume Settings (refer to Section 3.8.7 on page 39)

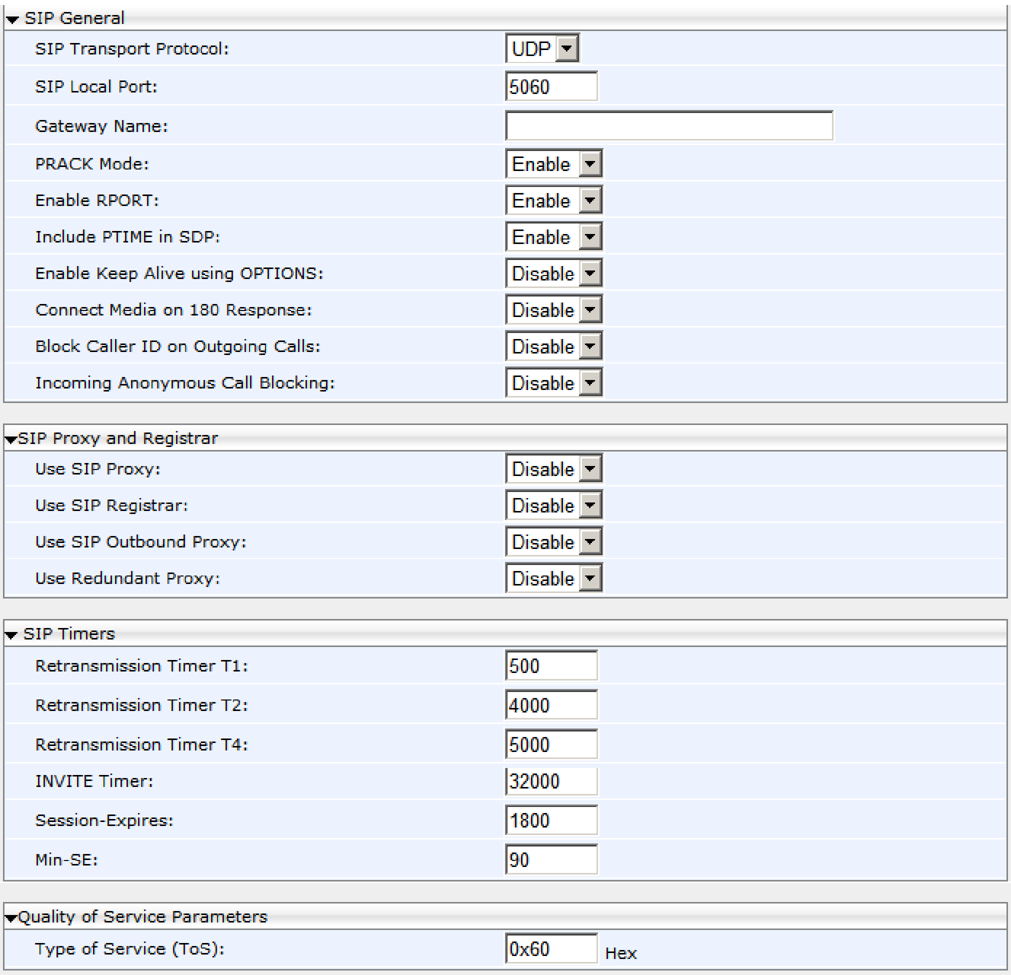
3.8.1 Configuring SIP

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 3-10: Signaling Protocol Page



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
SIP General Parameters		
SIP Transport Protocol	**voip/signalling/sip/transport_protocol**	Refer to Section 4.2.5.5.1
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/timeout**	
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**	
SIP Proxy and Registrar Parameters		
Use SIP Proxy	**voip/signalling/sip/use_proxy**	Refer to Section 4.2.5.5.2
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**	

Web Parameter	Configuration File Parameter	Description
Redundant Proxy Address	**voip/signalling/sip/redundant_proxy/address**	
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**	
SIP Timers		
Retransmission Timer T1	**voip/signalling/sip/sip_t1**	Refer to Section 4.2.5.5.3
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**	
SIP QoS		
Type of Service (ToS)	**voip/signalling/sip/tos**	Refer to Section 4.2.5.5.4

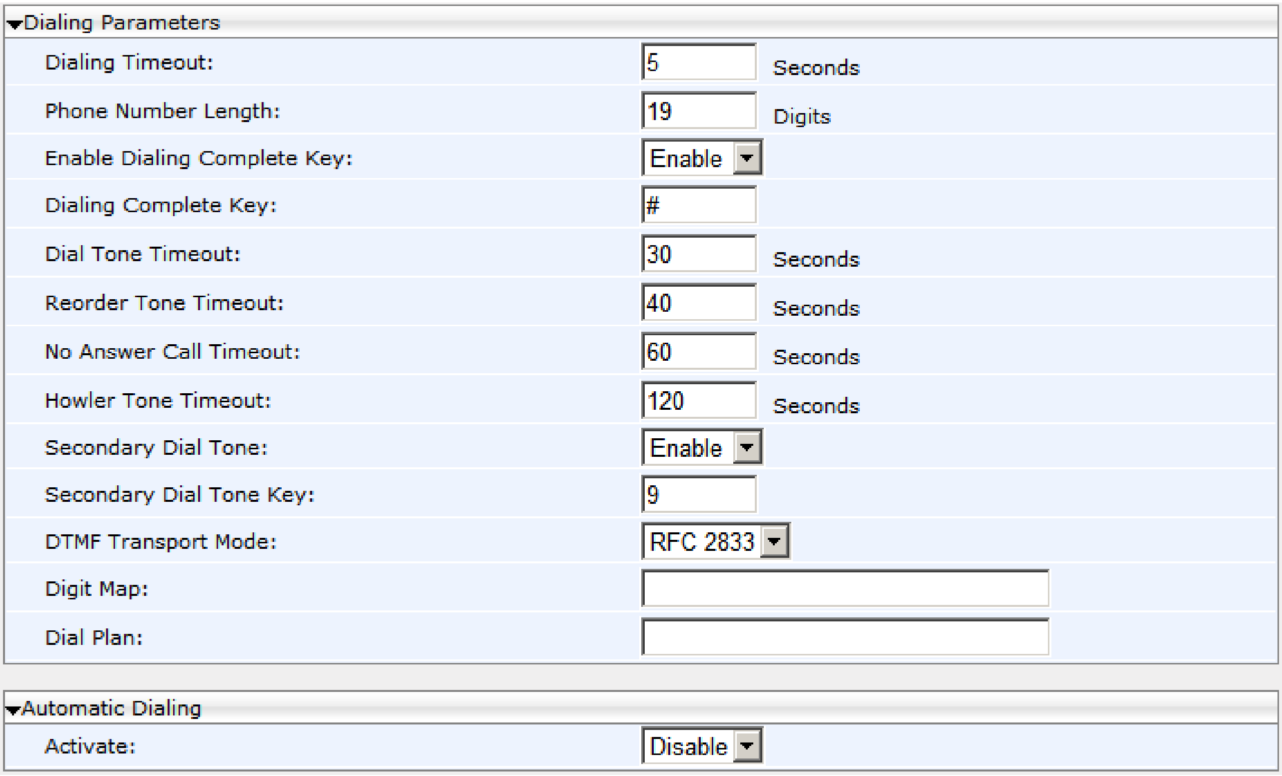
3.8.2 Configuring Dialing

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 3-11: Dialing Page



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 4.2.5.6
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**	

Web Parameter	Configuration File Parameter	Description
Secondary Dial Tone Key	**voip/dialing/secondary_dial_tone/key_sequence**	
DTMF Transport Mode	**voip/media/out_of_band_dtmf**	
Digit Map	**voip/signalling/sip/digit_map**	
Dial Plan	**voip/signalling/sip /number_rules**	
Automatic Dialing		
Activate	**voip/dialing/auto_dialing/enabled**	Refer to Section 4.2.5.6
Timeout	**voip/dialing/auto_dialing/timeout**	
Destination Phone Number	**voip/dialing/auto_dialing/destination**	

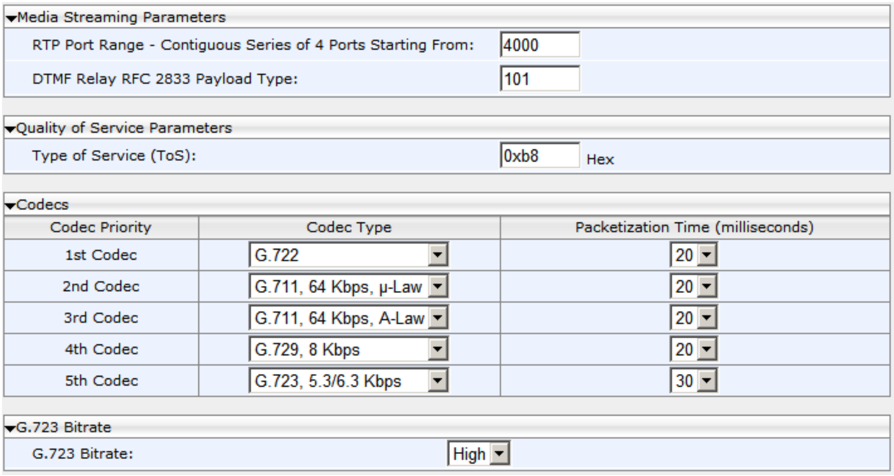
3.8.3 Configuring Media Streaming

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 3-12: Media Streaming Page



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 4.2.5.4
DTMF Relay RFC 2833 Payload Type	**voip/media/dtmf_payload**	
Quality of Service (QoS)		
Type of Service (ToS)	**voip/media/media_tos**	Refer to Section 4.2.5.4
Codecs		
Codec Type	**voip/codec/codec_info /%d/name**	Refer to Section 4.2.5.3
Packetization Time	**voip/codec/codec_info /%d/ptime**	
G.723 Bitrate		
G.723 Bitrate	**voip/codec/g723_bitrate**	Refer to Section 4.2.5.3

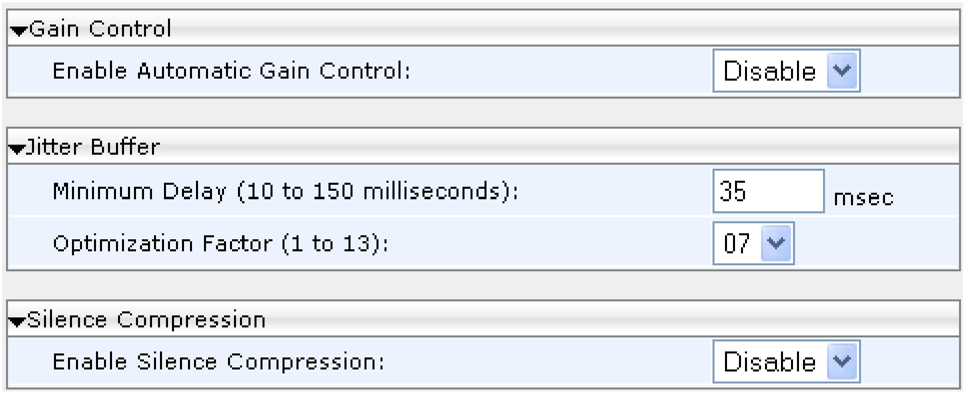
3.8.4 Configuring Voice

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 3-13: Voice Page



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
Gain Control		
Enable Automatic Gain Control	**voip/audio/gain/automatic_gain_control/enabled**	Refer to Section 4.2.5.2
Automatic Gain Control Direction	**voip/audio/gain/automatic_gain_control/direction**	
Target Energy	**voip/audio/gain/automatic_gain_control/target_energy**	
Jitter Buffer		
Minimum Delay	**voip/audio/jitter_buffer/min_delay**	Refer to Section 4.2.5.7
Optimization Factor	**voip/audio/jitter_buffer/optimization_factor**	
Silence Compression		
Enable Silence Compression	**voip/audio/silence_compression/enabled**	Refer to Section 4.2.5.7

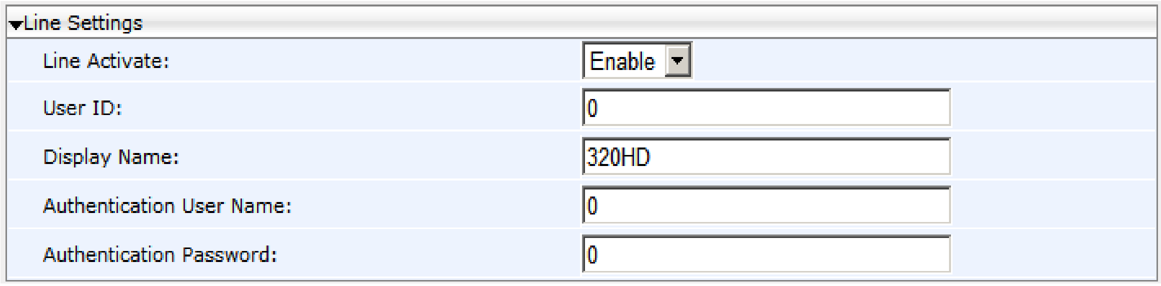
3.8.5 Configuring Lines

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 3-14: Line Settings Page



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	**voip/line/%d/enabled**	Refer to Section 4.2.5.1
User ID	**voip/line/%d/id**	
Display Name	**voip/line/%d/description**	
Authentication User Name	**voip/line/%d/auth_name**	
Authentication Password	**voip/line/%d/auth_password**	

