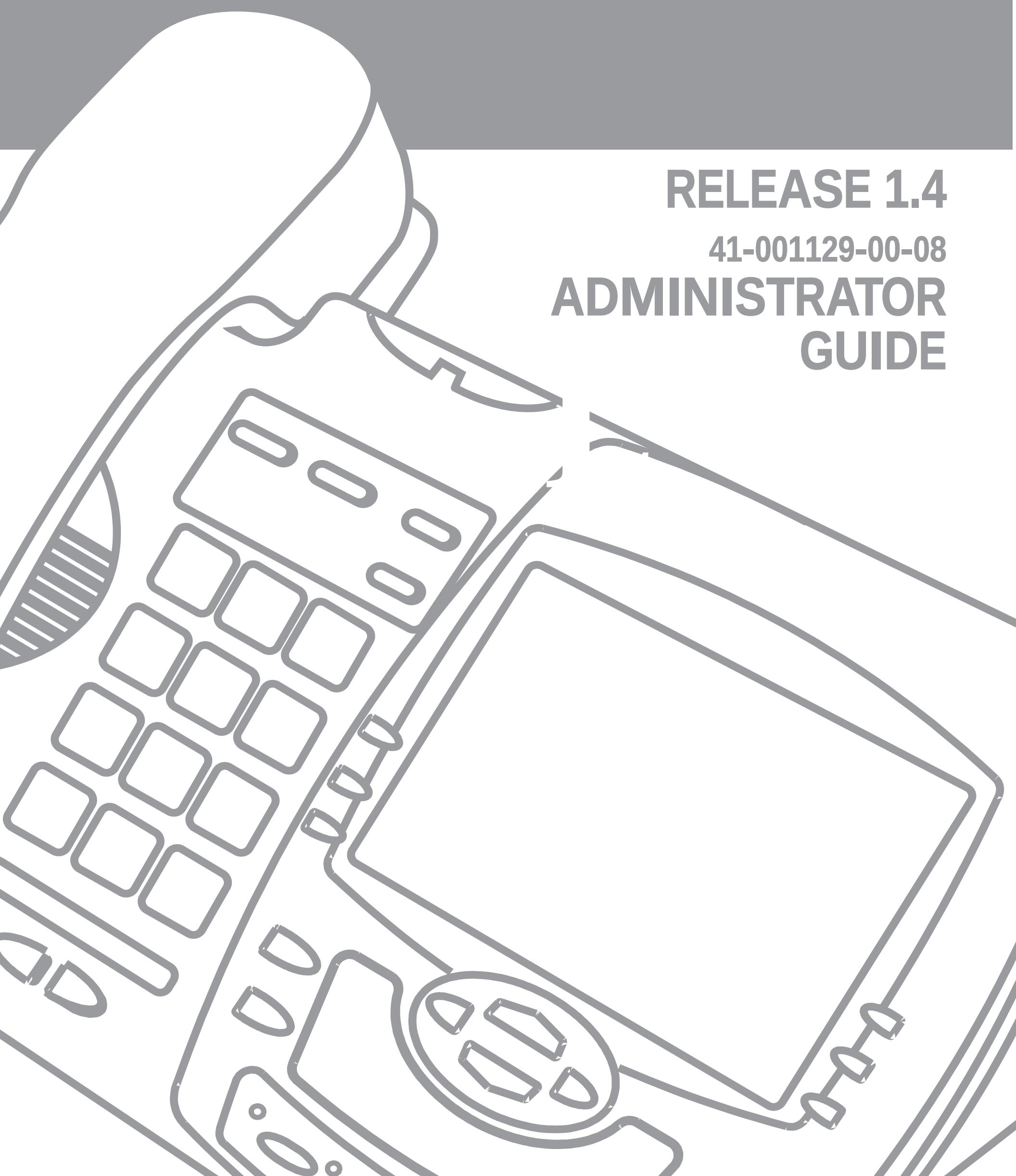
MODEL

480i, 480i CT, 9112i, 9133i

SIP IP PHONE



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ADMINISTRATOR GUIDE

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About this Guide

This guide targets network administrators, system administrators, developers and partners who need to understand how to install the IP phone on a SIP network. It also provides some user-specific information.

This guide contains information that is at a technical level, more suitable for system or network administrators. Prior knowledge of IP Telephony concepts is recommended.

This guide complements the Aastra product-specific *Installation Guide* and the Aastra product-specific *User Guide*.

- ***Installation Guide*** – contains installation and set-up instructions, information on general features and functions, and basic options list customization. Included with the phone.
- ***Administration Guide*** – explains how to set the phone up on the network, as well as advanced configuration instructions for the SIP IP phone. This guide contains information that is at a technical level more suitable for a system or network administrator.
- ***User Guide*** – explains the most commonly used features and functions for an end user.

Overview

This *SIP IP Phone Administrator Guide* provides information on the basic network setup required for the IP phones, Models 480i, 480i Cordless (480i CT), 9112i, and 9133i. It also includes details on the functioning and configuration of the IP phones.

Note: Features, characteristics, requirements, and configuration that are specific to a particular IP phone model are indicated where required in this guide.

Firmware Installation Methods

The firmware setup and installation for the IP phone can be done using any of the following:

- Phone keypad menu (Phone UI)
- Aastra Web-based user interface (Aastra Web UI)

When the IP phone is initialized for the first time, DHCP is enabled by default. Depending on the type of configuration server setup you may have, the IP phone may download a firmware version automatically, or you may need to download it manually.

Installation Considerations

The following considerations must be made before connecting the IP phone to the network:

- If you are planning on using dynamic IP addresses, make sure a DHCP server is enabled and running on your network.
- If you are not planning on using dynamic IP addresses, see "Configuring Network Settings" on page 23 for manually setting up an IP address.

To install the IP phone hardware and cabling, refer to the product-specific *Aastra Installation Guide*.

Installation Requirements

The following are general requirements for setting up and using your SIP IP phone:

- A SIP-based IP PBX system or network installed and running with a number created for the new IP phone.
- Adherence to SIP standard RFC 3261.
- Access to a configuration server where you can store the firmware image and configuration files.
- The IP phone must be configured for a specific type of protocol to use. TFTP is enabled by default. You can configure the following protocols on the IP phone:
  - **TFTP** (Trivial File Transfer Protocol)

Note: If you set TFTP, the configuration server must be able to accept connections anonymously.

  - **FTP** (File Transfer Protocol)
  - **HTTP** (Hypertext Transfer Protocol)
- A 802.3 Ethernet/Fast Ethernet LAN
- Category 5/5e straight through cabling
- Power over Ethernet (PoE) power supply (optional accessory – necessary only if no inline power is provided on the network). (Not applicable to 9112i)
- Power adapter (included for certain models of 9112i, 9133i, and 480i CT).

Configuration Server Requirement

A basic requirement for setting up the IP phone is to have a configuration server. The configuration server allows you to:

- Store the firmware images that you need to download to your IP phone.
- Stores configuration files for the IP phone
- Stores the software when performing software upgrades to the IP phone

Reference

For setting up your configuration server as a TFTP server, see "Appendix A: Configuration Server Protocol Setup."

Firmware/Configuration Files

When the IP phone is initialized for the first time, DHCP is enabled by default. Depending on the type of configuration server setup you may have, the IP phone may download a firmware version and configuration files automatically, or you may need to download it manually.

Note: Automatic download is dependant on your configuration server setup.

The firmware consists of a single file called:

- *<phone model>.st*

The configuration files consist of two files called:

- *aastra.cfg*
- *<mac>.cfg*

The following table provides the firmware for each Aastra IP phone model.

IP Phone Model	Associated Firmware
480i	480i.st
480i CT	480i Cordless.st
9112i	9112i.st
9133i	9133i.st

Configuration Methods

You can use the following to setup and configure the IP phone:

- IP phone UI
- Aastra Web UI
- Configuration files

Models 480i and 480i CT have 18 softkeys available to configure the IP phone. Model 9133i has 7 programmable keys. Model 9112i has 2 programmable keys.

<u>References</u>

For setting up and configuring the IP phone using either the IP phone UI, the Aastra Web UI, or the configuration files, see "Configuring the IP Phone" on page 22.

For information about the softkey and programmable key parameters, see "Softkey/ Programmable Key Parameters" on page 172.

Installing the Firmware/ Configuration Files

The following procedure describes how to install the firmware and configuration files.

<u>To install firmware/configuration files</u>:

1. If DHCP is disabled, manually enter the configuration server's IP address. For details on setting DHCP, see "DHCP" on page 22.
2. Copy the firmware file *<phone model>.st* to the root directory of the configuration server. This firmware file is downloaded only if it is different from the firmware currently loaded on the IP phone.

Note: The *<phone model>* attribute is the IP phone model (i.e., 480i.st, 9133i.st)

3. Copy the Aastra configuration files (*aastra.cfg* and *<mac>.cfg*) to the root directory of the configuration server.

Note: The *<mac>* attribute represents the actual MAC address of your phone. (i.e., *00085D030996.cfg*).

Configuration File Precedence

Aastra IP phones can accept two sources of configuration data:

- The server configuration most recently downloaded/cached from the configuration server files, *aastra.cfg*/*<mac>.cfg* (or the *aastra.tuz*/*<mac>.tuz* encrypted equivalents).
- Local configuration changes stored on the phone that were entered using either the IP phone UI or the Aastra Web UI

In the event of conflicting values set by the different methods, values are applied in the following sequence:

1. Default values hard-coded in the phone software
2. Values downloaded from the configuration server
3. Values stored locally on the phone

The last values to be applied to the phone configuration are the values that take effect.

For example, if a parameter's value is set in the local configuration (via Aastra Web UI or IP phone UI) and the same value was also set differently in one of the *<mac>.cfg*/*aastra.cfg* files on the configuration server, the local configuration value is the value that takes effect because that is the last value applied to the configuration.

IP Phone UI

The IP phone UI provides an easy way to access features and functions for using and configuring the IP phone. Hardkeys include **Hold (480i/480iCT), Swap (9112i), Redial, Options, Xfer, Conf, Icom (480i/480iCT)**, and **Services (480i/480iCT)**. Specific keys are also programmed to access the Directory List and the Callers List.

Reference

For more information on using the IP phone UI hardkeys, see "Hard Keys" on page 46. You can also refer to your model-specific IP Phone User Guide.

Options Key

The Options key allows you to access the "Options List". Accessible options in this list are for both user and administrator use. An administrator must enter a password for administrator options.

Note: An administrator has the option of enabling and disabling the use of password protection in the IP phone UI. This is configurable using the configuration files only. For more information about this feature, see "Password Settings" on page 114.

This document describes the administrator options only. For a description of the user options in the "Options List", see your model-specific IP Phone User Guide.

Using the Options Button

From the 480i/480iCT:

1. Press on the phone to enter the Options List.
2. Use the and to scroll through the list of options.
3. To select an option, press the **Show** softkey, press , or select the number on the keypad that corresponds to the option.
4. Use the **Change** softkey to change a selected option.
5. Press the **Done** softkey at any time to save the changes and exit the current option.
6. Press the **Cancel** softkey, press , or press Goodbye at any time to exit without saving changes.

From the 480i CT handset:

1. Press the key to enter the Options List when the phone is not in use.
2. Use the scroll keys and to scroll the options.
3. To select and change an option, press the keys.
4. Press when done.

From the 9112i and 9133i:

1. Press the **Options** key on the phone to enter the Options List.
2. Use the and to scroll the list of options, or enter the number on the keypad that corresponds to the option.
3. Press to enter an option.
4. Press to **Clear**, **Set**, or **Change** a value. The IP phone saves the settings immediately.
5. Press the **Options** key again to exit the Options List.

Aastra Web UI

An administrator can setup and configure the IP phone using the **Aastra Web UI**. The **Aastra Web UI** supports Internet Explorer and Gecko engine-based browsers like Firefox, Mozilla or Netscape.

<u>To access the Aastra Web UI</u>:

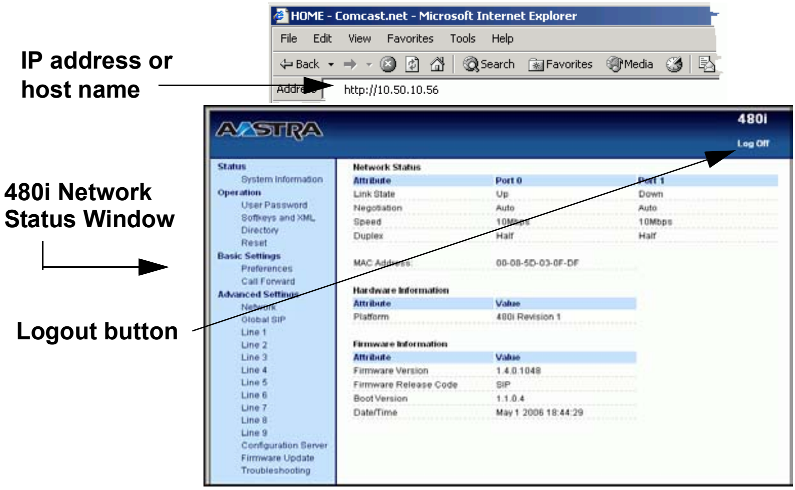
1. Open your web browser and enter the phone's IP address or host name into the address field.
2. At the prompt, enter your username and password and click OK.

Note: For an administrator, the default user name is "**admin**" and the password is "**22222**".
For a user, the default user name is "**user**" and the password field is left blank.

The Network Status window displays for the IP phone you are accessing.

3. You can logout of the Aastra Web UI at any time by clicking **LOGOFF**.

The following illustration is an example of a Network Status screen for the
480i IP phone.



The following categories display in the side menu of the Aastra Web UI: **Status, Operation, Basic Settings, Advanced Settings.**

Status

The **Status** section displays the network status and the MAC address of the IP phone. It also displays hardware and firmware information about the IP phone. The information in the Network Status window is read-only.

Operation

The **Operation** section provides the following options:

- **User Password** - Allows you to change user password
- **Programmable Keys (9112i and 9133i only)**- Allows you to configure up to 2 programmable keys on the 9112i and up to 7 on the 9133i.
- **Softkeys and XML (480i and 480i CT only)** - Allows you to configure up to 20 softkeys and load XML applications

- **Handset Keys (480i CT only)**- Allows you to configure up to 15 softkeys on the handset.
- **Directory** - Allows you to copy the Callers List and Directory List from your IP phone to your PC.
- **Reset** - Allows you to restart the IP phone when required.

Basic Settings

The **Basic Settings** section provides the following options:

- **Preferences** - Allows you to set General specifications on the IP phone such as , idle display name, local dial plan, park and pickup call settings, and enable/disable call waiting tone and stuttered dial tone. This section also allows you to set intercom settings, map conference and redial keys, set ring tones, set priority alerts, and enable directed call pickup.
- **Call Forward** - Allows you to set a phone number destination for where you want calls forwarded.

Advanced Settings

The **Advanced Settings** section provides the following options:

- **Network** - Allows you to set basic network settings such as, DHCP and IP address, and advanced network settings such as, Network Address Translation (NAT) and time servers. The Network subcategory also allows you to set Type of Service (ToS)/ Differentiated Services Code Point (DSCP), and VLAN settings.
- **Global SIP** - Allows you to set basic and advanced global SIP settings, and Real-time Transport Protocol (RTP) settings that apply to all lines on the IP phone.
- **Lines 1 through 9** (480i, 480i CT, and 9133i only) - Allows you to set SIP authentication settings, SIP network settings, and DTMF method to use on a specific line.
- **Configuration Server** - Allows you to set the protocol to use on the configuration server (TFTP (default), FTP, or HTTP), configure automatic firmware and configuration file updates, enable/disable auto-resync, and assign an XML push server list.
- **Firmware Update** - Allows you to manually perform a firmware update on the IP phone from the configuration server.
- **Troubleshooting** - Allows you to perform troubleshooting tasks whereby the results can be forwarded to Aastra Technical Support for analyzing and troubleshooting.

Enabling/Disabling the Aastra Web UI

The Aastra Web UI is enabled by default on the IP phones. A System Administrator can disable the Aastra Web UI on a single phone or on all phones if required using the configuration files. Use the following procedure to enable and disable the Aastra Web UI.

Configuration files

To disable the Aastra Web UI:

1. Using a text-based editing application, open the *<mac>.cfg* file if you want to disable the Web UI on a single phone. Open the *aastra.cfg* file to disable the Web UI on all phones.
2. Enter the following parameter:
   `web interface enabled: 0`

Note: A value of zero (**0**) disables the Web UI on the phone. A value of **1** enables the Web UI.

3. Save the changes and close the *<mac>.cfg* or the *aastra.cfg* file.
4. Restart the phone to apply the changes. The Aastra Web UI is disabled for a single IP phone or for all phones.

Administrator Level Options

Specific options on the IP phone allow an administrator to change or enter values as required. Depending on the IP phone model, you can use specific keys on the keypad to make option selections. For all models, you can also use the Aastra Web UI to change and enter values.

For the IP Phone UI, the administrator options in the "Options List" requires a password. The default password is "**22222**".

Note: An administrator has the option of enabling and disabling the use of password protection in the IP phone UI. This is configurable using the configuration files only. For more information about this feature, see "Password Settings" on page 114.

For the Aastra Web UI, administrator options require a user name and password. The default user name is "**admin**" and the default password is "**22222**".

Options via IP Phone UI

The following are administrator options in the "Options List" on the IP phone UI:

- **Phone Status->Factory Default**
- **Network**
- **SIP Settings**

Reference

For procedures on configuring the IP phone via the IP phone UI, see "Configuring the IP Phone" on page 22.

Options via Aastra Web UI

The following are administrator options in the Aastra Web UI:

- **Restore to Factory Defaults**
- **Basic Settings** (Idle Display Name, Dial Plan, Dial Plan Terminator, Digit Timeout, Outgoing Intercom, Key Mapping, Priority Alert, Directed Call Pickup)
- **Network**
- **Global SIP**
- **Line Settings**
- **Configuration Server**
- **Firmware Update**
- **Troubleshooting**

Reference

For procedures on configuring the IP phone via the Aastra Web UI, see "Configuring the IP Phone" on page 22.

Options via Configuration Files

A system administrator can enter specific parameters in the configuration files to configure the IP phones. All parameters can only be set by an administrator.

Reference

For a description of each configuration file parameter, see "Setting Parameters in Configuration Files" on page 111.

Using the Configuration Files

When you use the configuration files to configure the IP phones, you must use a text-based editing application to open the configuration file (aastra.cfg or <mac>.cfg).

Use the following procedure to add, delete, or change parameters and their settings in the configuration files.

Note: Apply this procedure wherever this Administrator Guide refers to configuring parameters using the configuration files.

Configuration files

1. Using a text-based editing application, open the configuration file for the phone, for which you want to configure the directory list (either aastra.cfg, <mac>.cfg or both).
2. Enter the required configuration parameters followed by the applicable value. For example,

```
directory 1: company_directory
directory 2: my_personal_directory
```

3. Save the changes and close the configuration file.
4. If the parameter requires the phone to be restarted in order for it to take affect, use the IP Phone UI or the Aastra Web UI to restart the phone.

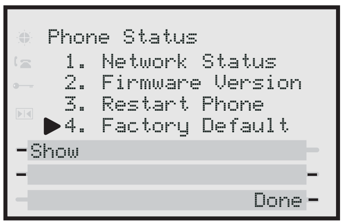
Phone Status

The **Phone Status** on the IP Phone displays the network status and firmware version of the IP phone. This option also allows you to restart the phone, and set the phone to factory defaults.

You can display phone status and reset the phone using the IP phone UI or the Aastra Web UI.

Phone Status via IP Phone UI

In the IP phone UI, the "**Network Status**", "**Firmware Version**", and "**Restart Phone**" options are available to the user and the administrator and do not require a password entry. The "**Factory Default**" option is for administrator use only.



The following information displays for phone status on the IP phone UI:

- **Network Port**
  (9112i only)
  Displays the network status of the Ethernet port at the back of the phone. Also displays the IP and MAC address of the phone. These fields are read-only.
- **Network Port 1 and Port 2**
  (9133i only)
  Displays the network status of the Ethernet ports at the back of the phone. Also displays the IP and MAC address of the phone. These fields are read-only.
- **Network Status**
  (480i and 480i CT only)
  Displays the network status of the Ethernet ports at the back of the phone. You can also view the phone’s IP and MAC addresses. These fields are read-only.

- **Firmware Version**
  Displays information about the firmware that is currently installed on the IP phone.
- **Restart Phone**
  This option lets you reboot the phone. A reset may be necessary when:
  - There is a change in your network, **OR**
  - To re-load modified configuration files, **OR**
  - If the settings for the IP phone on the IP PBX system have been modified.
- **Factory Default** (admin only)
  This option lets you reset the phone to its factory default settings. There are two options in setting the factory defaults on the IP phone:

  -All Defaults
  -Config Only

  The "**All Defaults**" option resets the factory defaults for all of the settings in the *aastra.cfg*, *<mac>.cfg*, and local configuration. Performing this option results in losing all user-modified settings.

  The "**Config Only**" option resets the settings on the local IP phone configuration only.

Phone Status via Aastra Web UI

In the Aastra Web UI, the "**Network Status**", "**Hardware Information**", and "**Firmware Information**" options are read only and also available to the user and administrator. Resetting the IP phone to factory defaults using the Aastra Web UI (**Operation->Reset->Current Settings**) is available to the administrator only.

The following information displays for phone status in the Aastra Web UI at the location **Status->System Information**. This information is available to the user and the administrator as read-only.

- **Network Satus**

Displays the network status of the Ethernet ports at the back of the phone. You can also view the phone's IP and MAC addresses. Information in this field includes Link State, Negotiation, Speed, and Duplex for Port 0 and Port 1.

- **Hardware Information**

Displays the current IP phone platform and the revision number.

- **Firmware Information**

Displays information about the firmware that is currently installed on the IP phone. Information in this field includes Firmware Version, Firmware Release Code, Boot Version, Release Date/Time.

Factory Default Feature

A user and administrator can restart the phone at **Operation->Reset->Phone.** However, only an administrator has access to restoring factory defaults to the IP phone at **Operation->Reset->Current Settings.**

There are two options for setting factory defaults using the Aastra Web UI:

- **Restore to Factory Defaults**
- **Remove Local Configuration Settings**

The "**Restore to Factory Defaults**" option resets the factory defaults for all of the settings in the *aastra.cfg*, *<mac>.cfg*, and local configuration. Performing this option results in losing all user-modified settings.

The "**Remove Local Configuration Settings**" option resets the settings on the local IP phone configuration only.

Reference

For procedures in setting factory defaults, see "Troubleshooting Solutions" on page 181.

Basic Preferences

(Aastra Web UI)

An administrator can configure the following basic preferences using the Aastra Web UI:

- **Idle Display Name 1 and 2**
  The names that display on the idle screen rather than the user name and phone number.
- **Local Dial Plan**
  A dial plan that describes the number and pattern of digits that a user dials to reach a particular telephone number.
- **Dial Plan Terminator**
  A dial plan terminator or timeout. When you configure the IP phone to use a dial plan terminator (such as the pound symbol (#)) the phone waits 4 or 5 seconds after you pick up the handset or press a key to make a call.
- **Digit Timeout**
  Represents the time, in seconds, to configure the timeout between consecutive key presses.
- **Park Call** (users and admin)
  The parking of a live call to a specific extension.
- **Pickup Parked Call**
  (users and admin)
  Picking up a parked call at the specified extension.
- **Incoming/Outgoing Intercom Calls**
  Specifies whether the IP phone or the server is responsible for notifying the recipient that an Intercom call is being placed. Also specifies the prefix code for server-side Intercom calls, and specifies the configuration to use when making the Intercom call.

Note: Users and administrators can configure incoming Intercom calls. Only administrators can configure outgoing Intercom calls.

- **Key Mapping**
  Allows you to set the **Redial** and/or **Conf** keys as speedial keys.
- **Priority Alerting**
  Enabling/disabling priority alert by setting specific ring tones for types of calls (Group, External, Internal, Emergency, Priority).
- **Directed Call Pickup**
  Enabling/disabling of directed call pickup feature and the playing of a ring tone splash.

References

For more information about parking calls and picking up parked calls, see "Park Calls/Pick Up Parked Calls" on page 66.

For more information about Idle Display Names, Local Dial Plan, Dial Plan Terminator, and Digit Timeout, see "SIP Local Dial Plan" on page 87.

For more information about incoming/outgoing Intercom calls, see "Incoming/Outgoing Intercom with Auto-Answer (Intercom applicable to 480i/480i CT only)" on page 90.

For more information about key mapping, directed call pickup, and priority alerting, see:

- "Setting Redial and Conf Keys as Speedials" on page 47

- "Directed Call Pickup (BLF Call Interception) (480i/480i CT/9133i)" on page 58

- "Priority Alerting" on page 96

Network

The following paragraphs describe the network parameters you can configure on the IP phone. Network settings are in two categories:

- **Basic network settings**
- **Advanced network settings**

Note: Specific parameters are configurable using the Aastra Web UI only and are indicated where applicable.

Basic Network Settings

If Dynamic Host Configuration Protocol (DHCP) is enabled, the IP phone automatically configures all of the Network settings. If the phone cannot populate the Network settings, or if DHCP is disabled, you can set the Network options manually.

- **DHCP**
  Enables or disables DHCP. When enabled, the phone may populate the following fields as read-only: IP Address, Subnet Mask, Gateway, Broadcast Address, Domain Name Servers (DNS), Trivial File Transfer Protocol (TFTP) Server, and Timer Servers.

Note: For DHCP to automatically populate the IP address or qualified domain name for the TFTP server, your DHCP server must support Option 66. For more information, see "DHCP" on page 22.

- **IP Address**
  IP address of the IP phone. To assign a static IP address, disable DHCP.
- **Subnet Mask**
  Subnet mask defines the IP address range local to the IP phone. To assign a static subnet mask, disable DHCP.
- **Gateway**
  The IP address of the network's gateway or default router IP address. To assign a static Gateway IP address, disable DHCP.
- **Primary DNS**
  Primary Domain Name Service. A service that translates domain names into IP addresses. To assign static DNS addresses, disable DHCP.
- **Secondary DNS**
  Secondary Domain Name Service. A service that translates domain names into IP addresses. To assign static DNS addresses, disable DHCP.

Note: If a host name is configured on the IP phone, you must also set a DNS.

Advanced Network Settings

- **NAT IP**
  Network Address Translator settings are used to map your firewall to an external NAT device. This is the IP address of the external network device that enforces NAT. Default is 0.0.0.0.
- **NAT Port**
  Hard-coded port number of the external network device that enforces NAT. Default is 0.
- **Nortel NAT Traversal Enabled**
  Enables or disables the phone to operate while connected to a network device that enforces NAT. Valid values are 0 (No) or 1 (Yes). Default is 0 (No).
- **Nortel NAT Timer (seconds)**
  The interval, in seconds, that the phone sends SIP ping requests to the Nortel proxy. Default is 30.
- **NTP Time Servers**
  Enables or disables the time server. This parameter affects time server1, time server2, and time server3. Valid values are 0 (enable) and 1 (disable). Default is 1 (disable).
- **Time Server 1, 2, and 3**
  The primary, secondary, and tertiary time server's IP address or qualified domain name. If the "NTP Time Server" parameter is enabled, and the primary and secondary time servers are not configured or cannot be accessed, the value for Time Server 3 is used to request the time.

Type of Service (ToS), DSCP

Network settings also allows you to set Type of Service (ToS) and Differentiated Services Code Point (DSCP).

For more information about ToS and DSCP see "Type of Service (ToS), Quality of Service (QoS), and DiffServ QoS" on page 30.

VLAN

You can also enable or disable VLAN and set specific VLAN IDs and priorities under Network Settings.

For more information about VLAN, see "Virtual LAN (optional)" on page 29.

SIP Settings

The following paragraphs describe the SIP parameters you can configure on the IP phone. SIP configuration consists of configuring:

- **Basic SIP Authentication Settings**
- **Basic SIP Network Settings**
- **Advanced SIP settings**
- **RTP Settings**

Note: Specific parameters are configurable using the Aastra Web UI only and are indicated where applicable.

If you have a proxy server or have a SIP registrar present at a different location than the PBX server, the SIP parameters may need to be changed. The SIP parameters can be set on a global or per-line basis.

Basic SIP Authentication Settings

- **Screen Name**
  Name that displays on the idle screen. Valid values are up to 20 alphanumeric characters. Configurable on a global and per-line basis.
- **Phone Number**
  (**User Name** in IP phone UI and configuration files) User name used in the name field of the SIP URI for the IP phone and for registering the phone at the registrar. Valid values are up to 20 alphanumeric characters. Configurable on a global and per-line basis.
- **Caller ID**
  (**Display Name** in IP phone UI and configuration files). Name used in the display name field of the "From SIP" header field. Some IP PBX systems use this as the caller's ID, and some may overwrite this with the string that is set at the PBX system. Valid values are up to 20 alphanumeric characters. Configurable on a global and per-line basis.
- **Authentication Name**
  Authorization name used in the username field of the Authorization header field of the SIP REGISTER request. Valid values are up to 20 alphanumeric characters. Configurable on a global and per-line basis.
- **Password**
  Password used to register the IP phone with the SIP proxy. Valid values are up to 20 alphanumeric characters. Passwords are encrypted and display as asterisks when entering. Configurable on a global and per-line basis.

- **BLA Number**
(not configurable via IP phone UI) Phone number that you assign to BLA lines that is shared across all phones (global configuration) or shared on a per-line basis (per-line configuration). For more information about BLA, see "Bridged Line Appearance (BLA) (480i/480i CT/9133i only)" on page 61.
- **Line Mode**
(**Sip Mode** in configuration files. Not configurable in IP phone UI). The mode-type that you assign to the IP phone on a global or per-line basis. Valid values are Generic (0), BroadSoft SCA (1), Nortel (2), or BLA (3). Default is Generic (0).

Basic SIP Network Settings

- **Proxy Server**
(**Proxy IP** in the configuration files). IP address of the SIP proxy server. Up to 64 alphanumeric characters. Configurable on a global and per-line basis.
- **Proxy Port**
SIP proxy server's port number. Default is 0. Configurable on a global and per-line basis.
- **Outbound Proxy Server**
Address of the outbound proxy server. All SIP messages originating from the phone are sent to this server. For example, if you have a Session Border Controller in your network, then you would normally set its address here. Default is 0.0.0.0. Configurable on a global and per-line basis.
- **Outbound Proxy Port**
The proxy port on the proxy server to which the IP phone sends all SIP messages. Default is 0. Configurable on a global and per-line basis.
- **Registrar Server**
(**Registrar IP** in the configuration files). IP address of the SIP registrar. Up to 64 alphanumeric characters. Enables or disables the phone to be registered with the Registrar. When Register is disabled globally, the phone is still active and you can dial using username and IP address of the phone. A message "No Service" displays on the idle screen and the LED is steady ON. If Register is disabled for a single line, no messages display and LEDs are OFF. Configurable on a global and per-line basis.
- **Registrar Port**
SIP registrar's port number. Default is 0. Configurable on a global and per-line basis.
- **Registration Period**
(Not configurable via IP Phone UI). The requested registration period, in seconds, from the registrar. Configurable on a global and per-line basis.

Advanced SIP Settings

In addition to the basic SIP settings, you can also configure the following advanced SIP parameters. These parameters are not configurable via the IP phone UI.

- **Explicit MWI Subscription**
If the IP phone has a message waiting subscription with the Service Provider, a Message Waiting Indicator (MWI) (LED or display icon) tells the user there is a message on the IP Phone. You can enable and disable MWI by setting this parameter to 0 (disable) or 1 (enable) in the configuration files or by checking the box for this field in the Aastra Web UI. Default is disabled.

- **Send MAC Address in REGISTER Message**
  Adds an "Aastra-Mac:" header to the SIP REGISTER messages sent from the phone to the call server, where the value is the MAC address of the phone.
- **Send Line Number in REGISTER Message**
  Adds an "Aastra-Line:" header to the SIP REGISTER messages sent from the phone to the call server, where the value is the line number that is being registered.
- **Session Timer**
  The time, in seconds, that the IP phone uses to send periodic re-*INVITE* requests to keep a session alive. The proxy uses these re-*INVITE* requests to maintain the status' of the connected sessions. See RFC4028 for details. Default is 0.
- **Timer 1 and Timer 2**
  The time, in milliseconds, that applies to an IP phone session. These timers are SIP transaction layer timers defined in RFC 3261.
  Timer 1 is an estimate of the round-trip time (RTT). Timer 2 represents the amount of time a non-INVITE server transaction takes to respond to a request.
- **Transaction timer**
  The amount of time, in milliseconds that the phone allows the callserver (registrar/proxy) to respond to SIP messages that it sends. If the phone does not receive a response in the amount of time designated for this parameter, the phone assumes the message has timed out. Valid values are 4000 to 64000. Default is 4000.
- **Transport Protocol**
  The protocol that the RTP port on the IP phone uses to send out RTP packets. Valid values are 0 (both), 1 (UDP), or 2 (TCP). Default is 1 (UDP).
- **Registration Retry Timer**
  Specifies the time, in seconds, that the phone waits between registration attempts when a registration is rejected by the registrar.
- **BLF Subscription Period**
  Specifies the time period, in seconds, that the BLF feature becomes active again after a software/firmware upgrade or after a reboot of the IP phone.

RTP Settings

You can configure the following RTP settings:

- **RTP Port**
  (**RTP Port Base** in IP Phone UI)
  The Real-Time Transport Protocol port base configured for the IP phone. Default is 3000.
- **Basic Codecs** (G.711 u-Law, G.711 a-Law, G.729)
  (not configurable via IP phone UI). Enables or disables basic codecs. Enabling this parameter allows the IP phone to use the basic Codecs when sending/receiving RTP packets. Valid values are 0 (disabled) and 1 (enabled). Default is 0 (disabled).
- **Force RFC2833 Out-of-Band DTMF**
  (not configurable via IP phone UI). Enables or disables out-of-band DTMF. Enabling this parameter forces the IP phone to use out-of-band DTMF according to RFC283. Valid values are 0 (disabled) and 1 (enabled). Default is 1 (enabled).
- **Customized Codec Preference List**
  (not configurable via IP phone UI). Specifies a customized Codec preference list which allows you to use the preferred Codecs for this IP phone. For valid values, see "RTP, Codec, DTMF Global Settings" on page 148.

- **DTMF Method**
(not configurable via IP phone UI). Sets the dual-tone multifrequency (DTMF) method to use on the IP phone on a global or per-line basis. Valid values are 0 (RTP), 1 (SIP INFO), or 2 (BOTH). Default is 0 (RTP). Configurable on a global and per-line basis.
- **Silence Suppression**
Enables or disables the phone to use the negotiated silence suppression setting.

Line Settings

An administrator can configure multiple lines on the IP phone with the same configuration (global) or a different configuration (per-line). The following table provides the number of lines available for each IP phone model.

IP Phone Model	Available Lines
480i	9
480i CT	9
9112i	1
9133i	9

A user or administrator can then assign the line to a specific softkey (480i/480i CT) or programmable key (9112i/9133i). The available softkeys also depends on the IP phone model as shown in the following table.

IP Phone Model	Softkeys Available	Programmable Keys Available
480i	20	-
480i CT	20	-
9112i	-	2
9133i	-	7

The softkey or programmable key can be set to use a specific function. Available functions depend on the IP phone model.

The functions you can configure are:

- line (480i, 480i CT, 9133i)
- speeddial
- do not disturb
- BLF (480i, 480i CT, 9133i)
- BLF/List (480i, 480i CT, 9133i)
- XML
- flash
- park
- pickup
- empty (480i, 480i CT)

References

For information about configuring softkeys, see Configuring Softkeys and Programmable Keys on page 51.

For more information about configuring lines on the IP phone, see "SIP Basic, Per-Line Settings" on page 138 and "DTMF Per-Line Settings" on page 150.

For more information about softkey functions see "Softkey/ Programmable Key Parameters" on page 172.

Configuration Server Settings

The configuration server stores the firmware images, configuration files, and software when performing software upgrades to the IP phone. An administrator can configure the following parameters for the configuration server:

- **Download Protocol**
  Protocol to use for downloading new versions of firmware and configuration files to the IP phone. Valid values are TFTP, FTP, and HTTP. Default is TFTP.
- **TFTP Server**
  IP address or qualified domain name of the TFTP server. You can select a primary or alternate TFTP server and then assign an IP address or qualified domain name to your selection. Set this option if TFTP is the download protocol selected.

Note: For DHCP to automatically populate the IP address or domain name for the TFTP server, your DHCP server must support Option 66. For more information, see "DHCP" on page 22.

- **FTP Server**
  IP address or network host name of the FTP server. If required, you can also assign a user name and password for access to the FTP server. Set this option if FTP is the download protocol selected. If you enter a network host name, DNS must also be set.
- **HTTP Server**
  IP address of the HTTP server. You can also assign an HTTP path to the HTTP server. Set this option if HTTP is the download protocol selected.
- **Mode**
  (not configurable via IP phone UI). Enables and disables the IP phone to be updated automatically (auto-resync) once a day at a specific time in a 24-hour period. Updating can be done to the configuration files only, the firmware only, or both. This feature works with TFTP, FTP, and HTTP servers. The auto update feature works with both encrypted and plain text configuration files.

Note: Any changes made using the Aastra Web UI or the IP phone UI are not overwritten by an auto-resync update. Auto-resync affects the configuration files only. However, the settings in the Aastra Web UI take precedence over the IP phone UI and the configuration files.

- **Time (24-hour)**
  (Not configurable via IP phone UI). Sets the time of day in a 24-hour period for the IP phone to be automatically updated (auto-resync). This parameter works with TFTP, FTP, and HTTP servers.

Note: Auto-Resync adds up to 15 minutes random time to the configured time. For example, if the auto resync time parameter is set to 02:00, the event takes place any time between 02:00 and 02:15.

- **XML Push Server List**
  (not configurable via IP phone UI). The HTTP server that is pushing XML applications to the IP phone.

Reference

For more information about configuring the configuration server, see "Configuration Server Protocol" on page 24.

Firmware Update Features

The IP phone uses a TFTP, FTP, or HTTP server (depending on the protocol configured on the IP phone) to download configuration files and firmware.

You can download the firmware stored on the configuration server in one of three ways:

- Manual firmware update using the Aastra Web UI (TFTP only).
- Manual update of firmware and configuration files (by restarting the phone via the IP phone UI or the Aastra Web UI).
- Automatic update of firmware, configuration files, or both at a specific time in a 24-hour period (via the Aastra Web UI or configuration files)

Reference

For more information about firmware update, see "Firmware Upgrade" on page 108.