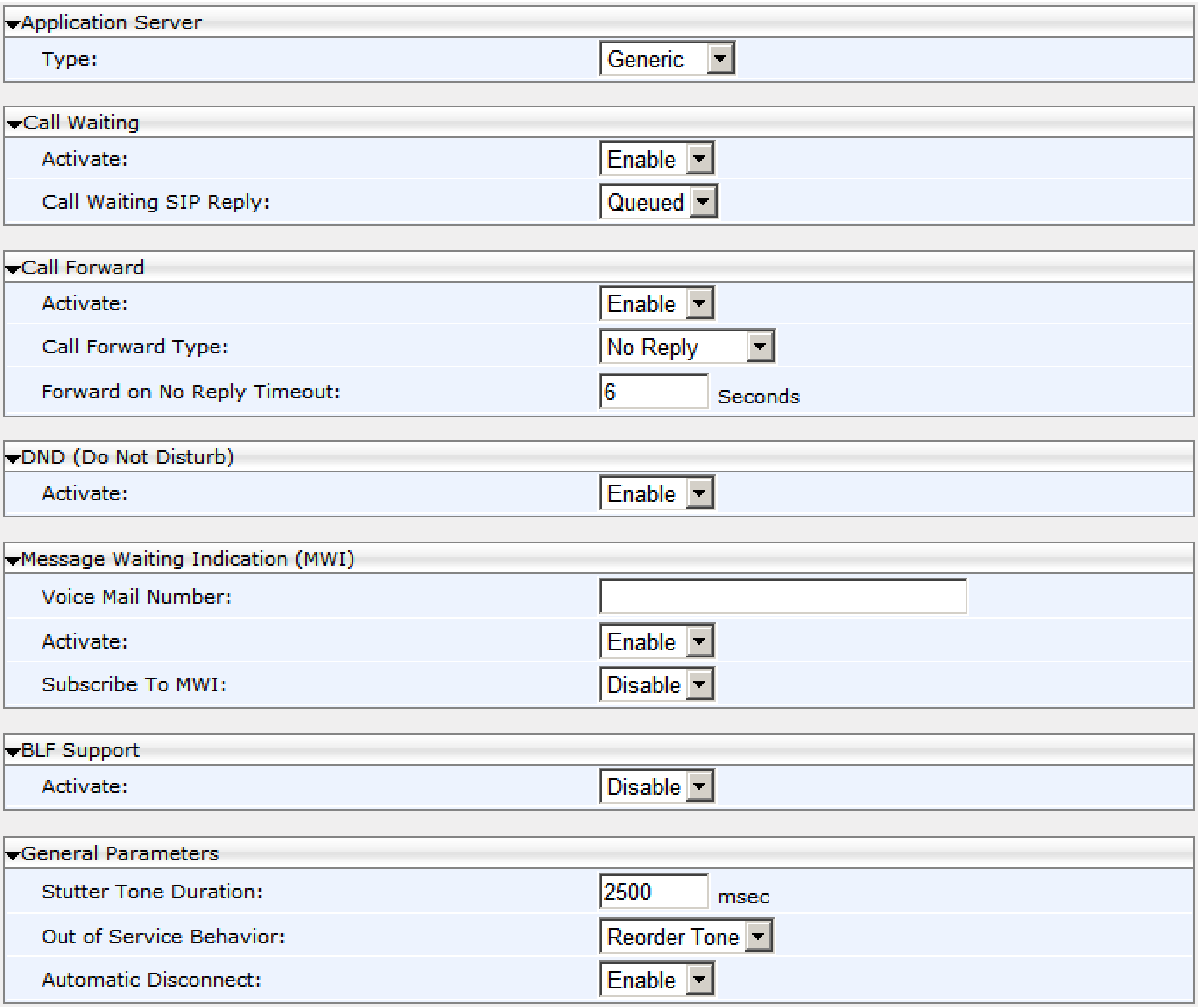
3.8.6 Configuring Supplementary Services

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

➢ **To define services:**

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

Figure 3-15: Services Page



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 4.2.5.8.1
Call Waiting		
Activate	**voip/services/call_waiting/enabled**	Refer to Section 4.2.5.8.2
Call Waiting SIP Reply	**voip/services/call_waiting/sip_reply**	
Call Forward		
Activate	**voip/services/call_forward/line/0/enabled**	Refer to Section 4.2.5.8.3
Call Forward Type	**voip/services/call_forward/line/0/type**	
Forward on No Reply Timeout	**voip/services/call_forward/line/0/timeout**	
DND (Do Not Disturb)		
Activate	**voip/services/do_not_disturb/enabled**	Refer to Section 4.2.5.8.4
Message Waiting Indication		
Voice Mail Number	**voip/services/msg_waiting_ind/voice_mail_number**	Refer to Section 4.2.5.8.5
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/expiraition_timeout**	
BLF Support		
Activate	**voip/services/busy_lamp_field/enabled**	Refer to Section 4.2.5.8.6
BLF Subscription Period	**voip/services/busy_lamp_field/subscription_period**	
User Resource List	**voip/services/busy_lamp_field/uri**	
Use Registrar as Application Server Address	**voip/services/busy_lamp_field/application_server/use_registrar**	
General Parameters		
Stutter Tone Duration	**voip/services/msg_waiting/stutter_tone_duration**	Refer to Section 4.2.5.8.1
Out of Service Behavior	**voip/services/out_of_service_bahavior**	Refer to Section 4.2.5.8.1
Automatic Disconnect	**voip/dialing/automatic_disconnect**	Refer to Section 4.2.5.6

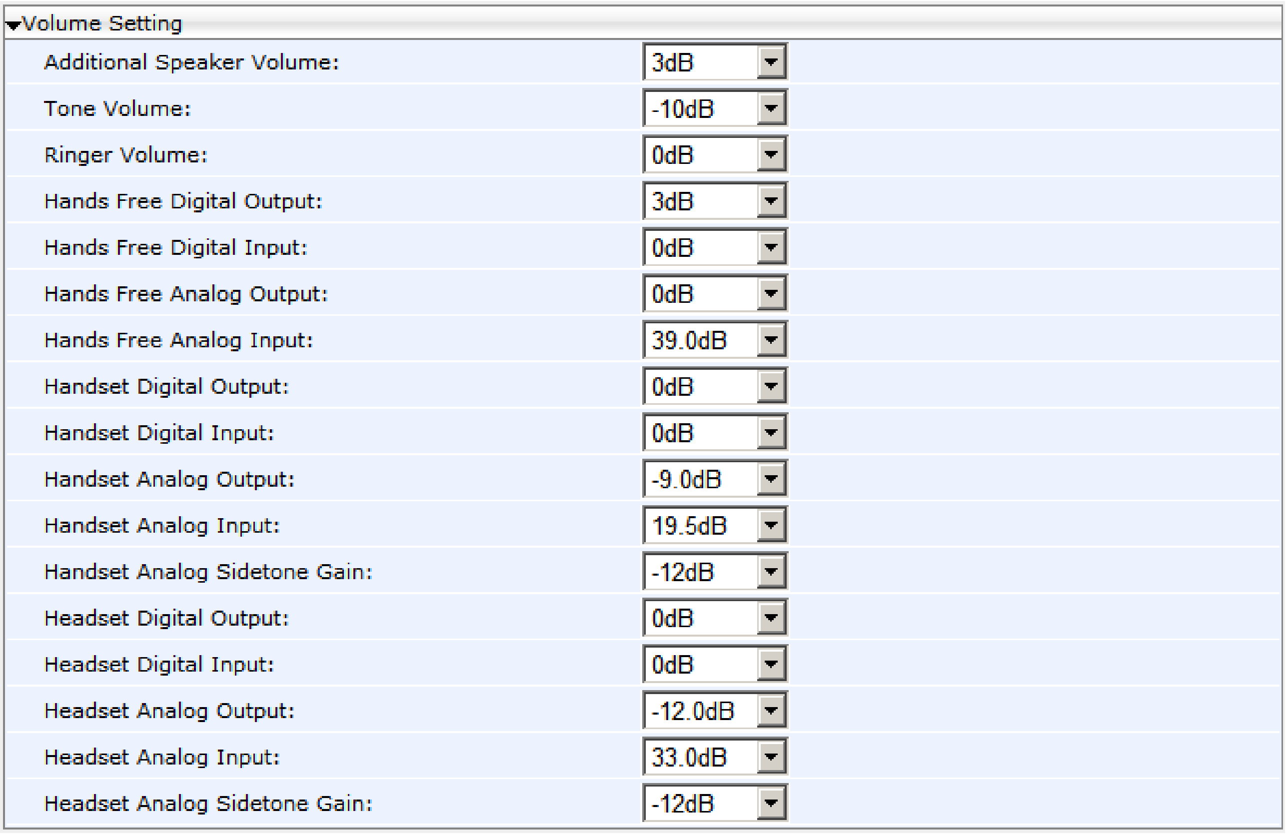
3.8.7 Configuring Volume Levels

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

➢ **To define volume settings:**

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

Figure 3-16: Volume Settings Page



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
Additional Speaker Gain	**voip/audio/gain/additional_speaker_gain**	Refer to Section 4.2.5.2.1
Tone Volume	**voip/audio/gain/tone_signal_level**	
Ringer Volume	**voip/audio/gain/ringer_signal_level**	
Hands-free Gain Parameters		
Hands Free Digital Output	**voip/audio/gain/handsfree_digital_output_gain**	Refer to Section 4.2.5.2.2
Hands Free Digital Input	**voip/audio/gain/handsfree_digital_input_gain**	

Web Parameter	Configuration File Parameter	Description
Hands Free Analog Output	**voip/audio/gain/handsfree_analog_output_gain**	
Hands Free Analog Input	**voip/audio/gain/handsfree_analog_input_gain**	
Handset Gain Parameters		
Handset Digital Output	**voip/audio/gain/handset_digital_output_gain**	Refer to Section 4.2.5.2.3
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**	
Headset Gain Parameters		
Headset Digital Output	**voip/audio/gain/headset_digital_output_gain**	Refer to Section 4.2.5.2.4
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**	

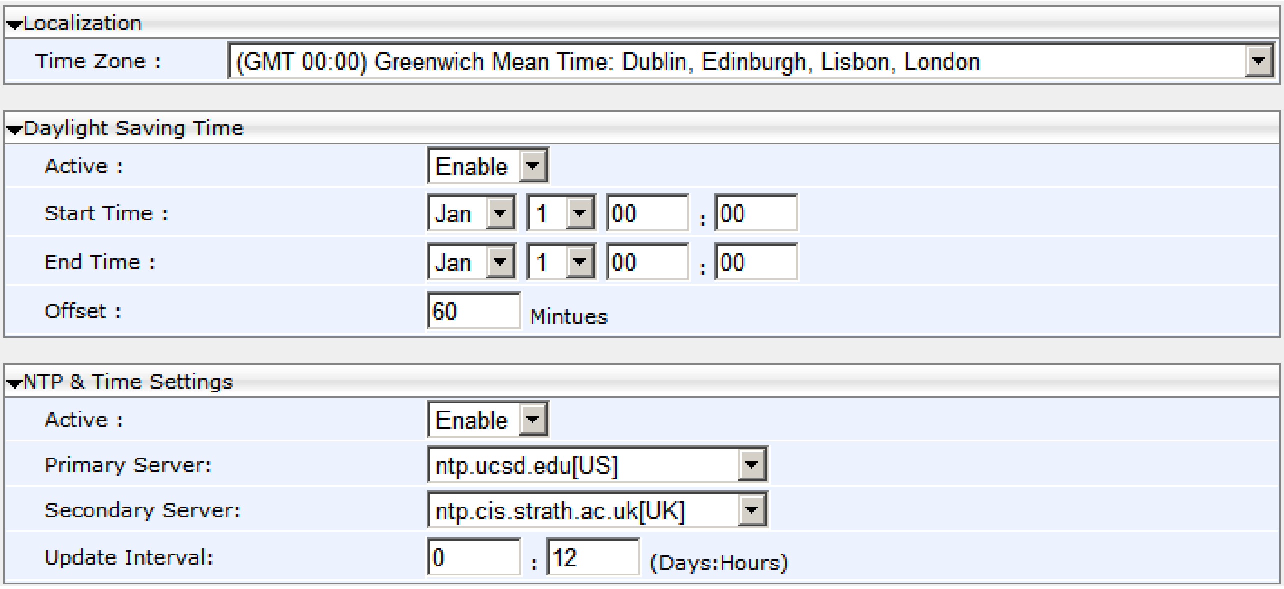
3.9 Configuring Date and Time

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. This configuration is done in the **Advanced Applications** menu.

➤ **To define the date and time:**

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

Figure 3-17: 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
Localization		
Time Zone	**system/ntp/gmt_offset**	Refer to Section 4.2.1.4
Daylight Saving Time		
Active	**system/ntp/daylight_saving/activate**	Refer to Section 4.2.1.4
Start Time	**system/ntp/daylight_saving/start_date**	
End Time	**system/ntp/daylight_saving/end_date**	
Offset	**system/ntp/daylight_saving/offset**	

Web Parameter	Configuration File Parameter	Description
NTP & Time Settings		
Active	**system/ntp/enabled**	Refer to Section 4.2.1.5
Primary Server	**system/ntp/primary_server_address**	
Secondary Server	**system/ntp/secondary_server_address**	
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 (refer to Section 3.10.1 on page 43)
- Manual update of firmware and configuration files (refer to Section 3.10.2 on page 45)

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.

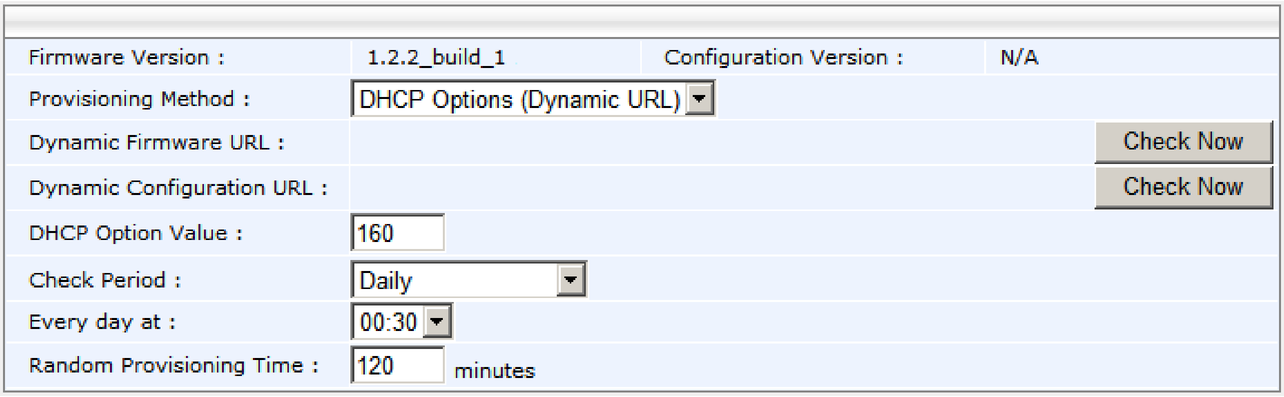
Notes:

- In the DHCP Discover message, the phone publishes its model name in Option fields 60 and 77 (e.g. 320HD). 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 46).
- 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 3-18: Automatic Update Page



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 4.2.3
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**	

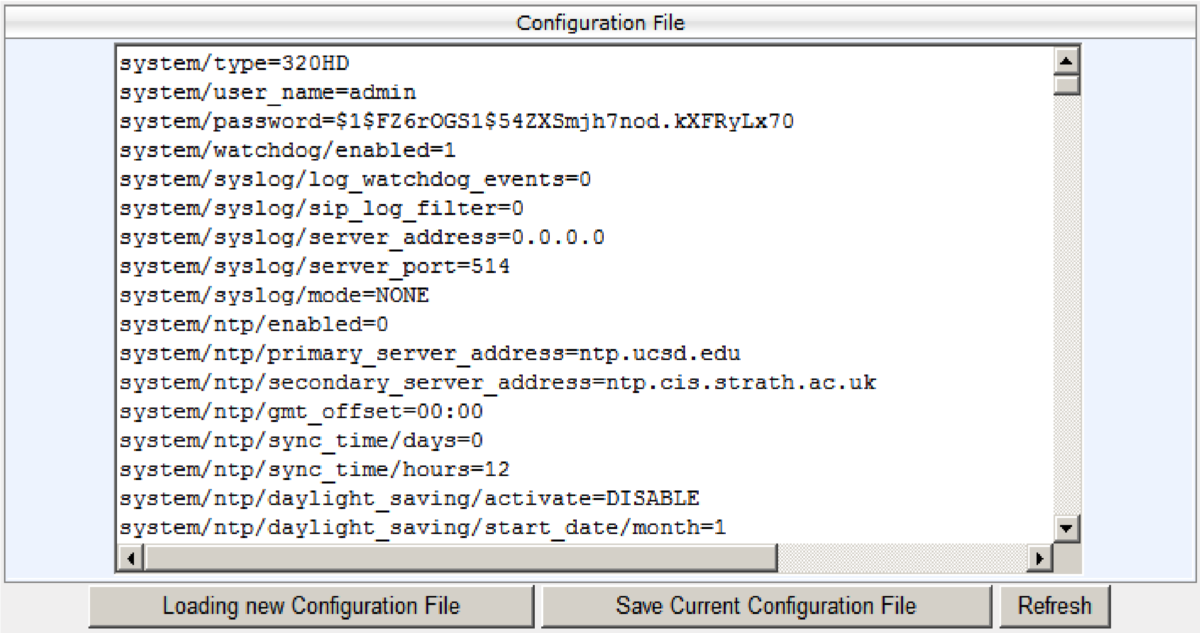
3.10.2 Updating (Manually) Configuration File

The phone enables you to view the current configuration file and save it to a folder on your PC (for backup). In addition, you can load a configuration file to your phone to replace the current one.

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

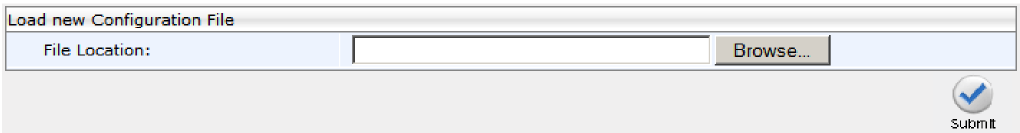
1. Access the 'Configuration File' page (**Management** tab > **Manual Update** menu > **Configuration File**). The current configuration file settings are displayed in the text pane.

Figure 3-19: Configuration File Page



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

Figure 3-20: 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 and that the version is later than the current one. 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 **Save 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's firmware, perform the procedure below.

➢ **To recover the phone's 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;
   a. 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.
   b. In the DHCP Discover message, the phone 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. 320HD_1.2.2_build_5.img).
   c. The phone then upgrades to the recovery firmware.
   d. 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 Window XP.

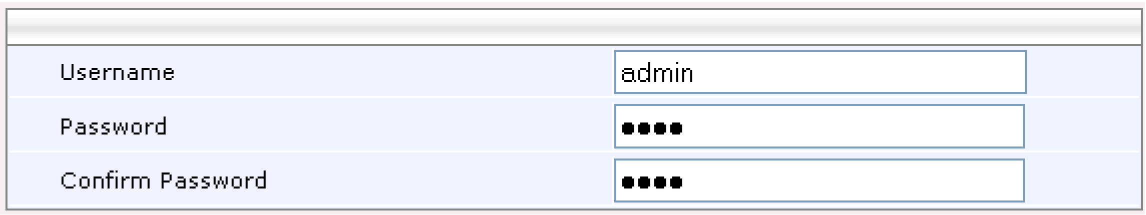
3.11 Changing Login Username and Password

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 3-21: System Authorization Page



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**, and then click **OK** at the confirmation prompt.

Table 3-14: Login Username and Password Parameters Description

Web Parameter	Configuration File Parameter	Description
Username	**system/user_name**	Refer to Section 4.2.1.2
Password	**system/password**	

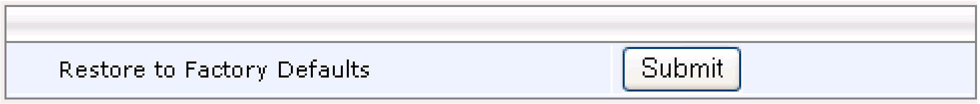
3.12 Restoring Phone Defaults

You can restore all your phone's settings to factory default settings.

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

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

Figure 3-22: Restore Defaults Page



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

Figure 3-23: Confirmation Box

3. Click **OK**.

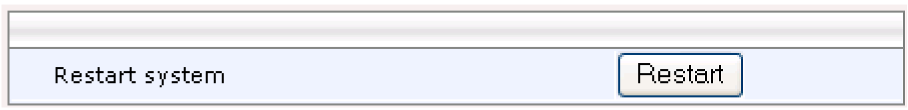
3.13 Restarting Phone

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 3-24: Restart System Page



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

Figure 3-25: Confirmation Box

3. Click **OK**.

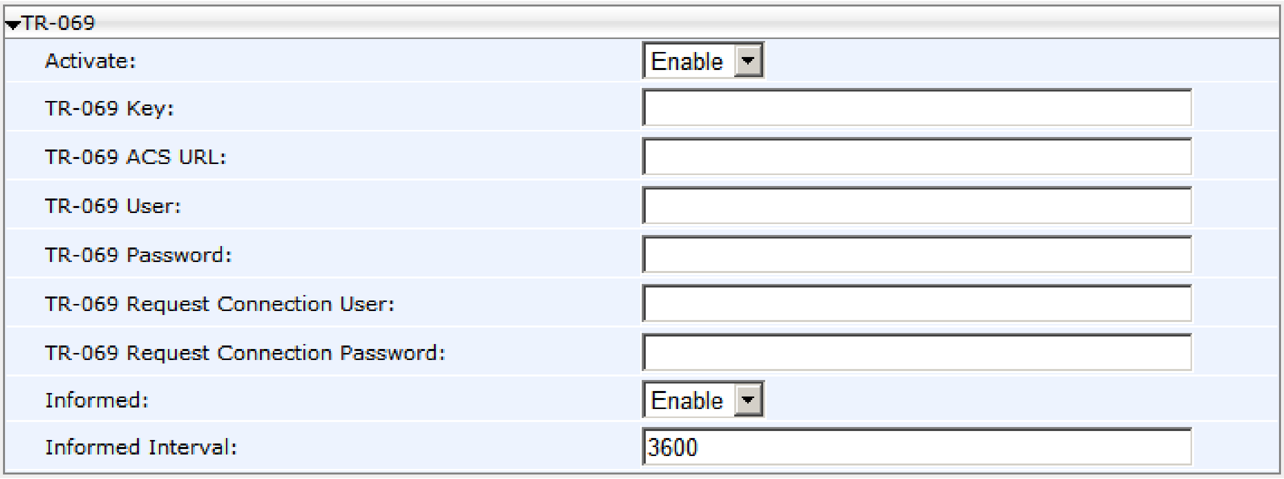
3.14 Configuring TR-069 Management

To connect to the remote management server, the phone's embedded TR-069 client must be configured.

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

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

Figure 3-26: TR-069 Page



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 3-27: 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 3-28: 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	☐
2	222	2000/01/02 Sunday 19:34:23	00:00:00	☐
3	anonymous	2000/01/01 Saturday 23:06:32	00:00:00	☐
4	anonymous	2000/01/01 Saturday 20:30:19	00:00:00	☐

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 3-29: System Information Page

Model Name	320HD
Firmware Version	1.2.2_build_5
Release Date	21 April 2010 10:11:16

3.16 Diagnostics

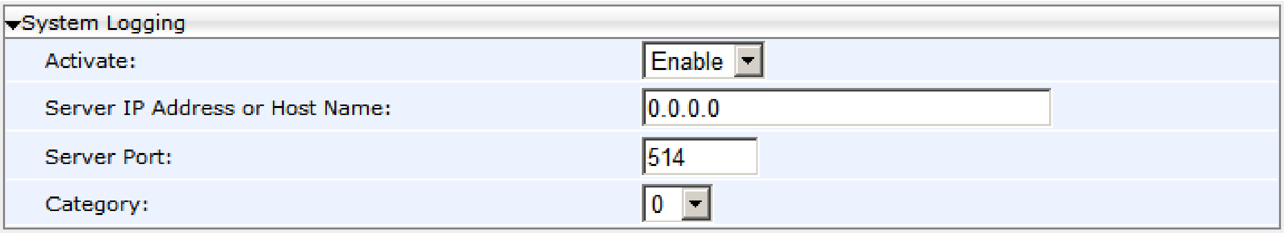
3.16.1 Configuring System Logging

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 3-30: System Logging Page



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	**system/watchdog/enabled**	Refer to Section 4.2.1.3
Server IP Address or Host Name	**system/syslog/server_address**	
Server Port	**system/syslog/server_port**	
Category	**system/syslog/sip_log_filter**	

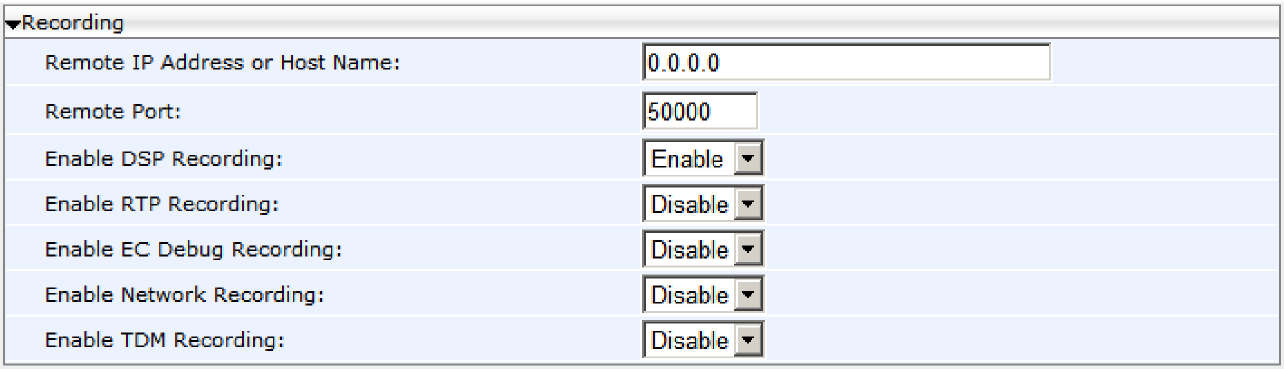
3.16.2 Configuring Packet Recording

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 3-31: Recording Page



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	**voip/packet_recording/remote_ip**	Refer to Section 4.2.7
Remote Port	**voip/packet_recording/remote_port**	
Enable DSP Recording	**/voip/packet_recording/enabled**	
Enable RTP Recording	**voip/packet_recording/rtp_recording/enabled**	
Enable EC Debug Recording	**voip/packet_recording/ec_debug_recording/enabled**	
Enable Network Recording	**voip/packet_recording/network_recording/enabled**	
Enable TDM Recording	**voip/packet_recording/tdm_recording/enabled**	