Troubleshooting (Aastra Web UI only)

This section describes tasks that a system administrator can perform on the IP phones for troubleshooting purposes. Using the Aastra Web UI, a system administrator can:

- Assign an IP address and IP port in which to save log files
- Save the current local configuration file to a specified location
- Save the current server configuration file to a specified location
- Show task and stack status

Aastra Technical Support can then use the information gathered to perform troubleshooting tasks.

Log Settings

Using the Aastra Web UI, you can specify the location for which to save files for troubleshooting purposes at **Advanced Settings ->Troubleshooting->Log Settings.**

You must designate the IP address of where you want to store the logs in the **Log IP** field, and the IP port to use for logging the data in the **Log Port** field.

Support Information

You can save the local and/or server configuration files of the IP phone to the location specified in the "**Log Settings**" section.

Performing this task allows Aastra Technical Support to view the current configuration of the IP phone and troubleshoot as necessary.

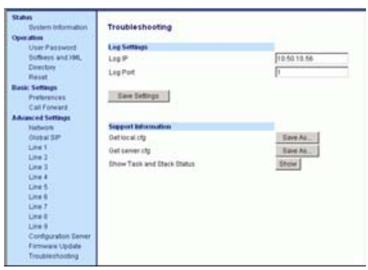
You can also display task and stack status information about the IP phone. Aastra Technical Support uses this information for troubleshooting the IP phone when required.

Performing Troubleshooting Tasks

Use the following procedure to perform troubleshooting on the IP phone.

Aastra Web UI

1. Click on **Advanced Settings->Troubleshooting.**



To set log settings:

2. In the "**Log IP**" field, enter the IP address of where you want log files to be stored.
3. In the "**Log Port**" field, enter the port number associated with the IP address specified in the "Log IP" field. This port passes the information from the IP phone to the IP address.
4. Click Save Settings to save your settings.
5. Click on **Operation->Reset.**
6. In the "**Restart Phone**" field click Restart to restart the IP phone.

To perform support tasks:

7. To store the local configuration file to the specified location, click on Save As... in the "**Get local.cfg**" field.
8. To store the server configuration file to the specified location, click on Save As... in the "**Get server.cfg**" field.
9. To display task and stack status information to the screen, click on Show in the "**Show Task and Stack Status**" field.

Note: The local and server configuration file information and the task and stack status information is for use by Aastra Technical Support for troubleshooting purposes.

Reference

For information that describes solutions to most common problems using the IP phones, see "Troubleshooting Solutions" on page 181.

Configuring the IP Phone

An administrator can configure the IP Phone Network and SIP options from the phone UI, from the Aastra Web UI, or the configuration files. Administrator level options are password protected in both the IP phone UI and the Aastra Web UI.

Note: An administrator has the option of enabling and disabling the use of password protection in the IP phone UI. This is configurable using the configuration files only. For more information about this feature, see "Password Settings" on page 114.

The procedures in this section include configuring from the IP phone UI and the Aastra Web UI. To configure the IP phones using the configuration files, see "Setting Parameters in Configuration Files" on page 111.

To configure the phone using the IP phone UI, you must enter an administrator password. To configure the phone using the Aastra Web UI, you must enter an administrator user name and password.

Note: In the IP phone UI, the default password is "**22222**". In the Aastra Web UI, the default admin user name is "**Admin**" and the default password is "**22222**".

References

For configuring the IP phone at the Asterisk IP PBX, see "Appendix B: Configuring the IP Phone at the Asterisk IP PBX."

For sample configuration files, see "Appendix C: Sample Configuration Files." These sample files include basic parameters required to register the IP phone at the PBX.

Basic Network Settings

This section describes the basic network settings on the IP phone.

DHCP

The IP phone is capable of querying a DHCP server, allowing a network administrator a centralized and automated method of configuring various network parameters for the phone. If DHCP is enabled, the IP phone requests the following network information:

- **Subnet Mask**
- **Gateway (i.e. router)**
- **Domain Name Server (DNS)**
- **Broadcast Address**
- **Network Time Protocol Server**
- **IP Address**
- **TFTP Server Name**

Note: For DHCP to automatically populate the IP address or domain name for the TFTP server, your DHCP server must support Option 66. Option 66 is responsible for forwarding the TFTP server IP address or domain name to the phone automatically. If your DHCP server does not support Option 66, you must manually enter the IP address or domain name for the TFTP server into your IP phone configuration.

The network administrator chooses which of these parameters (if any) are supplied to the IP phone by the DHCP server. The administrator must configure the phone manually to provide any required network parameters not supplied by the DHCP server.

Configuring DHCP

You can enable and disable DHCP using the configuration files, the IP phone UI, or the Aastra Web UI.

Configuration files

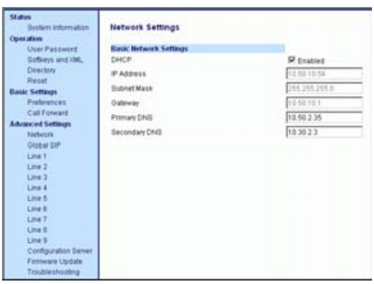
For specific parameters you can set in the configuration files, see "Network Settings" on page 112.

IP Phone UI

1. Press [▲] on the phone to enter the Options List.
2. Select **Network.**
3. Select option **DHCP.**
4. Press **Change** to set "Use DHCP" to "Yes" (enable) or "No" (disable).
5. Press **Done** to save the changes.

Aastra Web UI

1. Click on **Advanced Settings->Network->Basic Network Settings.**



2. Enable the "**DHCP**" field by checking the check box. (Disable this field by unchecking the box).
3. Click [Save Settings] to save your changes.

Configuring Network Settings

If you disable DHCP on your phone, you need to configure the network settings manually. You can configure the Network settings using the configuration files, the IP phone UI, or the Aastra Web UI.

Configuration files

For specific parameters you can set in the configuration files, see "Network Settings" on page 112.

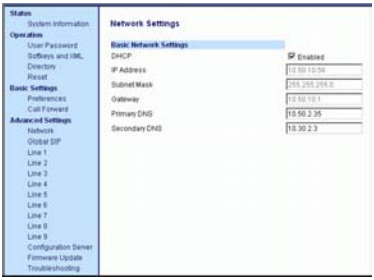
IP Phone UI

1. Press [Options] on the phone to enter the Options List.
2. Select **Network.**
3. Select **IP Address** and enter the IP address of the phone.
4. Select **Subnet Mask** and enter the subnet mask.
5. Select **Gateway** and enter the gateway address.
6. Select **DNS** and enter a Primary and/or Secondary DNS server.
7. Press **Done** to save the changes.

The IP phone is manually configured. You can now continue the phone configuration if required using the [Options] button on the IP phone.

Aastra Web UI

1. Click on **Advanced Settings->Network->Basic Network Settings**.



2. Enter an IP address of the phone in the **IP Address** field.
3. Enter a subnet mask in the **Subnet Mask** field.
4. Enter a gateway address in the **Gateway** field.
5. Enter a Primary DNS in the **Primary DNS** field, and/or a secondary DNS in the **Secondary DNS** field.
6. Click Save Settings to save your changes.

The IP phone is manually configured. You can now continue the phone configuration if required using the Aastra Web UI.

Configuration Server Protocol

You can download new versions of firmware and confguration files from the configuration server to the IP phone using any of the following types of protocols: TFTP, FTP, HTTP. The TFTP setting is the default download protocol. You can configure the type of protocol that the IP phone uses by setting it in the configuration files, the IP phone UI, or the Aastra Web UI.

Note: For DHCP to automatically populate the IP address or domain name for the TFTP server, your DHCP server must support Option 66. For more information, see "DHCP" on page 22.

Configuring the Configuration Server Protocol

Configuration files

For specific parameters you can set in the configuration files, see "Configuration Server Settings" on page 115.

IP Phone UI

1. Press Options on the phone to enter the Options List.
2. Select **Network**.
3. Select **Download Protocol**.
4. Select "**Use TFTP**", "**Use FTP**", or "**Use HTTP**". The IP phone uses the protocol you select to download new firmware and configuration files from the configuration server.
5. Press **Done** to save the change.
6. Select **TFTP Server, FTP Server,** or **HTTP Server**. Use the following table to configure the applicable server.

TFTP
- Select **TFTP**. - Select **Primary**. - Select **Primary TFTP**. - Enter the IP address or qualified domain name of the primary TFTP server - Press **Done** to save the change. **Optional**: You can also configure an alternate TFTP server if required. If Alternate TFTP is enabled, you must also enter an IP address or qualified domain name for the alternate TFTP server.
FTP
- Select **FTP Server**. - Enter the IP address of the FTP server. - Press **Done**. **Optional:** You can enter a username and password for accessing the FTP server if required: - Select **FTP Username**. - Enter a username for accessing the FTP server. - Press **Done**. - Select **FTP Password**. - Enter a password for accessing the FTP server. - Press **Done**.
HTTP
- Select **HTTP Server**. - Enter the IP address of the HTTP server. - Press **Done**. - Select **HTTP Path**. - Enter the HTTP sub-directory path name. If the IP phone's files are located in a sub-directory beneath the server's HTTP root directory, the relative path to that sub-directory should be entered in this field. - Press **Done**.

7. Press **Done** to finish configuring the configuration server protocol for the IP phone.

Note: The session prompts you to restart the IP phone to apply the configuration settings.

8. Select **Restart.**

Aastra Web UI

1. Click on **Advanced Settings->Configuration Server**.
2. Select the protocol from the "**Download Protocol**" list box. The IP phone uses the protocol you select to download new firmware and configuration files from the configuration server. Use the following table to configure the applicable server.

TFTP
- Enter an IP address or qualified domain name in the **"TFTP Server"** field. **Optional**: You can also configure an alternate TFTP server if required. If **"Use Alternate TFTP"** is enabled, you must also enter an IP address or qualified domain name for the alternate server in the **"Alternate TFTP"** field.
FTP
- Enter an IP address in the **"FTP Server"** field. **Optional**: You can enter a username and password for accessing the FTP server if required. - Enter a user name for a user that will access the FTP server in the **"FTP User Name"** field. - Enter a password for a user that allows access to the FTP server in the **"FTP Password"** field.
HTTP
- Enter an IP address in the **"HTTP Server"** field. - Enter a root sub-directory path for the HTTP server in the **"HTTP Path"** field. **Optional**: You can enter a list of users to be authenticated when they access the HTTP server in the **"HTTP POST Authentication List"** field.

3. Click Save Settings to save your changes.

Note: The session prompts you to restart the IP phone to apply the configuration settings.

4. Select **Operation->Reset** and click Restart.

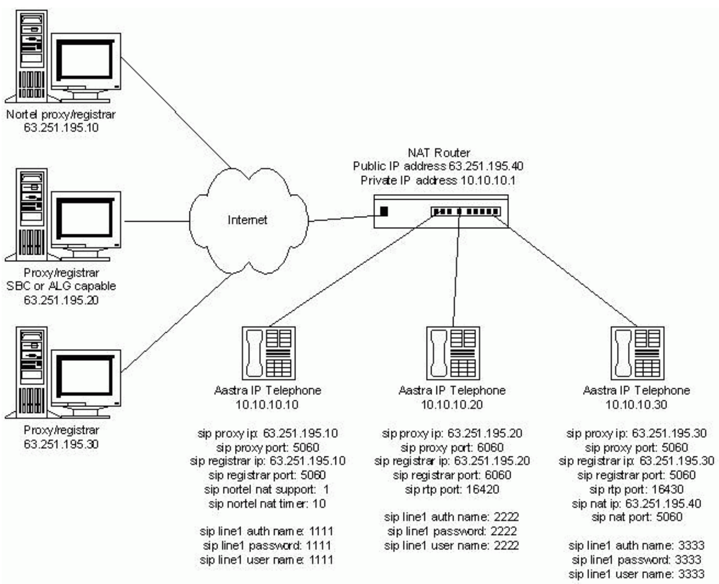
Advanced Network Settings

You can set advanced network settings on the IP phone such as, Network Address Translation (NAT), Nortel NAT, Network Time Protocol (NTP) Time Servers, Virtual LAN (VLAN), and Quality of Service (QoS) using the Aastra Web UI or the configuration files.

Note: The available advanced network parameters via the IP phone UI are NAT, Nortel NAT, VLAN, and QoS only.

Network Address Translation (NAT)

The protocols used by all IP phones do not interoperate completely with Network Address Translation (NAT). For the Aastra IP phones, specific configuration parameters allow the phone to operate while connected to a network device that enforces NAT. The following is a sample network using a NAT proxy and relevant IP phone configuration parameters.



Nortel Proxy/Registrar

The phone at IP address 10.10.10.10 is configured to register with the proxy at 63.251.195.10. Because it is a Nortel proxy, the configuration must additionally include the "sip nortel nat support" and "sip nortel nat timer" settings, telling the firmware to enhance the protocols with Nortel specific content.

Note: This IP phone uses RTP port 3000 (the default value) since an RTP port was not explicitly configured.

SBC or ALG proxy/registrar

The phone at IP address 10.10.10.20 is configured to register with the proxy at 63.251.195.20. Because the proxy/registrar has session border control (SBC) or application layer gateway (ALG) functionality, no additional IP phone configuration is required.

Other proxy/registrars

The phone at IP address 10.10.10.30 is configured to register with the proxy at 63.251.195.30. Because this proxy/registrar is not a Nortel proxy and has no SBC or ALG functionality, the configuration must additionally include the "sip nat ip" and "sip nat port" settings that contain the public ip address of the NAT router and the port used for call signaling messages. This information is embedded in protocol messages to allow the proxy/registrar to reach the IP phone on the NAT router private network.

NAT router configuration

You must configure the NAT router to allow signaling or media packets containing the various UDP port values to flow between the private and public networks that are separated by the NAT router. In the sample network, the NAT router must not filter packets using ports 3000, 5060, 6060, 16420, and 16430.

Nortel Networks NAT

Nortel Networks provides a proprietary solution to support connectivity to their proxies from phones placed behind devices (such as routers or firewalls) that use NAT. Nortel uses the SIP ping request/reply between the Nortel proxy and the phone in order to keep the connection through the router or firewall active. A SIP Nortel NAT timer is the interval, in seconds (default is 60), that the phone sends SIP ping requests to the Nortel proxy.

Configuring Nortel NAT (optional)

You can configure Nortel NAT using the configuration files, the IP phone UI, or the Aastra Web UI.

Configuration files

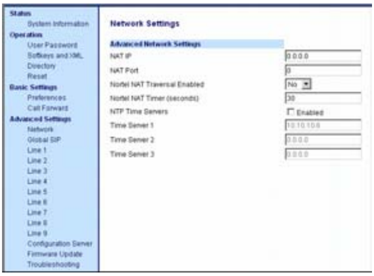
For specific parameters you can set in the configuration files, see "Network Address Translation (NAT) Settings" on page 125.

IP Phone UI

1. Press Options on the phone to enter the Options List.
2. Select **SIP Settings.**
3. Select **NAT.**
4. Select **Nortel.**
5. Select **NAT Enabled.**
6. Press **Change** to set Yes (enable) or No (disable) for NAT on a Nortel network.
7. Press **Done** to save the Nortel NAT settings.

Aastra Web UI

1. Click on **Advanced Settings ->Network->Advanced Network Settings.**



2. Select Yes (enable) or No (disable) in the "**Nortel NAT Traversal Enabled**" field to enable or disable NAT for a Nortel network.
3. Enter a time, in seconds, in the "**Nortel NAT timer**" field. Valid values are 0 to 2147483647. Default is 60.
4. Click Save Settings to save your changes.

Configuring NAT Address and Port (optional)

You can also configure a specific NAT address and port on the IP phone using the configuration files, IP Phone UI, or Aastra Web UI.

Configuration files

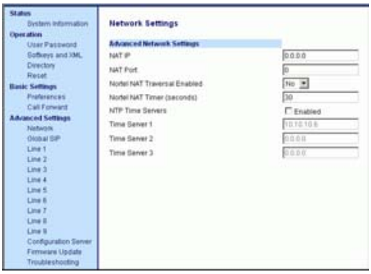
For specific parameters you can set in the configuration files, see "Network Address Translation (NAT) Settings" on page 125.

IP Phone UI

1. Press Options on the phone to enter the Options List.
2. Select **SIP Settings.**
3. Select **NAT.**
4. Select **NAT Settings.**
5. Select **NAT IP.**
6. Enter a public IP address of your NAT device in dotted-decimal format.
7. Press **Done** to save the NAT setting.
8. Select **NAT Port.**
9. Enter the public SIP signalling port number of your NAT device.
10. Press **Done** to save the NAT setting.

Aastra Web UI

1. Click on **Advanced Settings->Network->Advanced Network Settings**.



2. Enter a NAT IP address in the "**NAT IP**" field.
3. Enter a NAT port in the "**NAT Port**" field.
4. Click Save Settings to save your changes.

Virtual LAN (optional)

Virtual Local Area Network (VLAN) is a feature on the IP phone that allows for multiple logical Ethernet interfaces to send outgoing RTP packets over a single physical Ethernet as described in IEEE Std 802.3. On the IP phone, you configure a VLAN ID that associates with the physical Ethernet port.

By configuring specific VLAN parameters, the IP phones have the capability of adding and removing tags, and processing the ID and priority information contained within the tag.

Note: All latest VLAN functionality is backwards compatible with IP Phone Releases 1.3 and 1.3.1.

VLAN on the IP phones is disabled by default. When you enable VLAN, the IP phone provides defaults for all VLAN parameters. If you choose to change these parameters, you can configure them using the configuration files, the IP Phone UI, or the Aastra Web UI.

The following sections describe the VLAN features you can configure on the IP phones.

Type of Service (ToS), Quality of Service (QoS), and DiffServ QoS

ToS is an octet as a field in the standard IP header. It is used to classify the traffic of the different QoSs.

QoS provides service differentiation between IP packets in the network. This service differentiation is noticeable during periods of network congestion (for example, in case of contention for resources) and results in different levels of network performance.

Port 0 is the Ethernet connected to the network. Port 1 is the Ethernet used for passthrough to a PC (port 1 is not available on 9112i).

Differentiated Service (DiffServ) QoS is class-based where some classes of traffic receive preferential handling over other traffic classes.

The Differentiated Services Code Point (DSCP) value is stored in the first six bits of the ToS field. Each DSCP specifies a particular per-hop behavior that is applied to a packet.

The following parameters allow an administrator to configure ToS, QoS, and DiffServ QoS for VLAN:

tagging enabled
tos priority map
priority non-ip
VLAN idds
VLAN id port 1
QoS eth port 1 priority
tos sip
tos rtp
tos rtcp

Notes:

1. In order for the software to successfully maintain connectivity with a network using VLAN functionality, the IP phone reboots if you modify the "**tagging enabled**" (**VLAN enable** in the Web UI), "**VLAN id**", or "**VLAN id port 1**" parameters.
2. The "**QoS eth port 0 priority**" and "**QoS eth port smp priority**" parameters were applicable to software release 1.3.1 and earlier. They have no affect in software Release 1.4 and up.
3. When the Port 0 "VLAN id" and the Port 1 "VLAN id port 1" parameters have the same value, VLAN functionality is compatible with earlier IP phone software releases.

DSCP Range/VLAN Priority Mapping

DSCP bits in the ToS field of the IP header are set for RTP, RTCP, and SIP packets using either the default values or the values configured via the "**tos sip**", "**tos rtp**", and "**tos rtcp**" parameters.

When the VLAN global configuration parameter, "**tagging enabled**" is set to **1**, VLAN priority for IP packets is mapped to the DSCP value instead of a single priority for all packets. An administrator can also configure VLAN priority for non-IP packets using the "**priority non-ip**" parameter.

Since the default DSCP settings for SIP, RTP, and RTCP are 24, 32, and 32 respectively, this results in corresponding default VLAN priorities of 3 for SIP, 4 for RTP, and 4 for RTCP (based on the settings in the table "DSCP Range/ VLAN Priority" on page 31).

You can change the default parameters by modifying just the DSCP values, just the VLAN priority values, or by modifying all values.

The following table shows the DSCP range/VLAN piority mapping.

DSCP Range/VLAN Priority

DSCP Range	VLAN Priority
0-7	0
8-15	1
16-23	2
24-31	3
32-39	4
40-47	5
48-55	6
56-63	7

The following table identifies the default DSCP of protocols.

Protocol Name	Default DSCP Values in the ToS Field
rtp	32
rtcp	32
sip	24

Configuring Type of Service (ToS)/DSCP (optional)

Use the following procedures to configure ToS/DSCP on the IP phone.

Note: ToS/DSCP is enabled by default. The SIP, RTP, and RTCP parameters show defaults of 24, 32, and 32, respectively. Use the following procedures to change these settings if required.

Configuration files

For specific parameters you can set in the configuration files, see "Type of Service (ToS)/DSCP Settings" on page 121.

IP Phone UI

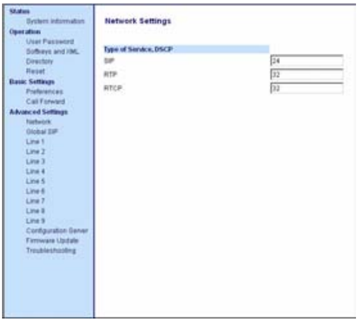
1. Press Options on the phone to enter the Options List.
2. Select **Network**.
3. Select **Type of Service**.
4. Select **SIP**, **RTP**, and/or **RTCP** and enter a value from **0** to **63**.

Note: If you change the ToS/DSCP setting for a Protocol, and VLAN is enabled, you will need to map the applicable priority to the Protocol setting as shown in the table "DSCP Range/VLAN Priority" on page 31 For more information, see the section "Configuring VLAN (optional)" on page 32.

5. Press **Done** to save the changes.
6. Press Done 2 more times to return to the Options List menu.
7. Select **Phone Status**.
8. Select **Restart Phone** and press **Restart** to reboot the phone for the ToS/DSCP settings to take affect.

Aastra Web UI

1. Click on **Advanced Settings->Network->Type of Service,DSCP.**



2. Choose a Protocol (SIP, RTP, and/or RTCP), and enter a value from **0** to **63**.

Note: If you change the ToS/DSCP setting for a Protocol, and VLAN is enabled, you will need to map the applicable priority to the Protocol setting as shown in the table "DSCP Range/VLAN Priority" on page 31 For more information, see the section "Configuring VLAN (optional)" on page 32.

3. Click **Save Settings** to save your changes.
4. Click on **Operation->Reset.**
5. In the "**Restart Phone**" field click **Restart** to restart the IP phone.

Configuring VLAN (optional)

Use the following procedures to configure VLAN on the IP phone.

Note: VLAN is disabled by default. When you enable VLAN, the IP phones use the default settings for each VLAN parameter. You can change the default settings if required using the following procedures..

Configuration files

For specific parameters you can set in the configuration files, see "Virtual Local Area Network (VLAN) Settings" on page 122.

IP Phone UI

1. Press **Options** on the phone to enter the Options List.
2. Select **Network.**
3. Select **VLAN.**

To globally enable/disable VLAN and set priority for non-IP packets:

4. Select **VLAN Enable,** and press **Change** to set VLAN Enable to **Yes** (or **No** to disable)
5. Press **Done** to save the change.
6. Select **Phone.**
7. Select **Priority.**
8. Select **Other** and enter a non-IP priority value from **0** to **7** for non-IP packets.
9. Press **Done** to save the change.
10. Press **Done** again to return to the VLAN Phone menu.

To set VLAN ID and priority for Port 0:

11. Select **VLAN ID** and enter a value from **1** to **4094** to specify the VLAN ID of Port 0.

12. Press **Done** to save the change.
13. Select **Priority**.
14. Select a VLAN Protocol (**SIP**, **RTP**, and/or **RTCP**).
15. Enter a VLAN priority value from **0** to **7** for the associated Protocol.
16. Press **Done** to save the changes.
17. Press **Done** again to return to the VLAN Settings menu.

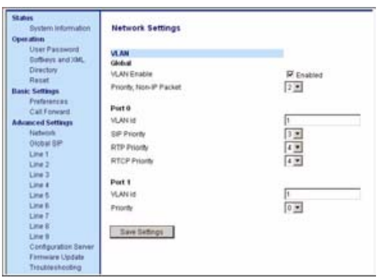
To set VLAN ID and priority for Port 1 (passthrough port):

18. Select **Passthrough**.
19. Select **VLAN ID** and enter a value from **1** to **4094** to specify the VLAN ID of Port 1.
20. Press **Done** to save the change.
21. Select **Priority**.
22. Select a VLAN priority value from **0** to **7** for Port 1.
23. Press **Done** to save the change.
24. Press **Done** 3 more times to return to the Options List main menu.
25. Select **Phone Status**.
26. Select **Restart Phone** and press **Restart** to reboot the phone for the VLAN features to take affect.

.

Aastra Web UI

1. Click on **Advanced Settings->Network->VLAN**.



Globally enable/disable VLAN and set priority for non-IP packets:

2. Enable VLAN by checking the **VLAN Enable** field check box. (Disable this field by unchecking the check box).
3. With VLAN enabled, select the priority (**0** to **7**) for non-IP packets in the **Priority, Non-IP Packet** field.

To set VLAN ID and priority for Port 0:

4. For Port 0, enter a VLAN ID value from **1** to **4094** in the **VLAN id** field.
5. Choose a VLAN Protocol (**SIP Priority**, **RTP Priority**, and/or **RTCP Priority**), and select a priority for the associated Protocol.

To set VLAN ID and priority for Port 1 (passthrough port):

6. For Port 1, enter a VLAN ID value from **1** to **4094** in the **VLAN id** field.
7. Select a VLAN priority value from **0** to **7** for Port 1 in the **Priority** field.
8. Click [Save Settings] to save your changes.
9. Click on **Operation->Reset**.
10. In the "**Restart Phone**" field click [Restart] to restart the IP phone.

Network Time Servers

Network Time Protocol (NTP) is a protocol that the IP phone uses to synchronize the phone clock time with a computer (configuration server) in the network.

To use NTP, you must enable it using the configuration files or the Aastra Web UI. You can specify up to 3 time servers in your network.

Note: The IP phones support NTP version 1.

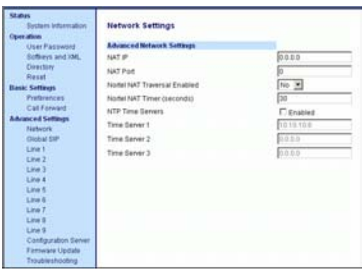
Configuring NTP Servers (optional)

Configuration files

For specific parameters you can set in the configuration files, see "Time Server Settings" on page 126.

Aastra Web UI

1. Click on **Advanced Settings->Network->Advanced Network Settings.**



2. Enable the "**NTP Time Servers**" field by checking the check box. (Disable this field by unchecking the box).
3. Enter an IP address or qualified domain name in the "**Time Server 1**", "**Time Server 2**", and/or "**Time Server 3**" field(s) to specify the location of the NTP time server.
4. Click Save Settings to save your changes.

Session Initiation Protocol (SIP) Settings

The IP phone uses the information in the Session Initiation Protocol (SIP) settings to register at the IP PBX.

The IP phone configuration defines network and user account parameters that apply **globally** to all SIP lines. Since not all SIP lines are necessarily hosted using the same IP-PBX/server or user account, additional sets of **per-line** parameters can also be defined for network and user account.

You configure and modify these parameters and associated values using the configuration files, the IP phone UI, or the Aastra Web UI. The Aastra Web UI and configuration file methods configure global and per-line SIP settings on the IP phone. The IP phone UI configures global SIP settings only.

The global SIP configuration parameters are:

- sip proxy ip
- sip proxy port
- sip registrar ip
- sip registrar port
- sip registration period
- sip outbound proxy
- sip outbound proxy port

The global user account authentication parameters are:

- sip mode
- sip vmail
- sip screen name
- sip user name
- sip display name
- sip auth name
- sip password
- sip bla number

The per-line SIP configuration parameters are:

- sip lineN proxy ip
- sip lineN proxy port
- sip lineN registrar ip
- sip lineN registrar port
- sip lineN registration period
- sip lineN outbound proxy
- sip lineN outbound proxy port

The per-line user account authentication parameters are:

- sip lineN mode
- sip lineN vmail
- sip lineN screen name
- sip lineN user name
- sip lineN display name
- sip lineN auth name
- sip lineN password
- sip lineN bla number

Note: The "sip vmail" and "sip lineN vmail" parameters are configurable using the configuration files only. To configure voicemail see "Voicemail (480i/480i CT only)" on page 82.

Specific sets of SIP parameters are inter-dependent with each other. To prevent conflicting parameter values from being applied, per-line values always take precedence over the corresponding set of global values.

For example, if a parameter value is configured for one of the per-line sets, all parameters from that set are applied and all parameters from the corresponding global section are ignored, even if some of the parameters within the global set are not defined in the per-line set.

SIP Precedence Example

The following example shows the SIP proxy feature and example schema for storage and parsing of the SIP configuration parameters.

The following SIP configuration is assumed:

```
# SIP network block
sip proxy ip:  10.30.11.154
sip proxy port: 5060
sip registrar ip:
10.44.122.37
sip registrar port: 4020
sip line3 proxy ip:
siparator.vonage.com
sip line3 proxy port: 0
```

Line3 specifies per-line values for proxy IP address and proxy port, so the phone uses those parameter values for SIP calls made on that line. However, because those parameters are part of the SIP network block, the phone does not apply any of the global SIP network block parameters. So even though the global parameters configure a SIP registrar, Line3 on the phone ignores all global network block parameters. Since line3 does not contain a per-line SIP registrar entry, the phone does not use a registrar for that line.

Note: Global SIP parameters apply to all lines unless overridden per-line.

Per-line settings are configurable for lines 1 through 7.

SIP Server (SRV) Lookup

The SIP SRV Lookup feature allows you to configure the IP phone to
perform a DNS server lookup on a SIP proxy, a SIP registrar, or a SIP outbound proxy.

The IP phone performs an SRV lookup when the IP address of the server is FQDN and the corresponding port is 0.

For example, if the phone is configured with **sip proxy ip of "ana.aastra.com"**, and **sip proxy port** of "**0**", the SRV lookup may return multilple servers, based on the priorities if one is selected as primary and others are selected as secondary.

However, if the IP address is an FQDN and the corresponding server port is non-zero, then the phone performs a DNS "A" Name Query to resolve the FQDN into dot notation form.

If the IP address is a valid dot notation and the port is zero, then a default port 5060 is used.

You can configure SRV lookup using the configuration files (*aastra.cfg* and *<mac>.cfg*) only. The parameters to use are:

- sip proxy ip
- sip proxy port

Configuring Basic SIP Settings (optonal)

You can configure SIP settings using the configuration files, the IP Phone UI, or the Aastra Web UI.

Note: To configure the SIP settings per-line, use the configuration files or the Aastra Web UI. (The 9112i has only one line available to configure SIP settings.)

Configuration files

For specific parameters you can set in the configuration files, see "SIP Basic, Global Settings" on page 132 or "SIP Basic, Per-Line Settings" on page 138

IP Phone UI

Note: You can set global configuration only using the IP Phone UI.

1. Press **Options** on the phone to enter the Options List.
2. Select **SIP Settings.**
3. Select **Proxy Server** (or **Proxy IP** for 480i/480i CT) and enter an IP address or fully qualified host name for the SIP proxy server.
4. Select **Proxy Port** and enter a port for accessing the SIP proxy server.
5. Select **Registrar Server** (or **Registrar IP** for 480i/480i CT) and enter an IP address or fully qualified host name for the SIP registrar server. A global value of 0.0.0.0 disables registration. However, the phone is still active and you can dial using username@ip address of the phone.

If the Registrar IP address is set to 0.0.0.0 for a per-line basis (i.e, line 1, line 2, etc.), then the register request is not sent, the "No Service" message does not display, and the message waiting indicator (MWI) does not come on.

6. Select **Registrar Port** and enter a port number for accessing the SIP registrar server.
7. Select **Register** and press **Change** to set Register to "**Yes**" (enable) or "**No**" (disable). This parameter enables/disables the IP phone to register on the network.
8. Select **User Name** to enter the user name in the name field of the SIP URI for the IP phone, and for registering the phone at the registrar.
9. Select **Display Name** to enter the name used in the display name field of the "From SIP" header field.
10. Select **Screen Name** to enter the name that displays on the idle screen.
11. Select **Auth Name** to enter the authorization name used in the username field of the Authorization header field of the SIP REGISTER request.
12. Select **Password** to enter the password used to register the IP phone with the SIP proxy.
13. Press **Done** to save the SIP settings.

Aastra Web UI

1. For global configuration, click on **Advanced Settings->Global SIP->Basic SIP Settings**.

or
For per-line configuration, click on **Advanced Settings ->Line *N* (1-9)**.

2. In the "**Screen Name**" field, enter the screen name that displays on the idle screen.
3. In the "**Phone Number**" field, enter the phone number of the IP phone.
4. In the "**Caller ID**" field, enter the phone number of the IP phone.

5. In the "**BLA Number**" field, enter the Bridge Line Appearance (BLA) number to be shared across all IP phones.
6. In the "**Line Mode**" field, select "Generic" for normal mode, "BroadSoft SCA" for a BroadWorks network, or "Nortel" for a Nortel network.
7. In the "**Proxy Server**" field, enter an IP address or fully qualified host name of the SIP proxy server.
8. In the "**Proxy Port**" field, enter a port number for accessing the SIP proxy server.
9. In the "**Outbound Proxy Server**" field, enter the SIP outbound proxy server IP address or fully qualified domain name. This parameter allows all SIP messages originating from a line on the IP phone, to be sent to an outbound proxy server.

Note: If you configure an outbound proxy and registrar for a specific line, and you also configure a global outbound proxy and registrar, the IP phone uses the global configuration for all lines except line 1. Line 1 uses the outbound proxy and registrar that you configured for that line.

10. In the "**Outbound Proxy Port**" field, enter the port on the IP phone that allows SIP messages to be sent to the outbound proxy server.
11. In the "**Registrar Server**" field, enter an IP address or fully qualified host name for the SIP registrar server. A global value of 0.0.0.0 disables registration. However, the phone is still active and you can dial using username@ip address of the phone.

    If the Registrar IP address is set to 0.0.0.0 for a per-line basis (i.e, line 1, line 2, etc.), then the register request is not sent, the "No Service" message does not display, and the message waiting indicator (MWI) does not come on.
12. In the "**Registrar Port**" field, enter the port number associated with the Registrar.
13. In the "**Registration Period**" field, enter the requested registration period, in seconds, from the registrar.
14. In the "**Authentication Name**" field, enter the name used in the username field of the Authorization header of the SIP REGISTER request.
15. In the "**Password**" field, enter the password used to register the IP phone with the SIP proxy
16. Click Save Settings to save your changes.

Configuring Advanced SIP Settings (optional)

Using the configuration files or the Aastra Web UI, you can set more advanced SIP settings on the IP phone such as:

- Explicit MWI Subscription Enabled
- Session Timer
- T1 Timer
- T2 Timer
- Transaction Timer
- Transport Protocol

Configuration files

Note: You can configure Advanced SIP settings on a global-basis only.

For specific parameters you can set in the configuration files, see "Advanced SIP Settings" on page 145.

Aastra Web UI

Note: You can configure Advanced SIP settings on a global-basis only.

1. Click on **Advanced Settings->Global SIP ->Advanced SIP Settings**.

2. Enable the "**Explicit MWI Subscription**" field by checking the check box. (Disable this field by unchecking the check box).
   If the IP phone has a message waiting subscription with the Service Provider, a Message Waiting Indicator (MWI) (LED or display icon) tells the user there is a message on the IP Phone.

3. In the "**Session Timer**" field, enter the time, in seconds, that the IP phone uses to send periodic re-*INVITE* requests to keep a session alive. The proxy uses these re-*INVITE* requests to maintain the status' of the connected sessions. See RFC4028 for details.

4. In the "**Timer 1** and **Timer 2**" fields, enter a time, in milliseconds, that will apply to an IP phone session. These timers are SIP transaction layer timers defined in RFC 3261.

   Timer 1 is an estimate of the round-trip time (RTT). Default is 500 msec.

   Timer 2 represents the amount of time a non-INVITE server transaction takes to respond to a request. Default is 4 seconds.

5. In the "**Transaction Timer**" field, enter the amount of time, in milliseconds, that the phone allows the callserver (registrar/proxy) to respond to SIP messages that it sends. Valid values are 4000 to 64000. Default is 4000.

Note: If the phone does not receive a response in the amount of time designated for this parameter, the phone assumes the message has timed out.

6. In the "**Transport Protocol**" field, select a transport protocol to use when sending SIP Real-time Transport Protocol (RTP) packets. Valid values are User Datagram Protocol (UDP), Transmission Control Protocol (TCP), or both.

7. Click Save Settings to save your changes.

Real-time Transport Protocol (RTP) Settings

Real-time Transport Protocol (RTP) is used as the bearer path for voice packets sent over the IP network. Information in the RTP header tells the receiver how to reconstruct the data and describes how the bit streams are packetized (i.e. which codec is in use). Real-time Transport Control Protocol (RTCP) allows endpoints to monitor packet delivery, detect and compensate for any packet loss in the network. Session Initiation Protocol (SIP) and H.323 both use RTP and RTCP for the media stream, with User Datagram Protocol (UDP) as the transport layer encapsulation protocol.

Note: If RFC2833 relay of DTMF tones is configured, it is sent on the same port as the RTP voice packets.

RTP Port

RTP is described in RFC1889. The UDP port used for RTP streams is traditionally an even-numbered port, and the RTCP control is on the next port up. A phone call therefore uses one pair of ports for each media stream.

On the Aastra IP phone, the initial port used as the starting point for RTP/RTCP port allocation can be configured using "**RTP Port Base**". The default RTP base port on the IP phones is 3000.

For example, if the RTP base port value is 5000, the first voice patch sends RTP on port 5000 and RTCP on port 5001. Additional calls would then use ports 5002, 5003, etc.

You can configure the RTP port on a global-basis only, using the configuration files, the IP Phone UI, or the Aastra Web UI.

Basic Codecs

CODEC is an acronym for **CO**mpress-**DEC**ompress. It consists of a set of instructions that together implement one or more algorithms. In the case of IP telephony, these algorithms are used to compress the sampled speech data, to decrease the content's file size and bit-rate (the amount of network bandwidth in kilobits per second) required to transfer the audio. With smaller file sizes and lower bit rates, the network equipment can store and stream digital media content over a network more easily.

Aastra IP phones support the International Telecommunications Union (ITU) transmission standards for the following CODECs:

- **Waveform CODECs**: G.711 pulse code modulation (PCM) with a-Law or u-Law companding
- **Parametric CODEC**: G.729a conjugate structure - algebraic code excited linear prediction (CS_ACELP).

All Codecs have a sampling rate of 8,000 samples per second, and operate and operate in the 300 Hz to 3,700 Hz audio range. The following table lists the default settings for bit rate, algorithm, packetization time, and silence suppression for each Codec, based on a minimum packet size.

Default Codec Settings.