Reader's Notes

4 Configuration File-based Management

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). 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=320HD
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 bolded font):

```
system/type=320HD
include 320HD <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=320HD
include http://10.10.10.10/320HD <MAC> voip.cfg
include https://remote-pc/vlan_conf.cfg
include tftp://10.10.10.10/320HD <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 arewritten outside the square brackets, for example, [1] Enable.

4.2.1 System Parameters

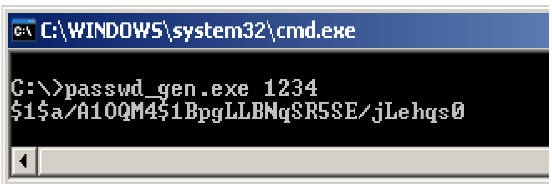
4.2.1.1 General Parameters

Table 4-1: General System Configuration File Parameters

Parameter	Description
Note: To add a value to these parameters, enter **system/** followed by the parameter name, equal sign and then the value (e.g. `system/type`=310HD).	
system/type	The phone model. The default value is 320HD.

4.2.1.2 Username and Password Parameters

Table 4-2: Username and Password Configuration File Parameters

Parameter	Description
Note: To add a value to these parameters, enter **system/** followed by the parameter name, equal sign and then the value (e.g. `system/type`=310HD).	
system/type	The phone model. The default value is 320HD.
system/user_name	The phone user name. The default value is admin. **Note:** This parameter is applicable only to the Web and Telnet interfaces.
system/password	The encrypted phone password. The default value is 1234. To generate an encrypted password for the **system/password** parameter, run the password generator (**passwd_gen**) 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 **system/password**.  **Note:** This parameter is applicable only to the Web and Telnet interfaces, and LCD display.

4.2.1.3 System Logging (Syslog) Parameters

Table 4-3: Syslog Configuration File Parameters

Parameter	Description
Note: To add a value to these parameters, enter **system/** followed by the parameter name, equal sign and then the value (e.g. `system/type`=310HD).	
system/watchdog/enabled	Enables the system watch dog. ▪ **[0]** Disable ▪ **[1]** Enable (default) **Note:** It is recommended to leave this parameter at its default value.
system/syslog/log_watchdog_events	Enables the watchdog logs to be sent to the address specified in system/syslog/server_address. ▪ **[0]** (default) - Disable ▪ **[1]** - Enable
system/syslog/sip_log_filter	Filters the type of the VoIP application logging. ▪ **[0]** (default) = Logging is disabled ▪ **[1]** - SIP Call Control ▪ **[2]** - SIP Stack ▪ **[3]** - SIP Call Control and SIP Stack ▪ **[4]** - User Application ▪ **[5]** - SIP Call Control and User Application ▪ **[6]** - SIP Stack and User Application ▪ **[7]** - SIP Call Control, SIP Stack and User Application
system/syslog/server_address	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.
system/syslog/server_port	Defines the UDP port of the Syslog server. The valid range is 0 to 65,535. The default port is 514.
system/syslog/mode	Defines the output direction of the Syslog information. ▪ **[NONE]** (default) - Disable ▪ **[UDP]** - Toward the network

4.2.1.4 Daylight Saving Time Parameters

Table 4-4: Daylight Saving Time Configuration File Parameters

Parameter	Description
system/ntp/gmt_offset	The GMT offset. The format of this value is + or - xx:yy (e.g., :+02:00). The default offset is 00:00.
system/ntp/daylight_saving/activate	Determines whether the phone automatically detects the Daylight Saving setting for selected time zones. ▪ **[Disable]** Disable (default) ▪ **[Enable]** Enable
system/ntp/daylight_saving/start_date	This sub-section defines the starting day for the daylight saving offset. ▪ **[month]** - defines the specific month in a year ▪ **[day]** - defines the specific day in a month ▪ **[hour]** - defines the specific hour in a day ▪ **[minute]** - defines the specific minute in an hour For example: To configure the phone to start daylight savings with a specific offset on February 22nd at 14:30, set the following: **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**
system/ntp/daylight_saving/start_date/month	The month in a year. The valid range is 1 to 12.
system/ntp/daylight_saving/start_date/day	The day in a month. The valid range is 1 to 31.
system/ntp/daylight_saving/start_date/hour	The hour in the day. The valid range is 0 to 23.
system/ntp/daylight_saving/start_date/minute	The minute in an hour. The valid range is 0 to 59.
system/ntp/daylight_saving/end_date	This sub-section defines the ending day for the daylight saving offset. ▪ **[month]** - defines the specific month in a year ▪ **[day]** - defines the specific day in a month ▪ **[hour]** - defines the specific hour in a day ▪ **[minute]** - defines the specific minute in an hour For example: To configure the phone to end the daylight savings on July 16th at 22:15, set the following: **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**

Parameter	Description
system/ntp/daylight_saving/end_date/month	The month in a year. The valid range is 1 to 12.
system/ntp/daylight_saving/end_date/day	The day in a month. The valid range is 1 to 31.
system/ntp/daylight_saving/end_date/hour	The hour in the day The valid range is 0 to 23.
system/ntp/daylight_saving/end_date/minute	The minute in an hour. The valid range is 0 to 59.
system/ntp/daylight_saving/offset	The offset value for the daylight saving. The valid range is 0 to 180. The default offset is 60.

4.2.1.5 Network Time Protocol (NTP) Server

Table 4-5: Network Time Protocol (NTP) Server Configuration File Parameters

Parameter	Description
system/ntp/enabled	Enables the NTP server from which the phone retrieves the date and time. ▪ **[0]** Disable (default) ▪ **[1]** Enable – obtains the time information from a configured NTP server
system/ntp/primary_server_address	Defines the address of the main NTP server (this can be a domain name, for example, tick.nap.com.ar).
system/ntp/secondary_server_address	Defines the address of the secondary NTP server.
system/ntp/sync_time	This sub-section defines how often the phone must perform an update with the NTP server. ▪ **[days]** - defines the number of days ▪ **[hours]** - defines the number of hours For example: To configure the phone to perform an update with an NTP server every 1 day and 6 hours, set the following: **system/ntp/sync_time/days=1** **system/ntp/sync_time/hours=6**
system/ntp/sync_time/days	The number of days. The valid range is 0 to 7. The default of days is 0.
system/ntp/sync_time/hours	The number of hours. The valid range is 0 to 24. The default is 12.

4.2.2 LAN Parameters

Table 4-6: Network Configuration File Parameters

Parameter	Description
Note: To add a value to these parameters, enter **network/** followed by the parameter name, equal sign and then the value (e.g. `network/lan_type`=DHCP).	
network/lan_type	Defines the IP addressing method: ▪ **[DHCP]** Automatic IP DHCP (default) - Phone's IP address is acquired automatically from a DHCP server ▪ **[STATIC]** Static IP - Phone's IP address is defined manually
network/lan/fixed_ip	This sub-section defines the parameters which are relevant in case **lan_type** is configured to "STATIC".
network/lan/fixed_ip/ip_address	The LAN IP address. **Note:** This parameter is applicable only when the phone is assigned a static IP address.
network/lan/fixed_ip/netmask	The subnet mask address. **Note:** This parameter is applicable only when the phone is assigned a static IP address.
network/lan/fixed_ip/gateway	The IP address of the default gateway. **Note:** This parameter is applicable only when the phone is assigned a static IP address.
Domain Name Server (DNS)	
network/lan/fixed_ip/primary_dns	The primary DNS server address. **Note:** This parameter is applicable only when the phone is assigned a static IP address.
network/lan/fixed_ip/secondary_dns	The secondary DNS server address. The phone connects to this server if the primary DNS server is unavailable. **Note:** This parameter is applicable only when the phone is assigned a static IP address.
VLAN	
network/lan/vlan/mode	Determines the VLAN mode of operation. ▪ **[Disable]** Disable ▪ **[Manual]** Manual Configuration of LAN - Static configuration of VLAN ID and priority ▪ **[Automatic]** Automatic Configuration of VLAN/CDP (default) - VLAN discovery mechanism based on Cisco Discovery Protocol (CDP)
network/lan/vlan/id	The VLAN ID. The valid range is 0 to 4096. The default VLAN ID is 0.
network/lan/vlan/priority	The priority of traffic pertaining to this VLAN. The valid range is 0 to 7 (where 7 is the highest priority). The default VLAN priority is 0.

4.2.3 Provisioning Parameters

Table 4-7: Provisioning Configuration File Parameters

Parameter	Description
Note: To add a value to these parameters, enter `provisioning/` followed by the parameter name equals the value (e.g. `provisioning/method=dynamic`).	
provisioning/method	Defines the provisioning method: ▪ **[Disable]** Disable - Automatic update is disabled. The phone attempts to upgrade its firmware and configuration ▪ **[Dynamic]** DHCP Options (Dynamic URL) (default) - Using DHCP options 160 or 66/67 for provisioning ▪ **[Static]** Static URL - Using Static URL for provisioning
provisioning/firmware/url	The static URL for checking the firmware file. The URL must be entered using one of the following syntax options: ▪ <protocol>://<server IP address or host name> ▪ <protocol>://<server IP address or host name>/<firmware file name> Where <protocol> can be one of the following protocols: "ftp", "tftp", "http" or "https". For example: ▪ tftp://192.168.2.1 – retrieved firmware file is 320HD.img ▪ ftp://192.168.2.1/Different_Firmware_Name.img - retrieved firmware file is Different_Firmware_Name.img **Note:** This parameter is applicable only when **method** is configured to "Static".
provisioning/configuration/url	The static URL for checking the configuration file. The URL must be entered using one of the following syntax options: ▪ <protocol>://<server IP address or host name> ▪ <protocol>://<server IP address or host name>/<configuration file name> Where <protocol> can be one of the following protocols: "ftp", "tftp", "http" or "https". For example: ▪ http://192.168.2.1 - configuration file name is <MAC Address>.cfg, for example, 001122334455.cfg ▪ https://192.168.2.1/320HD_<MAC>_conf.cfg - retrieved configuration file name is 320HD_<MAC Address>_conf.cfg, for example, 320HD_001122334455_conf.cfg **Note:** This parameter is applicable only when **method** is configured to "Static".
provisioning/period/type	Defines the period type for automatic provisioning: ▪ **[hourly]** Hourly **-** Sets an interval in hours. ▪ **[daily]** Daily (default) - Sets an hour in the day. ▪ **[weekly]** Weekly - Sets a day in the week and an hour in the day. ▪ **[powerup]** On Power-up Only - The phone tries to upgrade only after power-up.
provisioning/period/hourly/hours_interval	The interval in hours for automatically checking for new firmware and configuration files. The valid range is 1 to 168. The default is 24. **Note:** This parameter is applicable only when **`type`** is configured to "hourly".

Parameter	Description
provisioning/period/daily/time	The hour in the day for automatically checking for new firmware and configuration files. The format of this value is **hh:mm**, where **hh** is hour and **mm** is minutes. For example, `00:30`. The default time is **00:00**. **Note:** This parameter is applicable only when **type** is configured to “daily”.
provisioning/period/weekly/day	The day in the week for automatically checking for new firmware and configuration files. ▪ **[Sunday]** Sunday (default) ▪ **[Monday]** Monday ▪ **[Tuesday]** Tuesday ▪ **[Wednesday]** Wednesday ▪ **[Thursday]** Thursday ▪ **[Friday]** Friday ▪ **[Saturday]** Saturday **Note:** This parameter is applicable only when **type** is configured to “weekly”.
provisioning/period/weekly/time	The hour in the day for automatically checking for new firmware and configuration files. The format of this value is: **hh:mm**, where **hh** is hour and **mm** is minutes. For example: `00:30` The default time is **00:00**. **Note:** This parameter is applicable only when **type** is configured to “weekly”.
provisioning/url_option_value	Determines the DHCP option number to be used for receiving the URL for provisioning. The default value is 160. 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. The following syntax is available for DHCP option 160: ▪ <protocol>://<server IP address or host name> ▪ <protocol>://<server IP address or host name>/<firmware file name> ▪ <protocol>://<server IP address or host name>/<firmware file name>;<configuration file name> ▪ <protocol>://<server IP address or host name>/<configuration file name> Where <protocol> can be one of the following: “ftp”, ”tftp”, ”http” or “https”. For example: ▪ **ftp://192.168.2.1** – retrieved firmware file is *320HD.img* and the configuration file name is <MAC address>.cfg. For example, 001122334455.cfg ▪ **tftp://192.168.2.1/different_firmware_name.img** - retrieved firmware file is Different_Firmware_Name.img and the configuration file name is <MAC address>.cfg. For example, 001122334455.cfg ▪ **http://192.168.2.1/different_firmware_name.img;320HD_<MAC>_conf.cfg** - retrieved firmware file is different_firmware_name.img and the configuration file name is

Parameter	Description
	320HD_<MAC address>_conf.cfg. For example, **320HD_001122334455_conf.cfg** ▪ **https://192.168.2.1/320HD_<MAC>_conf.cfg** - In this case the retrieved firmware file is 320HD.img and the configuration file name is 320HD_<MAC Address>_conf.cfg. For example, 320HD_001122334455_conf.cfg The following syntax is available for DHCP Options 66/67: ▪ Option 66 must be a valid IP address or host name of a TFTP server only. ▪ Option 67 must be the firmware name. If Option 67 is absent, the phone requests for the 320HD.img image file. For example: ▪ **Option 66:** 192.168.2.1 or myTFTPServer ▪ **Option 67:** 320HD_1.2.2_build_5.img **Notes:** ▪ This parameter is applicable only when **method** is configured to “Dynamic”. ▪ It is recommended to leave the parameter at its default value to avoid conflict with other DHCP options settings.
provisioning/random_provisioning_time	Defines the maximum random number to start the provisioning process. 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 **random_provisioning_time** and performs the check only after the random time. The valid range is 0-65535. The default value is 120.
provisioning/XXX_uri	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). For example, in the below setting, *YYY.ext* is retrieved from the server whose information was provided during provisioning (e.g. DHCP options 66/67 or 160: **provisioning/XXX_uri=YYY.ext** Alternatively, the full URL and protocol can be set as follows: **provisioning/XXX_uri=<protocol>://<full path>/YYY.ext** For example: ▪ provisioning/XXX_uri=tftp://10.10.10.10/YYY.ext ▪ provisioning/XXX_uri=ftp:// 10.10.10.10/YYY.ext ▪ provisioning/XXX_uri=http:// 10.10.10.10/YYY.ext ▪ provisioning/XXX_uri=https:// 10.10.10.10/YYY.ext

Parameter	Description
provisioning/ring_tone_uri	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. For example: **provisioning/ring_tone_uri=tones.tgz** **Notes:** ▪ The ringtone file is downloaded only after boot up, and not periodically. ▪ If the tones file is new, the phone updates the information, but does not reboot.
provisioning/corporate_directory_uri	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. For example: **provisioning/corporate_directory_uri=corporate_dir.txt** **Notes:** ▪ The corporate directory file is downloaded after boot up and periodically. ▪ If the corporate directory file is new, the phone updates the information, does not reboot. ▪ For creating a Corporate Directory file, refer to Section 4.3.
provisioning/speed_dial_uri	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. For example: **provisioning/speed_dial_uri=speed_dial_list.txt** **Notes:** ▪ The speed dial file is downloaded after boot up and periodically. ▪ If the speed dial file is new, the phone reboots. ▪ For creating a Speed Dial file, refer to Section 4.4.

4.2.4 TR-069 Management Parameters

Table 4-8: TR-069 Management Configuration File Parameters

Parameter	Description
Note: To add a value to these parameters, enter **management/** followed by the parameter name, equal sign and then the value (e.g. `management/tr069/enabled`=0).	
management/tr069/enabled	Enables TR-069. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note**: This feature can be enabled only if the valid key is provided in the **feature_key** field.
management/tr069/feature_key	Feature key to enable the TR-069.
management/tr069/acs_url	URL for connecting the device to the ACS. **Note:** This parameter must be in the form of a valid HTTP or HTTPS URL.
management/tr069/user_name	Username used to authenticate the device when making a connection to the ACS.

Parameter	Description
management/tr069/password	Password used to authenticate the device when making a connection to the ACS.
management/tr069/connection_request/user_name	Username to authenticate an ACS making a Connection Request to the device.
management/tr069/connection_request/password	Password to authenticate an ACS making a Connection Request to the device.
management/tr069/inform/enabled	Determines whether or not the device must periodically send device information to the ACS using the **Inform method** call. ▪ **[0]** Disable ▪ **[1]** Enable (default)
management/tr069/inform/interval	The duration (in seconds) of the interval for which the device must attempt to connect with the ACS and call the **Inform method** if TR-069 is enabled (**management/tr069/inform/enabled**). The valid range is 0 to 65535. The default is 3600.

4.2.5 VoIP Parameters

4.2.5.1 Line Parameters

Table 4-9: Line Configuration File Parameters

Parameter	Description
Note: The value *%d* refers to the line number.	
voip/line/%d/enabled	Enables the line. ▪ **[0]** Disable ▪ **[1]** Enable The default for Line-0 is 1 (enabled); for the other lines it is 0 (disabled).
voip/line/%d/id	Lines VoIP user's ID for identification to initiate and accept calls. The user's ID can be up to 30 characters.
voip/line/%d/description	Arbitrary name to intuitively identify the line and that is displayed to remote parties as your caller ID.
voip/line/%d/auth_name	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.

Parameter	Description
voip/line/%d/auth_password	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.
voip/line/%d/do_not_disturb/activated	Activates the Do Not Disturb (DnD) feature, if enabled, using the parameter **voip/services/do_not_disturb/enabled**. ▪ **[0]** (default) - Activate ▪ **[1]** - Deactivate **Notes:** DnD can also be activated using the LCD screen interface (more common).

4.2.5.2 Gain Parameters

4.2.5.2.1 General Parameters

Table 4-10: General Gain Configuration File Parameters

Parameter	Description
Hands-free Gain Parameters **Note:** Values are in decibels (dB) and represented as follows: ▪ Negative values: use the word “minus” (e.g. =minus9db). ▪ Positive values: use the word “plus” (e.g. =plus9db). ▪ Decimal places: use underscore instead of period (e.g. plus19_5db).	
voip/audio/gain/additional_speaker_gain	Additional parameter for speaker gain configuration. ▪ **[0]** 0dB ▪ **[1]** 1dB ▪ **[2]** 2dB ▪ **[3]** 3dB (default)
voip/audio/gain/tone_signal_level	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).
voip/audio/gain/ringer_signal_level	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: ▪ **[0]** - Mute ▪ **[1]** - -31 dB ▪ **[32]** - 0 dB ▪ **[63]** - 31 dB

4.2.5.2.2 Hands-Free Gain Parameters

Table 4-11: Hands-Free Gain Configuration File Parameters

Parameter	Description
Hands-free Gain Parameters **Note:** Values are in decibels (dB) and represented as follows: ▪ Negative values: use the word "minus" (e.g. =minus9db). ▪ Positive values: use the word "plus" (e.g. =plus9db). ▪ Decimal places: use underscore instead of period (e.g. plus19_5db).	
voip/audio/gain/handsfree_digital_output_gain	Digital output gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 3 (3dB).
voip/audio/gain/handsfree_digital_input_gain	Digital input gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).
voip/audio/gain/handsfree_analog_output_gain	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 is **0db**.
voip/audio/gain/handsfree_analog_input_gain	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 is **plus39db**.

4.2.5.2.3 Handset Gain Parameters

Table 4-12: Handset Gain Configuration File Parameters

Parameter	Description
Handset Gain Parameters **Note:** Values are in decibels (dB) and represented as follows: ▪ Negative values: use the word “minus” (e.g. =minus9db). ▪ Positive values: use the word “plus” (e.g. =plus9db). ▪ Decimal places: use underscore instead of period (e.g. plus19_5db).	
voip/audio/gain/handset_digital_output_gain	Digital output gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).
voip/audio/gain/handset_digital_input_gain	Digital input gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).
voip/audio/gain/handset_analog_output_gain	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 is **minus9db**.
voip/audio/gain/handset_analog_input_gain	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 is **plus19_5db**.
voip/audio/gain/handset_analog_sidetone_gain	Analog side tone gain (in db). Valid values: **[minus9db]**, **[minus12db]** (default), **[minus15db]**, **[minus18db]**, **[minus21db]**, **[minus24db]**, **[minus27db]**, **[MUTE]** The default is **minus12db**.

4.2.5.2.4 Headset Gain Parameters

Table 4-13: Headset Gain Configuration File Parameters

Parameter	Description
Headset Gain Parameters **Note:** Values are in decibels (dB) and represented as follows: ▪ Negative values: use the word "minus" (e.g. =minus9db). ▪ Positive values: use the word "plus" (e.g. =plus9db). ▪ Decimal places: use underscore instead of period (e.g. plus19_5db).	
voip/audio/gain/headset_digital_output_gain	Digital output gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).
voip/audio/gain/headset_digital_input_gain	Digital input gain (in dB). The valid range is (-32) to 31 (dB), where -32 is mute. The default value is 0 (0dB).
voip/audio/gain/headset_analog_output_gain	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 is **minus12db**.
voip/audio/gain/headset_analog_input_gain	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 is **plus33db**.
voip/audio/gain/headset_analog_sidetone_gain	Analog side tone gain (in db). Valid values: **[minus9db]**, **[minus12db]**, **[minus15db]**, **[minus18db]**, **[minus21db]**, **[minus24db]**, **[minus27db]**, **[MUTE]** The default is **minus12db**.

4.2.5.2.5 Automatic Gain Control (AGC) Parameters

Table 4-14: AGC Configuration File Parameters

Parameter	Description
voip/audio/gain/automatic_gain_control/enabled	Enables the AGC. AGC automatically adjusts the phone's voice volume to compensate for weak or loud signals. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/audio/gain/automatic_gain_control/direction	Determines whether AGC is located before the encoder input (CTL_LOCAL) or after the decoder output (CTL_REMOTE). ▪ **[CTL_REMOTE]** For Remote User (default) - AGC is located after the Decoder output ▪ **[CTL_LOCAL]** For Local User **-** AGC is located before the Encoder input
voip/audio/gain/automatic_gain_control/target_energy	The required output energy (in dBm) of the AGC. The valid range is -31 to 0 (dB). The default value is -19.

4.2.5.3 Codec Parameters

Table 4-15: Codec Configuration File Parameters

Parameter	Description
voip/codec/codec_info/%d/ enabled	Determines the codecs that you want to implement and their priority. Up to five codecs can be configured, where the first codec (i.e., **voip/codec/0/…**) 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. 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. The *%d* variable stands for the priority: ▪ **[0]** - Disabled ▪ **[1]** (default) - Enabled
voip/codec/codec_info /%d/name	Name of the codec. The variable *%d* depicts the index number of the codec entry and its priority, where the first codec (i.e. **voip/codec/codec_info/0/name=…**) 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. ▪ **[G722]** G.722 ▪ **[PCMA]** G.711 A-Law ▪ **[PCMU]** G.711 Mu-Law ▪ **[G729]** G.729 ▪ **[G723]** G.723 For example, **voip/codec/codec_info/0/name=G722**.
voip/codec/codec_info /%d/ptime	Packetization time - length of the digital voice segment that each packet holds. The default is 20 millisecond packets, excluding G.723 which is 30 millisecond packets.
voip/codec/g723_bitrate	Low or high bit rate for G.723. ▪ **[LOW]** Low ▪ **[HIGH]** High (default)
voip/codec/g722_bitrate	G.722 bit rate. ▪ **[G722_64K]** (default) ▪ **[G722_56K]** ▪ **[G722_48K]** **Note:** Currently, only 64bps is supported.

4.2.5.4 Media Streaming Parameters

Table 4-16: Media Streaming Configuration File Parameters

Parameter	Description
voip/media/dtmf_payload	Defines the RTP payload type used for RFC 2833 DTMF relay packets. The valid range is 96 to 127. The default value is 101.
voip/media/media_port	Defines the starting port range for Real Time Protocol (RTP) voice transport. The valid range is 1024 to 65535. The default value is 4000.
voip/media/rtp_mute_on_hold	Mute sending RTP packets to remote in HOLD state. ▪ **[0]** - Disabled. RTP packets are sent to remote end when in HOLD state. ▪ **[1]** (default) - Enabled. RTP packets are not sent to remote end when in HOLD state.
voip/media/out_of_band_dtmf	DTMF transport mode. ▪ **[INBAND]** Inband ▪ **[RFC2833]** RFC 2833 (default) ▪ **[VIA_SIP]** Via SIP
Quality of Service (QoS)	
voip/media/media_tos	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 **0xb8**. Values can be set in decimal (e.g. 184) or hexadecimal (e.g. 0xb8).

4.2.5.5 SIP Signaling Parameters

4.2.5.5.1 General Parameters

Table 4-17: SIP General Configuration File Parameters

Parameter	Description
voip/signalling/sip/transport_protocol	Determines the transport layer for outgoing SIP calls initiated by the phone. ▪ **[UDP]** UDP (default) ▪ **[TCP]** TCP ▪ **[TLS]** TLS
voip/signalling/sip/port	Defines the local SIP port (UDP, TCP or TLS) for SIP messages. The valid range is 1024 to 65535. The default value is 5060.
voip/signalling/sip/proxy_gateway	Assigns a name to the phone. The name is used as the host part of the SIP URI in the From header. **Notes:** ▪ Ensure that the name you choose is the one with which the Proxy is configured to identify the phone. ▪ If not specified, the phone's IP address is used (default).

Parameter	Description
voip/signalling/sip/prack/enabled	Determines whether the phone sends PRACK (Provisional Acknowledgment) messages upon receipt of 1xx SIP reliable responses. ▪ **[0]** Disable ▪ **[1]** Enable (default)
voip/signalling/sip/rport/enabled	Determines whether the phone adds the 'rport' parameter to the relevant SIP message (in the SIP Via header). ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/signalling/sip/sdp_include_ptime	Determines whether the phone adds the PTIME parameter to the SDP message body. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/signalling/sip/keepalive_options/enabled	Determines whether keep-alive is performed using SIP OPTIONS messages sent to the Proxy. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/signalling/sip/keepalive_options/timeout	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.
voip/signalling/sip/connect_media_on_180	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. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/signalling/sip/block_callerid_on_outgoing_calls	When enabled, the outgoing INVITE message is sent with an anonymous From header and P-Asserted-Identity header. ▪ **[0]** Disable (default) ▪ **[1]** Enable For example: ▪ FROM header contains anonymous URI: *From: "Anonymous" sip:anonymous@anonymous.invalid* ▪ P-Asserted-Identity header: *P-Asserted-Identity:* "1001" 1115551001@proxy.net
voip/signalling/sip/anonymous_calls_blocking	When enabled, incoming INVITE messages with anonymous From header are rejected. ▪ **[0]** Disable (default) ▪ **[1]** Enable For example: *From:"Anonymous"<sip:anonymous@anonymous.invalid>* The phone responds with a SIP 403 "Forbidden" response.

Parameter	Description
voip/signalling/sip/auth_ retries	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.
voip/signalling/sip/displa y_name_in_registration_ msg/enabled	Sets the Display Name in the "To" and "From" fields of the SIP REGISTER message. ▪ **[0]** Disable (default) ▪ **[1]** Enable

Parameter	Description
voip/signalling/sip/digit_map	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. The valid value can be up to 256 characters. There are two main formats for the digit map configuration. The formats are distinguished by the separator ‘;’ or ’|’. ▪ **Using ‘|’ separator:** The following constructs can be used in each numbering scheme: ✓ **Digit:** A digit from "0" to "9". ✓ **DTMF:** A digit, or one of the symbols "A", "B", "C", "D", "#", or "*". Extensions may be defined. ✓ **Wildcard:** The symbol "x" which matches any digit ("0" to "9"). ✓ *** Range:** One or more DTMF symbols enclosed between square brackets ("[" and "]"). ✓ **Sub range:** 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 "]". ✓ **Position:** A period (".") which matches an arbitrary number, including zero, of occurrences of the preceding construct. **For example:** [2-9]11|0|100|101|011xxx.|9011xxx.|1[2-9]xxxxxxxxx|91[2-9]xxxxxxxxx|9[2-9]xxxxxx|*xx|[8]xxxx|[2-7]xxx This example includes the following rules: ✓ **[2-9]11:** 911 rule: 211, 311, 411, 511, 611, 711, 811, 911 are dialed immediately ✓ **0:** Local operator rule: After dialing "0" the phone waits T seconds and then completes the call automatically ✓ **100:** Auto-attendant default extension ✓ **101:** Voicemail default extension ✓ **011xxx.:** International rule without prefix ✓ **9011xxx.:** International rule with prefix ✓ **1[2-9]xxxxxxxxx:** LD rule without prefix ✓ **91[2-9]xxxxxxxxx:** LD rule with prefix ✓ **9[2-9]xxxxxx:** Local call with prefix ✓ ***xx:** 2-digit star codes ✓ **[1-7]xx:** A regular 3 digit extension that does not start with 9 or 8 is dialed immediately ✓ **[2-7]xx:** A regular 3 digit extension that does not start with 9 or 8 or 1 is dialed immediately ✓ **[2-7]xxx:** A regular 4 digit extension that does not start with 9 or 8 or 1 is dialed immediately ✓ **[8]xxx:** A 3 digit extension prefixed with an 8 (routes calls directly to voicemail of extension xxx) ✓ **[8]xxxx:** A 4 digit extension prefixed with an 8 (routes calls directly to voicemail of extension xxxx) ✓ **T**: Refers to the **Dialing Timeout.** ▪ **Using ‘;’ separator:** 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.

Parameter	Description
voip/signalling/sip /number_rules	This parameter works in conjunction with the parameter **voip/signalling/sip/digit_map** 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 ';'. The valid value can be up to 256 characters. For example: '4xxx=Line_\\\@10.1.2.3' 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.

4.2.5.5.2 SIP Proxy and Registrar Parameters

Table 4-18: SIP Proxy and Registrar Configuration File Parameters

Parameter	Description
voip/signalling/sip/use_proxy	Determines whether to use a SIP Proxy server. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/signalling/sip/proxy_address	The IP address or host name of the SIP proxy server. The default value is 0.0.0.0.
voip/signalling/sip/proxy_port	The UDP or TCP port of the SIP proxy server. The valid range is 1024 to 65535. The default value is 5060.
voip/signalling/sip/proxy_timeout	The SIP proxy server registration timeout (in seconds). The valid range is 0 to 86400. The default value is 3600.
voip/signalling/sip/sip_registrar/enabled	Determines whether the phone registers to a separate SIP Registrar server. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/signalling/sip/use_proxy_ip_port_for_registrar	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. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/signalling/sip/sip_registrar/addr	The IP address or host name of the Registrar server. The default value is 0.0.0.0.
voip/signalling/sip/sip_registrar/port	The UDP or TCP port of the Registrar server. The valid range is 1024 to 65535. The default value is 5060.
voip/signalling/sip/sip_outbound_proxy/enabled	Determines whether an outbound SIP proxy server is used (all SIP messages are sent to this server as the first hop). ▪ **[0]** Disable (default) ▪ **[1]** Enable