CODEC	Bit Rate	Algorithm	Packetization Time	Silence Suppression
G.711 a-law	64 Kb/s	PCM	30 ms	enabled
G.711 u-law	64 Kb/s	PCM	30 ms	enabled
G.729a	8 Kb/s	CS-ACELP	30 ms	enabled

You can enable the IP phones to use a default "basic codec" set, which consists of the set of codecs and packet sizes shown above.

Or you can instead configure a custom set of codecs and attributes instead of using the defaults.

Note: The basic and custom codec paramters apply to all calls, and are configured on a global-basis only using the configuration files or the Aastra Web UI.

Customized Codec Preference List

You can also configure the IP phones to use preferred Codecs. To do this, you must enter the payload value (**payload**), the packetization time in milliseconds (**ptime**), and enable or disable silence suppression (**silsupp**).

Payload is the codec type to be used. This represents the data format carried within the RTP packets to the end user at the destination. You can enter payload values for G.711 a-law, G.711 u-law, and G.729a.

Ptime (packetization time) is a measurment of the duration of PCM data within each RTP packet sent to the destination, and hence defines how much network bandwidth is used for transfer of the the RTP stream. You enter the ptime values for the customized Codec list in milliseconds. (See table below).

Silsupp is used to enable or disable silence suppression. Voice Activity Detection (VAD) on the IP phones is used to determine whether each individual packet contains useful speech data. Enabling **silsupp** results in decreased network bandwidth, by avoiding sending RTP packets for any frame where no voice energy was detected by the VAD.

You must enter the values for this feature in list form as shown in the following example:

payload=8;ptime=10;silsupp=on;
payload=0;ptime=10;silsupp=off

The valid values for creating a Codec preference list are as follows.

Customized Codec Settings

Attribute	Value
payload	0 for G.711 u-Law 8 for G.711 a-Law 18 for G.729a
ptime (in milliseconds)	5, 10, 15, 20.......90
silsupp	on off

You can specify a customized Codec preference list on a global-basis using the configuration files or the Aastra Web UI.

Out-of-Band DTMF

The IP phones support out-of-band Dual-Tone Multifrequency (DTMF) mode according to RFC2833. In the Aastra Web UI, you can enable or disable this feature as required. The "out-of-band DTMF" is enabled by default.

In out-of-band mode, the DTMF audio is automatically clamped (muted) and DTMF digits are not sent in the RTP packets.

You can configure out-of-band DTMF on a global-basis using the configuration files or the Aastra Web UI.

DTMF Method

A feature on the IP phone allows you to select the DTMF method that the phone uses to send DTMF digits from the IP phone via INFO messages. You can set the DTMF method as Real-Time Transport Protocol (RTP), SIP info, or both.

You can configure the DTMF method on a global or per-line basis using the configuration files or the Aastra Web UI.

Silence Suppression

In IP telephony, silence on a line (lack of voice) uses up bandwidth when sending voice over a packet-switched system. Silence suppression is encoding that starts and stops the times of silence in order to eliminate that wasted bandwidth.

Silence suppression is enabled by default on the IP phones. The phone negotiates whether or not to use silence suppression. Disabling this feature forces the phone to ignore any negotiated value.

You can configure this parameter via the configuration files or the Aastra Web UI.

Configuring RTP Features

Use the following procedures to configure RTP, basic Codecs, customized Codecs, DTMF, and silence suppression on the IP phone.

Configuration files

For specific parameters you can set in the configuration files, see:

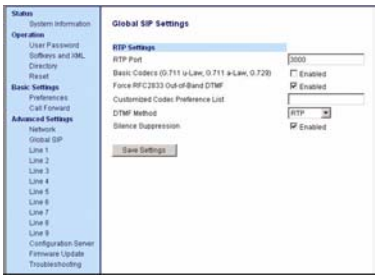
- "RTP, Codec, DTMF Global Settings" on page 148
- "DTMF Per-Line Settings" on page 150
- "Silence Suppression Settings" on page 150.

IP Phone UI

1. Press **Options** on the phone to enter the Options List.
2. Select **SIP Settings**.
3. Select **RTP Port Base** to change the RTP port base setting. Default is 3000.
4. Press **Done** to save the RTP Port Base setting.
5. Press **Restart** at the prompt to restart the IP phone

Aastra Web UI

1. Click on **Advanced Settings->Global SIP->RTP Settings**.



2. Enter an RTP Port Base in the **RTP Port** field. Default is 3000.
3. Enable the "**Basic Codecs (G.711 u-Law, G.711 a-Law, G.729)**" field by checking the check box. (Disable this field by unchecking the box). Enabling this parameter allows the IP phone to use the basic Codecs when sending/receiving RTP packets.
4. Enable the "**Force RFC2833 Out-of-Band DTMF**" field by checking the check box. (Disable this field by unchecking the box). Enabling this parameter forces the IP phone to use out-of-band DTMF according to RFC2833.

5. Enter a "**Customized Codec Preference List**" which allows you to use the preferred Codecs for this IP phone.

   For example, enter the following on one line:
   `payload=8;ptime=10;`
   `silsupp=on;`
   `payload=0;ptime=10;`
   `silsupp=off`.

   Valid values are:

Attribute	Value
payload	0 for G.711 u-Law 8 for G.711 a-Law 18 for G.729a
ptime (in milliseconds)	5, 10, 15, 20.....90
silsupp	on off

6. Select a method to use from the **DTMF Method** list box. Valid values are **RTP**, **SIP Info**, **Both**.

Note: You can also configure the DTMF Method on a per-line basis at **Advanced Settings-> Line N (1-9)**.

7. Silence suppression is enabled by default. If required, disable the "**Silence Suppression**" field by unchecking the check box.
8. Click Save Settings to save your changes.
9. Click on **Operation->Reset**.
10. In the "**Restart Phone**" field click Restart to restart the IP phone.

Operational Features

The IP phone has the following operational features:

- User Passwords - Allows you to change user passwords on the IP phone.
- Administrator Passwords - Allows you to change the administrator passwords on the IP phone (via configuration files only)
- Hard Keys - Allows you enable or disable the use of the Redial, Conference (Conf), and Transfer (Xfer) hard keys on the IP phone. Also allows you to set the Redial and Conf keys as speed-dials.
- Softkeys/Programmable Keys - Allows you to configure softkeys (480i/480i CT) or programmable keys (9112i/9133i) with specific settings such as speeddial, do not disturb, or BLF.
- Suppressing DTMF Playback- Allows you to enable or disable the suppression of DTMF playback when a softkey or programmable key is pressed to dial a number.
- Busy Lamp Field (BLF) (480i/480i CT/9133i only) - Softkey setting that allows extensions to be monitored for state changes.
- Directed Call Pickup (BLF Call Interception) (480i/480i CT/9133i) - Allows you to enable or disable the use of the Directed Call Pickup feature.
- BLF Subscription Period (480i/480i CT/9133i) - Allows you to set the time period that the phone resubscribes the BLF subscription service after a software/firmware upgrade or after a reboot of the IP phone.
- Do Not Disturb (DND) - Programmable key setting that allows you to set the phone to "do not disturb".
- Bridged Line Appearance (BLA) (480i/480i CT/9133i only)- Allows you to assign a phone number to lines that are shared across all phones (global configuration) or shared on a per-line basis (per-line configuration).
- Call Forwarding - Allows incoming calls on the IP phone to be forwarded to another destination. You can also enable or disable the ability to configure the Call Forward feature.
- Callers List - Displays a list of callers that have called the IP phone. You can also enable or disable the Caller List feature.
- Missed Calls Indicator - Displays the number of missed calls on the IP phone. You can also enable or disable the Missed Calls Indicator feature.
- Directory List - Displays a list of names and phone numbers in a directory listing. You can add to this list and edit existing entries. You can also enable or disable the Directory List feature.
- Voicemail (480i/480i CT only) - Allows the IP phone to forward incoming calls to a voicemail service.
- XML Customized Services - Allows you to customize the IP phone UI using XML applications
- SIP Local Dial Plan - Allows the IP phone to use a specific dial plan and dial plan terminator settings. Also allows you to set an idle display name.
- SIP Registration Retry Timer - Allows you to specify a time that the phone waits between registration attempts when a registration is rejected by the registrar.
- Park Calls/Pick Up Parked Calls - Allows you to configure the parking of a live call to a specific extension. You can then pick up the parked call using the call pickup feature.

- Incoming/Outgoing Intercom with Auto-Answer (Intercom applicable to 480i/480i CT only) - Allows you to press the Icom button and enter the number you want to call to initiate an Intercom call. The call can be controlled either locally (phone-side) or by the SIP server (server-side). You can also enable/disable auto-answer and mute/unmute the microphone.
- Audio Transmit and Receive Gain Adjustments - Allows you to adjust the default audio transmit and receive gain settings for the handset, headset, and speaker-phone.
- Ring Tones and Tone Sets - Allows you to set the type of ring tone and ring tone set to use on the IP phone. Ring tones can be configured on a global or per-line basis. Ring tone sets are configurable on a global-basis only.
- Priority Alerting - Allows you to enable or disable priority alert settings. Priority alerting allows incoming calls to trigger pre-defined ringing or call waiting alert tones. Also allows you to set Sylantro-only settings for priority alerting.
- Stuttered Dial Tone - Allows you to enable or disable the playing of a stuttered dial tone when there is a message waiting on the IP phone.
- Call Waiting Tone - Allows you to enable or disable the playing of a call waiting tone when a caller is on an active call and a new call comes into the phone.
- Language - Allows you to set the language to display on the IP phone UI. For the 480i/9133i/9112i, valid languages are English (default), French, Spanish, German, and Italian. For the 480i CT, valid languages are English (default), French, and Spanish.

The following paragraphs describe each of these features.

User Passwords

A user or an administrator can change the user passwords on the phone using the configuration files, the IP phone UI, or the Aastra Web UI. Use the following procedures to change the user password.

Configuration files

For specific parameters you can set in the configuration files, see "Password Settings" on page 114.

IP Phone UI

1. Press **Options** on the phone to enter the Options List.
2. Select **User Password**.
3. Enter the current user password.
4. Enter the new user password.
5. Re-enter the new user password.
6. Press **Enter** to save the new password. A message, "Password Changed" displays on the screen.

Aastra Web UI

1. Click on **Operation->User Password**.

2. In the "**Current Password**" field, enter the current user password.
3. In the "**New Password**" field, enter the new user password.
4. In the "**Confirm Password**" field, enter the new user password again.
5. Click Save Settings to save your changes.

Administrator Passwords

An administrator can change the administrator passwords on the phone using the configuration files only.

An administrator can also assign a password for using the Options key on the IP phone. You turn this feature on and off by entering the "*options password enabled*" parameter followed by a valid value in the configuration files. Valid values are 0 (false; Options key not password protected), or 1 (true; Options key password protected). If this parameter is set to 1, a user has to enter a password at the IP phone UI. If the password is entered correctly, the user is allowed to gain access to the Options Menu and no more password prompts display for other password protected screens. If the user fails to enter the correct password in three attempts, access to the Options Mneu is denied and the IP phoen returns to the idle screen.

Use the following procedures to change the administrator password.

Configuration files

For specific parameters you can set in the configuration files, see "Password Settings" on page 114.

Hard Keys

There are hard keys on your phone, such as **Hold, Redial, Xfer, Icom** and **Conf** (Hold and Icom not available on the 9112i and 9133i) that are configured for specific call-handling features. (See the product-specific User Guide for more information about the hard key functions).

Enabling/Disabling Redial, Xfer, and Conf Keys

You can enable or disable the **Redial**, **Xfer**, and **Conf** hard keys as required using the following parameters in the configuration files:

- **redial disabled**
- **conference disabled**
- **call transfer disabled**

Valid values for this parameter are **0** (enabled) and **1** (disabled).

If this parameter is set to **1**, the key is not active and is ignored if pressed by the user. For "**redial disabled**" the value of 1 does not save the dialed number to the "Redial List".

If this parameter is set to **0**, the key is active and can be pressed by the user.

Use the following procedure to enable/disable the **Redial**, **Xfer**, and **Conf** keys.

Configuration files

For specific parameters you can set in the configuration files, see "Hard Key Parameters" on page page 170.

Setting Redial and Conf Keys as Speedials

You can set the **Redial** and **Conf** hard keys on the IP phone to use as speeddial keys. When the **Redial** or **Conf** key is pressed, the number configured for the key automatically speed dials. If no number is configured, the **Redial** and **Conf** keys return to their original functionality.

You can configure this feature using the configuration files or the Aastra Web UI.

Note: If you configure the **Redial** and **Conf** keys for speeddialing on the 480i CT Base Station, the **Redial** and **Conf** keys on the 480i CT handset retain their original functionality. The **Redial** and **Conf** keys on the handset are not configured for speeddial.

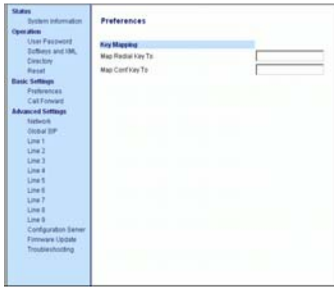
Use the following procedures to set the Redial and Conf keys as redial keys.

Configuration files

For specific parameters you can set in the configuration files, see "Hard Key Parameters" on page page 170.

Aastra Web UI

1. Click on **Basic Settings-> Preferences.**



2. In the **Key Mapping** section, enter a number in the "**Map Redial Key To**" field, that the IP phone will use to speedial when the Redial key is pressed.
3. Enter a number in the "**Map Conf Key To**" field, that the IP phone will use to speedial when the Conf Key is pressed.
4. Click Save Settings to save your changes. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
5. Click on **Operation->Reset.**
6. In the "**Restart Phone**" field click Restart to restart the IP phone.

Softkeys/Programmable Keys

You can configure the softkeys (480i/480i CT) and programmable keys (9112i/9133i) to perform specific functions on the IP phones.

Available configuration options include:

- **none -** Indicates softkey or programmable key is disabled (option for Web UI only).
- **line** (480i, 480i CT, 9133i) - Indicates softkey or programmable key is configured for line use.
- **speeddial** - Indicates softkey or programmable key is configured for speeddial use.
- **do not disturb (dnd)** - Indicates programmable key is configured for "do not disturb" use. (For more information on DND, see Do Not Disturb (DND) on page 60.)
- **BLF** (480i/480i CT/9133i) - Indicates softkey or programmable key is configured for Busy Lamp Field (BLF) use. A user can dial out on a BLF configured softkey.(For more information on BLF, see Bridged Line Appearance (BLA) (480i/480i CT/9133i only) on page 61.)
- **BLF\List** - (480i/480i CT/9133i) - Indicates softkey is configured for BLF list use. A user can dial out on a BLF\List configured softkey. (For more information on BLF List, see Bridged Line Appearance (BLA) (480i/480i CT/9133i only) on page 61.)
- **XML** - Indicates the programmable key is configured to accept an XML application for accessing customized XML services. (For more information on BLF, see XML Customized Services on page 83.)
- **flash** - Indicates the softkey is set to generate a flash event when it is pressed on the 480i and 480i CT, when a programmable key is pressed on the 9112i and 9133i, or a feature key is pressed on the 480i CT handset.

Note: The IP phone generates flash events only when a call is connected and there is an active RTP stream (for example, when the call is not on hold).

- **park-** Indicates the softkey is set to be used as a park key to park an incoming call.
- **pickup** - Indicates the softkey is set to be used as a pickup key to pick up a parked call.
- **empty** (480i, 480i CT) - Indicates the softkey is configured to force a blank entry on the IP phone display for a specific softkey. The soft keys are added in order (from softkey1 to softkey20) after any hard-coded keys have been added. If a particular soft key is not defined, it is ignored.

State-Based Softkeys

Users and administrators can configure a specific state to display when a softkey is being used. Available states you can configure for each softkey include:

- **idle** - The phone is not being used.
- **connected** - The line currently being displayed is in an active call (or the call is on hold)
- **incoming** - The phone is ringing.
- **outgoing** - The user is dialing a number, or the far-end is ringing.

By default, the softkeys display in the states of idle, connected, incoming, and outgoing. All states are enabled.

You can enable or disable the softkey states using the configuration files or the Aastra Web UI.

In the Aastra Web UI, the operational states for each softkey display enabled. To disable a state, you uncheck the box for that operational state.

In the configuration files, you use the following parameters to enable and disable operational states:

```
softkeyN states
```

You can enter multiple values (**idle, connected, incoming, outgoing**) for the "softkeyN state" parameter. For example:

```
softkeyN states: idle connected
```

You must associate the softkeyN state parameter with a specific softkey. In the following example, the softkeyN states parameter is associated with softkey 12:

```
softkey12 type: speeddial
softkey12 label: voicemail
softkey12 value *89
softkey12 states: outgoing
```

Note: Note: The IP phone idle screen condenses the softkeys. So in the previous example, softkey 12 will appear in position 1 if no other softkeys are set. A softkey type of "empty" does not display on the idle screen at all. (For more information about the softkey type of "empty" see "Softkey Settings for 480i and 480i CT" on page 172.

Configuration Example

The following example illustrates the use of the "softkeyN states" parameter, and the "softkeyN type" parameter with a value of **empty**. For clarity purposes, only the "softkeyN type" and "softkeyNstates" parameters are shown.

```
softkey1 type: line
softkey1 states: idle connected

softkey3 type: dnd
softkey3 states: idle

softkey4 type: line

softkey5 type: empty
softkey5 states: connected

softkey6 type: speeddial
softkey6 states: connected
```

The following table shows how the keys in the example above would display on the IP Phone UI.

Note: The "**empty**" key type allows a softkey to be removed quickly by deleting the softkey information from the configuration file.

Softkey	Idle	Connected	Notes
softkey1	Key 1	Key 2	Line displays for softkey1. Key 1 in connected state is the Drop key. Idle and connected display as applicable.
softkey2	(not used)	(not used)	Softkey2 is not displayed.
softkey3	Key 2	(not used)	DND displays for softkey3. Idle displays as applicable.
softkey4	Key 3	Key 3	Line displays for softkey4. Default state values (idle, connected, incoming, outgoing) display as applicable.
softkey5	(not used)	Key 4 (blank)	A blank displays for softkey5. Connected displays as applicable.
softkey6	(not used)	Key 5	Speeddial displays for softkey6. Connected displays as applicable.

Softkeys and programmable keys are configurable using the Aastra Web UI or the configuration files.

The following table provides the number of softkeys and programmable keys you can configure, and the number of lines available for each type of phone.

IP Phone Model	Softkeys Available	Programmable Keys Available	Lines Available	Handset Keys Available
480i	20	-	9	-
480i CT	20	-	9	15
9112i	-	2	1	-
9133i	-	7	9	-

Configuring Softkeys and Programmable Keys

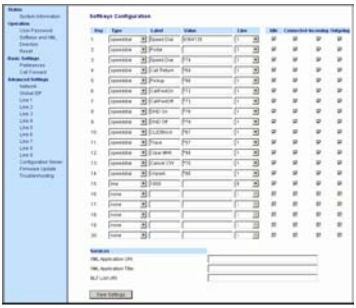
Use the following procedures to configure the softkeys and programmable keys on the IP phone.

Configuration files

For specific parameters you can set in the configuration files, see "Softkey/Programmable Key Parameters" on page 172.

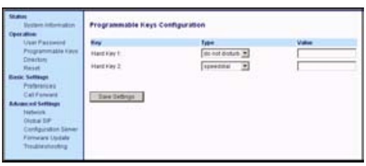
Aastra Web UI

1. For the 480i/480i CT, click on **Operation->Softkeys and XML.**



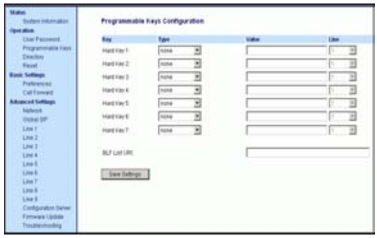
For the 9122i/9133i, click on **Operation->Programmable Keys**

9112i screen.



Note: Only two programmable keys are available on the 9112i.

9133i screen



2. In the "**Type**" field, select the type of softkey or programmable key you want to configure. For available type values on each IP phone model, see "Softkey/Programmable Key Parameters" on page 172.
3. For the 480i/480i CT, in the "**Label**" field, enter a label for the softkey.
4. In the "**Value**" field, enter a value to associate with the softkey or programmable key. For example, for a speeddial value, you can enter *1.
5. For the 480i/480i CT and 9133i, in the "**Line**" field, select the line for which you want to associate the softkey or programmable key.
6. For the 480i/480i CT, all operational states are enabled by default. The operational states display to the IP phone when a softkey is used. To disable an operational state, click the "**Idle**", "**Connected**", "**Incoming**", or "**Outgoing**" fields to uncheck the box.
7. Click Save Settings to save your changes.
8. Click on **Operation->Reset.**
9. In the "**Restart Phone**" field click Restart to restart the IP phone.

480i Cordless (CT) Feature Keys

In addition to the softkeys on the 480i CT, this phone also has handset keys you can configure with specific features. You can use the Aastra Web UI to configure the handset keys.

Note: You configure the handset keys using the Aastra Web UI (**Operation->Handset Keys**) or by pressing the "**F**" button on the handset.

You can program up to 15 feature keys on the 480i CT handset with specific functions using the Aastra Web UI.

The following table identifies the functions available for all 15 handset keys and the default functions for each key.

Key Function	Description	Default for:
Line 1	Line 1 key - Selects line one	Handset Key 1
Line 2	Line 2 key - Selects line two	Handset Key 2
Line 3	Line 3 key - Selects line three	Handset Key 3
Line 4	Line 4 key - Selects line four	Handset Key 4
Icom	Intercom key – Enter handset list to select handset to call	Handset Key 5
Dir	Directory key – Activate directory feature	Handset Key 6
Callers	Callers key – Activate callers feature	Handset Key 7
Xfer	Transfer key - Activate transfer feature	Handset Key 8
Conf	Conference key - Activate conference feature	Handset Key 9
Public	Public key – Toggle between public & private call mode	Handset Key 10
None	No function is selected – this key is empty, no label.	Handset Key 11 and 12
Line 5	Line 5 key (if available) - Selects line five.	Handset Key 13
Line 6	Line 6 key (if available) - Selects line six.	Handset Key 14
Line 7	Line 7key (if available) - Selects line seven.	Handset Key 15
Line 8	Line 8 key (if available) - Selects line eight	
Line 9	Line 9 key (if available) - Selects line nine	

Feature Key Programming Guidelines

The following are guidelines to use when programming the feature keys on the handset:

- All handsets paired with the same Base Station have the same programmed functions since the web interface applies the functions to all the handsets paired with that base.
- A newly registered handset or handset that was out-of-range during the programming needs to perform an "off-hook and on-hook" sequence in order for the newly programmed function to be broadcasted to the affected handsets. Simply press the key from the idle state to go off-hook. Then, press the key to go back on-hook.
- Duplicate functions can exist in the feature key as there is no filtering or duplicate checking done on the handset or the base.
- If no line keys are programmed for the feature key, the handset is restricted to intercom calls only.
- If all 12 programmable functions have been programmed to "None", the user is presented with a `List empty` message when the feature key is pressed.

- For security reasons, the user has 180 seconds (3 minutes) to complete the programming. Otherwise, the phone displays the following error: `** Error **: Session expired, Please reload page.`
- For security reasons, the user must submit the page from the same browser that was used to load the page. If the user tries to submit the page from any other IP address, the following error displays: `** Error ** Session invalid. Different Client IP Addresses.` — `Please reload page`

Programming Feature Keys

You can program up to 15 feature keys on the 480i CT IP phone using the Aastra Web UI. Use the following procedure to program the feature keys on your 480i CT Base Station and all paired handsets.

Aastra Web UI

1. Open your web browser and enter the phone's IP address or host name into the address field.
2. At the prompt, enter your user-name and password and click OK.

Note: For a user, the default user name is "**user**" and the password field is left blank

3. Click on **Operation->Handset Keys**..

4. Select the handset key you want to program.
5. Select the function for that handset key from the "**Key Function**" field.
6. Click Save Settings to save the function you selected to the handset key. The key programming information is sent to the 480i Base Station and to all the cordless handsets associated with that Base Station. Any key programmed to "None" does not appear in the handset's list.

Suppressing DTMF Playback

A feature on the IP phones allows users and administrators to enable or disable the suppression of DTMF playback when a number is dialed from the softkeys and programmable keys.

When suppression of DTMF playback is disabled, and you press a softkey or programmable key, the IP phone dials the stored number and displays each digit as dialed in the LCD window.

When the suppression of DTMF playback is enabled, the IP phone dials the stored number and displays the entire number immediately in the LCD window, allowing the call to be dialed much faster.

DTMF playback suppression is disabled by default. Suppressing DTMF playback can be configured using the Aastra Web UI and the configuration files.

Configuration files

For specific parameters you can set in the configuration files, see "Suppress DTMF Playback Settings" on page 162.

Aastra Web UI

1. Click on **Basic Settings-> Preferences.**

2. Go to the "**General**" section.
3. Enable the "**Suppress DTMF Playback**" field by checking the check box. (Disable this field by unchecking the box). Default is disabled.
4. Click Save Settings to save your settings. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
5. Click on **Operation->Reset.**
6. In the "**Restart Phone**" field click Restart to restart the IP phone and apply the changes.

Busy Lamp Field (BLF) (480i/480i CT/9133i only)

The BLF feature on the IP phones allows a specific extension to be monitored for state changes. BLF monitors the status (busy or idle) of extensions on the IP phone.

Note: The BLF setting is applicable to the Asterisk server only.

Example

A Supervisor configures BLFs on his phone for monitoring the status of a worker's phone use (busy or idle). When the worker picks up his phone to make a call, a busy indicator on the Supervisor's phone shows that the worker's phone is in use and busy.

BLF Setting (For use with Asterisk)

On the 480i and 480i CT, the busy and idle indicators show on the IP phone screen display next to the softkey programmed for BLF functionality. When the monitored user is idle, an icon with the handset on-hook shows next to the BLF softkey. When the monitored user is on an active call, a small telephone icon is shown with the handset off-hook.

On the 9133i, the LED lights next to each BLF programmable key illuminate steady to indicate the monitored line is off-hook or unregistered. The LED goes off when the line is idle.

BLF\List Setting (For use with the BroadSoft Broadworks Rel 13 or higher platform only)

The BLF\List feature on the IP phones is specifically designed to support the BroadSoft Broadworks Rel 13 Busy Lamp Field feature. This feature allows the IP phone to subscribe to a list of monitored users defined through the BroadWorks web portal.

In addition to monitoring the idle and busy state, the BLF\List feature also supports the ringing state. When the monitored user is idle, there is a small telephone icon shown with the handset on-hook. When the monitored user is in ringing state, there is a small bell icon shown. When the monitored user is on an active call then a small telephone icon is shown with the handset off-hook.

On the 9133i phone, the LED lights next to each BLF programmable key illuminate steady to indicate the monitored line is off-hook or unregistered. The LED goes off when the is idle. When the monitored extension is ringing, the LED flashes.

The Broadworks BLF feature is not the same as the Broadworks Shared Call Appearance (SCA) feature and does not permit call control over the monitored extension.

Example

A receptionist has a 480i running Broadsoft firmware that subscribes to a list of extensions from the BroadWorks Application Server. Each monitored extension in the list shows up individually on the 480i screen next to a softkey button. The softkey icons on the screen change depending on the state of the extensions.

On the 9133i running Broadsoft firmware, the programmable key LEDs illuminate either flashing, solid, or turn off depending on the state of those extensions.

Asterisk BLF Configuration

You can enable the BLF feature on Asterisk to enable monitoring for specific extensions. BLF on Asterisk is possible through the "hint" extension parameter.

Add the following in the Asterisk *extensions.conf* file for each target extension being monitored.

For example:

```
exten -> 9995551212,hint,SIP/
9995551212
```

Add the following in the Asterisk *sip.conf* file for each subscriber if it is not defined already.

For example:

```
[9995551212]
Subscribecontext=sip
```

BroadSoft BLF Configuration

You can enable the BLF feature on BroadSoft BroadWorks Rel 13 or higher through the BroadWorks Web Portal. Each user must have the Busy Lamp Field service enabled for their user. The user must add each desired extension to the "Monitored Users List" on the Busy Lamp Field service page and also enter in a list name for the monitored users BLF list on the same page.

Changes to the "Monitored Users List" is dynamic and the Aastra IP phones are automatically updated without requiring a restart.

Reference

For sample BLF configurations, see "Appendix D: Sample BLF Softkey Settings" on page 231.

Configuring BLFs

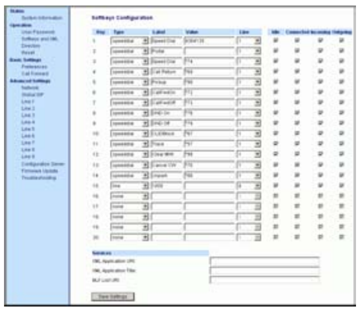
You can set BLF and BLF\List using the Aastra Web UI or the configuration files.

Configuration files

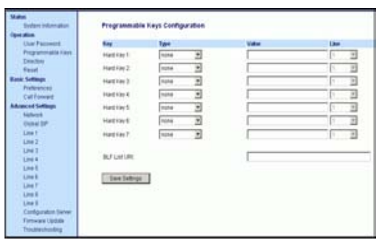
To set BLF or BLF\List in the configuration files, see "Softkey/ Programmable Key Parameters" on page 172.

Aastra Web UI

1. For the 480i/480i CT, click on **Operation->Softkeys and XML.**



For the 9133i, click on **Operation->Programmable Keys.**



Note: You can configure up to 18 softkeys on the 480i/480i CT. You can configure up to 7 programmable keys on the 9133i.

2. Select a softkey or programmable key to configure.

3. For the 480i, 480i CT, or 9133i, in the "**Type**" field, select "BLF" (Asterisk) or "BLF\List" (BroadSoft BroadWorks).
4. For the 480i/480i CT, in the "**Label**" field, enter the name of the person who's extension you are monitoring (Asterisk only).

Note: If BLF\List type is selected, no label value is required. The BroadWorks BLF List name is configured in the "BLF List URI" field instead.

5. In the "**Value**" field, enter a value to associate with the softkey or programmable key. For example, for BLF, the value is the extension you want to monitor. For BLF\List, the value is an identifier for the list of numbers you are monitoring.
6. Click [Save Settings] to save your changes. Changes are dynamic and take effect immediately on the softkey or programmable key on the IP phone.
7. In the "**Line**" field, select a line number that is actively registered to the appropriate SIP proxy you are using.
8. In the "**BLF List URI**" field, enter the name of the BLF list defined on the BroadSoft BroadWorks Busy Lamp field page for your particular user. For example, my480i-blf-list@as.broadworks.com.

Note: The value of the BLF\List URI parameter must match the list name configured. Otherwise, no values display on the 480i screen and the feature is disabled.

9. Click [Save Settings] to save your changes.

Directed Call Pickup (BLF Call Interception) (480i/480i CT/9133i)

Directed call pickup is an enhancement to the existing BLF feature found in 480i/480i CT and 9133i. The existing BLF feature allows a softkey/programmable key to monitor the states of an extension. The extension states can be one of three states: "busy", "ringing" and "idle". If the monitored extension is in the "ringing" state with an incoming call, and "Directed call pickup" is enabled, pressing the BLF key can pick up the incoming call on the monitored extension.

Note: The Asterisk and Epygi Quadro 4x/16x IP PBX servers support this feature. For details about Asterisk support, contact Aastra Technical Support.

You can also enable or disable the playing of a short "call waiting tone" when there is an incoming call on the BLF monitored extension. If the host tone is idle, the tone plays a "ring splash".

You can enable/disable Directed Call Pickup using the configuration files or the Aastra Web UI.

Configuring Directed Call Pickup

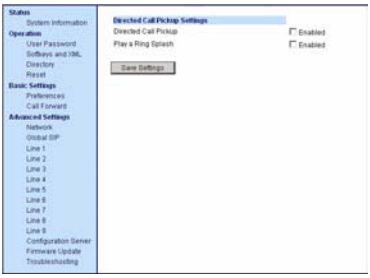
Use the following procedure to enable or disable the Directed Call Pickup feature on the IP phone.

Configuration files

To enable/disable Directed Call Pickup on the IP phone (480i/480i CT/9133i) using the configuration files, see "Directed Call Pickup (BLF Call Interception) Settings" on page 169.

Aastra Web UI

1. Click on **Basic Settings->Preferences->Directed Call Pickup Settings.**



2. Enable the "**Directed Call Pickup**" field by checking the check box. (Disable this field by unchecking the box). Default is disabled.
3. Enable the "**Play a Ring Splash**" field by checking the check box. (Disable this field by unchecking the box). Default is disabled.

   The IP phone plays a short "call waiting tone" when there is an incoming call on the BLF monitored extension. If the "**Play a Ring Splash**" parameter is enabled, and the host tone is idle, the tone plays a "ring splash".
4. Click [Save Settings] to save your changes.
5. Click on **Operation->Reset.**
6. In the "**Restart Phone**" field click [Restart] to restart the IP phone.

BLF Subscription Period (480i/480i CT/9133i)

On the IP phones, you can set the time period, in seconds, that the IP phone resubscribes the BLF subscription service after a software/firmware upgrade or after a reboot of the IP phone.

In the configuration files, you enter the following parameter with a valid value to set the BLF subscription period:

```
sip blf subscription
period: <value in seconds>
```

The minimum value for this 120 seconds (2 minutes). The default is 3600 (1 hour).

Setting this parameter to a value lower than 3600 allows the configured BLF feature to become active more quickly after a software/firmware upgrade or after a reboot of the IP phone. If you enter a value lower than 120 for this parameter, the default value (3600) will be used by the IP phone.

You can configure this feature using the configuration files or the Aastra Web UI.

Configuring BLF Subscription Period

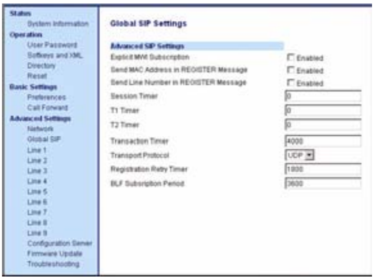
Use the following procedure to configure the BLF subscription period on the IP phone.

Configuration files

To configure the BLF subscription period on the IP phones using the configuration files, see "BLF Subscription Period Settings" on page 169.

Aastra Web UI

1. Click on **Advanced Settings-> Global SIP->Advanced SIP Settings.**



2. Enter a value, in seconds, from 120 (2 min) to 3600 (1 hour) in the "**BLF Subscription Period**" field.
3. Click [Save Settings] to save your changes.
4. Click on **Operation->Reset.**
5. In the "**Restart Phone**" field click [Restart] to restart the IP phone.

Do Not Disturb (DND)

The IP phones have a feature you can enable called "Do not Disturb (DND). You can configure DND on softkeys and programmable keys using the Aastra Web UI or the configuration files.

If DND is configured on the phone, the softkey or programmable key switches DND ON and OFF. If the phone shares a line with other phones, only the phone that has DND configured is affected.

The second line on the screen of the IP phone shows when DND is configured. When a call comes in on the line, the caller hears a busy signal or recorded message, depending on the server configuration.

Configuring DND

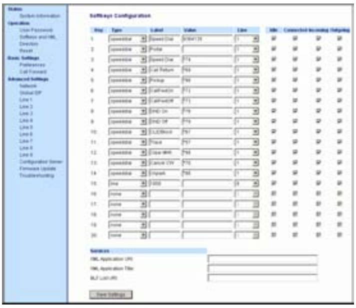
Use the following procedure to configure DND on the IP phone.

Configuration files

For specific softkey and programmable key parameters you can set in the configuration files, see "Softkey/Programmable Key Parameters" on page 172.

Aastra Web UI

1. Click on **Operation->Programmable Keys.**



2. Select a hard key to configure.

3. In the "**Type**" field, select "**do not disturb**".

Note: You do not need to set the "Value" for DND. DND is applied to the hard key only.

4. Click Save Settings to save your changes. These changes are not dynamic. You must restart your IP phone for the changes to take affect.

5. Click on **Operation->Reset.**

6. In the "**Restart Phone**" field click Restart to restart the IP phone.

Bridged Line Appearance (BLA) (480i/480i CT/9133i only)

A SIP bridge line appearance (BLA) on the IP phones allows multiple devices to share a single directory address (DA).

For example, people working at a technical support department could be located in different places. If their desktop phones are configured for BLA DA, when customer calls come in, all the phones with the BLA DA would ring but the call can only be answered by one of them.

Once the call is answered, the rest of the phones reflect the status of the call. If the call was put on "hold" by the original recipient, any one from the group can pick up the call.

Note: 1. This feature is dependent on the IP telephony system to which the IP phone is registered and according to draft-anil-sipping-bla-02.txt.
2. Interactive Intelligence and Sylantro servers support the single BLA group with single line appearance feature only.

You can apply BLA on the IP phones as follows:

- **As a single BLA group** - One BLA DA is shared among multiple phones. Only one phone at a time can pick up an incoming call or initiate an outgoing call on the BLA DA. All phones reflect the usage of the BLA DA. If the call is put on "hold", any one from the group can pick up the "held" call.
- **As a multiple BLA group** - On one single phone, multiple BLA DA can be associated with different line appearances. Every BLA DA is independent from each other and follows the same rules as "a single BLA group".
- **As multiple instances of a BLA DA** - A "x-line-id" parameter was defined in draft-anil-sipping-bla-02.txt to present the incoming call to or place an outgoing call on the specified line appearance instance. The parameter is carried in "Alert-Info" header field over the request-URI (INVITE e.g.) or in the NOTIFY messages to report the status of a dialog.

BLA DA can be configured on a global basis or on a per-line basis on the IP phones using the Aastra Web UI or the configuration files.

The following table shows the number of lines that can be set to BLA for each model phone.

IP Phone Model	Possible # of BLA Lines
480i	9
480i CT	9
9133i	9

Configuring BLA

You can configure BLA on a global or per-line basis using the configuration files or the Aastra Web UI.

Global BLA

You configure BLA on a global basis in the configuration files using the following parameters:

```
sip mode
sip user name
sip bla number
```

You configure BLA on a global basis in the Aastra Web UI using the following fields at **Advanced Settings->Global SIP->Basic SIP Settings:**

- Line Mode
- Phone Number
- BLA Number

Per-Line BLA

You configure BLA on a per-line basis in the configuration files using the following parameters:

```
sip lineN mode
sip lineN username
sip lineN bla number
```

You configure BLA on a per-line basis in the Aastra Web UI using the following fields at **Advanced Settings->Line 1 thru Line 9:**

- Line Mode
- Phone Number
- BLA Number

Sylantro servers and ININ servers require specific configuration methods for per-line configurations.