Parameter	Description
voip/signalling/sip/sip_outbound_proxy/addr	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. The default value is 0.0.0.0.
voip/signalling/sip/sip_outbound_proxy/port	The port on which the outbound proxy listens. The valid range is 1024 to 65535. The default value is 5060.
voip/signalling/sip/redundant_proxy/enabled	Enables the redundant proxy mechanism. ▪ **[0]** Disable (default) - Phone doesn't use redundant proxy (default). ▪ **[1] Enable** - Phone registered to redundant proxy if the primary proxy does not respond to SIP signaling messages.
voip/signalling/sip/redundant_proxy/address	The IP address or host name of the redundant proxy. The default value is 0.0.0.0. **Note:** This parameter is applicable only if the parameter **use_redundant_proxy** is set to 1.
voip/signalling/sip/redundant_proxy/port	The UDP or TCP port of the redundant proxy server. The valid range is 1024 to 65535. The default value is 5060. **Note:** This parameter is applicable only if the parameter **use_redundant_proxy** is set to 1.
voip/signalling/sip/redundant_proxy/keepalive_period	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. **Note**: This parameter is applicable only if the parameter **use_redundant_proxy** is set to 1.

Parameter	Description
voip/signalling/sip/redundant_proxy/symmetric_mode	Determines the proxy redundancy mode. ▪ **[0]** (default) - Asymmetric mode. ▪ **[1]** - Symmetric mode. 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. The Redundant Proxy feature includes two operational modes: ▪ **Asymmetric mode:** 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. ▪ **Symmetric mode:** 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 In both modes, the following applies: ▪ 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. ▪ 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. **Note**: This parameter is applicable only if the parameter **use_redundant_proxy** is set to 1.

4.2.5.5.3 SIP Timers Parameters

Table 4-19: SIP Timers Configuration File Parameters

Parameter	Description
voip/signalling/sip/sip_t1	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). The valid range is 100 to 60000. The default value is 500. **Note:** 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): ▪ The first retransmission is sent after 500 msec. ▪ The second retransmission is sent after 1000 (2*500) msec. ▪ The third retransmission is sent after 2000 (2*1000) msec. ▪ The fourth retransmission and subsequent retransmissions until SIPMaxRtx are sent after 4000 (2*2000) msec..
voip/signalling/sip/sip_t2	The maximum interval (in msec) between retransmissions of SIP messages (according to RFC 3261). The valid range is 4000 to 60000. The default value is 200. **Note:** The time interval between subsequent retransmissions of the same SIP message starts with SipT1Rtx and is multiplied by two until SipT2Rtx.
voip/signalling/sip/sip_t4	The SIP T4 retransmission timer according to RFC 3261. The valid range is 5000 to 60000. The default value is 1000.
voip/signalling/sip/sip_invite_timer	The SIP INVITE timer according to RFC 3261. The valid range is 0 to 65535. The default value is 32000.
voip/signalling/sip/session_timer	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. The valid range is 0 to 65535. The default value is 1800.
voip/signalling/sip/min_session_interval	The minimum value for the session interval that the application is willing to accept. The valid range is 0 to 65535. The default value is 900.

4.2.5.5.4 SIP QoS Parameters

Table 4-20: SIP QoS Configuration File Parameters

Parameter	Description
voip/signalling/sip/tos	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. The default value is 0x60. Values can be set in decimal (e.g. 96) or hexadecimal (e.g. 0x60).

4.2.5.6 Dialing Parameters

Table 4-21: Dialing Configuration File Parameters

Parameter	Description
voip/dialing/timeout	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). The valid range is 0 to 10. The default value is 5.
voip/dialing/phone_number_max_size	The maximum length of shortcut numbers that you can enter and the maximum number of digits that you can dial The valid range is 0 to 32. The default value is 19.
voip/dialing/dial_complete_key/enabled	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 dialled digits even if there is no match in the dial plan or digit map. ▪ **[0]** Disable ▪ **[1]** Enable (default) **Note**: This parameter is available only if the parameter **voip/dialing/dial_complete_key/enabled** is set to 1.
voip/dialing/dial_complete_key/key	Defines the Dialing Complete key. The valid value is a single character. The default value is the pound (#) key.
voip/dialing/dialtone_timeout	The maximum duration of the dial tone (in seconds) after which the dial tone stops and a reorder tone is played. The valid range is 0 to 300. The default value is 30.
voip/dialing/warning_tone_timeout	The maximum duration of the reorder tone (in seconds) after which the reorder tone stops and a howler tone is played. The valid range is 0 to 300. The default value is 40.
voip/dialing/unanswered_call_timeout	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 The valid range is 0 to 300. The default value is 60.

Parameter	Description
voip/dialing/offhook_tone_timeout	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. The valid range is 0 to 300. The default value is 120.
voip/dialing/secondary_dial_tone/enabled	Enables the secondary dial tone. ▪ **[0]** Disable - Phone doesn't use secondary dial tone. ▪ **[1]** Enable (default) - Phone plays secondary dial tone if the secondary dial tone key is pressed (first digit).
voip/dialing/secondary_dial_tone/key_sequence	Secondary dial tone is played if this is the first key pressed. The valid range is 0 to 9. The default value is 9. **Note**: This parameter is available only if the parameter **voip/dialing/secondary_dial_tone/enabled** is set to 1.
voip/dialing/automatic_disconnect	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. ▪ **[0]** Disable ▪ **[1]** Enable(default)
Automatic Dialling	
voip/dialing/auto_dialing/enabled	Determines whether automatic dialing is enabled (i.e., phone number is automatically dialed when you off-hook the phone). ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/dialing/auto_dialing/timeout	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.
voip/dialing/auto_dialing/destination	The number that is automatically dialed when the phone is off-hooked. The valid value can be up to 32 characters.

4.2.5.7 Voice Parameters

Table 4-22: Voice Configuration File Parameters

Parameter	Description
voip/audio/jitter_buffer/min_delay	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.
voip/audio/jitter_buffer/optimization_factor	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.
voip/audio/echo_cancellation/enabled	Enables echo cancellation. ▪ **[0]** - Disable ▪ **[1]** (default) – Enable **Note:** Disabling echo cancellation should be done for testing purposes only.
voip/audio/silence_compression/enabled	Enables silence compression for reducing network bandwidth consumption. ▪ **[0]** Disable (default) ▪ **[1]** Enable

4.2.5.8 Supplementary Services Parameters

4.2.5.8.1 General Parameters

Table 4-23: General Supplementary Services Configuration File Parameters

Parameter	Description
voip/services/application_server_type	Defines the type of the Application server to which the device is registered. ▪ **[Generic]** Generic (default) ▪ **[Asterisk]** Asterisk ▪ **[BSFT]** Broadsoft **Note:** In addition to the general supplementary services parameters, parameters (if any) unique to the selected Application server become applicable. For a listed of the unique parameters per Application server, refer to Section 5 on page 93.
voip/services/out_of_service_bahavior	Determines whether a reorder tone is played instead of a dial tone if you configured a Registrar IP address and the registration failed. ▪ **[NONE]** No Tone ▪ **[REORDER_TONE]** Reorder Tone (default)

4.2.5.8.2 Call Waiting Parameters

Table 4-24: Call Waiting Configuration File Parameters

Parameter	Description
voip/services/call_waiting/enabled	Enables the Call Waiting feature. ▪ **[0]** Disable ▪ **[1]** Enable (default)
voip/services/call_waiting/sip_reply	Determines the SIP response that is sent when another call arrives while a call is in progress: ▪ **[RINGING] Ringing** - 180 Ringing ▪ **[QUEUED] Queued** (default) - 182 Queued

4.2.5.8.3 Call Forward Parameters

Table 4-25: Call Forward Configuration File Parameters

Parameter	Description
voip/services/call_forward/line/0/enabled	Enables the Call Forward feature. ▪ **[0]** Disable ▪ **[1]** Enable (default)
voip/services/call_forward/line/0/type	Determines the condition upon which incoming calls are forwarded to another destination: ▪ **[Unconditional]** Unconditional - incoming calls are forwarded independently of the status of the line. ▪ **[Busy]** Busy - incoming calls are forwarded only if the phone is busy. ▪ **[No_Reply]** No Reply (default) - incoming calls are forwarded only if the phone does not answer before a user-defined timeout.
voip/services/call_forward/line/0/timeout	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.
voip/services/call_forward/line/0/destination	The destination to which the call is directed when call forward is activated.
voip/services/call_forward/line/0/active	Activates call forwarding, if it has been enabled (using the parameter **voip/services/call_forward/line/0/enabled**). ▪ **[0]** (default) - Disable ▪ **[1]** - Enable **Note:** Call forwarding can also be activated using the LCD screen (common).

4.2.5.8.4 Do Not Disturb Parameters

Table 4-26: Do Not Disturb Configuration File Parameters

Parameter	Description
voip/services/do_not_disturb/enabled	Enables the Do not Disturb feature. ▪ **[0]** Disable ▪ **[1]** Enable (default)

4.2.5.8.5 Message Waiting Indication Parameters

Table 4-27: Message Waiting Indication Configuration File Parameters

Parameter	Description
voip/services/msg_waiting_ind/enabled	Enables the MWI feature. ▪ **[0]** Disable ▪ **[1]** Enable (default)
voip/services/msg_waiting_ind/subscribe	Determines whether the phone registers to an MWI server. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/services/msg_waiting_ind/subscribe_port	The port number of the MWI server. The valid range is 1024 to 65535. The default value is 5060.
voip/services/msg_waiting_ind/subscribe_address	The IP address or host name of the MWI server. The default value is 0.0.0.0.
voip/services/msg_waiting_ind/expiraition_timeout	The interval between the MWI Subscribe messages. The valid range is 0 to 86400. The default value is 3600.
voip/services/msg_waiting/stutter_tone_duration	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.
voip/services/msg_waiting_ind/voice_mail_number	Defines the extension number for accessing your voice mail messages. The valid value is up to 64 characters.

4.2.5.8.6 Busy Lamp Field (BLF) Parameters

Table 4-28: Busy Lamp Field (BLF) Configuration File Parameters

Parameter	Description
voip/services/busy_lamp_field/enabled	Enables the BLF feature: ▪ **[0]** Disable ▪ **[1]** Enable (default)
voip/services/busy_lamp_field/subscription_period	The interval between BLF and SIP SUBSCRIBE messages. The valid range is 0 to 86400. The default value is 3600.
voip/services/busy_lamp_field/uri	The user resource list. This must be the username (not the domain name). For example, if the URI resource list is **mylist@server.com**, then only the value **mylist** must be entered. The valid value is up to 64 characters. **Note:** This parameter is applicable only if the parameter **voip/services/application_server_type** is set to **BSFT**.
voip/services/busy_lamp_field/application_server/use_registrar	Determines whether to use the registrar as the application server address. When enabled, there is no need to configure the application server address or domain name separately. ▪ **[0]** Disable (default) ▪ **[1]** Enable **Note:** This parameter is applicable only if the parameter **voip/services/application_server_type** is set to **BSFT**.
voip/services/busy_lamp_field/application_server/addr	The IP address or host name of the application server. The valid value is up to 64 characters. The default value is 0.0.0.0. **Note:** This parameter is applicable only if the parameter **voip/services/application_server_type** is set to **BSFT** and **voip/services/busy_lamp_field/application_server/use_registrar** is set to 0.

4.2.6 Regional Settings and Call Progress Tones Parameters

Table 4-29: Regional Settings Configuration File Parameters

Parameter	Description
voip/regional_settings/selected_country	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. Supported countries: ▪ **[Israel]** Israel ▪ **[China]** China ▪ **[France]** France ▪ **[Germany]** Germany ▪ **[Netherlands]** Netherlands ▪ **[UK]** UK ▪ **[Brazil]** Brazil ▪ **[Italy]** Italy ▪ **[Argentina]** Argentina ▪ **[Portugal]** Portugal ▪ **[USA]** USA
voip/regional_settings/use_config_file_values	Enables the user-defined CPT. When this parameter is enabled, the **selected_country** parameter is not relevant and the below Call Progress Tones values can be determined by the user. ▪ **[0]** (default) - Disable ▪ **[1]** - Enable
Call Progress Tones (CPT) **Note:** Up to 10 CPT's can be configured (**voip/regional_settings/call_progress_tones/0…9**).	
voip/regional_settings/call_progress_tones/%d/enabled	Enables the specific CPT. ▪ **[0]** - Disable ▪ **[1]** - Enable
voip/regional_settings/call_progress_tones/%d/name	Defines the name of the CPT.
voip/regional_settings/call_progress_tones/%d/cadence	Defines the cadence type of the tone. ▪ **[0]** - Continuous signal ▪ **[1]** - Cadence signal ▪ **[2]** - Burst signal
voip/regional_settings/call_progress_tones/%d/frequency_a	Defines the low frequency (in Hz) of the tone. The valid value range is 300 to 1980 Hz, in steps of 1 Hz. Unused frequencies must be set to zero.
voip/regional_settings/call_progress_tones/%d/frequency_b	Defines the high frequency (in Hz) of the tone. The valid value range is 300 to 3000 Hz, in steps of 1 Hz. Unused frequencies must be set to zero.

Parameter	Description
voip/regional_settings/call_progress_tones/%d/frequency_a_level	Output level of the low frequency tone (in -dBm) in Call Progress generation. The valid range is 0 to 63, where 63 is mute.
voip/regional_settings/call_progress_tones/%d/frequency_b_level	Output level of the low frequency tone (in -dBm) in Call Progress generation. The valid range is 0 to 63, where 63 is mute.
voip/regional_settings/call_progress_tones/%d/tone_on_0	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.
voip/regional_settings/call_progress_tones/%d/tone_off_0	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 4-30: Packet recording Configuration File Parameters

Parameter	Description
voip/packet_recording/enabled	Activates the packet recording mechanism. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/packet_recording/remote_ip	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.
voip/packet_recording/remote_port	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.
voip/packet_recording/rtp_recording/enabled	Activates the DSP RTP recording. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/packet_recording/ec_debug_recording/enabled	Activates the Echo Canceller Debug recording. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/packet_recording/network_recording/enabled	Activates the DSP network (TDM Out) recording. ▪ **[0]** Disable (default) ▪ **[1]** Enable
voip/packet_recording/tdm_recording/enabled	Activates the DSP TDM (TDM In) recording. ▪ **[0]** Disable (default) ▪ **[1]** Enable

4.2.8 LCD Display Parameters

Table 4-31: LCD Display Configuration File Parameters

Parameter	Description
personal_settings/language	Determines the LCD display user interface language. ▪ **[English]** English (default) ▪ **[Spanish]** Spanish ▪ **[Russian]** Russian ▪ **[Portuguese]** Portuguese ▪ **[German]** German ▪ **[Ukraine]** Ukraine ▪ **[French]** French
personal_settings/lcd_contrast	Determines the LCD contrast. The valid range is 0 to 14. The default value is 7.

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>(Speed
Dial or Speed Dial+BLF)
```