For Sylantro Server

When configuring the BLA feature on a per-ine basis for a Sylantro server, the value set for the "sip lineN bla number" parameter shall be the same value set for the "sip lineN user name" parameter for all the phones in the group. For example, if sip lineN user name is 1010, you would configure BLA on a per-line basis for the Sylantro server as follows:

`sip line 1 mode: 3`

`sip line1 user name:` **`1010`** `(#` for all the phones)

`sip line1 bla number:` **`1010`**

For ININ Server

When configuring the BLA feature on an ININ server, the value set for the sip lineN bla number parameter shall be the same value set for the sip lineN user name parameter without the incremented digit added to the phone #. For example, if the sip lineN user name for the first phone is 10101, and the sip lineN user name for the second phone is 10102, etc., you would configure BLA on a per-line basis for the ININ server as follows:

(# for phone 1 with appearance of phone 3)

```
sip line1 mode: 3
sip line1 user name: 10101
sip line1 bla number: 1010
```

(# for phone 2 with appearance of phone 3)

```
sip line1 mode: 3
sip line1 user name: 10102
sip line1 bla number: 1010
```

(# for phone 3)

```
sip line1 mode: 3
sip line1 user name: 1010
sip line1 bla number: 1010
```

Note: The original phone number which has the bridged line appearance on other phones, will have the "sip lineN user name" parameter the same as the "sip lineN bla number" (1010 in the above example on Phone 3).

Use the following procedure to configure BLA on the IP phone.

Configuration files

For specific **global** parameters you can set in the configuration files, see "SIP Basic, Global Settings" on page 132.

For specific **per-line** parameters you can set in the configuration files, see "SIP Basic, Per-Line Settings" on page 138.

Aastra Web UI

1. For global configuration, click on **Advanced Settings->Global SIP->Basic SIP Authentication Settings.**

or

For per-line configuration, click on **Advanced Settings ->Line *N* (1-9).**

2. In the "**Line Mode**" field, select the **BLA** option.
3. In the "**Phone Number**" field, enter the phone number of the IP phone.
4. In the "**BLA Number**" field, enter the Bridge Line Appearance (BLA) number to be shared across all IP phones.

5.Click Save Settings to save your changes. These changes are not dynamic. You must restart your IP phone for the changes to take affect.

6.Click on **Operation->Reset.**

7.In the "**Restart Phone**" field click Restart to restart the IP phone.

Using a BLA Line on the IP Phone

If you have either a global or per-line BLA configuration, and you want to share a call on the line with a BLA group, you need to press the Hold button before sharing the call with the group.

For example, if line 1 is configured for BLA, and you pick up a call on line 1, you must press the Hold button to share the call with the BLA group.

If you pick up a call on line 1 configured for BLA, and another call comes in on line 2, you can pick up line 2 without putting line 1 on hold. The line 1 call will be on hold automatically; however it is on hold locally only. The line 1 call cannot be shared with the BLA group.

Note: The Hold button must be pressed for a call on a BLA line to be shared with the BLA group.

Park Calls/Pick Up Parked Calls

The IP phones (including the 480i CT handset) have a park and pickup call feature that allows you to park a call and pickup a call when required. There are two ways a user or adminstrator can configure this feature:

- Using a static configuration
- Using a programmable configuration

Note: The IP phones accept both methods of configuration. However, to avoid redundancy, Aastra Telecom recommends you configure either a static configuration or a programmable configuration.

The IP phones support the Park/Pickup feature on the Asterisk, BroadWorks, Sylantro, and ININ PBX servers.

The following paragraphs describe the park and pickup methods of configuration on the IP phones.

Park/Pickup Static Configuration (480i/480i CT only)

You can configure a static configuration for parking and picking up a call using the Aastra Web UI at **Basic Settings-> Preferences**. By entering the appropriate value in the "**Park Call**" and "**Pickup Parked Call**" fields, you tell the phone where to park a live call and where to pickup the parked call.

On the IP phone UI, the static configuration method displays the following:

- When a call comes in, and you pickup the handset, the default label of "**Park**" displays on the Phone UI.
- After pressing the "**Park**" softkey to park the call, the default label of "**Pickup**" displays on the phone UI.

Note: On the 480i CT handset, pressing displays the "Park" and "Pickup" labels.

The values you enter in the Aastra Web UI for the Park/Pickup call feature are dependant on your type of server. The following table provides the values you enter for the "**Park Call**" and "**Pickup Parked Call**" fields in the Aastra Web UI.

Park/Pickup Call Server Configuration Values

Server	Park Values*	Pickup Values*
Aasterisk	700	700
Sylantro	*98	*99
BroadWorks	*68	*88
ININ PBX	callpark	pickup

*Leave "value" fields blank to disable the park and pickup feature.

Configuring Park /Pickup using Static Configuration (480i/480i CT only)

Use the following procedure to configure the Park/Pickup call feature using the static configuration method.

Note: Aastra recommends you configure either the static or the programmable configuration, but not both.

Aastra Web UI

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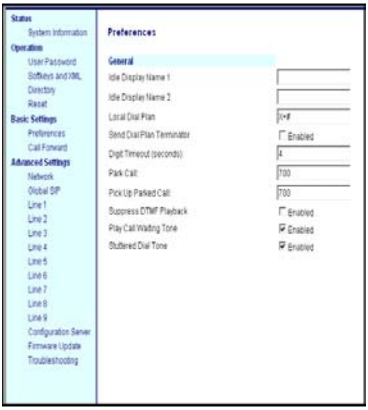
1. Click on **Basic Settings->Preferences->General.**



2. Enter a server value in the **Park Call** field to which incoming live calls will be parked.

Note: For values to enter in this field, see the table "Park/Pickup Call Server Configuration Values" on page 66.

3. Enter a server value in the **"Pickup Parked Call"** field.

Note: For values to enter in this field, see the table "Park/Pickup Call Server Configuration Values" on page 66.

4. Click Save Settings to save your changes.

5. Click on **Operation->Reset.**

6. In the **"Restart Phone"** field click Restart to restart the IP phone.

Park/Pickup Programmable Configuration

The programmable method of configuration creates park and pickup softkeys or programmable keys that you can configure on the IP phones (480i/480i CT/9112i/9133i).

For the 480i/480i CT you can set a softkey as "Park" or "Pickup" and then:

- specify a customized label to display on the Phone UI
- specify a value
- specify which line to use
- specify the state of the park and/or pickup keys

For the 9112i/9133i, you can set a programmable key as "Park" or "Pickup" and then:

- specify a value
- specify a line to use (9133i only)

On 480i/480i CT

On the IP phone UI, the Park/Pickup feature displays the following:

- When a call comes in, and you pickup the handset, the custom label that you configured for the Park softkey displays on the Phone UI.
- After the call is parked, the label that you configured for the Pickup softkey displays on other phones in the network. You can then press the "Pickup" softkey, followed by the applicable value to pickup the call on another phone in your network.

- On the 480i CT, the customized labels apply to the base unit only. On the 480i CT handset, pressing Ï displays the default labels of "Park" and "Pickup".

Notes:

1. On the 480i CT, the customized labels apply to the base unit only. On the 480i CT handset, pressing displays the default labels of "Park" and "Pickup".
2. On the 480i/480i CT, the old softkey labeled "Pickup" has been renamed to "Answer". This softkey uses the old functionality - when you pickup the handset, you see a softkey labeled "Answer". You can then press this key to pick up an incoming call. Do no confuse this feature with the new Park/Pickup configuration feature.

On 9112i/9133i

- When a call comes in, and you pickup the handset, you can press the applicable "Park" programmable key to park the call.
- After the call is parked, you can press the "Pickup" programmable key, followed by the applicable value to pickup the call.

You can configure a Park and Pickup programmable configuration using the configuration files or the Aastra Web UI.

Programmable Configuration Using Config Files

In the configuration files, you configure Park/Pickup using the softkey parameters. You must specify the"softkeyN value" and "prgkeyN value" as **<server type;server-specific value>**. The following examples show Park/ Pickup configurations using specific servers.

Model 480i/480 CT Examples

Server	Park Configuration	Pickup Configuration
Asterisk	softkeyN type: **park** softkeyN label: **parkCall** softkeyN value: **asterisk;700** softkeyN line: **1** softkeyN states: **connected***	softkeyN type: **pickup** softkeyN label: **pickupCall** softkeyN value: **asterisk;700** softkeyN line: **1** softkeyN states: **idle,outgoing****
Sylantro	softkeyN type: **park** softkeyN label: **parkCall** softkeyN value: **sylantro;*98** softkeyN line: **1** softkeyN states: **connected***	softkeyN type: **pickup** softkeyN label: **pickupCall** softkeyN value: **sylantro;*99** softkeyN line: **1** softkeyN states: **idle,outgoing****
BroadWorks	softkeyN type: **park** softkeyN label: **parkCall** softkeyN value: **broadworks;*68** softkeyN line: **1** softkeyN states: **connected***	softkeyN type: **pickup** softkeyN label: **pickupCall** softkeyN value: **broadworks;*88** softkeyN line: **1** softkeyN states: **idle,outgoing****
ININ PBX	softkeyN type: **park** softkeyN label: **parkCall** softkeyN value: **inin;callpark** softkeyN line: **1** softkeyN states: **connected***	softkeyN type: **pickup** softkeyN label: **pickupCall** softkeyN value: **inin;pickup** softkeyN line: **1** softkeyN states: **idle,outgoing****

* When you configure a softkey as "Park", you must configure the state of the softkey as "connected".

** When you configure a softkey as "Pickup", you can configure the state of the softkey as "idle, outgoing", or just "idle", or just "outgoing".

Model 9133i Examples

Server	Park Configuration	Pickup Configuration
Asterisk	prgkeyN type: **park** prgkeyN value: **asterisk;700** prgkeyN line: **1**	prgkeyN type: **pickup** prgkeyN value: **asterisk;700** prgkeyN line: **1**
Sylantro	prgkeyN type: **park** prgkeyN value: **sylantro;*98** prgkeyN line: **1**	prgkeyN type: **pickup** prgkeyN value: **sylantro;*99** prgkeyN line: **1**
BroadWorks	prgkeyN type: **park** prgkeyN value: **broadworks;*68** prgkeyN line: **1**	prgkeyN type: **pickup** prgkeyN value: **broadworks;*88** prgkeyN line: **1**
ININ PBX	prgkeyN type: **park** prgkeyN value: **inin;callpark** prgkeyN line: **1**	prgkeyN type: **pickup** prgkeyN value: **inin;pickup** prgkeyN line: **1**

Model 9112i Examples

Server	Park Configuration	Pickup Configuration
Asterisk	prgkeyN type: **park** prgkeyN value: **asterisk;700**	prgkeyN type: **pickup** prgkeyN value: **asterisk;700**
Sylantro	prgkeyN type: **park** prgkeyN value: **sylantro;*98**	prgkeyN type: **pickup** prgkeyN value: **sylantro;*99**
BroadWorks	prgkeyN type: **park** prgkeyN value: **broadworks;*68**	prgkeyN type: **pickup** prgkeyN value: **broadworks;*88**
ININ PBX	prgkeyN type: **park** prgkeyN value: **inin;callpark**	prgkeyN type: **pickup** prgkeyN value: **inin;pickup**

Note: The 9112i and 9133i do not allow for the configuration of labels and states.

Programmable Configuration Using the Aastra Web UI

On the 480i/480i CT, you configure a Park and/or Pickup key at **Operation->Softkeys and XML.** You enter a key label, and value for a specific line on the phone. The default state of the Park configuration is "**connected**". The default state of the Pickup configuration is "**idle, outgoing**".

The 480i CT handsets use the park/pickup configuration enabled at **Operation->Handset Keys** in the Aastra Web UI. If Park or Pickup are enabled on more than one line on the base unit, the 480i handset uses the first programmable configuration.

For example, if line 1 and line 6 are configured for park, the 480i CT handset uses the configuration set for line 1 to park a call.

On the 9112i/9133i, you configure a Park and/or Pickup key at **Operation->Programmable Keys**, and then enter the appropriate value (and specify a line for the 9133i).

Note: Applicable values depend on the server in your network (Asterisk, BroadWorks, Sylantro, ININ PBX. See the table "Park/Pickup Call Server Configuration Values" on page 66.

Configuring Park/Pickup of Programmable Configuration

Use the following procedures to configure the Park/Pickup call feature using the programmable configuration method.

Configuration files

For specific parameters you can set in the configuration files, see "Softkey Settings for 480i and 480i CT" on page 172 and "Programmable Key Settings for 9112i and 9133i" on page 176.

Aastra Web UI

For 480i/480i CT

1. Click on **Operation-> Softkeys and XML.**

2. Pick a softkey to configure for Parking a call.
3. In the "**Type**" field, select **Park.**
4. In the "**Label**" field, enter a label for the Park softkey.
5. In the "**Value**" field, enter the approriate value based on the server in your network.

Note: For values to enter in this field, see the table "Park/Pickup Call Server Configuration Values" on page 66.

6. In the "**Line**" field, select a line for which to apply the Park configuration.
7. Pick a softkey to configure for Picking up a call.
8. In the "**Label**" field, enter a label for the Pickup softkey.
9. In the "**Value**" field, enter the approriate value based on the server in your network.

Note: For values to enter in this field, see the table "Park/Pickup Call Server Configuration Values" on page 66.

10. Click Save Settings to save your changes.
11. Click on **Operation->Reset.**
12. In the "**Restart Phone**" field click Restart to restart the IP phone.

For 9112i/9133i

1. Click on **Operation-> Programmable Keys and XML.**

9112i Screen.

9133i Screen.

2. Pick a programmable key to configure for Parking a call.
3. In the "**Type**" field, select **Park.**

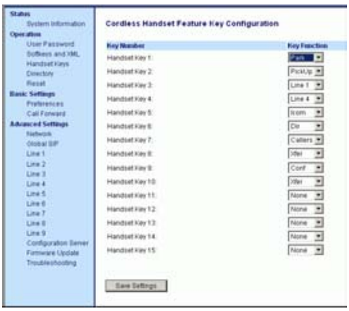
4. In the "**Value**" field, enter the approriate value based on the server in your network.

Note: For values to enter in this field, see the table "Park/Pickup Call Server Configuration Values" on page 66.

5. Click Save Settings to save your changes.
6. Click on **Operation->Reset**.
7. In the "**Restart Phone**" field click Restart to restart the IP phone.

For 480i CT Handset

1. Click on **Operation->Handset Keys**.



2. Pick a handset key to configure for parking a call.
3. In the "**Key Function**" field, select **Park**.
4. Pick another handset key to configure picking up a call.
5. In the "**Key Function**" field, select **Pickup**.
6. Click Save Settings to save your changes.
7. Click on **Operation->Reset**.
8. In the "**Restart Phone**" field click Restart to restart the IP phone.

Using the Park Call/Pickup Parked Call Feature

Use the following procedure on the IP phones to park a call and pick up a parked call.

Park a Call

1. While on a live call, press the "**Park**" softkey.
2. Perform the following for your specific server:

For Asterisk Server
- Server announces the extension number where the call has been parked. Once the call is parked, press the Goodbye key to complete parking.
For BroadWorks Server
- After you hear the greeting from the CallPark server, enter the extension where you want to park the call.
For Sylnatro Server
- Enter the extension number where you want to park the call, followed by "**#**" key.
For ININ Server
- Enter the extension number where you want to park the call, followed by "**#**" key.

If the call is parked successfully, the response is either a greeting voice confirming that the call was parked, or a hang up occurs. The parked call party will get music on hold.

3. If the call fails, you can pick up the call (using the next procedure) and press the "**Park**" softkey again to retry step 2.

Pickup a Parked Call

1. Pick up the handset on the phone.
2. Enter the extension number where the call was parked.
3. Press the "**Pickup**" softkey.

   If the call pick up is successful, you are connected with the parked call.

Call Forwarding

The call forwarding feature on the IP phone allows incoming calls to be forwarded to another destination. The phone sends the SIP message to the SIP proxy, which then forwards the call to the assigned destination.

Call forwarding is disabled by default. You can configure call forwarding on a phone-wide basis or on multi-line phones (480i/480i CT/9133i), on a per-line basis. If you have configured call forwarding on an individual line, then the settings for this line are used; otherwise, the phone-wide call forward settings are used.

You can configure call forwarding on all phones (global settings) or on specific lines (local settings) of a single phone.

For call forwarding you can set the following:

- **Call forward mode**
- **Destination number**
- **Number of rings before forwarding the call** (from 1 to 9 rings)

The following are the call forward modes you can set:

Call Forward Mode	Description
Off	Disables call forward
All	Phone forwards all incoming calls immediately to the specified destination.
Busy	Phone forwards incoming calls if the line is already in use.
No Answer	Phone forwards the call if it is not answered in the specified number of rings
Busy No Answer	Phone forwards the call if either the line is already in use or the call is not answered in the specified number of rings.
Global	Phone uses the phone-wide call forward setting. This is only valid when setting the mode of individual lines.

The following table shows the IP phone model and the number of lines for which you can configure call forwarding.

IP Phone Model	Available Lines for Call Forwarding
480i	9
480i CT	9
9112i	1
9133i	9

Enabling/Disabling the Ability to Configure Call Forwarding

Using the configuration files, you can enable or disable the ability to configure Call Forwarding in the Aastra Web UI and the IP Phone UI. You use the following parameter to enable/disable this feature:

- **call forward disabled**

Valid values for this parameter are **0** (disabled) and **1** (enabled). If this parameter is set to **0**, a user and administrator can configure Call Fowarding via the Aastra Web UI and the IP Phone UI using the "Call Foward" options. If this parameter is set to **1**, all "Call Forward" options are removed from the Aastra Web UI and the IP Phone UI, preventing the ability to configure Call Fowarding.

Configuration files

For specific parameters you can set in the configuration files for enabling/disabling Call Forwarding, see "Call Forward Settings" on page 153.

Configuration Method for Call Forwarding

The method you use to configure call forwarding depends on the model phone you are configuring.

You can set the phone-wide call forward settings using the IP phone UI or the Aastra Web UI. However, you must use the Aastra Web UI to set the per-line call forward settings. The per-line settings override the settings for global call forwarding.

You can set global and per-line settings on the 480i/480i CT/9133i. You set global settings for the 9112i only.

Configuring Call Forwarding

Use the following procedure to configure phone-wide call forwarding.

IP Phone UI

1. Press Options on the phone to enter the Options List.
2. Select **Call Forward.**
3. Enter the "**Call Forward**" number destination for which you want your incoming calls to be forwarded.

Note: If you leave the "Number" field blank, call forwarding is disabled.

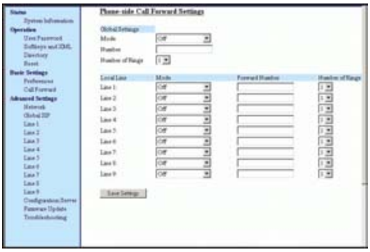
4. Enter the "**Call Mode**" that you want to set on your phone.
5. Enter the "**Number of Rings**" you want to set before the call is forwarded. Valid values are 1 to 9.

Note: "**Number of Rings**" field applies to **No Answer** and **Busy No Answer** modes only.

6. Press **Enter** to confirm the settings.

Aastra Web UI

1. Click on **Basic Settings->Call Forward**.



The 9112i has only one line for Call Forward settings.

For Global Call Forward Settings:

2. In the "**Mode**" field, select the mode you want to set on your phone.

Note: To disable call forwarding in the Aastra Web UI, set the mode to **OFF** and remove the phone number in the "**Number" field**.

3. In the "**Number**" field, enter the call forward number for which you want your calls to be call forwarded.
4. In the "**Number of Rings**" field, enter the number of rings you want to set before the call is forwarded. Valid values are 1 to 9.
5. Click Save Settings to save the Call Forward settings. The changes are dynamic and are immediately applied to the phone.

For Per-Line Call Forward Settings (not applicable to the 9112i):

1. Select a line to set Call Forwarding on.
2. In the "**Mode**" field, select the mode you want to set on this line.

Note: To disable call forwarding in the Aastra Web UI, set the mode to **OFF** and remove the phone number in the "**Forward Number**" field.

Note: To force a line to use the global settings, set the "**Mode**" field to **Global**.

3. In the "**Forward Number**" field, enter the call forward number for which you want your calls on this line to be call forwarded.
4. In the "**Number of Rings**" field, select the number of rings on the line before the call is forwarded. Valid values are 1 to 9.
5. Click Save Settings to save the Call Forward settings. The changes are dynamic and are immediately applied to the phone.

Callers List

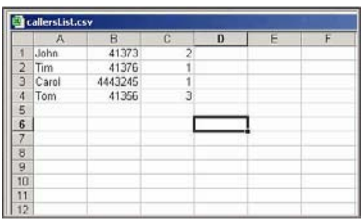
The IP phones have a "Callers List" feature that store the name, phone number, and incremental calls, for each call received by the phone.

You can enable and disable the Callers List feature using the configuration files. When disabled, the Callers List does not disaply on the IP phone UI and the Caller List key is ignored when pressed.

When enabled, you can view, scroll, and delete line items in the Callers List from the IP phone UI. You can also directly dial from a displayed line item in the Callers List. You can download the Callers List to your PC for viewing using the Aastra Web UI.

When you download the Callers List, the phone stores the *callerlist.csv* file to your computer in comma-separated value (CSV) format.

You can use any spreadsheet application to open the file for viewing. The following is an example of a Callers List in a spreadsheet application.


callersList.csv

	A	B	C	D	E	F
1	John	41373	2			
2	Tim	41376	1			
3	Carol	4443245	1			
4	Tom	41356	3			
5						
6						
7						
8						
9						
10						
11						
12						

The file displays the name, phone number, and the line that the call came in on.

Enabling/Disabling Callers List

You can enable and disable user access to the Callers List on the IP phones using the following parameter in the configuration files:

- **callers list disabled**

Valid values for this parameter are **0** (enabled) and **1** (disabled). If this parameter is set to **0**, the Callers List can be accessed by all users. If this parameter is set to **1**, the IP phone does not save any caller information to the Caller List. For 480i and 480i CT phones, the "Caller List" option on the IP phone is removed from the Services menu, and the Caller List key is ignored if pressed by the user.

Configuration files

For specific parameters you can set in the configuration files for enabling/disabling the Callers List, see "Callers List Settings" on page 153.

Using the Callers List

Use the following procedure to use the Callers List.

IP Phone UI

***On the 9112i/9133i*:**

1. Press ☏ on the phone to enter the Callers List.
2. Use the ▲ and ▼ to scroll through the line items in the Callers List.

Note: To the left of a line item, a ☏ icon displays with the handset ON or OFF the receiver. The ON receiver indicates the call came in as a missed call. The OFF receiver indicates the call came in and was answered.

3. To delete all entries in the Callers list, press the ◄ **Delete** key at the "Callers List" header.

   To delete a line item from the Callers List, select the line item you want to delete and press the ◄ **Delete** key.
4. To cancel a delete function, press the ▲ or the ▼ **Scroll** keys.
5. To save a line item to a programmable key for speeddialing, press the ◄ **Save** key and enter the line number at the "Save to?" prompt that is already configured for speeddialing at a programmable key.
6. To dial a displayed entry from the Callers List, pick up the handset, press the handsfree key, or press a line key.
7. To exit the Callers List, press the ☎ key.

On the 480i/480i CT:

1. Press **Services** on the phone to display the **Services** menu.
   or
   Press the ▼ key to enter the Callers List directly. (skip to step 3)
2. From the **Services** menu, select "**Callers List**".
3. Use the ▲ and ▼ to scroll through the line items in the Callers List.

Note: To the left of a line item, a ☎ icon displays with the handset ON or OFF the receiver. The ON receiver indicates the call came in as a missed call. The OFF receiver indicates the call came in and was answered.

4. To delete all entries in the Callers list, press the ◄ **Delete** softkey at the "Callers List" header.

   To delete a line item from the Callers List, select the line item you want to delete and press the ◄ **Delete** softkey.
5. To cancel a delete function, press the ▲ or the ▼ **Scroll** keys.
6. To save a line item to a programmable key for speeddialing, press the ◄ **Save** softkey and enter the line number at the "Save to?" prompt that is already configured for speeddialing at a softkey.
7. To dial a displayed entry from the Callers List, pick up the handset, press the handsfree key, or press a line key.
8. To exit the Callers List, press the **Services** key.

Downloading the Callers List

Use the following procedure to download the Callers List using the Aastra Web UI.

Aastra Web UI

1. Click on **Operation->Directory**.

2. In the Callers List field, click on Save As... .
   A File Download message displays.
3. Click **OK**.
4. Enter the location on your computer where you want to download the Callers List and click **SAVE**.
   The *callerslist.csv* file downloads to your computer.
5. Use a spreadsheet application to open and view the Callers List.

Missed Calls Indicator

The IP phone has a "missed calls" indicator that increments the number of missed calls to the phone. This feature is accessible from the IP phone UI only.

You can enable and disable the Missed Calls Indicator feature using the configuration files. When disabled, the Missed Calls Indicator does not increment as calls come into the IP phone.

When enabled, the number of calls that have not been answered increment on the phone's idle screen as **"<number> New Calls".** As the number of unanswered calls increment, the phone numbers associated with the calls are stored in the Callers List. The user can access the Callers List and clear the call from the list. Once the user accesses the Callers List, the "<number> New Calls" on the idle screen is cleared.

Enabling/Disabling Missed Calls Indicator

You can enable (turn on) and disable (turn off) the Missed Calls Indicator on the IP phones using the following parameter in the configuration files:

- **missed calls indicator disabled**

Valid values for this parameter are **0** (enabled) and **1** (disabled). If this parameter is set to **0**, the indicator increments as unanswered calls come into the IP phone. If set to **1**, the indicator does not increment the unanswered calls.

Configuration files

For specific parameters you can set in the configuration files for enabling/disabling the Missed Calls Indicator, see "Missed Calls Indicator Settings" on page 154.

Accessing and Clearing Missed Calls

Use the following procedure to access and clear missed calls from the Callers List. Once you display the Callers List, the "<number> New Calls" indicator clears.

IP Phone UI

On the 9112i/9133i:

1. Press ☏ on the phone to enter the Callers List.
2. Use the ▲ and ▼ to scroll through the line items in the Callers List to find the line items that have the ☏ icon with the receiver ON. These are the missed calls to the phone.
3. To clear the line item from the Callers List, select the line item you want to clear and press the ◀ **Delete** key.
4. To cancel a delete function, press the ▲ or the ▼ **Scroll** keys.
   The line item is deleted from the Callers List.

On the 480i/480i CT:

1. Press Services on the phone to display the **Services** menu.
   or
   Press the ▼ key to enter the Callers List directly.
   (skip to step 3)
2. From the **Services** menu, select **"Callers List"**.
3. Use the ▲ and ▼ to scroll through the line items in the Callers List to find the line items that have the ☏ icon with the receiver ON. These are the missed calls to the phone.
4. To clear a line item from the Callers List, select the line item you want to delete and press the **Clear** softkey.

   The line item is deleted from the Callers List.

Directory List

The IP phones have a "**Directory List**" feature that allows you to store frequently used numbers on the phone. You can also dial directly from a directory entry.

The Directory feature also includes a quick-search feature that allows you to enter the first letter that corresponds to a name in the Directory to find specific line items. The phone displays the first name with this letter.

Note: The quick-search feature in the Directory List works only when the Directory is first accessed.

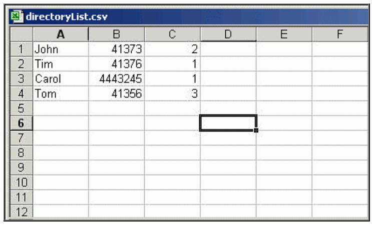
You can enable and disable access to the Directory List using the configuration files. When disabled, the Directory List does not display on the IP phone UI and the Directory List key is ignored when pressed. On the 480i and 480i CT the "**Directory**" option is also removed from the "**Services**" menu.

If the Directory List is enabled, you can view, add, change, and delete entries to/from the Directory List using the IP phone UI. You can also directly dial a number from the Directory List.

For the 480i CT, a public and private softkey can be used at a Directory List line item. The **Private** key toggles a number in the Directory List to private. The **Public** key allows a number in the Directory List to be sent to the handsets. A 480i CT accepts a maximum of 50 entries with the public attribute.

You can download the Directory List to your PC via the Aastra Web UI. The phone stores the *directorylist.csv* file to your PC in comma-separated value (CSV) format.

You can use any spreadsheet application to open the file for viewing. The following is an example of a Directory List in a spreadsheet application.


directoryList.csv

	A	B	C	D	E	F
1	John	41373	2			
2	Tim	41376	1			
3	Carol	4443245	1			
4	Tom	41356	3			
5						
6						
7						
8						
9						
10						
11						
12						

The file displays the name, phone number, and line number for each Directory entry.

Enabling/Disabling Directory List

You can enable and disable user access to the Directory List on the IP phones using the following parameter in the configuration files:

- **directory disabled**

Valid values for this parameter are **0** (enabled) and **1** (disabled). If this parameter is set to **0**, the Directory List can be accessed by all users. If this parameter is set to **1**, the Directory List does not display on the IP phone and the Directory key is disabled. On the 480i and 480i CT the "Directory" option is also removed from the "Services" menu.

Configuration files

For specific parameters you can set in the configuration files for enabling/disabling the Directory List, see "Directory Settings" on page 152.

Server to IP Phone Download

You can populate your IP phone Directory List with server directory files. To activate this feature, you need to add the following parameters to the configuration files:

- **directory 1: company_directory**
- **directory 2 : my_personal_directory'**

The IP phone recognizes the following characters in a Directory List:

Character	Description
'#'	Pound character; any characters appearing after the # on a line are treated as a comment
','	Comma character; used to separate the name, URI number, line, and mode fields within each directory entry.
'"'	Quotation mark; when pound and comma characters are found between quotes in a name field or URI number field, they are treated as regular characters.

A valid directory entry has a name, a URI number, and optional line number, and an optional mode attribute, all separated by commas. If a line number is not present, the entry is assigned to line 1. If a mode attribute (public or private) is not present, the entry is assigned to "**Private**".

The following directory entries are considered valid:

```
# our company's directory
# updated 1 jan 2012

# mode = private, by default
#
joe foo bar, 123456789, 6

# line = 1, by default
# mode = private, by default
#
snidley whiplash, 000111222

# the parser ignores the COMMA
# in the name
# mode = private, by default
#
"manny, jr", 093666888, 9

# the parser ignores the POUND
# chars in the URI number
# mode = private, by default
#
hello dolly, "12#34#7", 2
```

Server to IP Phone Download Behavior

The software that reads directory files from the server, loads the file's contents into the phone's NVRAM when the phone is booting. Directory entries in the NVRAM that originate from a server directory file are 'owned' by the server.

During the boot process both directory files are read, combined into a single list, and any duplicate entries are deleted from the list. Any entries in this list that are not already in the phone's NVRAM are added to the NVRAM and flagged as being owned by the server.

Likewise, any entries in the NVRAM that are owned by the server, but are no longer in one of the server's directory files, are removed from the NVRAM. Entries made from the IP phone UI are never touched.

Directory List Limitations

The following table indicates the maximum characters for each line and field in the Directory List.

Directory List Limitations	
Maximum length of a line	255 characters
Maximum length of a name	15 characters
Maximum length of a URI	45 characters
Maximum number directory entries in the NVRAM	200 entries
Maximum number directory entries in the NVRAM with the "public" attribute (480i CT only)	50 entries

Using the Directory List

Use the following procedures to access the Directory List.

IP Phone UI

***On the 9112i/9133i*:**

1. Press **Directory** on the phone to enter the Directory List.

Note: If no key is pressed within 3 seconds, the phone prompts you to press the first letter in the name of the required directory entry. The phone finds and displays the first name with this letter.

2. Use the and to scroll through the line items in the Directory List.
3. To delete all entries in the Directory list, press the **Delete** key at the "Directory List" header.

   To delete a line item from the Directory List, select the line item you want to delete and press
   the **Delete** key.
4. To cancel a delete function, press the or the **Scroll** keys.
5. To add a new entry to the list, press the **Save** key at the "Directory" header screen and perform step 6.
   or
   Press the **Save** key at a line item and press the **Directory** key again to perform step 6.
6. Enter a phone number, name, and line number and press the **Save** key after each field entry.
7. To save an entry to a programmable key for speeddialing , press the **Save** key and enter the line number at the "Save to?" prompt that is already configured for speeddialing at a programmable key.
8. To edit an entry, use the key for each field you are editing. Press the **Save** key to move to each field.
9. To dial a displayed entry from the Directory List, pick up the handset, press the / handsfree key, or press a line key.
10. To exit the Directory List, press the **Directory** key again.

***On the 480i/480i CT*:**

1. Press Services on the phone to display the **Services** menu.
   or
   Press the key to enter the Directory List directly. (skip to step 3)
2. From the **Services** menu, select "**Directory List**".
3. Use the and to scroll through the line items in the Directory List.
4. To delete all entries in the Directory list, press the **DeleteList** softkey at the "Directory List" header.

   To delete a line item from the Directory List, select the line item you want to delete and press the **Delete** softkey.

5. To cancel a delete function, press the ▲ or the ▼ Scroll keys.
6. To add a new entry to the list, press the **Add New** softkey at the "Directory List" header screen and perform step 7.
   or
   Press the **Add New** softkey at a line item and perform step 7.
7. Enter a phone number, name, and line number and press the ◀ **Save** softkey after each field entry.

Note: The 480i/480i CT allows up to 200 directory entries.

8. For the 480i CT, press the **Public/Private** softkeys to toggle between making the new entry public or private. The entry is set to **Private** by default. If the entry is made **Public**, the entry is sent to the handsets. A 480i CT accepts a maximum of 50 entries with the public attribute.
9. To edit an entry, use the **Change** softkey. A screen displays allowing you to edit the name, phone number, and line number, as well as the public/private setting.
10. To dial a displayed entry from the Directory List, pick up the handset, press the handsfree key, or press the **Dial** softkey.
11. To exit the Directory List, press the **Quit** softkey.

Downloading from the Server to the IP Phone

You can use the configuration files to download the Directory List from the configuration server to the IP phone.

Note: You must use TFTP to download the Directory List.

Configuration files

For specific parameters you can set in the configuration files for downloading the Directory List, see "Directory Settings" on page 152.

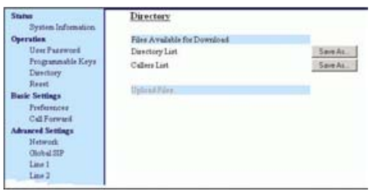
Downloading from the IP Phone to the Server

You can use the Aastra Web UI to download the Directory List from the IP phone to the configuration server.

Note: You must use TFTP to download the Directory List.

Aastra Web UI

1. Click on **Operation->Directory**.



2. In the Directory List field, click on Save As... .
   A File Download message displays.
3. Click **OK**.
4. Enter the location on your computer where you want to download the Directory List and click **SAVE**.
   The *directorylist.csv* file downloads to your computer.
5. Use a spreadsheet application to open and view the Directory List.

Voicemail (480i/480i CT only)

The Voicemail feature on the 480i/ 480i CT IP phones allow you to configure lines with phone numbers so the phone can dial out to connect to a voicemail server. You associate the Voicemail numbers with the phone numbers configured on each line (1 - 9 lines).

For each assigned Voicemail number, there can be a minimum of 0 or a maximum of 1 Voicemail access phone number.

The Voicemail list displays a list of phone numbers assigned to the 480i/480i CT that have registered voicemail accounts associated with them.

Note: The Voicemail list does not display the voicemail access number.

The phone displays up to 99 voicemails for an account even if the number of voicemails exceeds the limit.

Registered account numbers/URIs that exceed the length of the screen, either with or without the voicemail icon and the message count, are truncated with an ellipse character at the end of the number/URI string.

The end of the Voicemail list displays the number of new voicemail messages (if any exist).

Configuring Voicemail (480i/480i CT only)

You configure Voicemail in the configuration files to dial a specific number to access an existing voicemail account. The user then follows the voicemail instructions for listening to voicemails.

Note: The phone must have a registered voicemail account from a server for this feature to be enabled.
When no registered voicemail accounts are registered to the phone, the display shows "List Empty".

To configure the Voicemail feature on the 480i/480i CT, you must enter the following parameter in the configuration files:

- **sip lineN vmail:**

You can enter up to 9 Voicemail numbers associated with each of the 9 lines on the phone.

For example:

sip line1 vmail: *97
sip line2 vmail: *95

Note: In the above example, the user would dial *97 to access the voicemail account for line 1, and *95 to access the voicemail account for line 2.

Configuration files

For specific parameters you can set in the configuration files, see "Voicemail Settings" on page 151.

Using Voicemail (480i480i CT only)

Use the following procedure to access voicemail.

IP Phone UI

1. Press Services on the phone to display the **Services** menu.
2. Select "**Voicemail**".
3. Use the and to scroll through the line items in the Voicemail list.

4.When you have selected a line item, press the / handsfree key, **Scroll Right** key, or press a line softkey to make an outgoing call using the voicemail access phone number associated with the line for which the voicemail account is registered.

From a selected item in the Voicemail list, you can also lift the handset (go offhook) to make an outgoing call using the voicemail access phone number.

XML Customized Services

Extensible Markup Language (XML) is a markup language much like HTML. HTML was designed to display data and to focus on how data looks. XML was designed to describe data and to focus on what data is.

The following are characteristics of XML:

- XML tags are not predefined. You must define your own tags
- XML uses a Document Type Definition (DTD) or an XML Schema to describe the data
- XML with a DTD or XML Schema is designed to be self-descriptive
- XML is a W3C Standard Recommendation

XML and the IP Phones

The XML application for the IP phones allows users to create custom services they can use via the phone's keyboard and display. These services include things like weather and traffic reports, contact information, company info, stock quotes, or custom call scripts.

The IP phone XML application supports four proprietary objects that allow the creation of menu screens, message screens, input screens, and directory screens:

- Text Menu object
- Text Screen object
- UserInput object
- Directory object

Creating XML Customized Screens

For specific information about creating XML menu screens, message screens, user input screens, and directory screens, see "Appendix F: How to Create an XML Application".

XML Object Requests from IP Phone

Users can access XML applications via the "**Services**" menu on the 480i/480i CT and via a programmable key on the 9112i/9133i IP phones. The phone performs an HTTP GET on the URI configured in the Aastra Web UI or configuation files.

You configure the following parameters for object requests:

- **xml application URI**
- **xml application title**

The xml application URI is the application you are loading into the IP phone.

The xml application title is the name of the XML application that displays on the Services menu in the IP Phone UI (as option #4).

XML Push Requests

In addition to initiating a request to an XML application from the Services menu, an HTTP server can push an XML object to the phone via HTTP Post. When the phone sees a PUSH request containing an XML object, it tries to authenticate the request. It does so by checking the IP address or host name of the requesting host against a list of trusted hosts (or domain names) configured via the Aastra Web UI (parameter called **XML Push Server List**) or the configuration files (parameter called **xml application post list**). If the request is authenticated, the XML object is handled by the IP phone accordingly, and displays the information to the screen.