The *type* variable represents speed dial (enter the value “0”) or speed dial with BLF (enter the value “1”).

For example:

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

Reader's Notes

5 Application Server-Specific Configurations

This chapter describes configurations specific to third-party Application servers. These configurations use unique parameters.

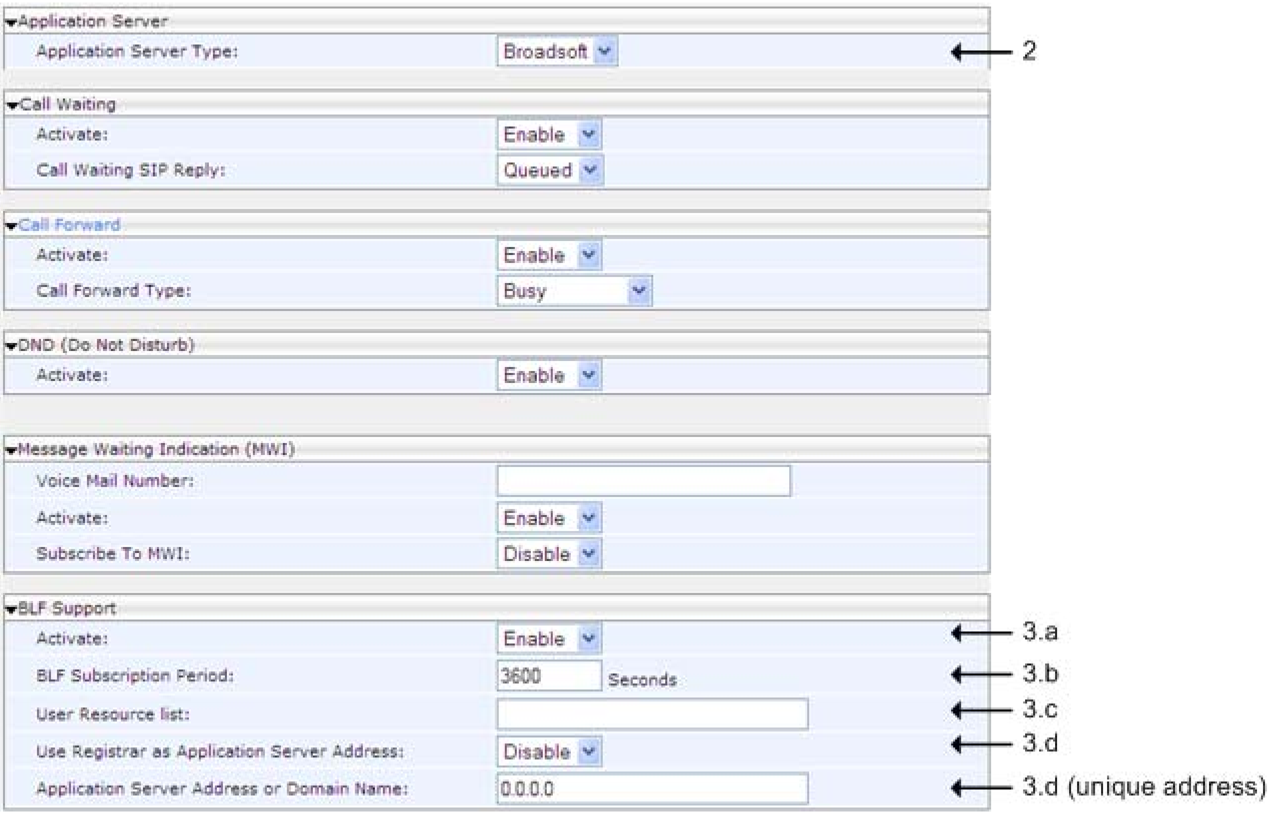
5.1 Configuring BLF for BroadWorks™

The configuration of the BLF feature is unique when the selected application server is BroadWorks.

➢ **To configure BLF for BroadWorks:**

1. Access the Services page (**Configuration** tab > **Voice Over IP** menu > **Services**). The figure's callouts below correspond to the steps in this procedure:

Figure 5-1: BLF Configuration for BroadWorks



2. From the 'Type' drop-down list, select "Broadsoft".
3. In the **BFL Support** group, do the following:
   a. From the 'Activate' drop-down list (*voip/services/busy_lamp_field/enabled* parameter), select "Enable".
   b. In the 'BLF Subscription Period' field (*voip/services/busy_lamp_field/subscription_period* parameter), enter the interval between BLF and SIP SUBSCRIBE messages.

c. In the ‘User Resource List’ field (*voip/services/busy_lamp_field/uri* parameter), enter the resource list URI to which the phone can subscribe to in order to get the BLF information from the application server.

Note: For the URI resource list, enter only the username (not the domain name). For example, if the URI resource list is **mylist@server.com**, then only the value **mylist** must be entered. The application server’s address can be the same as the SIP Registrar address or specifically defined for the application server, as described below.

d. Define the application server’s address:

- **Same address as SIP Registrar:** From the ‘Use Registrar as Application Server Address’ (*voip/services/busy_lamp_field/application_server/use_registrar* parameter) drop-down list, select “Enable”. The Registrar’s address is used as the application server’s address, defined by the parameter *voip/signalling/sip/sip_registrar/addr* (refer to Section 4.2.5.5.2).
- **Unique address:** From the ‘Use Registrar as Application Server Address’ drop-down list, select “Disable”, and then in the “Application Server Address or Domain Name” field (*voip/services/busy_lamp_field/application_server/addr* parameter), enter the IP address or domain name of the application server.

4. Click **Submit**.

5. Define speed dial keys with the BLF feature (refer to Section 3.5).

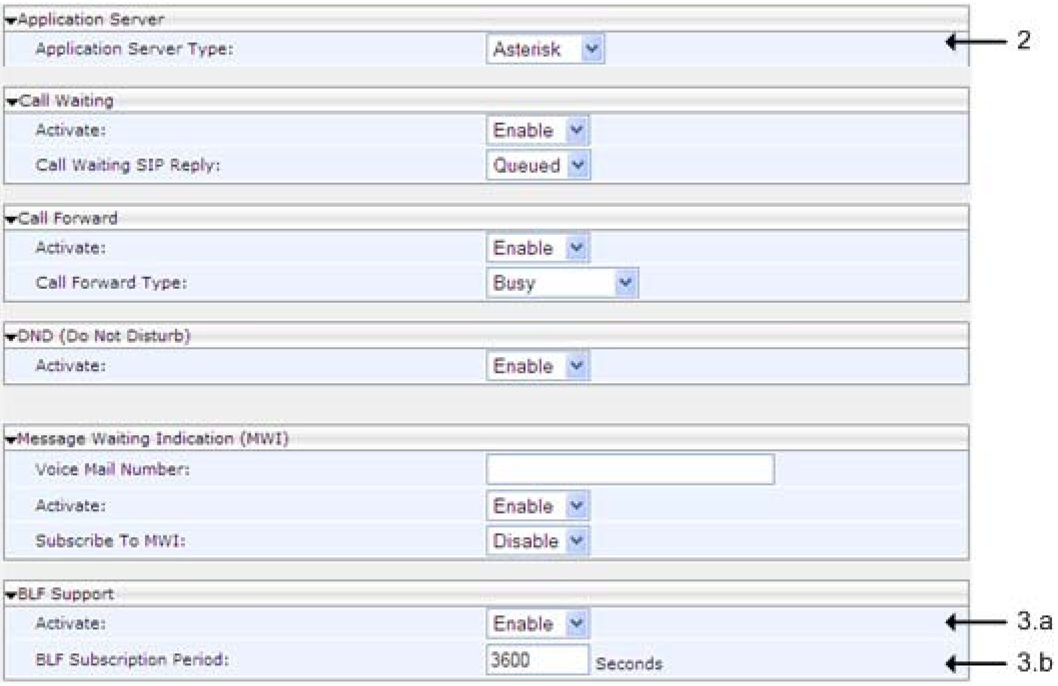
5.2 Configuring BLF for Asterisk

The configuration of the BLF feature is unique when the selected application server is Asterisk.

➢ **To configure BLF for Asterisk application server:**

1. Access the Services page (**Configuration** tab > **Voice Over IP** menu > **Services**). The figure's callouts below correspond to the steps in this procedure:

Figure 5-2: BLF Configuration for Asterisk



2. From the 'Type' drop-down list, select "Asterisk".
3. In the **BFL Support** group, configure the following:
   a. From the 'Activate' drop-down list (*voip/services/busy_lamp_field/enabled* parameter), select "Enable".
   b. (Optional) In the 'BLF Subscription Period' field (*voip/services/busy_lamp_field/subscription_period* parameter), enter the interval between BLF and SIP SUBSCRIBE messages.

Note: The application server's address is the same as the SIP Registrar address defined by the parameter *voip/signalling/sip/sip_registrar/addr* (refer to Section 4.2.5.5.2).

4. Click **Submit**.
5. Define speed dial keys with the BLF feature (refer to Section 3.5).

Reader's Notes

6 Built-in Test Plan

The phone provides a built-in test for verifying its operational integrity. This test can be performed before deployment and in cases of troubleshooting. The built-in test is comprised of numerous tests that are 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
- Green-color LEDs test
- Red-color LEDs 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

6.1 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 6.2 on page 98.

6.2 Keypad and Off-Hook/On-Hook Test

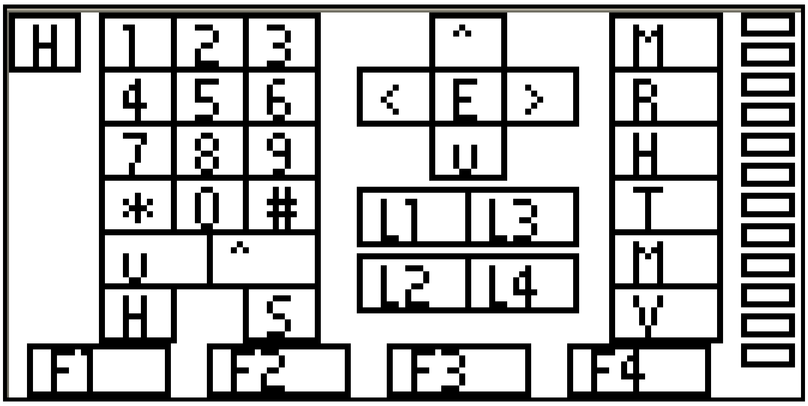
The Keypad and Off-Hook/On-Hook tests checks the responsiveness and correct functioning of the following:

- All the keys on the phone's keypad
- Handset when picked up (off-hook) and placed back on the receiver (on-hook)

➢ **To test the keypad and on-hook state:**

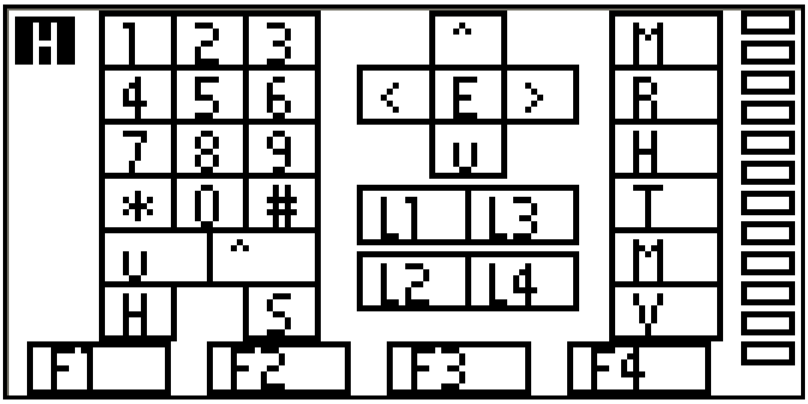
1. Using the keypad, dial the number "0123456789" and then press the star ("*") key; the phone's LCD screen displays a graphical display of the keypad, where each key has a corresponding indicator on the LCD screen.

Figure 6-1: Keypad Test



2. Off-hook the handset; the hook indicator ("H") on the LCD screen turns white on a black background.

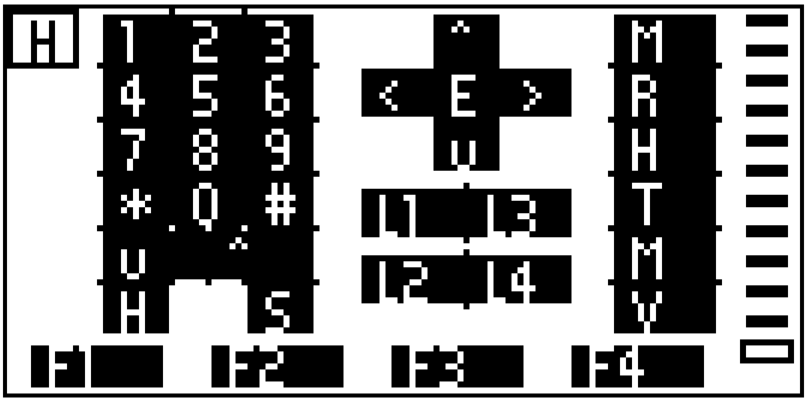
Figure 6-2: Keypad Test – Off-Hook



3. On-hook the handset; the hook indicator on the LCD screen turns white again.

4. Press each key on the phone's keypad; each key that is pressed turns its corresponding indicator on the LCD screen to the color black.

Figure 6-3: Keypad Test - Keys



5. Continue with the Green-color LEDs test in Section 6.3 on page 100.

6.3 Green-Color LEDs Test

Upon the successful completion of the Keypad and On-Hook/Off-Hook test (in the previous section), the LCD screen displays the message “Green LEDs are on”. Consequently, all LEDs light up green (except the VOICE MAIL, HEADSET, SPEAKER, and MUTE LEDs, which are red only), as shown in the figure below:

Figure 6-4: Green LEDs On

Continue with the Red-Color LEDs test in Section 6.4 on page 100.

6.4 Red-Color LEDs Test

Upon the completion of the Green-Color LEDs test (in the previous section), perform the Red-color LEDs test for verifying that the red light is functioning for the phones LEDs.

➢ **To test the LEDs for red-color lighting:**

1. Press any key; all the LEDs turn red, as shown in the figure below:

Figure 6-5: Red LEDs Test

2. Continue with the Handset test in Section 6.5 on page 101.

6.5 Handset Test

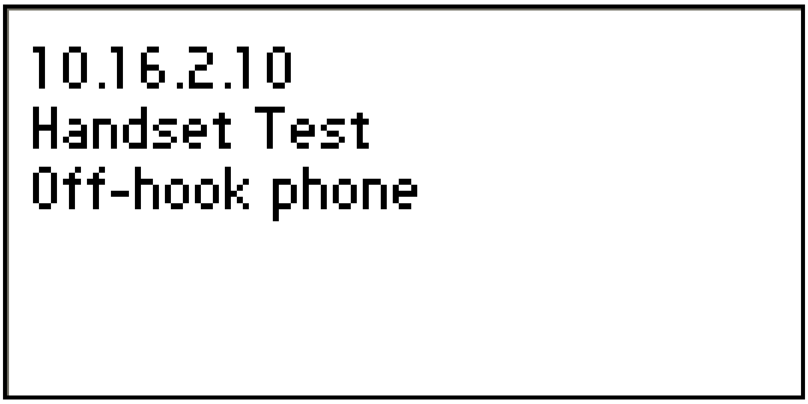
Upon successful completion of the Red-color LEDs test (in the previous section), perform the Handset test. This test verifies the correct functioning of the handset, which includes the following:

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

➢ **To test the handset:**

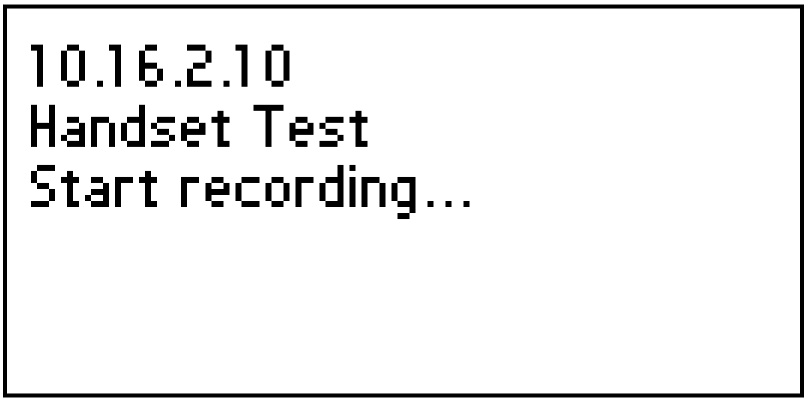
1. Press any key; the LCD screen displays the message "Prepare to test handset. Off hook to start".