Note: The HTTP Post must contain HTTP packets that have an "xml" line in the message body. For more information aboutn adding "xml" lines in HTTP packets, see “Appendix F: How to Create an XML Application”.

Example XML Configuration

The following example shows the parameters you enter in the configuration files to configure an XML application:

```
xml application URI:
http://172.16.96.63/aastra/
internet.php

xml application title:
Aastra Telecom

xml application post list:
10.50.10.53,
dhcp10-53.ana.aastra.com
```

Configuring XML

After creating an XML application, an administrator can configure the IP phone to use the application using the configuration files or the Aastra Web UI.

Configuration files

For specific parameters you can set in the configuration files, see "XML Settings" on page 155.

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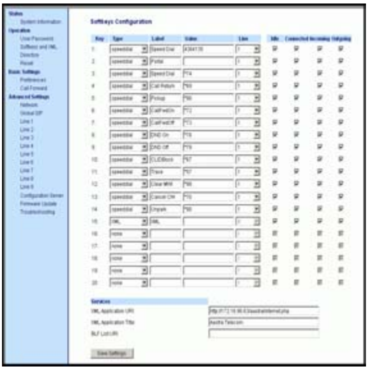
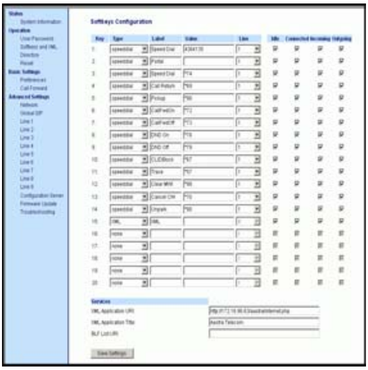
Aastra Web UI

On the 480i/480i CT

1. Click on **Operation-> Softkeys and XML.**



2. Select a key from keys 1 through 20.
3. In the "**Type**" field, select **XML** from the list box.
4. In the "**Label**" field, enter a label that displays on the IP phone for the softkey. For example, "**XML**".
5. In the "**XML Application URI**" field, enter the HTTP server path or qualified domain name of the XML application you want to load to the IP phone. For example, the following illustration shows an XML application called "`http://172.16.96.63/aastra/internet.php`" in the applicable field.



6. In the "**XML Application Title**" field, enter the name of the XML application that you want to display on the IP phone Services Menu. In the illustration above, the XML Application Title is "**Aastra Telecom**".

7. Click Save Settings to save your changes. The XML application is dynamically applied to the IP phone you are configuring.

When the XML application is pushed to the phone via an HTTP POST, a host IP address or domain name server is required.

8. (optional) Click on **Advanced Settings->Configuration Server**.

9. In the "**XML Push Server List (Approved IP Addresses)**" field, enter the host IP address and/or domain name server. You can enter multiple IP address and/or domain name servers (separated by commas). For example, the following illustration shows a host IP address of "10.50.10.53" in the applicable field.

10. Click Save Settings to save your changes. These changes are not dynamic. You must restart your IP phone for the changes to take affect.

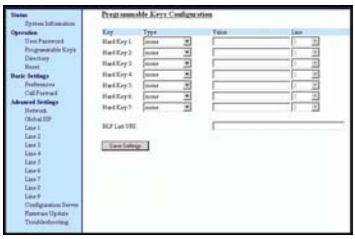
11. Click on **Operation->Reset.**

12. In the "**Restart Phone**" field click Restart to restart the IP phone and push the HTTP Server List.

Note: No posting is performed if a session times out.

On the 9112i/9133i

1. Click on **Operation-> Programmable Keys**.



2. Select a programmable key.
3. In the "**Type**" field, select XML from the list box.
4. In the "**Value**" field, the IP address or qualified domain name of the XML application.
5. Click Save Settings to save your settings. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
6. Click on **Operation->Reset.**
7. In the "**Restart Phone**" field click Restart to restart the IP phone and apply the XML application to the IP phone.

Using the XML Customized Service

After you create, save, and configure the IP phone with an XML application, the customized service is ready for you to use.

Use the following procedure to use the XML feature on the IP phone.

IP Phone UI

On the 480i/480i CT:

1. Press Services on the phone to display the **Services** menu.
2. Select "**Custom Features**".
3. Use the and to scroll through the line items in a menu-driven and directory "**Custom Features**" screen.
   Message services display to the screen after selecting the "**Custom Features**" option.
   For user input services, follow the prompts as appropriate.
4. To exit from the "**Customized Features**" screen, press **Exit.**

***On the 9112i/9133i*:**

1. Press the programmable key configured on the phone for XML.
   A "**Custom Features**" screen displays.
2. Use the and to scroll through the customized features.
3. For menu and directory services, select a service to display the information for that customized service.
   Message services display to the screen after pressing the programmable key.
   For user input services, follow the prompts as appropriate.
4. To exit from the "**Customized Features**" screen, press the XML programmable key again.

SIP Local Dial Plan

A dial plan describes the number and pattern of digits that a user dials to reach a particular telephone number. Access codes, area codes, specialized codes, and combinations of the number of digits dialed are all part of a dial plan. For instance, the North American Public Switched Telephone Network (PSTN) uses a 10-digit dial plan that includes a 3-digit area code and a 7-digit telephone number. Most PBXs support variable length dial plans that use 3 to 11 digits. Dial plans must comply with the telephone networks to which they connect. Only totally private voice networks that are not linked to the PSTN or to other PBXs can use any dial plan.

The IP phones have local dial plan capacity. You configure the SIP Local Dial Plan using the Aastra Web UI or the configuration files.

The IP phone SIP local dial plan available symbols are as follows:

Symbol	Description
0, 1, 2, 3, 4, 5, 6, 7, 8, 9	Digit symbol
X	Match any digit symbol (wildcard)
*, #, .	Other keypad symbol
|	Expression inclusive OR
+	0 or more of the preceding digit symbol or [] expression
[]	Symbol inclusive OR
-	Used only with [], represent a range of acceptable symbols; For example, [2-8]

Dial Plan Example

An example of a SIP Local Dial Plan is:

[01]XXX|[2-8]XXXX|91XXXXXXXXXX|X+.|*XX

The dial plan in the above example can accept any 4-digit dial strings that begin with a '0' or '1', any 5-digit dial strings that begin with a '2' up to '8', any 12-digit dial strings that begin with '91', any non-empty digit string that ends with a '.' or any 2-digit code that begins with a '*'.

SIP Dial Plan Terminator

The IP phone allows the configuration of a dial plan terminator. When you configure the IP phone to use a dial plan terminator or timeout (such as the pound symbol (#)) the phone waits 4 or 5 seconds after you pick up the handset or press a key to make a call.

You can configure the dial plan terminator using Aastra Web UI or the configuration files.

Configuring the SIP Local Dial Plan

Use the following procedures to configure the SIP Local Dial Plan using the configuration files or the Aastra Web UI.

Configuration files

For specific parameters you can set in the configuration files, see "SIP Local Dial Plan Settings" on page 129.

Aastra Web UI

1. Click on **Basic Settings-> Preferences.**

2. In the "**Local Dial Plan**" field, enter a valid local dial plan for the IP phone. Default is X+#|XX+* .
3. Enable the "**Send Dial Plan Terminator**" field by checking the check box. (Disable this field by unchecking the box). Default is disabled.
4. In the "**Digit Timeout (in seconds)**" field, enter a timeout value. This is the length of time the phone waits before dialing. Default is 4 seconds.
5. Click Save Settings to save your settings. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
6. Click on **Operation->Reset.**
7. In the "**Restart Phone**" field click Restart to restart the IP phone and apply the dial plan to the IP phone.

For the 480i/480i CT (optional)

In addition to configuring the dial plan and dial plan terminator above, the 480i/480i CT also allows you to configure names that are displayed on the idle screen rather than the user name and phone number, respectively.

1. In the "**Idle Display Name 1**" field, enter a name that displays on the IP phone when the phone is idle.
2. In the "**Idle Display Name 2**" field, enter another name that displays on the IP phone when the phone is idle.
3. Click Save Settings to save your settings. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
4. Click on **Operation->Reset.**
5. In the "**Restart Phone**" field click Restart to restart the IP phone and apply the display names to the IP phone.

SIP Registration Retry Timer

You can set the number of seconds that the phone waits between registration attempts when a registration is rejected by the registrar. The default value is 1800 seconds (30 minutes) and the minimum value is 30 seconds.

You can configure this timer via the configuration files or the Aastra Web UI.

Configuring SIP Registration Retry Timer

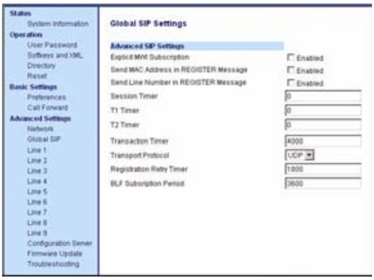
Use the following procedure to configure the SIP registration retry timer on the IP phones..

Configuration files

To configure the SIP registration retry timer using the configuration files, see "SIP Registration Retry Timer Setting" on page 131.

Aastra Web UI

1. Click on **Advanced Settings->Global SIP->Advanced SIP Settings.**



2. Enter a time, in seconds, from 30 to 1800 in the "**Registration Retry Timer**" field.

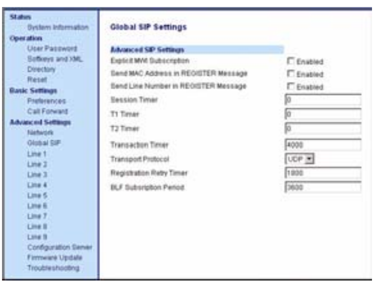
3. Click Save Settings to save your changes.

4. Click on **Operation->Reset.**

5. In the "**Restart Phone**" field click Restart to restart the IP phone.

Aastra Web UI

1. Click on **Advanced Settings->Global SIP->Advanced SIP Settings.**



2. Enter a value, in seconds, from 120 (2 min) to 3600 (1 hour) in the "**BLF Subscription Period**" field.

3. Click Save Settings to save your changes.

4. Click on **Operation->Reset.**

5. In the "**Restart Phone**" field click Restart to restart the IP phone.

Incoming/Outgoing Intercom with Auto-Answer (Intercom applicable to 480i/480i CT only)

The Intercom feature on the IP phones allows you to press the Icom button (Icom) and then enter the number you want to call to initiate an intercom call. Intercom calls can be controlled either locally (phone-side) or by the SIP server (server-side).

Note: Auto-answer is applicable to all IP phone models, but the Intercom feature is applicable to the 480i and 480i CT only.

Outgoing Intercom Calls

On outgoing intercom calls, an available unused line is found when the Icom button is pressed. Since this line has no configuration, the phone applies an existing configuration ("Outgoing Intercom Settings", Line, default is Line 1) to this line in preparation for placing the intercom call. For example, an outgoing intercom call can use the configuration of line 1 but places the actual intercom call using line 9.

A **phone-side** Intercom call indicates the phone is responsible for telling the recipient that an intercom call is being placed, while a **server-side** intercom call means the SIP server is responsible for informing the recipient. Server-side calls require additional configuration of a **prefix code.** After pressing the Icom button and entering the number to call, the phone automatically adds the prefix to the called number and sends the outgoing call via the server.

Incoming Intercom Calls

On incoming intercom calls, you can enable (turn ON) or disable (turn OFF) the microphone on the IP phone. You can also enable or disable auto-answer. If auto-answer is enabled on the IP phone, the phone plays a tone to alert the user before answering the intercom call. If auto-answer is disabled, the phone rejects the incoming intercom call and sends a busy signal to the caller. By default, the microphone is disabled and auto-answer is enabled.

Configuring Intercom Calls and Auto-Answer

You can configure the Intercom feature and auto-anaswer using the configuration files or the Aastra Web UI.

Note: An administrator can configure the incoming and outgoing Intercom feature. A user can configure the incoming Intercom feature only.

Use the following procedures to configure Intercom calls and auto-answer on the IP phone.

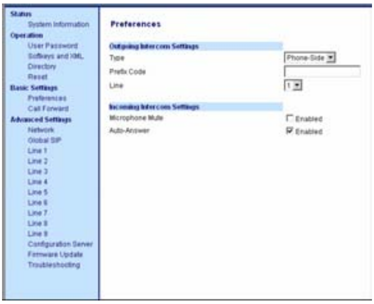
Configuration files

For specific parameters you can set in the configuration files, see "Intercom and Auto-Answer Settings" on page 163.

Aastra Web UI

Outgoing Intercom Settings

1. Click on **Basic Settings->Preferences->Outgoing Intercom Settings.**



2. Select an Intercom type for outgoing Intercom calls from the **Type** list box. Valid values are **Phone-Side, Server-Side, Off.** Default is **Off.**
3. If Server-Side is selected, enter a prefix to add to the phone number in the **"Prefix Code"** field.

Note: For Sylantro servers, enter *96.

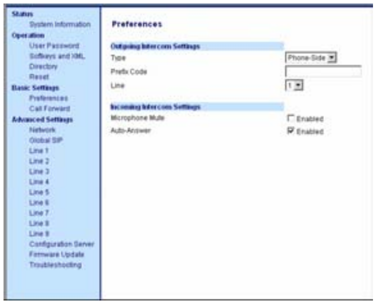
4. If Phone-Side or Server-Side is selected, select a line from the **Line** list box for which you want the IP phone to use as its configuration on the Intercom call.

Note: The IP phone uses the configuration from the line you select from this list box. The call itself is made using the first available line at the time of the call.

5. Click Save Settings to save your changes.
6. Click on **Operation->Reset.**
7. In the "**Restart Phone**" field click Restart to restart the IP phone.

Incoming Intercom Settings

1. Click on **Basic Settings->Preferences->Incoming Intercom Settings.**



2. The "**Microphone Mute**" field is enabled by default. The microphone is muted on the IP phone for Intercom calls made by the originating caller. To disable this field, uncheck the box.
3. The "**Auto-Answer**" field is enabled by default. The automatic answering feature is turned on for the IP phone for answering Intercom calls. To disable this field, uncheck the box.

Note: If the Auto-Answer field is not checked, the phone rejects the incoming intercom call and sends a busy signal to the caller.

4. Click Save Settings to save your changes.
5. Click on **Operation->Reset.**
6. In the "**Restart Phone**" field click Restart to restart the IP phone.

Audio Transmit and Receive Gain Adjustments

The audio gain properties for the IP phone handset, headset, and speakerphone is adjusted to reduce side-tone and echo on the local and far-end equipment. You can adjust these settings from -10 db to +10 db to best suit your comfort level and deployment environment by using the following parameters in the configuration files:

- **headset tx gain**
- **headset sidetone gain**
- **handset tx gain**
- **handset sidetone gain**
- **handsfree tx gain**
- **audio mode**

The default setting for these parameters is 0 (zero).

Note: Aastra Telecom recommends you leave the default of 0 (zero) as the settings for these parameters.

Headset tx gain is the increased (+db) or decreased (-db) amount of signal transmitted from the headset microphone to t he far-end party.

Headset sidetone gain is the increased (+db) or decreased (-db) amount of sidetone signal from the headset microphone to the headset speaker.

Handset tx gain is the increased (+db) or decreased (-db) amount of signal transmitted from the handset microphone to the far-end party.

Handset sidetone gain is the increased (+db) or decreased (-db) amount of sidetone signal from the handset microphone to the handset speaker.

Handsfree tx gain is the increased (+db) or decreased (-db) amount of signal transmitted from the base microphone to the far-end party.

Audio mode allows you to configure how the / key (handsfree key) works. Audio mode has 4 options.

Audio Mode Option	Description
0	**Speaker** - Calls can be made or received using the handset or handsfree speakerphone and can be switched between the two modes by pressing the / key. When on speaker, you can return to using the handset by placing the handset on the cradle and picking it up again.
1	**Headset** - Calls can be made or received using the headset. Calls can be switched between the headset and handset by pressing the / key.
2	**Speaker/Headset** - Incoming calls are sent to the speakerphone . By pressing the / key, you can switch between the handsfree speakerphone, the headset, and the handset.
3	**Headset/Speaker** - Incoming calls are sent to the headset. By pressing the / key, you can switch between the headset, the handsfree speakerphone, and the handset.

Configuring Audio Transmit and Receive Gain Adjustments

You can configure the audio transmit and gain adjustments using the configuration files only.

Use the following procedure to configure this feature.

Configuration files

For specific parameters you can set in the configuration files, see "Audio Transmit and Receive Gain Adjustment Settings" on page 166.

Ring Tones and Tone Sets

You can configure ring tones and ring tone sets on the IP phones.

Ring Tones

There are several distinct ring tones a user or administrator can select from to set on the IP phones. You can enable/disable these ring tones on a global basis or on a per-line basis.

The following table identifies the valid settings and default values for eac h type of configuration method.

Ring Tone Settings Table

Configuration Method	Valid Values	Default Value
Configuration Files	Global: 0 (Tone1) 1 (Tone 2) 2 (Tone 3) 3 (Tone 4) 4 (Tone 5) 5 (Silent) Per-Line: -1 (global) 0 (Tone1) 1 (Tone 2) 2 (Tone 3) 3 (Tone 4) 4 (Tone 5) 5 (Silent)	Global: 0 (tone 1) Per-Line: -1 (global)
IP Phone UI	Global: Tone 1 Tone 2 Tone 3 Tone 4 Tone 5 Silent	Global: Tone 1
Aastra Web UI	Global: Tone 1 Tone 2 Tone 3 Tone 4 Tone 5 Silent Per-Line: Global Tone 1 Tone 2 Tone 3 Tone 4 Tone 5 Silent	Global: Tone 1 Per-Line: Global

Ring Tone Sets

In addition to ring tones, you can configure ring tone sets on a global-basis on the IP phones. Ring tone sets consist of tones customized for a specific country. The ring tone sets you can configure on the IP phones are:

- **US** (Default - also used in Canada)
- **United Kingdom**
- **Italy**
- **Germany**
- **France**
- **Europe** (generic tones)
- **Australia**

When you configure the country's tone set, the country-specific tone is heard on the phone for the following:

- dial tone
- secondary dial tone
- ring tone
- busy tone
- congestion tones
- call waiting tone
- ring cadence pattern

You configure ring tones and tone sets using the Aastra Web UI, IP Phone UI, or configuration files. However, when using the IP phone UI, you can set global configuration only.

Configuring Ring Tones and Tone Sets

Use the following procedures to configure ring tones and tone sets on the IP phones.

Configuration files

For specific parameters you can set in the configuration files for ring tones, see "Ring Tone and Tone Set Global Settings" on page 157 or "Ring Tone Per-Line Settings" on page 158.

IP Phone UI

(global configuration only)

1. Press [Options key] on the phone to enter the Options List.
2. Select **Tones.**
3. Select **Set Ring Tone.**
4. Select the type of ring tone (**Tone 1** through **Tone 5**, or **Silent**).
5. Select **Set** and then select **Next.**
6. Select **Tone Set.**
7. Select the country for which you want to apply the tone set. Valid values are **Australia**, **Europe**, **France**, **Germany**, **Italy**, **UK**, and **US**. Default is **US**.
8. Select **Set.**
   The ring tone and tone set you select is immediately applied to the IP phone.

Aastra Web UI

1. Click on **Basic Settings-> Preferences.**

For global configuration:

2. In the "**Ring Tones**" section, select a country from the "**Tone Set**" field.
3. Select a value from the "**Global Ring Tone**" field.

Note: See the Ring Tone Settings Table on page 94 for valid values.

For per-line configuration:

4. In the "**Ring Tone**" section, select a line for which you want to set ring tone.
5. Select a value from the "**LineN**" field.

Note: See the Ring Tone Settings Table on page 94 for valid values.

6. Click Save Settings to save your settings. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
7. Click on **Operation->Reset.**
8. In the "**Restart Phone**" field click Restart to restart the IP phone and apply the ring tone.

Priority Alerting

Priority alerting on the IP phones is a feature that allows incoming calls to trigger pre-defined ringing or call waiting alert tones.

You can enable or disable priority alerting on the IP phone for the Asterisk, Broadworks, and Sylantro servers using the configuration files and the Aastra Web UI. Configuration of priority alerting is on a global-basis only.

How Priority Alerting Works

When the IP phone detects an incoming call, the phone firmware inspects the INVITE request in the IP packet for an "Alert-Info" header.

If it contains an "Alert-Info" header, the firmware strips out the URL and keyword parameter and maps it to the appropriate Bellcore tone.

If there is no keyword parameter in the "Alert-Info" header, or the INVITE message contains no "Alert-Info" header, then the IP phone firmware uses the Bellcore standard ring tone.

Asterisk/Broadworks Servers

The ring tone keywords that can display in the "Alert-Info" header for an Asterisk and Broadworks server are:

Asterisk/Broadworks Sesrver Ring Tone Keywords
Bellcore-dr2 Bellcore-dr3 Bellcore-dr4 Bellcore-dr5

When the ring tone keywords appear in an "Alert-Info" header from an Asterisk or Broadworks server, the IP phone maps the keywords to the default ring tone patterns.

Sylantro Servers

The ring tone keywords that can display in the "Alert-Info" header for a Sylantro server are:

Sylantro Sesrver Ring Tone Keywords
alert-group alert-external alert-internal alert-emergency alert-priority

When the ring tone keywords appear in an "Alert-Info" header from a Sylantro server, the keyword is mapped to the ring tone pattern based on the configuration you set in the Aastra Web UI or the configuration files.

Ring Tone Patterns

In IP Telephony, different ringing patterns have different frequencies and cadences. Ring cadence is the ringing pattern heard by the called party, before they pick up the call.

On the IP phones, if you enable priority alerting when using an Asterisk or Broadworks server, the IP phone uses the following Bellcore-specified tones by default:

Ring Tone Pattern (Asterisk/Broadworks Servers)

Call Criteria	Bellcore Tones
internal calls	Bellcore-dr2
external calls	Bellcore-dr3
calls with contact list	Bellcore-dr4
calls with specific time frames	Bellcore-dr5

If you enable priority alerting when using a Sylantro server, you can specify the Bellcore tone to be used for the following configurable criteria:

Ring Tone Pattern (Sylantro Servers)

Call Criteria	Bellcore Tones
alert group	Normal ringing (default) Bellcore-dr2 Bellcore-dr3 Bellcore-dr4 Bellcore-dr5 Silent
alert external	Normal ringing (default) Bellcore-dr2 Bellcore-dr3 Bellcore-dr4 Bellcore-dr5 Silent
alert internal	Normal ringing (default) Bellcore-dr2 Bellcore-dr3 Bellcore-dr4 Bellcore-dr5 Silent
alert emergency	Normal ringing (default) Bellcore-dr2 Bellcore-dr3 Bellcore-dr4 Bellcore-dr5 Silent
alert priority	Normal ringing (default) Bellcore-dr2 Bellcore-dr3 Bellcore-dr4 Bellcore-dr5 Silent

The following table identifies the different Bellcore ring tone patterns and cadences.

Bellcore Tone	Pattern ID	Pattern	Cadence	Minimum Duration (ms)	Nominal Duration (ms)	Maximum Duration (ms)
(Standard)	1	Ringing Silent	2s On 4s Off	1800 3600	2000 4000	2200 4400
Bellcore-dr2	2	Ringing Silent Ringing Silent	Long Long Long	630 315 630 3475	800 400 800 4000	1025 525 1025 4400
Bellcore-dr3	3	Ringing Silent Ringing Silent Ringing Silent	Short Short Long	315 145 315 145 630 2975	400 200 400 200 800 4000	525 525 525 525 1025 4400
Bellcore-dr4	4	Ringing Silent Ringing Silent Ringing Silent	Short Long Short	200 145 800 145 200 2975	300 200 1000 200 300 4000	525 525 1100 525 525 4400
Bellcore-dr5	5	Ringing		450	500	550

Note: If the "Do Not Disturb" (DND) or the "Call Forward" (CFWD) feature is enabled on the server-side, and the user is still waiting for a call, the "Bellcore-dr5" is a ring splash tone that reminds the user that these are enabled.

Call Waiting Tones

Call Waiting is a feature that tells you if a new caller is trying to contact you when you are already on the phone.

A discreet tone alerts you to the new caller, so you can answer your second incoming call by putting your first caller on hold.

The IP phones use the following Bellcore-specified call waiting tones.

Bellcore Call-Waiting Tone	Pattern ID	Pattern	Minimum Duration (ms)	Nominal Duration (ms)	Maximum Duration (ms)
CallWaitingTone 1	1	Tone On	270	300	330
Bellcore-dr2 CallWaitingTone2	2	Tone On Tone Off Tone On	90 90 90	100 100 100	110 110 110
Bellcore-dr3 CallWaitingTone3	3	Tone On Tone Off Tone On Tone Off Tone On	90 90 90 90 90	100 100 100 100 100	110 110 110 110 110
Bellcore-dr4 CallWaitingTone4	4	Tone On Tone Off Tone On Tone Off Tone On	90 90 270 90 90	100 100 300 100 100	110 110 330 110 110

For Asterisk and Broadworks servers, call waiting tones are specified by the default Bellcore tones indicated in the table Ring Tone Pattern (Asterisk/ Broadworks Servers) on page 97.

For Sylantro servers, call waiting tones are specified by the Bellcore tones you configure in the Aastra Web UI or the configuration files. See the table Ring Tone Pattern (Sylantro Servers) on page 97.

Configuring Priority Alerting

Use the following procedures to configure priority alerting on the IP phones.

Configuration files

For specific parameters you can set in the configuration files for priority alerting, see "Priority Alert Settings" on page 159.

Aastra Web UI

1. Click on **Basic Settings-> Preferences.**

2. In the "Priority Alerting Settings" section, enable the "**Enable Priority Alerting**" field by checking the check box. (Disable this field by unchecking the box).

For Sylanto Servers

3. Select a ring tone pattern for the "**Group**" field.
4. Select a ring tone pattern for the "**Internal**" field.
5. Select a ring tone pattern for the "**External**" field.
6. Select a ring tone pattern for the "**Emergency**" field.
7. Select a ring tone pattern for the "**Priority**" field.
8. Click Save Settings to save your settings. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
9. Click on **Operation->Reset.**
10. In the "**Restart Phone**" field click Restart to restart the IP phone and apply the ring tone.

Stuttered Dial Tone

You can enable or disable the playing of a stuttered dial tone when there is a message waiting on the IP phone.

You can configure this feature using the configuration files and the Aastra Web UI.

Configuring Stuttered Dial Tone

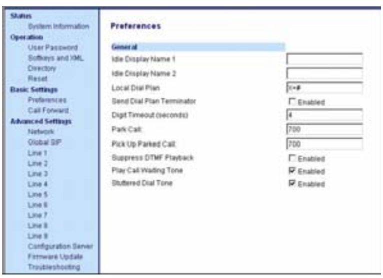
Use the folllowing procedures to configure stuttered dial tone on the IP phones.

Configuration files

For specific parameters you can set in the configuration files for enabling/disabling stuttered dial tone, see "Stuttered Dial Tone Setting" on page 158.

Aastra Web UI

1. Click on **Basic Settings-> Preferences->General.**



2. Stuttered dial tone is enabled by default. If required, disable the "**Stuttered Dial Tone**" field by unchecking the check box.
3. Click Save Settings to save your settings.
4. Click on **Operation->Reset.**
5. In the "**Restart Phone**" field click Restart to restart the IP phone.

Call Waiting Tone

You can enable or disable the playing of a call waiting tone when a caller is on an active call and a new call comes into the phone.

You can configure this feature using the configuration files and the Aastra Web UI.

Configuring Call Waiting Tone

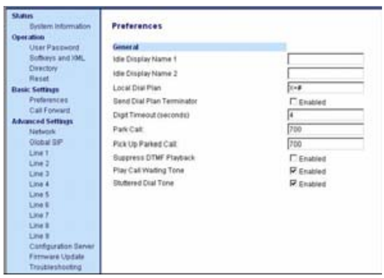
Use the folllowing procedures to configure a call waiting tone on the IP phones.

Configuration files

For specific parameters you can set in the configuration files for enabling/disabling a call waiting tone, see "Call Waiting Tone Setting" on page 159.

Aastra Web UI

1. Click on **Basic Settings-> Preferences->General.**



2. A call waiting tone is enabled by default. If required, disable the "**Play Call Waiting Tone**" field by unchecking the check box.
3. Click Save Settings to save your settings.
4. Click on **Operation->Reset.**
5. In the "**Restart Phone**" field click Restart to restart the IP phone.

Language

Using the IP phone UI or the configuration files, you can set the phones to use a specific language to display in the IP Phone UI.

When you set the language to use on the phone, all of the display screens (menus, services, options, etc.) display in that language.

Valid languages for the 480i/9112i/9133i include English, French, Spanish, German, and Italian.

Valid languages for the 480i CT include English, French, and Spanish. Default language for all model phones is English.

Configuring Language

You configure the language on the IP phone using the **Options** key on the IP phone. To configure language using the configuration files, enter the following parameter:

- **language: <language to set>**

Use the following procedures to configure the language for the IP phone.

Configuration files

For specific parameters you can set in the configuration files for setting languages, see "Language Settings" on page 162.

IP Phone UI

1. Press on the phone to enter the Options List.
2. Select **Language**.
3. Select **In English** (English), **En francais** (French), **En espanol** (Spanish), **In Deutsch** (German), or **In italiano** (Italian).

Note: Valid values for the 480i CT are English, French, and Spanish only.

4. On the 480i/480i CT, select **Done**.
   On the 9112i/9133i, select **Enter** and then **Set** to confirm the change.

   The language you select is immediately applied to the IP phone UI display.

Advanced Operational Features

This section provides the following information about advanced features of the IP phones:

- MAC Address/Line Number in REGISTER Messages - Allows you to enable or disable the sending of the MAC address and line number from the IP phone to the call server, in a REGISTER message.
- SIP Message Sequence for Blind Transfer - Allows you to enable or disable the phone to use the Blind Transfer method available in software prior to release 1.4.
- Update Caller ID During a Call - Allows you to enable or disable the updating of the Caller ID information during a call.
- Boot Sequence Recovery Mode - Allows you to enable or disable Web recovery mode and set the maximum boot count on the IP phone.

MAC Address/Line Number in REGISTER Messages

The IP phones can send the MAC address and line number in the REGISTER packets making it easier for the call server when a user configures the phones via the Aastra Web UI or the IP Phone UI. The following two configurable headers send this information to the call server:

Aastra-Mac: <mac address>
Aastra-Line: <line number>

The MAC address is sent in uppercase hex numbers, for example, 00085D03C792. The line number is a number between 1 and 9.

The following parameters allow you to enable/disable the sending of MAC address and line number to the call server:

sip send mac:
sip send line:

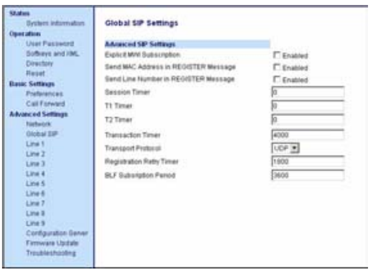
These parameters are disabled by default. The parameters are configurable via the configuration files or the Aastra Web UI.

Configuration files

For specific parameters you can set in the configuration files for enabling/disabling MAC address and line number, see "Advanced Operational Parameters" on page 178.

Aastra Web UI

1. Click on **Advanced Settings->Global SIP->Advanced SIP Settings.**



2. Enable the "**Send MAC Address in REGISTER Message**" field by checking the check box. (Disable this field by unchecking the box).
3. Enable the "**Send Line Number in REGISTER Message**" field by checking the check box. (Disable this field by unchecking the box).
4. Click Save Settings to save your settings. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
5. Click on **Operation->Reset.**
6. In the "**Restart Phone**" field click Restart to restart the IP phone and apply the changes.

SIP Message Sequence for Blind Transfer

The SIP message sequence for Blind Transfer avoids the transfer target having two simultaneous calls. Prior to release 1.4, a CANCEL message was sent to the transfer target (if it was in a ringing state) after sending a REFER to the transferee to complete the transfer. In the 1.4 and later releases, the CANCEL is now sent before the REFER message.

The following parameter allows the system administrator to force the phone to use the Blind Transfer method available in software versions prior to 1.4:

- **sip cancel after blind transfer**

This parameter is configurable via the configuration files only.

Configuration files

For the specific parameter you can set in the configuration files for enabling/disabling the blind transfer method, see "Blind Transfer Setting." on page 179.

Update Caller ID During a Call

It is possible for a proxy or call server to update the Caller ID information that displays on the phone during a call, by modifying the SIP Contact header in the 200 OK message and/or in a re-INVITE message. The phone displays the updated name and number information contained within the Contact header.

The following parameter allows the system administrator to enable or disable this feature:

- **sip update callerid**

This parameter is configurable via the configuration files only.

Configuration files

For the specific parameter you can set in the configuration files for enabling/disabling the update of caller ID during a call, see "Update Caller ID Setting." on page 179.

Boot Sequence Recovery Mode

You can force the IP phone into recovery mode by pressing the **1** and **#** keys during boot up when the logo displays. This feature is enabled by default on the IP phone.

You can disable this feature using the following parameter in the configuration files:

- **force web recovery mode disabled**

Valid values for this parameter are 0 (false) and 1 (true). Default is 0 (false).

A boot counter increments after each faulty boot. When the counter reaches a predetermined value, it forces Web recovery mode. The counter is reset to zero upon a successful boot.

The predetermined value is set using the following parameter in the configuration files:

- **max boot count**

A zero (0) value disables this feature. The default value is 10.

You can configure the boot sequence recovery mode parameters using the configuration files only.

Configuring Boot Sequence Recovery Mode

You configure the boot sequence recover mode using the configuration files.

Configuration files

For the specific parameters you can set in the configuration files for boot sequence recovery mode, see "Boot Sequence Recovery Mode." on page 180.

Encryption and the IP Phone

An encryption feature for the IP phone allows Service Providers the capability of storing encrypted files on their server to protect against unauthorized access and tampering of sensitive information (i.e., user accounts, login passwords, registration information). Service Providers also have the capability of locking a phone to use a specific server-provided configuration only.

Configuration File Encryption Method

Only a System Administrator can encrypt/decrypt the configurations files for an IP Phone.

System Administrators use a password distribution scheme to manually pre-configure or automatically configure the phones to use the encrypted configuration with a unique key.

From a Microsoft Windows command line, the System Administrator uses an Aastra-supplied encryption tool called "*anacrypt.exe*".

Note: Aastra also supplies encryption tools to support Linux platforms (*anacrypt.linux*) and Solaris platforms (*anacrypt.sunos*) if required.

This tool processes the plain text *<mac>.cfg* and *aastra.cfg* files and creates triple-DES encyrpted versions called *<mac>.tuz* and *aastra.tuz*. Encryption is performed using a secret password that is chosen by the administrator.

The encryption tool is also used to create an additional encrypted tag file called *security.tuz*, which controls the decryption process on the IP phones. If *security.tuz* is present on the TFTP/FTP/HTTP server, the IP phones download it and use it locally to decrypt the configuration information from the *aastra.tuz* and *<mac>.tuz* files. Because only the encrypted versions of the configuration files need to be stored on the server, no plain-text configuration or passwords are sent across the network, thereby ensuring security of the configuration data.

To make changes to the configuration files, the System Administrator must decrypt the files, make the changes, and re-encrypt the files. The encrypted files must then be downloaded to the IP phones again.

Note: If the use of encrypted configuration files is enabled (via *security.tuz* or pre-provisioned on the IP phone) the *aastra.cfg* and *<mac>.cfg* files are ignored, and only the encrypted equivalent files *aastra.tuz* and *<mac>.tuz* are read.

Procedure to Encrypt/Decrypt Configuration Files

To encrypt the IP phone configuration files:

1 . Open a command line window application (i.e., DOS window).

1. At the prompt, enter ***anacrypt.exe*** and press <Return>.

```
C:\> anacrypt.exe -h
Provides encryption and decryption of the configuration
files used for the family of Aastra IP phones, using 56bit
triple-DES and site-specific keys.
Copyright (c) 2005, Aastra Technologies, Ltd.
Copyright (c) 1999, Philip J. Erdelsky
Usage:
anacrypt infile.{cfg|tuz} [-o outfile] [-p password] [-h]
[-v] Display version number
[-h] Display program help text
[-o [device:][path]] Writes output file on specific device
or path
[-p password] Password used to generate the cryptographic
key

Restrictions:
Infile extension determines operation, .cfg=plaintext to
be encrypted, .tuz=ciphertext to be decrypted. Outfile
extension is opposite of input.
Filenames may optionally include any non-wildcard subset
of [device:][\path\].
If -p is omitted, user is prompted to interactively enter
the password.
Note: 3DES does not validate decryption, incorrect
password produces garbage. For site-specific keyfile
security.cfg the plaintext must match password.
```

Examples:

The following examples illustrate the use of the anacrypt.exe file.

Example 1

Encrypt aastra.cfg into aastra.tuz using password 1234abcd:

```
C:\> anacrypt aastra.cfg -p 1234abcd
```

Example 2

Decrypt aastra.tuz into aastra.cfg prompting user for password:

```
C:\> anacrypt aastra.tuz
```

Example 3

Decrypt mac.tuz using password 1234abcd, display plaintext on console:

```
C:\> anacrypt aastra.tuz -o CON: -p 1234abcd
```

Example 4

Encrypt a site-specific keyfile prompting user for password and write the encrypted file directly into the TFTP server root directory:

```
C:\> anacrypt security.cfg -o d:\tftp\root
```

Example 5

Encrypt all config files in C:\data using password 1234abcd and write the encrypted files directly into the TFTP server root directory:

```
C:\> FOR %a IN (C:\data\*.cfg) DO "anacrypt %a -o
d:\tftp\root -p 1234abcd"
```

Example 6

Decrypt all config files in the TFTP root directory using password 1234abcd and write the resulting plaintext into the Windows temporary directory:

```
C:\> FOR %a IN (d:\tftp\root\*.tuz) DO "anacrypt %a -o
%TEMP% -p 1234abcd"
```

Example 7

Use the "-v" variable to display version number.

```
C:\> anacrypt -v
The encryption tag format supported by this anacrypt is:
Tuzo v1.3 rev1
The corresponding IP phone firmware build is: 20051017
```

Firmware Upgrade

The IP phone uses a TFTP, FTP, or HTTP server (depending on the protocol configured on the IP phone) to download configuration files and firmware.

The configuration server should be ready and be able to accept connections. For information on setting up the configuration server, see **"Configuration Server Requirement"** on page 2.

You can download the firmware stored on the configuration server in one of three ways:

- Manual firmware update using the Aastra Web UI (TFTP only).
- Manual update of firmware and configuration files (by restarting the phone via the IP phone UI or the Aastra Web UI).
- Automatic update of firmware, configuration files, or both at a specific time in a 24-hour period (via the configuration files or the Aastra Web UI).

Manual Firmware Update (TFTP only)

Use the following procedure to activate a firmware download using TFTP.

Warning: Do not reset or turn off the phone until the download is complete.

Aastra Web UI

(TFTP only)

Note: This procedure allows you to download the *<phone model.st>* file from a TFTP server even if your phone is configured to use HTTP or FTP.

1. Click on **Advanced Settings->Firmware Update.**

2. Enter the TFTP server IP address or qualified domain name in the **"TFTP Server IP"** field.
3. Enter the firmware file name (*<phone model>.st*) that you want to download to your IP phone in the **"File Name"** field. For example, *9112i.st*.

Note: This file name must match the actual name of the firmware file residing on your configuration server.

4. Click Download Firmware.
   This starts the upgrade process. If the upgrade is successful the following message displays on the screen:
   "Firmware Upgrade Successful".

Manual Firmware and Configuration File Update

Restarting the phone forces the phone to check for both firmware and configuration files stored on the configuration server.

Warning: Do not reset or turn off the phone until the download is complete.

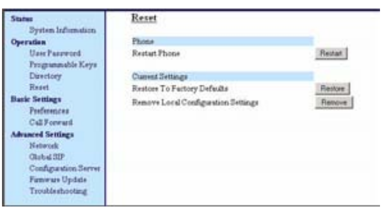
IP Phone UI

1. Press **Options** on the phone to enter the Options List.
2. Select **Phone Status** and press **Show**.
3. Select **Restart Phone**.
4. On the 480i and 480i CT, press **Restart** to restart the phone.
5. On the 9112i and 9133i, press **#** to restart the phone.

   The firmware and configuration files download from the configuration server.

Aastra Web UI

1. Click on **Operation->Reset**.



2. Click Restart .
3. Click **OK** at the confirmation prompt.

The firmware and configuration files download from the configuration server.

Automatic Update (auto-resync)

The auto-resync feature on the IP phones allows an administrator to enable the phone to be updated automatically once a day at a specific time in a 24-hour period if the files on the server have changed. This feature works with TFTP, FTP, and HTTP servers. An administrator can enable this feature using the Aastra Web UI or using the configuration files (aastra.cfg and <mac>.cfg).

Note: The automatic update feature works with both encrypted and plain text configuration files.

When configuring via the Aastra Web UI, the administrator sets the following parameters:

Mode
Time

The Mode parameter determines the type of update that the IP phone performs: configuration file only, firmware only, or both.

The **Time** parameter sets the period of time for which the IP phone is automatically updated.

When configuring via the configuration files, the following parameters must be set:

- **auto-resync mode**
- **auto-resync time**

Configuring Automatic Update

Use the following procedures to configure automatic update of the IP phone firmware, configuration files, or both.

Notes:

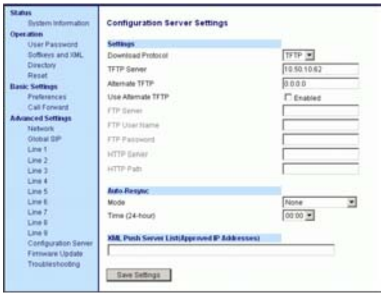
1. If a user is accessing the Aastra Web UI, they are not informed of an auto-reboot.
2. Any changes made using the Aastra Web UI or the IP phone UI are not overwritten by an auto-resync update. Auto-resync affects the configuration files only. However, the settings in the Aastra Web UI take precedence over the IP phone UI and the configuration files.
3. If the IP phone is in use (not idle) at the time of the resync check, the reboot occurs when the phone becomes idle.
4. The resync time is based on the local time of the IP phone.
5. Auto-Resync adds up to 15 minutes random time to the configured time. For example, if the auto resync time parameter is set to 02:00, the event takes place any time between 02:00 and 02:15. This prevents several phones from accessing the server at the exact same time.

Configuration files

For specific parameters you can set in the configuration files for automatic update, see "Configuration Server Settings" on page 115.

Aastra Web UI

1. Click on **Advanced Settings ->Configuration Server ->Auto-Resync.**.



2. Select the auto-resync mode from the **Mode** field. Valid values are **None, Configuration Files, Firmware, Both**. Default is **None**.
3. Select the time from the **Time (24-hour)** field that you want the update to take place. Valid values are 00:00 to 23:30 (in 30 minute increments).
4. Click Save Settings to save your settings. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
5. Click on **Operation->Reset.**
6. In the "**Restart Phone**" field click Restart to restart the IP phone and apply the update.

The update performs automatically at the time you designated.

For more information about setting automatic update on the IP phone, see the "auto resync mode" and "auto resync time" parameters on page 119 and page 120, respectively.

Setting Parameters in Configuration Files

You can set specific configuration parameters in the configuration files for configuring you IP phone. The *aastra.cfg* and *<mac>.cfg* files are stored on the server. The *aastra.cfg* file stores global IP phone configuration settings. The *<mac>.cfg* file stores configuration settings specific to the IP phone with that MAC address. When you restart the IP phone, these files are downloaded to the phone.

If you make changes to the phone configuration, the changes are stored in a local configuration on the phone (not on the server).

Configuration changes made to the *<mac>.cfg* file override the configuration settings in the *aastra.cfg* file.

Reference

For information about configuration file precedence, see "Configuration File Precedence" on page 4.

This section includes the following types of configurable parameters:

- Operational, Basic, and Advanced Parameters on page 112
- Hard Key Parameters on page 170
- Softkey/Programmable Key Parameters on page 172
- Advanced Operational Parameters on page 178

Operational, Basic, and Advanced Parameters

The following sections provide the configuration parameters you can configure on the IP phone. Each parameter table includes the name of the parameter, a description, the format, default value, range, and example. The table also provides the method for which the parameters can be configured (IP phone UI, Aastra Web UI, or configuration files).

Network Settings

Parameter – *dhcp* *DHCP* (in Web UI)	IP phone UI **Options->Network Settings** Aastra Web UI **Advanced Settings->Network** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enabling DHCP will populate most of the network information. The DHCP server should serve the network information that the IP phone requires. If the IP phone is unable to get any required information then it should be entered manually. Parameters affected: ip. Use "0" to disable DHCP and "1" to enable DHCP.
Format	Integer
Default Value	1
Range	0 or 1
Example	dhcp: 1

Parameter – *ip* *Ip Address* (in Web UI)	IP phone UI **Options->Network Settings** Aastra Web UI **Advanced Settings->Network** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This parameter assigns a static IP address to the IP phone device.
Format	IP address
Default Value	0.0.0.0
Range	Not Applicable
Example	ip: 192.168.0.25

Parameter – *subnet mask* *Subnet Mask* (in Web UI)	IP phone UI **Options->Network Settings** Aastra Web UI **Advanced Settings->Network** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Subnet mask defines the IP address range local to the IP phone.
Format	IP address
Default Value	255.255.255.0
Range	Not Applicable
Example	subnet mask: 255.255.255.224

Parameter – *default gateway* *Gateway* (in Web UI)	IP phone UI Aastra Web UI Configuration Files	**Options->Network Settings** **Advanced Settings->Network** **aastra.cfg, <mac>.cfg**
Description	The IP address of the network's gateway or default router IP address.	
Format	IP address	
Default Value	1.0.0.1	
Range	Not Applicable	
Example	default gateway: 192.168.0.1	

Parameter – *dns1* *Primary DNS* (in Web UI)	IP phone UI Aastra Web UI Configuration Files	**Options->Network Settings** **Advanced Settings->Network** **aastra.cfg, <mac>.cfg**
Description	Primary domain name server IP address. For any of the IP address settings on the IP phone a domain name value can be entered instead of an IP address. With the help of the domain name servers the domain names for such parameters can then be resolved to their corresponding IP addresses.	
Format	IP address	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	dns1: 192.168.0.5	

Parameter – *dns2* *Secondary DNS* (in Web UI)	IP phone UI Aastra Web UI Configuration Files	**Options->Network Settings** **Advanced Settings->Network** **aastra.cfg, <mac>.cfg**
Description	Secondary domain name servers' IP address.	
Format	IP address	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	dns2: 192.168.0.6	

Password Settings

Parameter – *admin password*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Allows you to set a new administrator password for the IP phone.
Format	Not Applicable
Default Value	22222
Range	Up to 63 alphanumeric characters
Example	admin password: 123456

Parameter – *user password* *Current Password* (in Web UI)	IP phone UI **Options->User Password** Aastra Web UI **Operation->User Password** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Allows you to set a new user password for the IP phone.
Format	Not Applicable
Default Value	Left Blank
Range	Up to 63 alphanumeric characters
Example	user password: 123456

Parameter – *options password enabled*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables password protection of the Options key on the IP phone. If enabled, upon pressing the Options key, a user has to enter a password at the IP phone UI. If the password is entered correctly, the user is allowed to gain access to the Options Menu and no more password prompts display for other password protected screens. If the user fails to enter the correct password in three attempts, access to the Options Menu is denied and the IP phone returns to the idle screen. **Note:** The password to enter is the administrator password configured for that phone.
Format	Boolean
Default Value	0
Range	0 (false; not password protected) 1 (true; password protected)
Example	options password enabled: 1

Aastra Web UI Settings

Parameter – *web interface enabled*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the Aastra Web UI for a single IP phone when placed in the *<mac>.cfg* file. Enables or disables the Aastra Web UI for all phones when placed in the *aastra.cfg* file.
Format	Boolean
Default Value	Not Applicable
Range	0 = Disable 1 = Enable
Example	web interface enabled: 1

Configuration Server Settings

Parameter – *download protocol* *Download Protocol* (in Web UI)	IP phone UI **Options->Network** Aastra Web UI **Advanced Settings-> Configuration Server** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Protocol to use for downloading new versions of software to the IP phone.
Format	Text
Default Value	TFTP
Range	TFTP, FTP, HTTP
Example	download protocol: FTP

Parameter – *tftp server* *TFTP Server* (in Web UI)	IP phone UI **Options->Network-> TFTP Server->Primary TFTP** Aastra Web UI **Advanced Settings-> Configuration Server** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The TFTP server's IP address or qualified domain name. If DHCP is enabled and the DHCP server provides the information, this field is automatically populated. Use this parameter to change the IP address or domain name of the TFTP server. This will become effective after this configuration file has been downloaded into the phone.
Format	IP address or qualified domain name
Default Value	0.0.0.0
Range	Not Applicable
Example	tftp server: 192.168.0.130

Parameter – *alternative tftp server* *Alternate TFTP* (in Web UI)	IP phone UI Aastra Web UI Configuration Files	**Options->Network-> TFTP Server->Alernate TFTP** **Advanced Settings-> Configuration Server** **aastra.cfg, <mac>.cfg**
Description	The alternate TFTP server's IP address or qualified domain name. This will become effective after this configuration file has been downloaded into the phone.	
Format	IP address or qualified domain name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	alternative tftp server: 192.168.0.132	

Parameter – *use alternative tftp server* *Use Alternate TFTP* (in Web UI)	IP phone UI Aastra Web UI Configuration Files	**Options->Network-> TFTP Server->Select TFTP** **Advanced Settings-> Configuration Server** **aastra.cfg, <mac>.cfg**
Description	Enables or disables the alternate TFTP server. Valid values are "0" disabled and "1" enabled.	
Format	Not Applicable	
Default Value	0	
Range	0 or 1	
Example	use alternative tftp server: 1	

Parameter – *ftp server* *FTP Server* (in Web UI)	IP phone UI Aastra Web UI Configuration Files	**Options->Network->FTP Server** **Advanced Settings-> Configuration Server** **aastra.cfg, <mac>.cfg**
Description	The FTP server's IP address or network host name. This will become effective after this configuration file has been downloaded into the phone. **Optional:** You can also assign a username and password for access to the FTP server. See the following parameters for setting username and password.	
Format	IP address or fully qualified Domain Name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	ftp server: 192.168.0.131	

Parameter – *ftp username* *FTP User Name* (in Web UI)	IP phone UI **Options->Network->FTP Server** Aastra Web UI **Advanced Settings->Configuration Server** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The username to enter for accessing the FTP server. This will become effective after this configuration file has been downloaded into the phone.
Format	Text
Default Value	Not Applicable
Range	Up to 63 alphanumeric characters
Example	ftp username: 480iaastra

Parameter – *ftp password* *FTP Password* (in Web UI)	IP phone UI **Options->Network->FTP Server** Aastra Web UI **Advanced Settings->Configuration Server** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The password to enter for accessing the FTP server. This will become effective after this configuration file has been downloaded into the phone.
Format	Text
Default Value	Not Applicable
Range	Up to 63 alphanumeric characters
Example	ftp password: 1234fcs

Parameter – *http server* *HTTP Server* (in Web UI)	IP phone UI **Options->Network->HTTP Server** Aastra Web UI **Advanced Settings->Configuration Server** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The HTTP server's IP address. This will become effective after this configuration file has been downloaded into the phone. **Optional**: You can also assign an HTTP relative path to the HTTP server. See the following parameter.
Format	IP address or fully qualified Domain Name
Default Value	0.0.0.0
Range	Not Applicable
Example	http server: 192.168.0.132

Parameter – *http path* *HTTP Path* (in Web UI)	IP phone UI Aastra Web UI Configuration Files	**Options->Network-> HTTP Server** **Advanced Settings-> Configuration Server** **aastra.cfg, <mac>.cfg**
Description	The HTTP sub-directory path name to enter. If the IP phone's files are located in a sub-directory beneath the server's HTTP root directory, the relative path to that sub-directory should be entered in this field.	
Format	dir/dir/dir	
Default Value	Not Applicable	
Range	Up to 63 alphanumeric characters	
Example	http path: ipphones/480i	

Parameter – *auto resync mode* *Mode* (in Web UI)	Aastra Web UI: **Advanced Settings-> Configuration Server-> Auto-Resync** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Enables and disables the phone to be updated automatically once a day at a specific time in a 24-hour period. This parameter works with TFTP, FTP, and HTTP servers. Valid values are: **None (0)** - Disable auto-resync **Configuration Files (1)** - Updates the configuration files on the IP phone automatically at the specified time if the files on the server have changed. **Firmware (2)** - Updates the firmware on the IP phone automatically at the specified time if the files on the server have changed. **Both (3)** - Updates the configuration files and firmware automatically at the specified time if the files on the server have changed. **Notes:** 1. If a user is accessing the Aastra Web UI, they are not informed of an auto-reboot. 2. Any changes made using the Aastra Web UI or the IP phone UI are not overwritten by an auto-resync update. Auto-resync affects the configuration files only. However, the settings in the Aastra Web UI take precedence over the IP phone UI and the configuration files. 3. The resync time is based on the local time of theIP phone. 4. If the IP phone is in use (not idle) at the time of the resync check, the reboot occurs when the phone becomes idle. 5. The automatic update feature works with both encrypted and plain text configuration files.
Format	Integer
Default Value	Aastra Web UI None Configuration Files 0
Range	Aastra Web UI None Configuration Files Firmware Both Configuration Files 0 (none) 1 (configuration files only) 2 (firmware only) 3 (configuration files and firmware)
Example	auto resync mode: 1

Parameter – *auto resync time* *Time (24-hour)* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings-> Configuration Server-> Auto-Resync** **aastra.cfg, <mac>.cfg**
Description	Sets the time of day in a 24-hour period for the IP phone to be automatically updated. This parameter works with TFTP, FTP, and HTTP servers. **Notes:** 1. The resync time is based on the local time of the IP phone. 2. The value of 00:00 is 12:00 A.M. 3. When selecting a value for this parameter in the Aastra Web UI, the values are in 30-minute increments only. 4. When entering a value for this parameter using the configuration files, the value can be entered using minute values from 00 to 59 (for example, the auto resync time can be entered as 02:56). 5. Auto-Resync adds up to 15 minutes random time to the configured time. For example, if the auto resync time parameter is set to 02:00, the event takes place any time between 02:00 and 02:15. 6. When the language on the phone is set to French or Spanish, you must enter the time in the format "00h00" (configuration files only).	
Format	hh:mm 00h00 (for French and Spanish configuration files)	
Default Value	Aastra Web UI 00:00 Configuration Files 00:00	
Range	Aastra Web UI 00:00 to 23:30 (in 30 minute increments) Configuration Files hh = 00 to 23 mm = 00 to 59	
Example	auto resync time: 03:24	

Type of Service (ToS)/DSCP Settings

Parameter – *tos sip*	IP phone UI **Options->Network-> Type of Service->SIP** Aastra Web UI **Advanced Settings->Network ->Type of Service,DSCP** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The Differentiated Services Code Point (DSCP) for SIP packets.
Format	Integer
Default Value	24
Range	0 to 63
Example	tos sip: 3

Parameter – *tos rtp*	IP phone UI **Options->Network-> Type of Service->RTP** Aastra Web UI **Advanced Settings->Network ->Type of Service,DSCP** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The Differentiated Services Code Point (DSCP) for RTP packets.
Format	Integer
Default Value	32
Range	0 to 63
Example	tos rtp: 2

Parameter – *tos rtcp*	IP phone UI **Options->Network-> Type of Service->RTCP** Aastra Web UI **Advanced Settings->Network ->Type of Service,DSCP** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The Differentiated Services Code Point (DSCP) for RTCP packets.
Format	Integer
Default Value	32
Range	0 to 63
Example	tos rtcp: 3

Virtual Local Area Network (VLAN) Settings

Parameter – *tagging enabled* *VLAN Enable* (in Web UI)	IP phone UI Aastra Web UI Configuration Files	**Options->Network->VLAN ->VLAN Enable** **Advanced Settings->Network ->VLAN->Global** **aastra.cfg, <mac>.cfg**
Description	Enables or disables VLAN on the IP phones.	
Format	Boolean	
Default Value	0 (false)	
Range	0 (false) 1 (true)	
Example	tagging enabled: 1	

Parameter – *vlan id* *VLAN id* (for Port 0 in Web UI)	IP phone UI Aastra Web UI Configuration Files	**Options->Network->VLAN ->Phone->VLAN ID** **Advanced Settings->Network ->VLAN->Port 0** **aastra.cfg, <mac>.cfg**
Description	VLAN is a feature on the IP phone that allows for multiple logical Ethernet interfaces to send outgoing RTP packets over a single physical Ethernet as described in IEEE Std 802.3. On the IP phone, you configure a VLAN ID that associates with the physical Ethernet Port 0.	
Format	Integer	
Default Value	1	
Range	1 to 4094	
Example	vlan id: 300	

Parameter – *tos priority map* *SIP Priority* *RTP Priority* *RTCP Priority* (for Port 0 in Web UI)	IP phone UI	**Options->Network->VLAN ->Phone->Priority->SIP** **Options->Network->VLAN ->Phone->Priority->RTP** **Options->Network->VLAN ->Phone->Priority->RTCP**
	Aastra Web UI	**Advanced Settings->Network ->VLAN->Port 0**
	Configuration Files	**aastra.cfg, <mac>.cfg**
Description	This parameter is based on the Type of Service (ToS), Differentiated Services Code Point (DSCP) setting for SIP (tos sip parameter), RTP (tos rtp parameter) and RTCP (tos rtcp parameter). It is the mapping between the DSCP value and the VLAN priority value for SIP, RTP, and RTCP packets. You enter the tos priority map value as follows: (DSCP_1,Priority_1)(DSCP_2,Priority_2).....(DSCP_64,Priority_64) where the DSCP value range is 0-63 and the priority range is 0-7. Mappings not enclosed in parentheses and separated with a comma, or with values outside the ranges, are ignored.	
Format	Integer	
Default Value	3 (based on the default ToS DSCP SIP setting of 24) 4 (based on the default ToS DSCP RTP setting of 32) 4 (based on the default ToS DSCP RTCP setting of 32)	
Range	0 to 63 (for DSCP) 0 to 7 (for SIP, RTP, and RTCP priorities)	
Example	tos priority map: (24,7)	

The following table identifies the default DSCP-to-priority mapping structure.

DSCP Range	DSCP Priority
0-7	0
8-15	1
16-23	2
24-31	3
32-39	4
40-47	5
48-55	6
56-63	7

Parameter – *priority non-ip* *Priority, Non-IP Packet* (in Web UI)	IP phone UI **Options->Network->VLAN ->Phone->Priority->Other** Aastra Web UI **Advanced Settings->Network ->VLAN->Global** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Specifies the priority value for non-IP packets.
Format	Integer
Default Value	5
Range	0 to 7
Example	priority non-ip: 7

Parameter – *QoS eth port 1 priority* *Priority* (for Port 1 in Web UI)	IP phone UI **Options->Network->VLAN ->Passthrough->Priority** Aastra Web UI **Advanced Settings->Network ->VLAN->Port 1** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Specifies the priority value used for passing VLAN packets through to a PC via Port 1. **Note:** Not available on 9112i.
Format	Integer
Default Value	0
Range	0 to 7
Example	QoS eth port 1 priority: 3

Parameter – *vlan id port 1* *VLAN id* (for Port 1in Web UI)	IP phone UI **Options->Network->VLAN ->Passthrough->VLAN ID** Aastra Web UI **Advanced Settings->Network ->VLAN->Port 1** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Specifies the VLAN ID used to pass packets through to a PC via Port 1. **Note:** Not available on 9112i.
Format	Integer
Default Value	1
Range	1 to 4094
Example	vlan id port 1: 3

Network Address Translation (NAT) Settings

Parameter – *sip nat ip* *NAT IP* (in Web UI)	IP phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Network** Configuration Files **aastra.cfg, <mac>.cfg**
Description	IP address the network device that enforces NAT.
Format	IP Address
Default Value	0.0.0.0
Range	Not Applicable
Example	sip nat ip: 192.245.2.1

Parameter – *sip nat port* *NAT Port* (in Web UI)	IP phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Network** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Port number of the network device that enforces NAT.
Format	Integer
Default Value	0
Range	Not Applicable
Example	sip nat port: 5060

Parameter – *sip nortel nat support* *Nortel NAT Traversal Enabled* (in Web UI)	IP phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Network** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the phone to operate while connected to a network device that enforces NAT.
Format	Integer
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	sip nortel nat support: 1

Parameter – *sip nortel nat timer* *Nortel NAT Timer* (in Web UI)	Aastra Web UI **Advanced Settings->Network** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The interval, in seconds, that the phone sends SIP ping requests to the Nortel proxy.
Format	Integer
Default Value	30
Range	0 to 2147483647
Example	sip nortel nat timer: 60

Time Server Settings

Parameter – *time server disabled* *NTP Time Servers* (in Web UI)	Aastra Web UI **Advanced Settings->Network** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This parameter enables or disables the time server. This parameter affects the time server1, time server2, and time server3 parameters. Use "0" to enable time server and "1" to disable time server.
Format	Integer
Default Value	0 (enabled)
Range	0 (enabled) 1 (disabled)
Example	time server disabled: 0

Parameter – *time server1* *Time Server 1* (in Web UI)	Aastra Web UI **Advanced Settings->Network** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The primary time server's IP address or qualified domain name. If the time server is enabled, the value for time server1 will be used to request the time from.
Format	IP address or qualified domain name
Default Value	0.0.0.0
Range	Not Applicable
Example	time server1: 192.168.0.5

Parameter – *time server2* *Time Server 2* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Network** **aastra.cfg, <mac>.cfg**
Description	The secondary time server's IP address or qualified domain name. If the time server is enabled, and the primary time server is not configured or cannot be accessed the value for time server2 will be used to request the time from.	
Format	IP address or qualified domain name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	time server2: 192.168.0.5	

Parameter – *time server3* *Time Server 3* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Network** **aastra.cfg, <mac>.cfg**
Description	The tertiary time server's IP address or qualified domain name. If the time server is enabled, and the primary and secondary time servers are not configured or cannot be accessed the value for time server3 will be used to request the time from.	
Format	IP address or qualified domain name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	time server3: 192.168.0.5	

Parameter – *date format*	IP phone UI **Options->Time and Date** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This parameter allows the user to change the date to various formats.
Format	Integer
Default Value	0
Range	0-7 Following table shows the format for the corresponding *date format* values: 0: WWW MMM DD 1: DD-MMM-YY 2: YYYY-MM-DD 3: DD/MM/YYYY 4: DD/MM/YY 5: DD-MM-YY 6: MM/DD/YY 7: MMM DD
Example	date format: 7

Parameter – *time format*	IP phone UI **Options->Time and Date** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This parameter changes the time to 12 hour or 24 hour format. Use “0” for the 12 hour format and “1” for the 24 hour format.
Format	Integer
Default Value	0
Range	0 or 1
Example	time format: 0

SIP Local Dial Plan Settings

(480i/480i CT only)

Parameter – *displayName1* *Idle Display Name 1* (in Web UI)	Aastra Web UI Configuration Files	**Basic Settings->Preferences** **aastra.cfg, <mac>.cfg**
Description	The name displayed on the idle screen rather than the user name and phone number.	
Format	Alphanumeric characters	
Default Value	Not Applicable	
Range	Up to 63 characters	
Example	displayName1: SIPphone	

(480i/480i CT only)

Parameter – *displayName2* *Idle Display Name 2* (in Web UI)	Aastra Web UI Configuration Files	**Basic Settings->Preferences** **aastra.cfg, <mac>.cfg**
Description	The name displayed on the idle screen rather than the user name and phone number.	
Format	Alphanumeric characters	
Default Value	Not Applicable	
Range	Up to 63 characters	
Example	displayName2: MYphone	

Parameter – *sip dial plan* *Local Dial Plan* (in Web UI)	Aastra Web UI Configuration Files	**Basic Settings->Preferences** **aastra.cfg, <mac>.cfg**
Description	A dial plan describes the number and pattern of digits that a user dials to reach a particular telephone number. The SIP local dial plan is as follows: **Symbol** — **Description** 0, 1, 2, 3, 4, 5, 6, 7, 8, 9 — Digit symbol X — Match any digit symbol (wildcard) *, #, . — Other keypad symbol | — Expression inclusive OR + — 0 or more of the preceding digit symbol or [] expression [] — Symbol inclusive OR - — Used only with [], represent a range pf acceptable symbols; For example, [2-8] **Note**: In the configuration files, you must enter the value using quotes. See the example below.	
Format	Alphanumeric characters	
Default Value	X+#|XX+*	
Range	Not Applicable	
Example	sip dial plan: "X+#|XXX+*"	

Parameter – *sip dial plan terminator* *Send Dial Plan Terminator* (in Web UI)	Aastra Web UI Configuration Files	**Basic Settings->Preferences** **aastra.cfg, <mac>.cfg**
Description	The IP phone allows the configuration of a dial plan terminator. When you configure the IP phone to use a dial plan terminator or timeout (such as the pound symbol (#)) the phone waits 4 or 5 seconds after you pick up the handset or press a key to make a call.	
Format	Boolean	
Default Value	0	
Range	"0" - Disable "1" - Enabled	
Example	sip dial plan terminator: 1	

Parameter – *sip digit timeout* *Digit Timeout* (in Web UI)	Aastra Web UI Configuration Files	**Basic Settings->Preferences** **aastra.cfg, <mac>.cfg**
Description	Represents the time, in seconds, to configure the timeout between consecutive key presses.	
Format	Integer	
Default Value	4	
Range	Not Applicable	
Example	sip digit timeout: 6	

SIP Registration Retry Timer Setting

Parameter – *sip registration retry timer* *Registration Retry Timer* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Global SI ->Advanced SIP Settings** **aastra.cfg, <mac>.cfg**
Description	Specifies the time, in seconds, that the phone waits between registration attempts when a registration is rejected by the registrar.	
Format	Integer	
Default Value	1800 (30 minutes)	
Range	30 to 1800	
Example	sip registration retry timer: 30	

SIP Basic, Global Settings

Parameter – *sip screen name* *Screen Name* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Global SIP->Basic SIP Settings** **aastra.cfg, <mac>.cfg**
Description	Used to display text on the screen of the phone. You may want to set this parameter to display the phone user's name.	
Format	Text	
Default Value	Not Applicable	
Range	Up to 20 alphanumeric characters	
Example	sip screen name: Joe Smith	

Parameter – *sip user name* *Phone Number* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Global SIP->Basic SIP Settings** **aastra.cfg, <mac>.cfg**
Description	Used in the name field of the *SIP URI* for the IP phone and for registering the IP phone at the registrar.	
Format	Text	
Default Value	Not Applicable	
Range	Up to 20 alphanumeric characters	
Example	sip user name: 1010	

Parameter – *sip display name* *Caller ID* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Global SIP->Basic SIP Settings** **aastra.cfg, <mac>.cfg**
Description	Used in the display name field of the *From* SIP header field. Some IP PBX systems use this as the caller's ID and some may overwrite this with the string that is set at the PBX system.	
Format	Text	
Default Value	Not Applicable	
Range	Up to 20 alphanumeric characters	
Example	sip display name: Joe Smith	

Parameter – *sip mode* *Line mode* (in Web UI)	Aastra Web UI: **Advanced Settings->Global SIP-> Basic SIP Settings** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Allows you to configure the mode of the line. Applicable values are: • **Generic** - Normal line • **BroadSoft SCA** - Shared Call/Line Appearances (SCA) line for BroadWorks network (call activity can go to more than one phone) • **Nortel** - Conference line for Nortel Networks (private - all call activity goes to one phone) • **BLA** - Bridged Line Appearance (BLA) line.
Format	Integer
Default Value	0
Range	Valid values are: 0 - Generic 1 - BroadSoft SCA 2 - Nortel 3 - BLA
Example	sip mode: 2

Parameter – *sip bla number* *BLA Number* (in Web UI)	Aastra Web UI: **Advanced Settings->Global SIP-> Basic SIP Settings** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Allows you to assign a phone number that is shared across all IP phones.
Format	Integer
Default Value	Not Applicable
Range	Not Applicable
Example	sip bla number: 1010

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Parameter – *sip proxy ip* *Proxy Server* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Global SIP-> Basic SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The IP address of the SIP proxy server for which the IP phone uses to send all SIP requests. A SIP proxy is a server that initiates and forwards requests generated by the IP phone to the targeted user.
Format	IP address or fully qualified Domain Name
Default Value	0.0.0.0
Range	Not applicable
Example	sip proxy ip: 192.168.0.101

Parameter – *sip proxy port* *Proxy Port* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Global SIP-> Basic SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The proxy server's port number
Format	Integer
Default Value	0
Range	Not Applicable
Example	sip proxy port: 5060

Parameter – *sip registrar ip* *Registrar Server* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Global SIP->Basic SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The address of the registrar for which the IP phone uses to send *REGISTER* requests. A SIP registrar is a server that maintains the location information of the IP phone. A global value of 0.0.0.0 disables registration. However, the phone is still active and you can dial using username@ip address of the phone. If the Registrar IP address is set to 0.0.0.0 for a per-line basis (i.e, line 1, line 2, etc.), then the register request is not sent, the "No Service" message does not display, and the message waiting indicator (MWI) does not come on.
Format	IP address or fully qualified Domain Name
Default Value	0.0.0.0
Range	Not Applicable
Example	sip registrar ip: 192.168.0.101

Parameter – *sip registrar port* *Registrar Port* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Global SIP->Basic SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The registrar's port number
Format	Integer
Default Value	0
Range	Not Applicable
Example	sip registrar port: 5060

Parameter – *sip registration period* *Registration Period* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP->Basic SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The requested registration period, in seconds, from the registrar.
Format	Integer
Default Value	0
Range	0 to 2147483647
Example	sip registration period: 3600

Parameter – *sip auth name*	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Global SIP->Basic SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Used in the username field of the Authorization header field of the *SIP REGISTER* request.
Format	Text
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip auth name: 5553456

Parameter – *sip password* *Password* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Global SIP->Basic SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg** **Note:** For the 9112i, this parameter is both a global and per-line setting.
Description	The password that will be used to register at the registrar.
Format	Text
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip password: 12345

Parameter – *sip outbound proxy* *outbound proxy server* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->Advanced SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the address of the outbound proxy server. All SIP messages originating from the phone are sent to this server. For example, if you have a Session Border Controller in your network, then you would normally set its address here.
Format	IP Address or fully qualified Domain Name
Default Value	0.0.0.0
Range	Not Applicable
Example	sip outbound proxy: 10.42.23.13

Parameter – *sip outbound proxy port* *outbound proxy port* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->Advanced SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The proxy port on the proxy server to which the IP phone sends all SIP messages.
Format	Integer
Default Value	0
Range	Not Applicable
Example	sip outbound proxy port: 5060

SIP Basic, Per-Line Settings

The following parameters are SIP per-line settings. The value of "*N*" is 1 - 9 for 480i, 480i CT, and 9133i. The value of "N" is 1 for 9112i.

Parameter – *sip lineN screen name* *Screen Name* (in Web UI)	Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files**aastra.cfg, <mac>.cfg**
Description	Used to display text on the screen of the phone. You may want to set this parameter to display the phone user's name.
Format	Text
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip line1 screen name: Joe Smith

Parameter – *sip lineN user name* *Phone Number* (in Web UI)	Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files**aastra.cfg, <mac>.cfg**
Description	Used in the name field of the *SIP URI* for the IP phone and for registering the IP phone at the registrar. **Note:** When configuring per-line BLA on an ININ server, the username must be incremented as shown in the example for the "sip lineN bla number" parameter on page 143.
Format	Text
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip line1 user name: 1010

Parameter – *sip lineN display name* *Caller ID* (in Web UI)	Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files**aastra.cfg, <mac>.cfg**
Description	Used in the display name field of the *From* SIP header field. Some IP PBX systems use this as the caller's ID and some may overwrite this with the string that is set at the PBX system.
Format	Text
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip line1 display name: Joe Smith

Parameter – *sip lineN mode* *Line Mode* (in Web UI)	Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files**aastra.cfg, <mac>.cfg**
Description	Allows you to configure the mode of the line. Applicable values are: • **Generic** - Normal line • **BroadSoft SCA** - Shared Call/Line Appearances (SCA) line for BroadWorks network (call activity can go to more than one phone) • **Nortel** - Conference line for Nortel Networks (private - all call activity goes to one phone) • **BLA** - Bridged Line Appearance (BLA) line. **Note:** If the softkeys on the 480i/480i CT or the programmable keys on the 9133i are set as line keys, and you configure that line key for BLA, the key is configured to use BLA.
Format	Integer
Default Value	0
Range	Valid values are: 0 - Generic 1 - BroadSoft SCA 2 - Nortel 3 - BLA
Example	sip line1 mode: 2

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Parameter – *sip lineN proxy ip* *Proxy Server* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files**aastra.cfg, <mac>.cfg**
Description	The IP address of the SIP proxy server for which the IP phone uses to send all SIP requests. A SIP proxy is a server that initiates and forwards requests generated by the IP phone to the targeted user.
Format	IP address or fully qualified Domain Name
Default Value	0.0.0.0
Range	Not applicable
Example	sip line1 proxy ip: 192.168.0.101

Parameter – *sip lineN proxy port* *Proxy Port* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files**aastra.cfg, <mac>.cfg**
Description	The proxy server's port number
Format	Integer
Default Value	0
Range	Not Applicable
Example	sip line1 proxy port: 5060

Parameter – *sip lineN registrar ip* *Registrar Server* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The address of the registrar for which the IP phone uses to send *REGISTER* requests. A SIP registrar is a server that maintains the location information of the IP phone. A global value of 0.0.0.0 disables registration. However, the phone is still active and you can dial using username@ip address of the phone. If the Registrar IP address is set to 0.0.0.0 for a per-line basis (i.e, line 1, line 2, etc.), then the register request is not sent, the "No Service" message does not display, and the message waiting indicator (MWI) does not come on.
Format	IP address or fully qualified Domain Name
Default Value	0.0.0.0
Range	Not Applicable
Example	sip line1 registrar ip: 192.168.0.101

Parameter – *sip lineN registrar port* *Registrar Port* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The registrar's port number
Format	Integer
Default Value	0
Range	Not Applicable
Example	sip line1 registrar port: 5060

Parameter – *sip lineN registration period* *Registration Period* (in Web UI)	Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files**aastra.cfg, <mac>.cfg**
Description	The requested registration period, in seconds, from the registrar.
Format	Integer
Default Value	0
Range	0 to 2147483647
Example	sip line1 registration period: 3600

Parameter – *sip lineN auth name* *Authentication Name* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files**aastra.cfg, <mac>.cfg**
Description	Used in the username field of the Authorization header field of the *SIP REGISTER* request.
Format	Text
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip line1 auth name: 5553456

Parameter – *sip lineN password* *Password* (in Web UI)	IP Phone UI **Options->SIP Settings** Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files**aastra.cfg, <mac>.cfg** **Note:** For the 9112i, this parameter is both a global and per-line setting.
Description	The password that will be used to register at the registrar.
Format	Text
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip line1 password: 12345

Parameter – *sip lineN bla number* *BLA Number*r (in Web UI)	Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Allows you to assign a phone number that is shared on specific lines on the IP phone. **For Sylantro Server:** When configuring the BLA feature on a Sylantro server, the value set for the **sip lineN bla number** parameter shall be the same value set for the **sip lineN user name** parameter for all the phones in the group. For example, if sip lineN user name is 1010, you would configure BLA on a per-line basis for the Sylantro server as follows: sip line1 user name: 1010(# for all the phones) sip line1 bla number: 1010 **For ININ Server:** When configuring the BLA feature on an ININ server, the value set for the **sip lineN bla number** parameter shall be the same value set for the **sip lineN user name** parameter without the incremented digit added to the phone #. For example, if the sip lineN user name for the first phone is 10101, and the sip lineN user name for the second phone is 10102, etc. you would configure BLA on a per-line basis for the ININ server as follows: sip line1 user name: 10101 (# for phone 1 with) sip line1 bla number: 1010 appearance of phone 3) sip line1 user name: 10102 (# for phone 2 with) sip line1 bla number: 1010 appearance of phone 3) sip line1 user name: 1010 (# for phone 3) sip line1 bla number: 1010 **Note:** The original phone number which has the bridged line appearance on other phones, will have the "sip lineN user name" parameter the same as the "sip lineN bla number" (1010 in the above example on Phone 3).
Format	Integer
Default Value	Not Applicable
Range	Not Applicable
Example	**Sylantro Server:** sip line1 bla number: 1010 **ININ Server:** sip line 1 bla number: 1010

Parameter – ***sip lineN outbound proxy*** *Outbound Proxy Server* (in Web UI)	Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the address of the outbound proxy server. All SIP messages originating from the phone are sent to this server. For example, if you have a Session Border Controller in your network, then you would normally set its address here.
Format	IP Address or fully qualified Domain Name
Default Value	0.0.0.0
Range	Not Applicable
Example	sip outbound proxy: 10.42.23.13

Parameter – ***sip lineN outbound proxy port*** *Outbound Proxy Port* (in Web UI)	Aastra Web UI **Advanced Settings->Line 1 thru 9 (480i/480i CT/9133i)** Aastra Web UI **Advanced Settings->Global SIP (9112i)** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The proxy port on the proxy server to which the IP phone sends all SIP messages.
Format	Integer
Default Value	0
Range	Not Applicable
Example	sip outbound proxy port: 5060