Figure 6-6: Handset Test



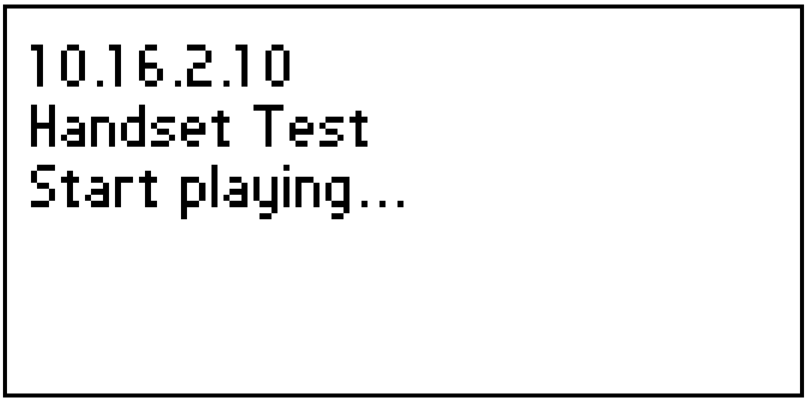
2. Off hook the phone; the LCD screen displays the message "start recording".

Figure 6-7: Handset Test - Recording



3. Speak into the handset microphone; after about five seconds, the LCD screen displays the message ”start playing”.

Figure 6-8: Handset Test- Playing



The voice message 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 6.6 on page 102.

6.6 Headset Test

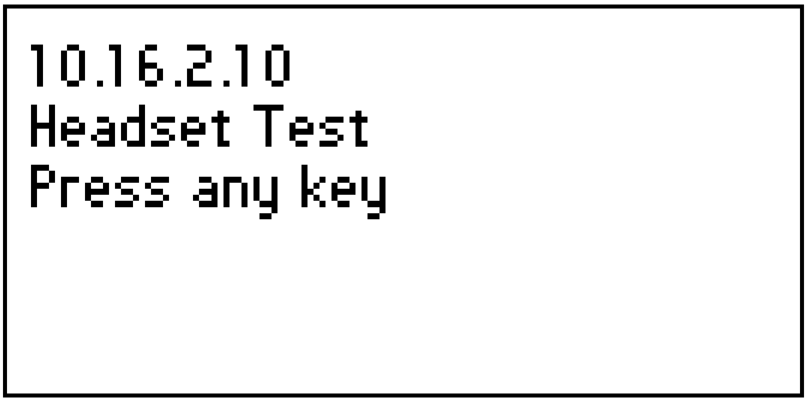
Upon successful completion of the Handset test (in the previous section), perform the Headset test. This test verifies the correct functioning of the headset (for hands-free operation), which includes the following:

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

➢ **To test the headset:**

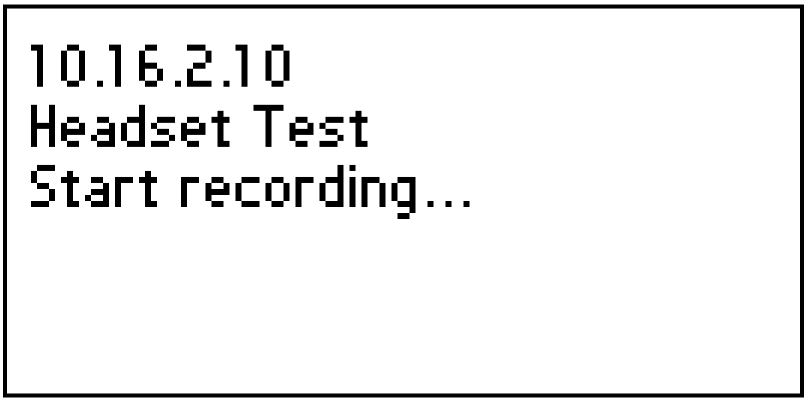
1. After about five seconds after your recorded voice is played in the Handset test (in the previous section), the LCD screen displays the message ”Prepare to test headset. Press any key to start”.

Figure 6-9: Headset Test



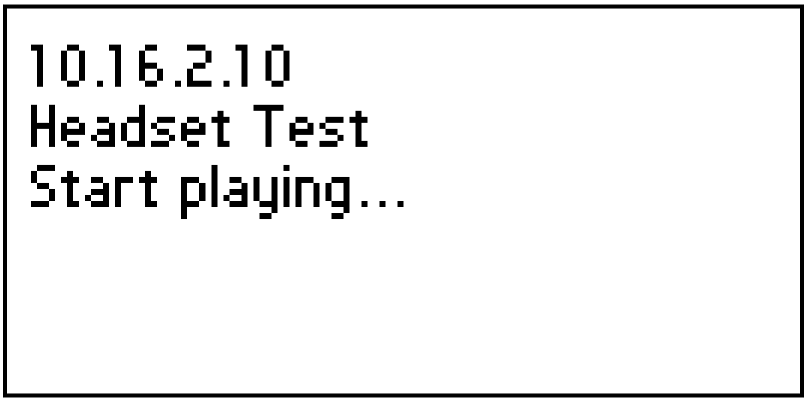
2. Press any key; the LCD screen displays the message ”start recording”.

Figure 6-10: Headset Test - Recording



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

Figure 6-11: Headset Test - Playing



4. Continue with the Speaker test in Section 6.7 on page 104.

6.7 Speaker Test

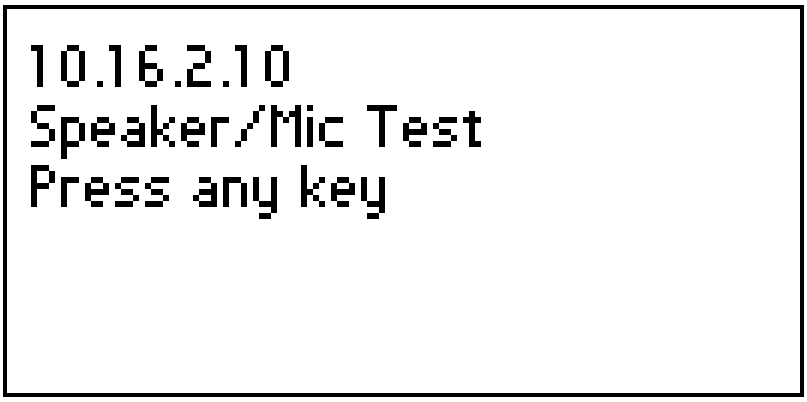
Upon successful completion of the Headset test (in the previous section), perform the Speaker test. This test verifies the correct functioning of the speakers, which includes the following:

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

➢ **To test the speaker:**

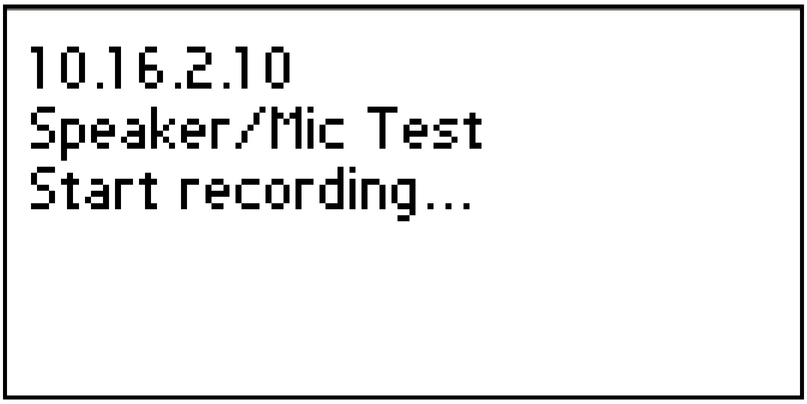
1. After about five seconds after your recorded voice is played in the Headset test (in the previous section), the LCD screen displays the message ”Speaker/Mic Test. Press any key”.

Figure 6-12: Speaker/Mic Test



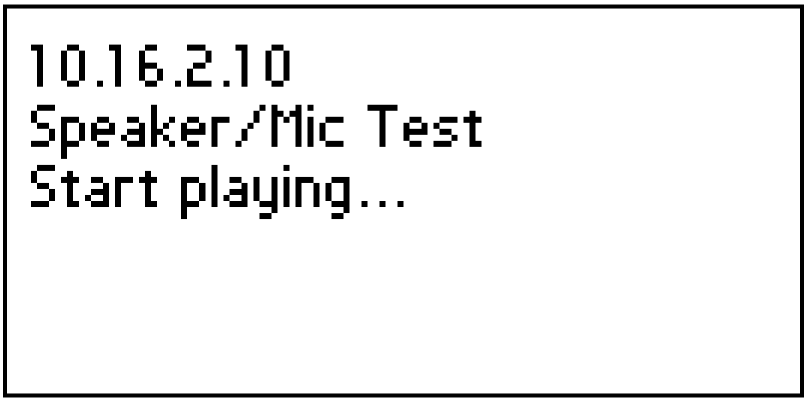
2. Press any key; the LCD screen displays the message ”Start recording”.

Figure 6-13: Speaker/Mic Test– Recording



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 6-14: Speaker/Mic Test – Playing



4. Continue with the MAC Address and Serial Number Verification test in Section 6.8 on page 106.

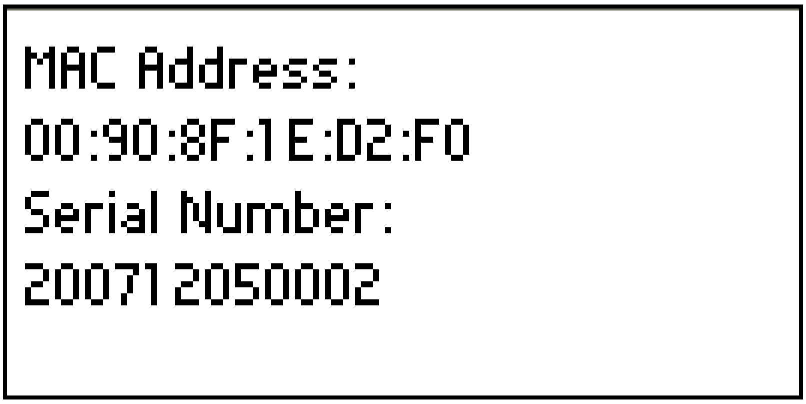
6.8 MAC Address and Serial Number Verification Test

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 6-15: MAC Address and Serial Number Verification Test



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

7 Specifications

Table 7-1: 320HD IP Phone Specifications

Feature	Details
VoIP Signaling Protocols	▪ SIP: RFC 3261, RFC 2327 (SDP)
Data Protocols	▪ IPv4, TCP, UDP, ICMP, ARP, DNS and DNS SRV for SIP Signaling ▪ SIP over TLS (SIPS) ▪ 802.1p/Q for Traffic Priority and QoS ▪ ToS (Type of Service) field, indicating desired QoS DHCP Client ▪ DHCP Client ▪ NTP Client
Media Processing	▪ Voice Coders: G.711, G.723.1, G.729A/B, G.722 ▪ Acoustic Echo Cancelation: G.168-2004 compliant, 64-msec tail length ▪ Adaptive Jitter Buffer 300 msec ▪ Voice Activity Detection ▪ Comfort Noise Generation ▪ Packet Lost Concealment ▪ RTP/RTCP Packetization (RFC 3550, RFC 3551) ▪ DTMF Relay (RFC 2833)
Telephony Features	▪ Call Hold / Un Hold ▪ Call Transfer ▪ 3-way Conference (with local mixing) ▪ Redial ▪ Caller ID Notification ▪ Call Waiting Indication ▪ Message Waiting Indication (including MWI LED) ▪ Local and Corporate Directories ▪ Automatic On-hook Dialing ▪ CWRR (Call Waiting Reminder Ring) ▪ Secondary Dial Tone ▪ Call Logs: Missed/Received Calls and Dialed Numbers ▪ Speed Dial (including BLF support for Asterisk and BroadWorks™) ▪ Dial Plan ▪ Call Forward (Unconditional / Busy / No answer) ▪ Redundant SIP Proxy Mechanism ▪ Multiple Line (Currently support up to 4 lines)

Feature	Details
Configuration/ Management	▪ TR-069 ▪ LCD Display User Interface Language Support (Various Languages) ▪ Web-based Management (HTTPS) ▪ Auto-Provisioning (via TFTP, FTP, HTTP, and HTTPS) for firmware and configuration file upgrade ▪ DHCP options (66, 67, and 160) for Auto-provisioning ▪ DHCP options (12, 60, and 77) for device information
Hardware	▪ **LCD screen:** Graphic LCD (132*64) mono ▪ **Connectors interfaces:** ✓ 2 x RJ-45 ports (10/100BaseT Ethernet) for WAN and LAN ✓ RJ-9 port (jack) for Headset ✓ RJ-9 port (jack) for Handset ▪ **Mounting:** ✓ Wall mounting ✓ Adjustable angle tilt for desktop mounting ▪ **Power:** ✓ DC jack adapter 12V ✓ Power supply AC 100 ~ 240V ✓ PoE: IEEE802.3af (optional) ▪ **Keys:** ✓ 12 x speed dial keys ✓ Line 1 ~ 2 (LED) ✓ 4 x softkeys ✓ VOICE MAIL message hotkey with LED indicator ✓ 4-way navigation keys with ENTER Key ✓ MENU ✓ REDIAL ✓ HOLD ✓ MUTE (LED) ✓ TRANSFER ✓ VOLUME control key ✓ HEADSET ✓ SPEAKER
Headset Compatibility	▪ GN.com 2100 – both single and dual ▪ Jabra BIZ 2400 – both single and dual ▪ GN.com 9350 ▪ Plantronics H261N ▪ Plantronics H251N

Reader's Notes

AudioCodes 300HD Series of High Definition IP Phones

HD VoIP

320HD IP Phone

Administrator's Manual

320HD IP Phone

Version 1.4.0

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