Advanced SIP Settings

Parameter – *sip explicit mwi subscription* *Explicit MWI Subscription* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Global SIP ->Advanced SIP Settings** **aastra.cfg, <mac>.cfg**
Description	If the IP phone has a message waiting subscription with the Service Provider, a Message Waiting Indicator (MWI) (LED or display icon) tells the user there is a message on the IP Phone. You can enable and disable MWI by setting this parameter to the following: • "**0**" to disable • "**1**" to enable	
Format	Boolean	
Default Value	0	
Range	0 (disable) 1 (enable)	
Example	sip explicit mwi subscription: 1	

Parameter – *sip session timer* *Session Timer* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Global SIP ->Advanced SIP Settings** **aastra.cfg, <mac>.cfg**
Description	The time, in seconds, that the IP phone uses to send periodic re-*INVITE* requests to keep a session alive. The proxy uses these re-*INVITE* requests to maintain the status' of the connected sessions. See RFC4028 for details.	
Format	Integer	
Default Value	0	
Range	Not Applicable	
Example	sip session timer: 30	

Parameter – *sip T1 timer* *T1 Timer* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->Advanced SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This timer is a SIP transaction layer timer defined in RFC 3261. Timer 1 is an estimate of the round-trip time (RTT). Default is 500 msec.
Format	Integer
Default Value	500
Range	Not Applicable
Example	sip T1 timer: 600

Parameter – *sip T2 timer* *T2 Timer* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->Advanced SIP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This timer is a SIP transaction layer timer defined in RFC 3261. Timer 2 represents the amount of time a non-INVITE server transaction takes to respond to a request. Defaults is 4 seconds.
Format	Integer
Default Value	4
Range	Not Applicable
Example	sip T2 timer: 8

Parameter – *sip transaction timer* *Transaction Timer* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Global SIP ->Advanced SIP Settings** **aastra.cfg, <mac>.cfg**
Description	The amount of time, in milliseconds that the phone allows the callserver (registrar/proxy) to respond to SIP messages that it sends. If the phone does not receive a response in the amount of time desginated for this parameter, the phone assumes the message has timed out.	
Format	Integer	
Default Value	4000	
Range	4000 to 64000	
Example	sip transaction timer: 6000	

Parameter – *sip transport protocol* *Transport Protocol* (in Web UI)	Aastra Web UI Configuration Files	**Advanced Settings->Global SIP ->Advanced SIP Settings** **aastra.cfg, <mac>.cfg**
Description	The protocol that the RTP port on the IP phone uses to send out RTP packets.	
Format	Boolean	
Default Value	1 (UDP)	
Range	Valid values are: 0 - Both 1 - UDP 2 - TCP	
Example	sip transport protocol: 2	

RTP, Codec, DTMF Global Settings

Parameter – *sip rtp port* *RTP Port Base* (in IP Phone UI) *RTP Port* (in Web UI)	IP Phone UI **Options->SIP Settings ->RTP Port Base** Aastra Web UI **Advanced Settings->Global SIP ->RTP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Indicates the port through which the RTP packets are sent. The RTP port is used for sending DTMF tones and for the audio stream. Your network administrator may close some ports for security reasons. You may want to use this parameter to send RTP data using a different port.
Format	Integer
Default Value	3000
Range	Not Applicable
Example	sip rtp port: 5000

Parameter – *sip use basic codecs* *Basic Codecs* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->RTP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables basic codecs. Enabling this parameter allows the IP phone to use the basic Codecs when sending/receiving RTP packets.
Format	Boolean
Default Value	0
Range	0 - Disable 1 - Enable
Example	sip use basic codecs: 1

Parameter – *sip out-of-band dtmf* *Force RFC2833 Out-of-Band DTMF* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->RTP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables out-of-band DTMF. Enabling this parameter forces the IP phone to use out-of-band DTMF according to RFC2833.
Format	Boolean
Default Value	1
Range	0 - Disable 1 - Enable
Example	sip out-of-band dtmf: 0

Parameter – *sip customized codec* *Customized Codec Preference List* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->RTP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Specifies a customized Codec preference list which allows you to use the preferred Codecs for this IP phone.
Format	Comma-separated list of semicolon-separated values
Default Value	Not Applicable
Range	Valid values for the syntax are: **payload** 0 for G.711 μ-Law 8 for G.711 a-Law 18 for G.729a **ptime** (in milliseconds) 5, 10, 15, 20........90 **silsupp** on, off
Example	sip customized codec: payload=8;ptime=10;silsupp=on,payload=0;ptime=10; silsupp=off

Parameter – *sip dtmf method* *DTMF Method* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->RTP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Sets the Dual-tone multifrequency (DTMF) method to use on the IP phone.
Format	Boolean
Default Value	0 (RTP)
Range	0 (RTP) 1 (SIP INFO) 2 (BOTH)
Example	sip dtmf method: 1

DTMF Per-Line Settings

Parameter – *sip lineN dtmf method* *DTMF Method* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->RTP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Sets the Dual-tone multifrequency (DTMF) method to use on the IP phone for a specific line.
Format	Integer
Default Value	0 (RTP)
Range	0 (RTP) 1 (SIP INFO) 2 (BOTH)
Example	sip line1 dtmf method: 1

Silence Suppression Settings

Parameter – *sip silence suppression* *Silence Suppression* (in Web UI)	Aastra Web UI **Advanced Settings->Global SIP ->RTP Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Silence suppression is enabled by default on the IP phones. The phone negotiates whether or not to use silence suppression. Disabling this feature forces the phone to ignore any negotiated value.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	sip silence suppression: 0

Voicemail Settings

Parameter – *sip lineN vmail* **Note:** The value of "*N*" is 1 - 9 for 480i, 480i CT, and 9133i. The value of "N" is 1 for 9112i.	Configuration Files**aastra.cfg, <mac>.cfg**
Description	Use this parameter in the *<mac>.cfg* file to configure the phone to dial a specific number to access an existing voicemail account on a Service Provider's server. The user then follows the voicemail instructions for listening to voicemails. **Note:** The phone must have a registered voicemail account from a server for this feature to be enabled. When no registered voicemail accounts are registered to the phone, the display shows "List Empty". The phone displays up to 99 voicemails for an account even if the number of voicemails exceeds the limit. Registered account numbers/URIs that exceed the length of the screen, either with or without the voicemail icon and the message count, are truncated with an ellipse character at the end of the number/URI string.
Format	Integer
Default Value	Not Applicable
Range	0 to 99
Example	sip line1 vmail: *97 **Note:** In the above example, the user would dial *97 to access the voicemail account.

Directory Settings

Parameter – *directory 1* *Directory List* (in Web UI)	Aastra Web UI **Operation->Directory** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The name of a directory list that you can download from the configuration server.
Format	Alphanumeric characters
Default Value	Not Applicable
Range	Not Applicable
Example	directory 1: companylist.csv

Parameter – *directory 2* *Directory List* (in Web UI)	Aastra Web UI **Operation->Directory** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The name of a directory list that you can download from the configuration server.
Format	Alphanumeric characters
Default Value	Not Applicable
Range	Not Applicable
Example	directory 2: personallist.csv

Parameter – *directory disabled*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the Directory on the IP phone. If this parameter is set to 0, users can access the Directory List via the IP phone UI. If this parameter is set to 1, the Directory List does not display on the IP phone and the Directory key is disabled. On the 480i and 480i CT the "Directory" option is also removed from the "Services" menu.
Format	Boolean
Default Value	0 (false)
Range	0 (false), 1 (true)
Example	directory disabled: 1

Callers List Settings

Parameter – *callers list disabled*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the Callers List. If this parame.ter is set to 0, the Callers List can be accessed by all users. If this parameter is set to 1, the IP phone does not save any caller information to the Caller List. For 480i and 480i CT phones, the "Caller List" option on the IP phone is removed from the Services menu, and the Caller List key is ignored if pressed by the user
Format	Boolean
Default Value	0 (false)
Range	0 (false), 1 (true)
Example	callers list disabled: 1

Call Forward Settings

Parameter – *call forward disabled*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the ability to configure Call Forwarding. If this parameter is set to 0, a user and administrator can configure Call Fowarding via the Aastra Web UI and the IP Phone UI using the "Call Foward" options. If this parameter is set to 1, all "Call Forward" options are removed from the Aastra Web UI and the IP Phone UI, preventing the ability to configure Call Fowarding.
Format	Boolean
Default Value	0 (false)
Range	0 (false), 1 (true)
Example	callers list disabled: 1

Missed Calls Indicator Settings

Parameter – *missed calls indicator disabled*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the Missed Calls Indicator. If the "missed calls indicator disabled" parame.ter is set to 0, the indicator increments as unanswered calls come into the IP phone. If the "missed calls indicator disabled" parameter is set to 1, the indicator is disabled and will NOT increment as unanswered calls come into the IP phone.
Format	Boolean
Default Value	0 (flase)
Range	0 (false), 1 (true)
Example	missed calls indicator disabled: 1

XML Settings

Parameter – *xml application URI* *XML Application URI* (in Web UI)	Aastra Web UI **Operation->Softkeys and XML->Services** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the XML application you are loading into the IP phone configuration.
Format	HTTP server path or fully qualified Domain Name
Default Value	Not Applicable
Range	Not Applicable
Example	xml application URI: http://172.16.96.63/aastra/internet.php

Parameter – *xml application title* *XML Application Title* (in Web UI)	Aastra Web UI **Operation->Softkeys and XML->Services** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This parameter allows you to rename the XML application in the IP phone UI (Services->4. Custom Feature). By default, when you load an XML application to the IP phone, the XML application title is called "Custom Feature". The "xml application title" parameter allows you to change that title. For example, if you are loading a traffic report XML application, you could change this parameter title to "Traffic Reports", and that title will display in the IP phone UI as Services->4. Traffic Reports.
Format	Alphanumeric characters
Default Value	Not Applicable
Range	Not Applicable
Example	xml application title: Traffic Reports

Parameter – *xml application post list* *XML Push Server List* (in Web UI)	Aastra Web UI **Advanced Settings-> Configuration Server** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The HTTP server that is pushing XML applications to the IP phone.
Format	IP address in dotted decimal format and/or Domain name address
Default Value	Not Applicable
Range	Not Applicable
Example	xml application post list: 10.50.10.53, dhcp10-53.ana.aastra.com

Ring Tone and Tone Set Global Settings

Parameter – *ring tone* *Global Ring Tone* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences->Ring Tones** IP Phone UI **Options->Tones->Set Ring Tone** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Globally sets the type of ring tone on the IP phone. Ring tone can be set to one of six distinct rings.
Format	Integer
Default Value	Aastra Web UI : Tone 1 IP Phone UI: Tone 1 Configuration Files: 0 (Tone 1)
Range	**Aastra Web UI & IP Phone UI** Tone 1 Tone 2 Tone 3 Tone 4 Tone 5 Silent **Configuration Files** 0 (Tone 1) 1 (Tone 2) 2 (Tone 3) 3 (Tone 4) 4 (Tone 5) 5 (Silent)
Example	ring tone: 3

Parameter – *tone set* *Tone Set* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences->Ring Tones** IP Phone UI **Options->Tones->Tone Set** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Globally sets a tone set for a specific country.
Format	Text
Default Value	US
Range	Australia Europe (generic tones) France Germany Italy UK US (also used in Canada)
Example	tone set: Germany

Ring Tone Per-Line Settings

Parameter – *lineN ring tone* *Line N* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences-> Ring Tones** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Sets the type of ring tone on the IP phone on a per-line basis. Ring tone can be set to one of six distinct rings.
Format	Integer
Default Value	Aastra Web UI : Global Configuration Files: -1 (Global)
Range	**Aastra Web UI** Global Tone 1 Tone 2 Tone 3 Tone 4 Tone 5 Silent **Configuration Files** -1 (Global) 0 (Tone 1) 1 (Tone 2) 2 (Tone 3) 3 (Tone 4) 4 (Tone 5) 5 (Silent)
Example	line1 ring tone 3

Stuttered Dial Tone Setting

Parameter – *stutter disabled* *Stuttered Dial Tone* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences-> General** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enable or disables the playing of a stuttered dial tone when there is a message waiting on the IP phone.
Format	Boolean
Default Value	0 (false)
Range	0 (false) 1 (true)
Example	stuttered disabled: 1

Call Waiting Tone Setting

Parameter – *call waiting tone* *Play Call Waiting Tone* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences->General** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enable or disables the playing of a call waiting tone when a caller is on an active call and a new call comes into the phone.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disable) 1 (enabled)
Example	call waiting tone: 0

Priority Alert Settings

Parameter – *prioity alerting enabled* *Enable Priority Alerting* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences->Priority Alerting Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables and disables distinctive ringing on the IP phone for incoming calls and call-waiting calls.
Format	Boolean
Default Value	1 (true)
Range	0 (false) 1 (true)
Example	priority alerting enabled: 0

For Sylantro Server only

Parameter – *alert group* *alert group* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences->Priority Alerting Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	When an "alert group" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone.
Format	Integer
Default Value	0 Normal ringing
Range	0 Normal ringing (default) 1 Bellcore-dr2 2 Bellcore-dr3 3 Bellcore-dr4 4 Bellcore-dr5 5 Silent
Example	alert group: 4

Parameter – *alert external* *alert external* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences-> Priority Alerting Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	When an "alert external" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone.
Format	Integer
Default Value	0 Normal ringing
Range	0 Normal ringing (default) 1 Bellcore-dr2 2 Bellcore-dr3 3 Bellcore-dr4 4 Bellcore-dr5 5 Silent
Example	alert external: 4

Parameter – *alert internal* *alert internal* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences-> Priority Alerting Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	When an "alert-internal" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone.
Format	Integer
Default Value	0 Normal ringing
Range	0 Normal ringing (default) 1 Bellcore-dr2 2 Bellcore-dr3 3 Bellcore-dr4 4 Bellcore-dr5 5 Silent
Example	alert internal: 4

Parameter – *alert emergency* *alert emergency* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences->Priority Alerting Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	When an "alert emergency" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone.
Format	Integer
Default Value	0 Normal ringing
Range	0 Normal ringing (default) 1 Bellcore-dr2 2 Bellcore-dr3 3 Bellcore-dr4 4 Bellcore-dr5 5 Silent
Example	alert emergency: 4

Parameter – *alert priority* *alert priority* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences->Priority Alerting Settings** Configuration Files **aastra.cfg, <mac>.cfg**
Description	When an "alert priority" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone.
Format	Integer
Default Value	0 Normal ringing
Range	0 Normal ringing (default) 1 Bellcore-dr2 2 Bellcore-dr3 3 Bellcore-dr4 4 Bellcore-dr5 5 Silent
Example	alert priority: 4

Language Settings

Parameter – *language*	IP Phone UI Configuration Files	**Options->Language** **aastra.cfg, <mac>.cfg**
Description	Used to set the language that the phone uses to display messages, etc. Valid values for 480i/9112i/9133i are: 0 (English) 1 (French) 2 (Spanish) 3 (German) 4 (Italian) Valid values for 480i CT are: 0 (English) 1 (French) 2 (Spanish)	
Format	Integer	
Default Value	0	
Range	0 to 4 (for 480i/9112i/9133i) 0 to 2 (for 480i CT)	
Example	language: 2	

Suppress DTMF Playback Settings

Parameter – *suppress dtmf playback* *Suppress DTMF Playback* (in Web UI)	Aastra Web UI Configuration Files	**Basic Settings->Preferences** **aastra.cfg, <mac>.cfg**
Description	Enables and disables suppression of DTMF playback when a number is dialed from the softkeys or programmable keys. When you disable the suppression of DTMF playback and you press a softkey or programmable key, the IP phone dials the stored number and displays each digit as dialed in the LCD window. When you enable the suppression of DTMF playback, the IP phone dials the stored number and displays the entire number immediately in the LCD window, allowing the call to be dialed faster.	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	suppress dtmf playback: 1	

Intercom and Auto-Answer Settings

Outgoing Intercom Settings

(480i/480i CT only)

Parameter – *sip intercom type* *Type* (in Web UI)	Aastra Web UI Configuration Files	**Basic Settings->Preferences-> Outgoing Intercom Settings** **aastra.cfg, <mac>.cfg**
Description	Determines whether the IP phone or the server is responsible for notifying the recipient that an Intercom call is being placed.	
Format	Integer	
Default Value	**For Aastra Web UI:** Off (480i, 480i CT and handsets) **For Configuration Files:** 3 Off (480i, 480i CT and handsets)	
Range	**For Aastra Web UI:** Phone-Side Server-Side Off (480i and 480i CT handsets) **For Configuration Files:** 1 - Phone-Side 2 - Server-Side 3 - Off	
Example	sip intercom type: 1	

Parameter – *sip intercom prefix code* *Prefix Code* (in Web UI)	Aastra Web UI Configuration Files	**Basic Settings->Preferences-> Outgoing Intercom Settings** **aastra.cfg, <mac>.cfg**
Description	The prefix to add to the phone number for server-side outgoing Intercom calls. This parameter is required for all server-side Intercom calls. **Note:** The example below shows *96 for the prefix code which is used for Sylantro servers.	
Format	String	
Default Value	N/A	
Range	N/A	
Example	sip intercom prefix code: *96	

Parameter – *sip intercom line* *Line* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences->Outgoing Intercom Settings** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Specifies the line for which the IP phone uses the configuration from, when making the Intercom call. The IP phone uses the first available line for physically making the call but uses the configuration from the line you set for this parameter. **Note:** The "*sip intercom type*" parameter must be set with the **Server-Side** option to enable the "*sip intercom line*" parameter.
Format	Integer
Default Value	1
Range	Line 1 through 9
Example	sip intercom line: 1

Incoming Intercom Settings

Parameter – *sip intercom mute mic* *Microphone Mute* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences->Incoming Intercom Settings** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Enables or disables the microphone on the IP phone for Intercom calls made by the originating caller.
Format	Integer
Default Value	1
Range	0 (false - microphone is not muted) 1 (true - microphone is muted)
Example	sip intercom mute mic: 1

Parameter – *sip allow auto answer* *Auto-Answer* (in Web UI)	Aastra Web UI Configuration Files	**Basic Settings->Preferences-> Incoming Intercom Settings** **aastra.cfg, <mac>.cfg**
Description	Enables or disables the IP phone to allow automatic answering for an Intercom call. If auto-answer is enabled on the IP phone, the phone plays a tone to alert the user before answering the intercom call. If auto-answer is disabled, the phone rejects the incoming intercom call and sends a busy signal to the caller.	
Format	Boolean	
Default Value	1 (true)	
Range	0 (false - do not allow auto-answer) 1 (true - allow auto-answer)	
Example	sip allow auto answer: 0	

Audio Transmit and Receive Gain Adjustment Settings

Parameter – *headset tx gain*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the increased (+db) or decreased (-db) amount of signal transmitted from the headset microphone to t he far-end party. The amount of Tx gain in the IP phone firmware has been reduced to avoid side-tone and echo on the local and far-end equipment. This parameter allows you to adjust the Tx gain settings to best suit your comfort level and deployment environment.
Format	Integer
Default Value	0
Range	-10 db to +10 db
Example	headset tx gain: -5

Parameter – *headset sidetone gain*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the is the increased (+db) or decreased (-db) amount of sidetone signal from the headset microphone to the headset speaker. The amount of sidetone gain in the IP phone firmware has been reduced to avoid side-tone and echo on the local and far-end equipment. This parameter allows you to adjust the sidetone gain settings to best suit your comfort level and deployment environment.
Format	Integer
Default Value	0
Range	-10 db to +10 db
Example	headset sidetone gain: -1

Parameter – *handset tx gain*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the increased (+db) or decreased (-db) amount of signal transmitted from the handset microphone to the far-end party. The amount of Tx gain in the IP phone firmware has been reduced to avoid side-tone and echo on the local and far-end equipment. This parameter allows you to adjust the Tx gain settings to best suit your comfort level and deployment environment.
Format	Integer
Default Value	0
Range	-10 db to +10 db
Example	handset tx gain: -5

Parameter – *handset sidetone gain*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the increased (+db) or decreased (-db) amount of sidetone signal from the handset microphone to the handset speaker. The amount of sidetone gain in the IP phone firmware has been reduced to avoid side-tone and echo on the local and far-end equipment. This parameter allows you to adjust the sidetone gain settings to best suit your comfort level and deployment environment.
Format	Integer
Default Value	0
Range	-10 db to +10 db
Example	handset sidetone gain: -1

Parameter – *handsfree tx gain*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the increased (+db) or decreased (-db) amount of signal transmitted from the base microphone to the far-end party. The amount of Tx gain in the IP phone firmware has been reduced to avoid side-tone and echo on the local and far-end equipment. This parameter allows you to adjust the Tx gain settings to best suit your comfort level and deployment environment. **Note:** The example below increases the speakerphone mic transmit gain by 10 db.
Format	Integer
Default Value	0
Range	-10 db to +10 db
Example	handsfree tx gain: 10

Parameter – *audio mode*	IP Phone UI **Options->Set Audio** Configuration Files **aastra.cfg, <mac>.cfg**
Description	Allows you to configure how the "handsfree" key on the IP phone operates.
Format	Integer
Default Value	0
Range	**0** **Speaker** - Calls can be made or received using the handset or handsfree speakerphone and can be switched between the two modes by pressing the / key. When on speaker, you can return to using the handset by placing the handset on the cradle and picking it up again. **1** **Headset** - Calls can be made or received using the headset. Calls can be switched between the headset and handset by pressing the / key. **2** **Speaker/headset** - Incoming calls are sent to the speakerphone . By pressing the / key, you can switch between the handsfree speakerphone, the headset, and the handset. **3** **Headset/speaker** - Incoming calls are sent to the headset. By pressing the / key, you can switch between the headset, the handsfree speakerphone, and the handset.
Example	audio mode: 2

Directed Call Pickup (BLF Call Interception) Settings

Parameter – *directed call pickup* *Directed Call Pickup* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences ->Directed Call Pickup Settings** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Enables or disables the use of "directed call pickup" feature.
Format	Boolean
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	directed call pickup: 1

Parameter – *play a ring splash* *Play a Ring Splash* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences ->Directed Call Pickup Settings** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Enables or disables the playing of a short "call waiting tone" when there is an incoming call on the BLF monitored extension. If the host tone is idle, the tone plays a "ring splash".
Format	Boolean
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	play a ring splash: 1

BLF Subscription Period Settings

Parameter – *sip blf subscription period* *BLF Subscription Period* (in Web UI)	Aastra Web UI: **Advanced Settings->Global SIP ->Advanced SIP Settings** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Specifies the time period, in seconds, that the IP phone resubscribes the BLF subscription service after a software/firmware upgrade or after a reboot of the IP phone.
Format	Integer
Default Value	3600
Range	120 (2 minutes minimum value)
Example	sip blf subscription period: 180

Hard Key Parameters

This section provides the hard key settings you can use to enable and disable the Redial, Conf, and Xfer keys on the IP phone.

Parameter – *redial disabled*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the Redial key on the IP phone. If this parameter is set to 0, the key is active and can be pressed by the user. If this parameter is set to 1, pressing the Redial key is ignored, and the dialed number is not saved to the "Redial List".
Format	Boolean
Default Value	0 (false)
Range	0 (false), 1 (true)
Example	redial disabled: 1

Parameter – *conference disabled*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the Conf key on the IP phone. If this parameter is set to 0, the key is active and can be pressed by the user. If this parameter is set to 1, pressing the Conf key is ignored.
Format	Boolean
Default Value	0 (false)
Range	0 (false), 1 (true)
Example	conference disabled: 1

Parameter – *call transfer disabled*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the Xfer key on the IP phone. If this parameter is set to 0, the key is active and can be pressed by the user. If this parameter is set to 1, pressing the Xfer key is ignored.
Format	Boolean
Default Value	0 (false)
Range	0 (false), 1 (true)
Example	call transfer disabled: 1

Parameter – *map redial key to* *Map Redial Key To* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences ->Key Mapping** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Sets the **Redial** key as a speedial key if a value is entered for this parameter. If you leave this parameter blank, the **Redial** key returns to its original functionality. **Note**: If you configure the **Redial** key for speeddialing on the 480i CT Base Station, the **Redial** key on the 480i CT handset retains its original functionality. The **Redial** key on the handset is not configured for speeddial.
Format	Integer
Default Value	N/A
Range	N/A
Example	map redial key to: 5551234

Parameter – *map conf key to* *Map Conf Key To* (in Web UI)	Aastra Web UI: **Basic Settings->Preferences ->Key Mapping** Configuration Files: **aastra.cfg, <mac>.cfg**
Description	Sets the **Conf** key as a speedial key if a value is entered for this parameter. If you leave this parameter blank, the **Conf** key returns to its original functionality. **Note**: If you configure the **Conf** key for speeddialing on the 480i CT Base Station, the **Conf** key on the 480i CT handset retains its original functionality. The **Conf** key on the handset is not configured for speeddial.
Format	Integer
Default Value	N/A
Range	N/A
Example	map conf key to: 5551267

Softkey/Programmable Key Parameters

This section provides the softkey and programmable key parameters you can configure on the IP phones. The following table provides the number of softkeys and programmable keys you can configure, and the number of lines available for each type of phone.

IP Phone Model	Softkeys Available	Programmable Keys Available	Feature Keys Available	Available Lines
480i	20	-	-	9
480i CT	20	-	-	9
480i CT Handset	-	-	15	9
9112i	-	2	-	1
9133i	-	7	-	9

Softkey Settings for 480i and 480i CT

The value of "*N*" for the following parameters is dependent on the number of softkeys available on the 480i/480i CT models. See the table above for applicable values.

Parameter – *softkeyN type* *Type* (in Web UI)	Aastra Web UI **Operation->Softkeys and XML** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The type of softkey to configure. Valid types are: • **line** - Indicates softkey is configured for line use. • **dnd** - Indicates softkey is configured for do not disturb on the phone. This option is "**do not disturb**" in the Aastra Web UI). • **speeddial** - Indicates softkey is configured for speeddial use • **blf** - Indicates softkey is configured for Busy Lamp Field (BLF) use. User can dial out on a BLF configured key. • **list** - Indicates softkey is configured for BLF list use. (This option is BLF\List in the Aastra Web UI). User can dial out on a BLF List configured key. • **xml** - Indicates the softkey is configured to accept an XML application for accessing customized XML services. • **flash** - Indicates the softkey is set to generate a flash event when it is pressed on the 480i and 480i CT, or a feature key is pressed on the 480i CT handset. **Note:** The IP phone generates flash events only when a call is connected and there is an active RTP stream (for example, when the call is not on hold). • **park** - Indicates the softkey is configured to park incoming calls when pressed. • **pickup** - Indicates the softkey is configured to pick up parked calls when pressed. • **empty** - Indicates the softkey is configured to force a blank entry on the IP phone display for a specific softkey. The soft keys are added in order (from softkey1 to softkey20) after any hard-coded keys have been added. If a particular soft key is not defined, it is ignored.
Format	Text

Default Value	none
Range	line dnd speeddial blf list ("BLF\List" in the Aastra Web UI) xml flash park pickup empty
Example	softkey1 type: line softkey2 type: speeddial

Parameter – *softkeyN label* *Label* (in Web UI)	Aastra Web UI **Operation->Softkeys and XML** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The text label that displays on the IP phone for the softkey. **Notes:** **1.** For the 480i and 480i CT phones, an icon appears beside the soft key label that indicates the status of the line. **2.** If the *softkeyN type* parameter is set to "**flash**", and no label value is entered for the *softkeyN label* parameter, the label of "**Flash**" is used.
Format	Text
Default Value	Not Applicable
Range	For line, blf types - Up to 9 characters. For speeddial type - Up to 11 characters.
Example	softkey1 label: “Line 9” softkey2 label: “info”

Parameter – *softkeyN value* *Value* (In WEb UI)	Aastra Web UI Configuration Files	**Operation->Softkeys and XML** **aastra.cfg, <mac>.cfg**
Description	This is the value you assign to the softkey. • For speedial type - Value is the phone number or extension to enter for the softkey. • For blf type - Value is the extension you want to monitor. • For Park, Pickup types - For valid values, see the "Park/Pickup Call Server Configuration Values" on page 66. For Park/Pickup examples, see "Model 480i/ 480 CT Examples" on page 68. **Note:** No values required for line, list, and xml types.	
Format	Integer	
Default Value	Not Applicable	
Range	N/A	
Example	softkey1 value: 9 softkey2 value: 411	

Parameter – *softkeyN line* *Line* (in Web UI)	Aastra Web UI Configuration Files	**Operation->Softkeys and XML** **aastra.cfg, <mac>.cfg**
Description	This is the line associated with the softkey you are configuring. The number of applicable lines available is dependent on the specific IP phone model.	
Format	Integer	
Default Value	1	
Range	1 through 9 (for 480i and 480i CT)	
Example	softkey1 line: 1 softkey2 line: 5	

Parameter – *softkeyN states*	Configuration Files **aastra.cfg, <mac>.cfg**

Description	Displays the status of the phone when a softkey is pressed. You can enter multiple values (**idle, connected, incoming, outgoing**) for the "softkeyN state" parameter.
	
	You must associate the softkeyN state parameter with a specific softkey. In the following example, the softkeyN states parameter is associated with softkey 12:
	`softkey12 type: speeddial`
	`softkey12 label: voicemail`
	`softkey12 value *89`
	`softkey12 states: outgoing`
	
	Note: The IP phone idle screen condenses the softkeys. So in the previous example, softkey 12 will appear in position 1 if no other softkeys are set. A softkey type of "empty" does not display on the idle screen at all.

Format	Text
Default Value	**For softkey types - Line, DND, speeddial, BLF, BLF List, XML, empty:** idle, connected, incoming, outgoing **For softkey type - Flash:** <blank> **For softkey type - Park:** connected **For softkey type - Pickup:** idle, outgoing
Range	Valid values are: **idle** The phone is not being used. **connected** The line currently being displayed is in an active call (or the call is on hold) **incoming** The phone is ringing. **outgoing** The user is dialing a number, or the far-end is ringing. **Note:** For softkey type, Pickup, values can be: idle, outgoing idle outgoing
Example	softkey1 states: idle incoming outgoing softkey2 states: connected

Programmable Key Settings for 9112i and 9133i

The value of "*N*" for the following parameters is dependent on the number of programmable keys available on the 9112i/9133i models. See the table on page 172 for the applicable values.

Parameter – *prgkeyN type* *Type* (in Web UI)	Aastra Web UI **Operation->Programmable Keys** Configuration Files **aastra.cfg, <mac>.cfg**
Description	The type of programmable key to configure. Valid types are: • **line** (9133i only) - Indicates softkey is configured for line use. • **speeddial** - Indicates programmable key is configured for speeddial use • **dnd** - Indicates programmable key is configured for do not disturb on the phone. This option is "**do not disturb**" in the Aastra Web UI). • **blf** (9133i only) - Indicates programmable key is configured for Busy Lamp Field (BLF) use. User can dial out on a BLF configured key. • **list** (9133i only; Aastra Web UI only) - Indicates programmable key is configured for BLF list use. User can dial out on a BLF List configured key. • **xml** - Indicates the programmable key is configured to accept an XML application for accessing customized XML services. • **flash** - Indicates the programmable key is set to generate a flash event when it is pressed on the 9112i and 9133i. **Note:** The IP phone generates flash events only when a call is connected and there is an active RTP stream (for example, when the call is not on hold). • **park** - Indicates the softkey is configured to park incoming calls when pressed. • **pickup** - Indicates the softkey is configured to pick up parked calls when pressed.
Format	Text
Default Value	Not Applicable
Range	line (9133i only) speeddial dnd ("do not disturb" in the Aastra Web UI) blf (9133i only) list (9133i only; Aastra Web UI only) xml flash park pickup
Example	prgkey1 type: speeddial

Parameter – *prgkeyN value* *Value* (in Web UI)	Aastra Web UI **Operation->Programmable Keys** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the value you assign to the programmable key. • For line type (9133i only) - Value is optional; for example L4. • For speedial type - Value is the phone number or extension to enter for the programmable key. • For blf type (9133i only) - Value is the extension you want to monitor. • xml - Value is IP address of the XML application. • For Park, Pickup types - For valid values, see the "Park/Pickup Call Server Configuration Values" on page 66. For Park/Pickup examples, see "Model 9133i Examples" and "Model 9112i Examples" on page 69. **Notes:** **1.** No values required for dnd and list types. **2.** If the *prgkeyN type* parameter is set to "**flash**", and no label value is entered for the *prgkeyN label* parameter, the label of "**Flash**" is used.
Format	Integer
Default Value	N/A
Range	N/A
Example	prgkey1 value: 411

Parameter (9133i only) – *prgkeyN line* *Line* (in Web UI)	Aastra Web UI **Operation->Programmable Keys** Configuration Files **aastra.cfg, <mac>.cfg**
Description	This is the line associated with the programmable key you are configuring. This parameter is for the 9133i only.
Format	Integer
Default Value	1
Range	1 - 9
Example	prgkey1 line: 1 prgkey2 line: 5

Advanced Operational Parameters

The following parameters in this section allow the system admininstrator to set advanced operational features on the IP phones.

MAC Address/Line Number

This section provides the parameters you can set to enable or disable the sending of the MAC address and line number in REGISTER messages to the call server.

Parameter – *sip send mac* *Send MAC Address in REGISTER Message* (in Web UI)	Aastra Web UI: Configuration Files	**Advanced Settings->Global SIP ->Advanced SIP Settings** **aastra.cfg, <mac>.cfg**
Description	Adds an "Aastra-Mac:" header to the SIP REGISTER messages sent from the phone to the call server, where the value is the MAC address of the phone.	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	sip send mac: 1	

Parameter – *sip send line* *Send Line Number in REGISTER Message* (in Web UI)	Aastra Web UI: Configuration Files	**Advanced Settings->Global SIP ->Advanced SIP Settings** **aastra.cfg, <mac>.cfg**
Description	Adds an "Aastra-Line:" header to the SIP REGISTER messages sent from the phone to the call server, where the value is the MAC address of the phone.	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	sip send line: 1	

Blind Transfer Setting.

Parameter – *sip cancel after blind transfer*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Forces the phone to use the Blind Transfer method available in software prior to release 1.4. This method sends the CANCEL message after the REFER message when blind transferring a call.
Format	Boolean
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	sip cancel after blind transfer: 1

Update Caller ID Setting.

Parameter – *sip update callerid*	Configuration Files **aastra.cfg, <mac>.cfg**
Description	Enables or disables the updating of the Caller ID information during a call.
Format	Boolean
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	sip update callerid: 1

Boot Sequence Recovery Mode.

Parameter – *force web recovery mode disabled*	Configuration Files	**aastra.cfg, <mac>.cfg**
Description	Enables or disables the forcing web recovery mode feature. If this parameter is set to "**1**", you cannot force web recovery. If this parameter is set to "**0**", press **1** and **#** keys during boot up when the logo displays to force the web recovery mode.	
Format	Boolean	
Default Value	0 (false)	
Range	0 (false) 1 (true)	
Example	force web recovery mode disabled: 1	

Parameter – *max boot count*	Configuration Files	**aastra.cfg, <mac>.cfg**
Description	Specifies the number of faulty boots that occur before the phone is forced into Web recovery mode.	
Format	Integer	
Default Value	10	
Range	0 to 32767 **Note:** Zero (0) disables the max boot count feature.	
Example	max boot count: 0	

Troubleshooting Solutions

This section describes solutions to some most common problems that can occur.

Why does my phone display “Application missing”?

If you have experienced networking issues while the phone was downloading the application from the TFTP server, it is possible that the phone can no longer retrieve the required firmware file. In the event that the phone is no longer able to communicate with the TFTP server in its attempt to re-download the firmware and the phone cannot locate the application locally, the message "Application missing" displays.

The phone also displays the following: “Recovery web-client at: *<IP Address>*”. The IP Address displayed is the IP address of the phone. If the phone is unable to receive an IP from the DHCP server or has lost its record of its static IP, the phone will auto-assign itself
the default IP 192.168.0.50.

To recover the firmware for your phone in this circumstance, please perform the following:

1. Launch your web browser on your computer.

Note: Your computer will need to be on the same network as your IP phone.

2. In the URL, type: “http://*<IP Address>*” (where IP Address is the IP Address displayed on the phone). Your browser will launch the **Aastra IP Phone Firmware Recovery** page.
3. Call Customer Support and request a *<phone model>.bin.gz* file.
4. Copy the file to your TFTP server.
5. Enter the *<phone model>.bin.gz* file that is ready for download.
6. Enter the IP address or qualified domain name of the TFTP server.
7. Press the Download Firmware button.

Please ensure that the TFTP server is running and accessible on the network. If the firmware file is correctly located on the running TFTP server, the phone will locate the file and reload the application onto the phone.

Why does my phone display the “No Service” message?

The phone displays the **“No Service”** message if the SIP settings have not been set up correctly.

The Registrar server could be set to 0.0.0.0. A global value of 0.0.0.0 disables registration. However, the phone is still active and you can dial using username@ip address of the phone. The phone displays "**No Service**".

If the Registrar IP address is set to 0.0.0.0 for a per-line basis (i.e, line 1, line 2, etc.), then the register request is not sent, the "**No Service**" message does not display, and the message waiting indicator (MWI) does not come on.

Check that the "**Registrar Server**" IP address in the Aastra Web UI at **Advanced Settings->Global SIP** is correct. Check the "**sip registrar ip**" parameter in the configuration files is correct.

Why does my phone display "Bad Encrypted Config"?

The IP phone displays "Bad Encrypted Config" because encrypted configuration files are enabled but the decryption process has failed. Specific cases where decryption fails are:

<u>Reason</u>:

The site-specific password in *security.tuz* does not match the password used to encrypt the *<mac>.tuz* or *aastra.tuz* files.

<u>Fix</u>:

Encrypt the *.cfg* files to *.tuz* using the correct password, or replace the *security.tuz* with the correct encrypted file.

<u>Reason</u>:

Neither of the *<mac>.tuz* and *aastra.tuz* files are present on the configuration server (TFTP/FTP/HTTP).

<u>Fix</u>:

Create the encrypted files using *anacrypt.exe* and copy them to the configuration server.

<u>Reason</u>:

The encrypted *<mac>.tuz* or *aastra.tuz* file is encrypted using a different version of *anacrypt.exe* than the phone firmware.

<u>Fix</u>:

Run "*anacrypt.exe -v*" and confirm that the correct version is reported, compared to the phone firmware version.

Why is my phone not receiving the TFTP IP address from the DHCP Server?

For DHCP to automatically populate the IP address or qualified domain name for the TFTP server, your DHCP server must support Option 66. Option 66 is responsible for forwarding the TFTP server IP address or domain name to the phone automatically. If your DHCP server does not support Option 66, you must manually enter the IP address or qualified domain name for the TFTP server into your IP phone configuration.

For procedures on configuring the TFTP server using the IP phone UI and the Aastra Web UI, see the section, "Configuring the Configuration Server Protocol" on page 24.

For specific protocol parameters you can set in the configuration files, see "Configuration Server Settings" on page 115.

How to restart the IP phone?

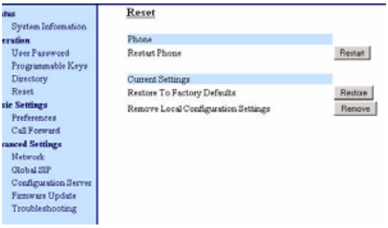
IP Phone UI 

1. Press Options on the phone to enter the Options List.
2. Select **Phone Status**.
3. Select **Restart Phone**.
4. Press **Restart** to restart the phone.

Note: To cancel without restarting the phone, press **Cancel**.

Aastra Web UI

1. Click on **Operation->Reset.**



2. Click Restart .
3. Click **OK** at the confirmation prompt.

How do I set the IP phone to factory default?

IP Phone UI

1. Press Options on the phone to enter the Options List.
2. Select **Phone Status.**
3. Select **Factory Default.**

To restore all factory defaults:

4. Select **All Defaults**. This option restores all factory defaults, and removes any saved configuration and directory list files.
5. Press **Default.**

Note: Press **Cancel** to cancel the operation.

6. Press **Restart** to restart the phone.

To restore the local configuration file factory defaults only:

1. Select **Config only** to restore all factory defaults to the local configuration file. This option removes the configuration file that contains saved parameters set from the Aastra Web UI or the IP phone UI.
2. Press **Default.**

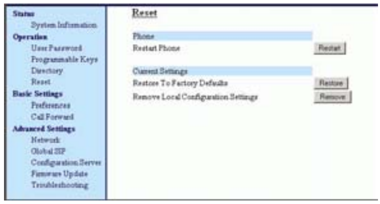
Note: Press **Cancel** to cancel the operation.

3. Press **Restart** to restart the phone.

:

Aastra Web UI

1. Click on **Operation-> Reset.**



To restore all factory defaults:

2. In the "**Restore to Factory Defaults**" field, click Restore .
   This restores all factory defaults, and removes any saved configuration and directory list files.
3. Press to Restart restart the phone.

To restore the local configuration file factory defaults only:

1. In the "**Remove Local Configuration Settings**" field, click Remove .
   This restores all factory defaults to the local configuration file. It removes the configuration file that contains saved parameters set from the Aastra Web UI or the IP phone UI.
2. Press Restart to restart the phone.

How to reset a user's password?

IP Phone UI

1. Press Options on the phone to enter the Options List.
2. Select **User Password**.
3. Enter the current user password.
4. Enter the new user password.
5. Re-enter the new user password.
6. Press **Enter** to save the new password. A message, "Password Changed" displays on the screen.

Aastra Web UI

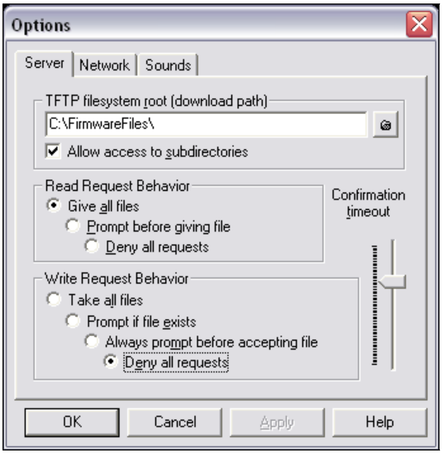
1. Click on **Operation->User Password.**

2. In the "**Current Password**" field, enter the current user password.
3. In the "**New Password**" field, enter the new user password.
4. In the "Confirm Password" field, enter the new user password again.
5. Click Save Settings to save your changes.

Appendix A: Configuration Server Protocol Setup

TFTP Server Set-up

There are a number of TFTP servers available. PumpKIN is one of such TFTP servers. Use the keywords “pumpkin TFTP server” on Google and you should get the web site where you can download the software from. Installing PumpKIN is straightforward. To configure the directory from where you would be serving the files, click on the Options button on PumpKIN’s main window as shown in the following figure.



It is important to select the “Give all files” radio button under the “Read Request Behavior” category. This makes the files to be served without any manual intervention when requested.

If you want to prevent users from writing files to the directory select the “Deny all requests” in the “Write Request Behavior” category. Click the OK button after you have entered all the required information. All the firmware files should be in the file system root directory. Currently we do not support downloads from files present in sub-directories. Consult PumpKIN’s documentation if you need more information on how to set-up the TFTP server.

Appendix B: Configuring the IP Phone at the Asterisk IP PBX

The following configuration illustrates how to create a user with an extension to make and receive calls using the Asterisk as the PBX. This configuration is defined in the *sip.conf* file present along with the other configuration files that are created when Asterisk is installed. Usually, the configuration files can be found at the */etc/asterisk* directory.

```
;This is used in the "extensions.conf" file to identify this
;physical phone when issuing Dial commands.
[phone1]

;The type to use for the 480i is "friend".
;"Peer" is used when the Asterisk is contacting a proxy,
;"user" is used for phones that can only make calls
;and "friend" acts as both a peer and a user.
type=friend

;If your host has an entry in your DNS then you just enter the
;machines name in the host= field.
host=dynamic
defaultip=192.168.1.1         ;default IP address that the phone is
                              ;configured to

;The password that phone1 will use to register with this PBX
secret=1234

dtmfmode=rfc2833              ;Choices are inband, rfc2833, or info
mailbox=1000                  ;Mailbox for message waiting indicator

;If a phone is not in a valid context you will not be
;able to use it. In this example' sip' is used. You can use
;whatever you like, but make sure they are the same, you will
;need to make an entry in your extensions.conf file (which we
;will get to later)
context=sip
callerid="Phone 1" <1234>
```

After this is defined in the "sip.conf" file, some information has to be entered in the "extensions.conf" file present in the same directory as the "sip.conf" file. The following definition in the file under the [sip]section/ context completes defining the extension for the 480i phone.

```
exten -> 1234,1,Dial(SIP/phone1,20)
```

This definition completes configuring the 480i phone at the IP PBX system.

To verify whether the extension has been successfully registered at the IP PBX system, enter the Asterisk console and reload Asterisk. Use the command "sip show peers" at the console. This will display the extensions that are registered at the IP PBX system.

```
Name/username     Host          Mask                  Port     Status
phone1/phone1 192.168.1.1   (D) 255.255.255.255      5060
Unmonitored
```

This completes the basic set-up for the 480i phone with 1234 extension at the Asterisk IP PBX system. Refer to Asterisk documentation for set-up on extended or advanced features such as voice mail and call forwarding, etc.

Appendix C: Sample Configuration Files

This section consists of the sample configuration files necessary to configure the IP phones. The general format is similar to configuration files used by several Unix-based programs. Any text following a number sign (#) on a line is considered to be a comment, unless the # is contained within double-quotes. Currently, Boolean fields use 0 for false and 1 for true.

480i Sample Configuration File

```
# Sample Configuration File
# =========================

# Date: October 20th, 2005

# Phone Model: 480i

# Notes:
#
# The general format used here is similar to configuration files
# used by several UNIX-based programs. Any text following a number
# sign (#) is considered to be a comment, unless the number sign is
# contained within double-quotes ("#") where it is considered to be
# a pound. For Boolean fields, 0 = false, 1 = true.

# Comments:
#
# This file contains sample configurations for the "aastra.cfg" or
# "<mac>.cfg" file.  The settings included here are examples only.
# You should change/comment the values to suit your requirements.
#
# Not all possible paramters are shown, refer to the admin guide for
# the full list of supported parameters, their defaults and valid
# ranges.
#
# The Aastra 480i, 480iCT, 9112i and 9133i phones will download 2
# configuration files from the TFTP server while restarting, the
# "aastra.cfg" file and  the "<mac>.cfg" file. These two
configuration
# files can be used to configure all of the settings of the phone
with
# the exception of assigning a static IP address to a phone and line
# settings, which should only be set in the "<mac>.cfg" file.
#
```

```
# The "aastra.cfg" file configures the settings server wide, while
# the "<mac>.cfg" file configures only the phone with the MAC address
# for which the file is named (for example, "00085d0304f4.cfg"). The
# settings in the "aastra.cfg" file will be overridden by settings
# which also appear in the "<mac>.cfg" file.
#-------------------------------------------------------------------


# DHCP Setting
# ============

#dhcp: 1 # DHCP enabled.

# DHCP:
# 0 = false, means DHCP is disabled.
# 1 = true, means DHCP is enabled.
#
# Notes:
#
#  DHCP is normally set from the Options list on the phone or
#  the web interface
#
#  If DHCP is disabled, the following network settings will
#  have to be configured manually either through the configuration
#  files, the Options List in the phone, or the Web Client: IP
#  Address (of the phone), Subnet Mask, Gateway, DNS, and TFTP
#  Server.
#--------------------------------------------------------------------


# Network Settings
## = = = = = = = =

# Notes: If DHCP is enabled, you do not need to set these network
# settings.  Although depending on you DHCP server configuration you
# may still have to set the dns address.

#ip:     # This value is unique to each phone on a server
         # and should be set in the "<mac>.cfg" file if
```

```
        # setting this manually.

#subnet mask:
#default gateway:
#dns1:
#dns2:


# Time Server Settings
## ====================

#time server disabled: 1  # Time server disabled.
#time server1:            # Enable time server and enter at
#time server2:            # least one time server IP address or
#time server3:          # qualified domain name

# Time Server Disabled:
#   0 = false, means the time server is not disabled.
#   1 = true, means the time server is disabled.



# NAT Settings
# ===============

# Option 1:
#
#  If you are connecting to a Nortel MCS call server and there is a
#  NAT device between the server and the phone, then you must set the
#  following two parameters for the phone to function correctly.

#sip nortel nat support: 1      # 1 = enabled
#sip nortel nat timer: 60       # seconds between keep alive messages


# Option 2:
#
#  If you are using a session border controller, you should set the
#  outbound proxy to the session border controller address
```

```
#sip outbound proxy: sbc.aastra.com
#sip outbound proxy port: 0              # a value of 0 enables SRV
                                        # lookups for the address of
                                         # the proxy.


# Option 3:
#
#  If you know the public IP address of your NAT device and and have
#  opened up a port for the SIP messages then you can statically
#  assign this information.

#sip nat ip: 67.123.122.90
#sip nat port: 5890
```

 # Additional Network Settings

 # ==========================

```
#sip rtp port: 3000     # Eg. RTP packets are sent to port 3000.



#-------------------------------------------------------------------
```

 # Configuration Server Settings

= = = = = = = = = = = = = = =

```
# Notes: This section defines which server the phone retrieves new
# firmware images and configuration files from.  Three protocols are
# supported TFTP, FTP and HTTP

download protocol: TFTP   # valid values are TFTP, FTP and HTTP

## TFTP server settings
tftp  server: 192.168.0.130
#alternative tftp server:
#use alternative tftp server: 1         # If your DHCP server assigns
                                        # a TFTP server address which
                                      # you do not use, you can use
```

```
                                        # the alternative tftp server.

## FTP server settings
#ftp server: 192.168.0.131   # can be IP or FQDN
#ftp username: aastra
#ftp password: 480iaastra


## HTTP server settings (for http://bogus.aastra.com/firmware/)
#http server: bogus.aastra.com   # can be IP or FQDN
#http path: firmware


#-------------------------------------------------------------------


# Dial Plan Settings
# ==================
#
# Notes:
#
#  As you dial a number on the phone, the phone will initiate a call
#  when one of the following conditions are meet:
#
#   (1) The entered number is an exact match in the dial plan
#   (2) The "#" symbol has been pressed
#   (3) A timeout occurs
#
#  The dial plan is a regular expression that supports the following
#  syntax:
#
#    0,1,2,3,4,5,6,7,8,9,*,#  : matches the keypad symbols
#    x                        : matches any digit (0...9)
#    +                        : matches 0 or more repetitions of the
#                             :  previous expression
#    []                       : matches any number inside the brackets
#                             :  can be used with a "-" to represent a
#                             :  range
#    ()                       : expression grouping
#    |                        : either or
```

```
#
#
#  If the dialled number doesn't match the dial plan then the call
#  is rejected.

sip digit timeout: 3       # set the inter-digit timeout in seconds

# Example dial plans
sip dial plan: "x+#|xx+*"   # this is the default dial string, note
                           # that is must be quoted since it contains
                            # a '#' character



#sip dial plan: [01]xxx|[2-8]xxxx|91xxxxxxxxxx
                            # accecpt any 4 digit number beginning
                            # with a 0 or 1, any 5 digit number
                           # beginning with a number between 2 and 8
                            # (inclusive) or a 12 digit number
                            # beginning with 91



#sip dial plan terminator: 1   # enable sending of the "#" symbol to
                               # to the proxy in the dial string

#-------------------------------------------------------------------



# General SIP Settings
# = = = = = = = = = = = =

#sip session timer: 30      # enable support of RFC4028, the default
                            # value of 0 disables this functionality


#sip transport protocol: 0  # use UDP (1), TCP (2) or both (0) for
sip
                            # messaging

#sip use basic codecs: 1    # limit codecs to G711 and G729
#sip out-of-band dtmf: 0    # turn off support for RFC2833 (on by
                            # default)
```

```
# Global SIP User Settings
# ======================
#
# Notes:
#   These settings are used as the default configuration for the hard
#   key lines on the phone.  That is:
#
#     L1 to L4 on the 480i and 480iCT
#     L1 to L3 on the 9133i
#     L1 on the 9112i
#
#   These can be over-ridden on a per-line basis using the per-line
#   settings.
#
#   See the Admin Guide for a detailed explaination of how this works

sip screen name: Joe Smith     # the name display on the phone's
screen
sip user name: 4256            # the phone number
sip display name: Joseph Smith # the caller name sent out when making
                               # a call.
sip vmail: *78                 # the number to reach voicemail on

sip auth name: jsmith          # account used to authenticate user
sip password: 12345            # password for authentication account

sip mode: 0                    # line type:
                               #   0 - generic,
                               #   1 - BroadSoft SCA line
                               #   2 - Nortel line

sip proxy ip: proxy.aastra.com # IP address or FQDN of proxy
sip proxy port: 5060           # port used for SIP messages on the
                               # proxy.  Set to 0 to enable SRV
                               # lookups

sip registrar ip: aastra.com   # IP address or FQDN of registrar
```

```
sip registrar port: 0         # as proxy port, but for the registrar
sip registration period: 3600  # registration period in seconds
```

Per-line SIP Settings

==================

```
# configure line 3 as the support Broadsoft SCA line
#   - the proxy and registrar settings are taken from the global
#     settings above

sip line3 screen name: Support
sip line3 user name: 4000
sip line3 display name: Aastra Support
sip line3 auth name: support
sip line3 password: 54321
sip line3 mode: 1
sip line3 vmail: *78


# configure line 5 (a soft key line) as an ordinary line
# of a test server

sip line5 screen name: Test 1
sip line5 user name: 5551001
sip line5 display name: Test 1
sip line5 auth name: 5551001
sip line5 password: 5551001
sip line5 mode: 0
sip line5 proxy ip: 10.50.10.102
sip line5 proxy port: 5060
sip line5 registrar ip: 10.50.10.102
sip line5 registrar port: 5060
sip line5 registration period: 60


#--------------------------------------------------------------------

# Softkey Settings
```

```
#

# Softkeys can be set either server wide or unique to each phone.
# Setting softkeys as line/call appearances should be done in the
# "<mac>.cfg" file, since these are unique to each phone.

# Notes:
#
#  There are a maximum of 18 softkeys that can be configured on the
#  480i or 480iCT phone. These can be set up through either of the 2
#  configuration files, depending on whether this is to be server wide
#  ("aastra.cfg") or phone specific ("<mac>.cfg").  Each softkey needs
#  to be numbered from 1 - 18, for example "softkey12 type:
#  speeddial". Softkeys can be set up as speeddials or as additional
#  call/line appearances and have a type, label and value associated
#  with it as seen here in the default softkey settings.

#  SOFTKEY TYPES: "line", "speeddial", "blf", "list", "dnd"
#  SOFTKEY LABEL: Alpha numeric name for the softkey. The maximum
#                 number of characters for this value is 10 for
#                 speeddials and dnd, 9 chars for lines, blf
#  SOFTKEY VALUE: If softkey type is a speeddial, any DTMFs (from
#                 0 - 9, *, "#") or a comma (,) for 500ms pause and
#                 'E' for On-hook can be set for the value.
#                If softkey type is blf it is the extension you want
#                 to monitor.
#  SOFTKEY LINE:  This is line associated with the softkey.  For line
#                 softkeys the value must be between 5 and 9 (1 - 4
#                are already hardcoded as the L1, L2, L3 and L4 hard
#                 key line/call appearances)

# Speed Dials
softkey1 type: speeddial
softkey1 label: "Ext Pickup"
softkey1 value: *8
softkey2 type: speeddial
softkey2 label: "Call Return"
softkey2 value: *69
```

```
# DND Key
softkey4 type: dnd
softkey4 label: DND

# Line appearance
softkey6 type: line
softkey6 label: Test 1
softkey6 line: 5

# blf
softkey8 type: blf
softkey8 label: Jane Doe
softkey8 value: 4559
softkey8 line: 1

# list
softkey11 type: list
softkey12 type: list
```

480i CT Sample Configuration File

```
# Sample Configuration File
# =========================

# Date: October 26th, 2005

# Phone Model: 480iCT

# Notes:
#
# The general format used here is similar to configuration files
# used by several UNIX-based programs. Any text following a number
# sign (#) is considered to be a comment, unless the number sign is
# contained within double-quotes ("#") where it is considered to be
# a pound. For Boolean fields, 0 = false, 1 = true.

# Comments:
#
# This file contains sample configurations for the "aastra.cfg" or
# "<mac>.cfg" file.  The settings included here are examples only.
# You should change/comment the values to suit your requirements.
#
# Not all possible paramters are shown, refer to the admin guide for
# the full list of supported parameters, their defaults and valid
# ranges.
#
# The Aastra 480i, 480iCT, 9112i and 9133i phones will download 2
# configuration files from the TFTP server while restarting, the
# "aastra.cfg" file and  the "<mac>.cfg" file. These two 
configuration
# files can be used to configure all of the settings of the phone 
with
# the exception of assigning a static IP address to a phone and line
# settings, which should only be set in the "<mac>.cfg" file.
#
# The "aastra.cfg" file configures the settings server wide, while 
the
# "<mac>.cfg" file configures only the phone with the MAC address for
```

```
# which the file is named (for example, "00085d0304f4.cfg"). The
# settings in the "aastra.cfg" file will be overridden by settings
# which also appear in the "<mac>.cfg" file.
#-------------------------------------------------------------------



# DHCP Setting
# ============

#dhcp: 1 # DHCP enabled.

# DHCP:
# 0 = false, means DHCP is disabled.
# 1 = true, means DHCP is enabled.
#
# Notes:
#
#  DHCP is normally set from the Options list on the phone or
#  the web interface
#
#  If DHCP is disabled, the following network settings will
#  have to be configured manually either through the configuration
#  files, the Options List in the phone, or the Web Client: IP
#  Address (of the phone), Subnet Mask, Gateway, DNS, and TFTP
#  Server.
#-------------------------------------------------------------------



# Network Settings
# ================

# Notes: If DHCP is enabled, you do not need to set these network
# settings.  Although depending on you DHCP server configuration you
# may still have to set the dns address.
```

```
#ip:     # This value is unique to each phone on a server
         # and should be set in the "<mac>.cfg" file if
         # setting this manually.

#subnet mask:
#default gateway:
#dns1:
#dns2:


# Time Server Settings
# ====================

#time server disabled: 1  # Time server disabled.
#time server1:            # Enable time server and enter at
#time server2:            # least one time server IP address or
#time server3:          # qualified domain name

# Time Server Disabled:
#   0 = false, means the time server is not disabled.
#   1 = true, means the time server is disabled.



# NAT Settings
#=============

# Option 1:
#
#  If you are connecting to a Nortel MCS call server and there is a
#  NAT device between the server and the phone, then you must set the
#  following two parameters for the phone to function correctly.

#sip nortel nat support: 1      # 1 = enabled
#sip nortel nat timer: 60       # seconds between keep alive messages
```

```
# Option 2:
#
#  If you are using a session border controller, you should set the
#  outbound proxy to the session border controller address

#sip outbound proxy: sbc.aastra.com
#sip outbound proxy port: 0               # a value of 0 enables SRV
                                         # lookups for the address of
                                          # the proxy.


# Option 3:
#
#  If you know the public IP address of your NAT device and and have
#  opened up a port for the SIP messages then you can statically
#  assign this information.

#sip nat ip: 67.123.122.90
#sip nat port: 5890



# Additional Network Settings
#==========================

#sip rtp port: 3000     # Eg. RTP packets are sent to port 3000.



#-------------------------------------------------------------------



# Configuration Server Settings
# ============================

# Notes: This section defines which server the phone retrieves new
# firmware images and configuration files from.  Three protocols are
# supported TFTP, FTP and HTTP

download protocol: TFTP   # valid values are TFTP, FTP and HTTP
```

```
## TFTP server settings
tftp  server: 192.168.0.130
#alternative tftp server:
#use alternative tftp server: 1        # If your DHCP server assigns
                                       # a TFTP server address which
                                       # you do not use, you can use
                                      # the alternative tftp server.

## FTP server settings
#ftp server: 192.168.0.131   # can be IP or FQDN
#ftp username: aastra
#ftp password: 480iaastra


## HTTP server settings (for http://bogus.aastra.com/firmware/)
#http server: bogus.aastra.com   # can be IP or FQDN
#http path: firmware



#-------------------------------------------------------------------


# Dial Plan Settings
# ==================
#
# Notes:
#
#  As you dial a number on the phone, the phone will initiate a call
#  when one of the following conditions are meet:
#
#   (1) The entered number is an exact match in the dial plan
#   (2) The "#" symbol has been pressed
#   (3) A timeout occurs
#
#  The dial plan is a regular expression that supports the following
#  syntax:
```

```
#
#    0,1,2,3,4,5,6,7,8,9,*,#  : matches the keypad symbols
#    x                        : matches any digit (0...9)
#    +                        : matches 0 or more repetitions of the
#                             :  previous expression
#    []                       : matches any number inside the brackets
#                             :  can be used with a "-" to represent a
#                             :  range
#    ()                       : expression grouping
#    |                        : either or
#
#
#  If the dialled number doesn't match the dial plan then the call
#  is rejected.

sip digit timeout: 3        # set the inter-digit timeout in seconds

# Example dial plans
sip dial plan: "x+#|xx+*"   # this is the default dial string, note
                            # that is must be quoted since it contains
                            # a '#' character



#sip dial plan: [01]xxx|[2-8]xxxx|91xxxxxxxxxx
                            # accecpt any 4 digit number beginning
                            # with a 0 or 1, any 5 digit number
                            # beginning with a number between 2 and 8
                            # (inclusive) or a 12 digit number
                            # beginning with 91



#sip dial plan terminator: 1   # enable sending of the "#" symbol to
                               # to the proxy in the dial string

#-------------------------------------------------------------------
```

```
# General SIP Settings
# ====================

#sip session timer: 30      # enable support of RFC4028, the default
                             # value of 0 disables this functionality

#sip transport protocol: 0  # use UDP (1), TCP (2) or both (0) for
sip
                             # messaging

#sip use basic codecs: 1    # limit codecs to G711 and G729
#sip out-of-band dtmf: 0    # turn off support for RFC2833 (on by
                             # default)



# Global SIP User Settings
# ========================
#
# Notes:
#   These settings are used as the default configuration for the hard
#   key lines on the phone.  That is:
#
#     L1 to L4 on the 480i and 480iCT
#     L1 to L3 on the 9133i
#     L1 on the 9112i
#
#   These can be over-ridden on a per-line basis using the per-line
#   settings.
#
#   See the Admin Guide for a detailed explaination of how this works

sip screen name: Joe Smith     # the name display on the phone's
screen
sip user name: 4256            # the phone number
sip display name: Joseph Smith # the caller name sent out when making
                               # a call.
sip vmail: *78                 # the number to reach voicemail on
```

```
sip auth name: jsmith          # account used to authenticate user
sip password: 12345            # password for authentication account

sip mode: 0                    # line type:
                               #   0 - generic,
                               #   1 - BroadSoft SCA line
                               #   2 - Nortel line

sip proxy ip: proxy.aastra.com # IP address or FQDN of proxy
sip proxy port: 5060           # port used for SIP messages on the
                               # proxy.  Set to 0 to enable SRV
                               # lookups

sip registrar ip: aastra.com   # IP address or FQDN of registrar
sip registrar port: 0         # as proxy port, but for the registrar
sip registration period: 3600  # registration period in seconds


# Per-line SIP Settings
# =====================

# configure line 3 as the support Broadsoft SCA line
#   - the proxy and registrar settings are taken from the global
#     settings above

sip line3 screen name: Support
sip line3 user name: 4000
sip line3 display name: Aastra Support
sip line3 auth name: support
sip line3 password: 54321
sip line3 mode: 1
sip line3 vmail: *78


# configure line 5 (a soft key line) as an ordinary line
# of a test server
```

```
sip line5 screen name: Test 1
sip line5 user name: 5551001
sip line5 display name: Test 1
sip line5 auth name: 5551001
sip line5 password: 5551001
sip line5 mode: 0
sip line5 proxy ip: 10.50.10.102
sip line5 proxy port: 5060
sip line5 registrar ip: 10.50.10.102
sip line5 registrar port: 5060
sip line5 registration period: 60


#------------------------------------------------------------------

# Softkey Settings
# ================

# Softkeys can be set either server wide or unique to each phone.
# Setting softkeys as line/call appearances should be done in the
# "<mac>.cfg" file, since these are unique to each phone.

# Notes:
#
#  There are a maximum of 18 softkeys that can be configured on the
#  480i or 480iCT phone. These can be set up through either of the 2
#  configuration files, depending on whether this is to be server
wide
#  ("aastra.cfg") or phone specific ("<mac>.cfg").  Each softkey
needs
#  to be numbered from 1 - 18, for example "softkey12 type:
#  speeddial". Softkeys can be set up as speeddials or as additional
#  call/line appearances and have a type, label and value associated
#  with it as seen here in the default softkey settings.

#  SOFTKEY TYPES: "line", "speeddial", "blf", "list", "dnd"
#  SOFTKEY LABEL: Alpha numeric name for the softkey. The maximum
```

```
#                 number of characters for this value is 10 for
#                 speeddials and dnd, 9 chars for lines, blf
#  SOFTKEY VALUE: If softkey type is a speeddial, any DTMFs (from
#                 0 - 9, *, "#") or a comma (,) for 500ms pause and
#                 'E' for On-hook can be set for the value.
#                 If softkey type is blf it is the extension you want
#                 to monitor.
#  SOFTKEY LINE:  This is line associated with the softkey.  For line
#                 softkeys the value must be between 5 and 9 (1 - 4
#                 are already hardcoded as the L1, L2, L3 and L4 hard
#                 key line/call appearances)

# Speed Dials
softkey1 type: speeddial
softkey1 label: "Ext Pickup"
softkey1 value: *8
softkey2 type: speeddial
softkey2 label: "Call Return"
softkey2 value: *69

# DND Key
softkey4 type: dnd
softkey4 label: DND

# Line appearance
softkey6 type: line
softkey6 label: Test 1
softkey6 line: 5

# blf
softkey8 type: blf
softkey8 label: Jane Doe
softkey8 value: 4559
softkey8 line: 1

# list
softkey11 type: list
```

```
softkey12 type: list


#---------------------------------------------------------------------


# Cordless Handset Feature Keys
# ============================


# Notes:
#
#  In addition to the configuration parameters that exist on the 480i
#  phone, following are the parameters specific to the 480i Cordless
#  phones' handset. These parameters can be defined either int the
#  aastra.cfg or the <mac>.cfg files.
#
#  The feature keys are displayed when the user presses the F button
#  on the cordless phone's hand set.  If any changes to the features
#  keys are made using these parameters the feature keys that exist
on
#  the hand set have to be refreshed. To refresh the feature keys
#  simply open a new line or press one of the feature keys that are
#  available from the hand set. After a couple of seconds the
cordless
#  should get the new list from the base set.  There are 15 feature
#  keys that can be configured for the cordless hand set. Each
feature
#  key has the following settings. N corresponds to the feature key
#  that is being configured for and ranges from 0-14.  Feature key N
#  En label: "String" Feature key N Fr label: "Fr-String" Feature key
#  N Sp label: "Sp-String" Feature key N control: 1     #Takes an
#  integer value Feature key N hs event: 1    #Takes an integer value
#  Feature key N base event: 1  #Takes an integer value



#key list version: 1
#  The parameter value has to be incremented by one whenever the
#  parameters that carry the feature keys change. The range is from
#  1-254. After reaching 254 start over from 1.
```

```
#Feature key 0 En label: "Line 1"
#  English label for the key. Displayed when the phone's language is
#  set to use English

#Feature key 0 Fr label: "Fr-Line 1"
#  French label for the key. Displayed when the phone's language
#  is set to use French

#Feature key 0 Sp label: "Sp-Line 1"
#  Spanish label for the key. Displayed when the phone's language
#  is set to use Spanish

Feature key 0 Gr label: "Gr-Line 1"
#  German label for the key. Displayed when the phone's language
#  is set to use German

Feature key 0 It label: "It-Line 1"
#  Italian label for the key. Displayed when the phone's language
#  is set to use Italian

#Feature key 0 control: 1
#  1 - Make the key configurable by the user through the phone and
#      the phone's web client
#  2 - Locks the key from user modifications. User cannot modify
#      this key from the handset or the phone's web client.
#  4 - Hide this key. Do not show it in the Feature keys list in the
#      cordless handset
#  6 - Lock and hide this key. Do not show it in the Feature keys
#      list in the cordless handset and do not let the user modify
#      this key using the phone or the web client.

#Feature key 0 hs event: 7
#  These events are for handset specific events. Events can be local
#  to the handset like directory/caller's list, intercom etc. or may
#  be an event that is sent to the base set for fruther processing.
#  When this key is configured as a base event then the base set
```

```
#  will process the value of this key in conjunction with the value
#  configured for the "Feature key N base event". Where N is the
#  feature key is being configured for.
#  In addition to the values listed below the valid values are
#  [7-23]. The values [7-23] indicate generic handset events. If
#  you are using values within this range make sure to use the value
#  only once.
#  The events local to the handset are as follows:
#
#    58 - Menu (Options)
#    59 - Feature Key
#    60 - Redial
#    61 - Directory
#    62 - Callers' list
#    63 - Services
#    86 - Icom

#Feature key 0 base event: 1
#  Indicates a corresponding action to perform on the base set when
#  the "Feature key N hs event" is set to any value between 7-23.
#
#   1 - Seize base set's line1
#   2 - Seize base set's line2
#   3 - Seize base set's line3
#   4 - Seize base set's line4
#   5 - Seize base set's line5
#   6 - Seize base set's line6
#   7 - Seize base set's line7
#   8 - Seize base set's line8
#   9 - Seize base set's line9
#   10 - Seize base set's line0
#   11 - Send the base set's transfer event
#   12 - Send the base set's conference event
#   13 - Make feature list public


# Example configuration
```

```
key list version: 1

Feature key 0 En label: "Line 1"
Feature key 0 Fr label: "Fr-Line 1"
Feature key 0 Sp label: "Sp-Line 1"
Feature key 0 control: 0
Feature key 0 hs event: 7
Feature key 0 base event: 1

Feature key 1 En label: "Conf."
Feature key 1 Fr label: "Fr-Conf."
Feature key 1 Sp label: "Sp-Conf."
Feature key 1 control: 1
Feature key 1 hs event: 8
Feature key 1 base event: 12

Feature key 2 En label: "Xfer"
Feature key 2 Fr label: "Fr-Xfer."
Feature key 2 Sp label: "Sp-Xfer."
Feature key 2 control: 2
Feature key 2 hs event: 9
Feature key 2 base event: 11

Feature key 3 En label: "Icom"
Feature key 3 Fr label: "Fr-Icom"
Feature key 3 Sp label: "Sp-Icom"
Feature key 3 control: 1
Feature key 3 hs event: 86
Feature key 3 base event: 13

Feature key 4 En label: "Opt"
Feature key 4 Fr label: "Fr-Opt"
Feature key 4 Sp label: "Sp-Opt"
Feature key 4 hs event: 58
Feature key 4 control: 1
Feature key 4 base event: 13
```

```
Feature key 5 En label: "Callers"
Feature key 5 Fr label: "Fr-Callers"
Feature key 5 Sp label: "Sp-Callers"
Feature key 5 hs event: 62
Feature key 5 control: 1
Feature key 5 base event: 13


Feature key 6 En label: "Top"
Feature key 6 Fr label: "Fr-Top"
Feature key 6 Sp label: "Sp-Top"
Feature key 6 hs event: 17
Feature key 6 control: 1
Feature key 6 base event: 13

Feature key 7 En label: "Redial"
Feature key 7 Fr label: "Fr-Redial"
Feature key 7 Sp label: "Sp-Redial"
Feature key 7 hs event: 60
Feature key 7 control: 4
Feature key 7 base event: 13

Feature key 8 En label: "Dir."
Feature key 8 Fr label: "Fr-Dir."
Feature key 8 Sp label: "Sp-Dir."
Feature key 8 hs event: 61
Feature key 8 control: 2
Feature key 8 base event: 13

Feature key 9 En label: "Services"
Feature key 9 Fr label: "Fr-Services"
Feature key 9 Sp label: "Sp-Services"
Feature key 9 hs event: 63
Feature key 9 control: 1
Feature key 9 base event: 13
```

9112i Sample Configuration File

```
# Sample Configuration File
# ========================

# Date: October 26th, 2005

# Phone Model: 9112i

# Notes:
#
# The general format used here is similar to configuration files
# used by several UNIX-based programs. Any text following a number
# sign (#) is considered to be a comment, unless the number sign is
# contained within double-quotes ("#") where it is considered to be
# a pound. For Boolean fields, 0 = false, 1 = true.

# Comments:
#
# This file contains sample configurations for the "aastra.cfg" or
# "<mac>.cfg" file.  The settings included here are examples only.
# You should change/comment the values to suit your requirements.
#
# Not all possible paramters are shown, refer to the admin guide for
# the full list of supported parameters, their defaults and valid
# ranges.
#
# The Aastra 480i, 480iCT, 9112i and 9133i phones will download 2
# configuration files from the TFTP server while restarting, the
# "aastra.cfg" file and  the "<mac>.cfg" file. These two 
configuration
# files can be used to configure all of the settings of the phone 
with
# the exception of assigning a static IP address to a phone and line
# settings, which should only be set in the "<mac>.cfg" file.
#
# The "aastra.cfg" file configures the settings server wide, while 
the
# "<mac>.cfg" file configures only the phone with the MAC address for
# which the file is named (for example, "00085d0304f4.cfg"). The
# settings in the "aastra.cfg" file will be overridden by settings
# which also appear in the "<mac>.cfg" file.
#-------------------------------------------------------------------
```

```
# DHCP Setting
# ============

#dhcp: 1 # DHCP enabled.

# DHCP:
# 0 = false, means DHCP is disabled.
# 1 = true, means DHCP is enabled.
#
# Notes:
#
#  DHCP is normally set from the Options list on the phone or
#  the web interface
#
#  If DHCP is disabled, the following network settings will
#  have to be configured manually either through the configuration
#  files, the Options List in the phone, or the Web Client: IP
#  Address (of the phone), Subnet Mask, Gateway, DNS, and TFTP
#  Server.
#-------------------------------------------------------------------


# Network Settings
# ================

# Notes: If DHCP is enabled, you do not need to set these network
# settings.  Although depending on you DHCP server configuration you
# may still have to set the dns address.

#ip:     # This value is unique to each phone on a server
         # and should be set in the "<mac>.cfg" file if
         # setting this manually.

#subnet mask:
#default gateway:
#dns1:
#dns2:


# Time Server Settings
# ====================

#time server disabled: 1  # Time server disabled.
#time server1:            # Enable time server and enter at
#time server2:            # least one time server IP address or
#time server3:          # qualified domain name.
```

```
# Time Server Disabled:
#   0 = false, means the time server is not disabled.
#   1 = true, means the time server is disabled.



# NAT Settings
#============

# Option 1:
#
#  If you are connecting to a Nortel MCS call server and there is a
#  NAT device between the server and the phone, then you must set the
#  following two parameters for the phone to function correctly.

#sip nortel nat support: 1      # 1 = enabled
#sip nortel nat timer: 60       # seconds between keep alive messages


# Option 2:
#
#  If you are using a session border controller, you should set the
#  outbound proxy to the session border controller address

#sip outbound proxy: sbc.aastra.com
#sip outbound proxy port: 0               # a value of 0 enables SRV
                                          # lookups for the address of
                                          # the proxy.

# Option 3:
#
#  If you know the public IP address of your NAT device and and have
#  opened up a port for the SIP messages then you can statically
#  assign this information.

#sip nat ip: 67.123.122.90
#sip nat port: 5890
```

```
# Additional Network Settings
#==========================

#sip rtp port: 3000     # Eg. RTP packets are sent to port 3000.


#--------------------------------------------------------------------


# Configuration Server Settings
# ===========================

# Notes: This section defines which server the phone retrieves new
# firmware images and configuration files from.  Three protocols are
# supported TFTP, FTP and HTTP

download protocol: TFTP   # valid values are TFTP, FTP and HTTP

## TFTP server settings
tftp  server: 192.168.0.130
#alternative tftp server:
#use alternative tftp server: 1        # If your DHCP server assigns
                                       # a TFTP server address which
                                       # you do not use, you can use
                                      # the alternative tftp server.

## FTP server settings
#ftp server: 192.168.0.131   # can be IP or FQDN
#ftp username: aastra
#ftp password: 480iaastra


## HTTP server settings (for http://bogus.aastra.com/firmware/)
#http server: bogus.aastra.com   # can be IP or FQDN
#http path: firmware


--------------------------------------------------------------------
```

```
# Dial Plan Settings
# ==================
#
# Notes:
#
#  As you dial a number on the phone, the phone will initiate a call
#  when one of the following conditions are meet:
#
#   (1) The entered number is an exact match in the dial plan
#   (2) The "#" symbol has been pressed
#   (3) A timeout occurs
#
#  The dial plan is a regular expression that supports the following
#  syntax:
#
#    0,1,2,3,4,5,6,7,8,9,*,#  : matches the keypad symbols
#    x                        : matches any digit (0...9)
#    +                        : matches 0 or more repetitions of the
#                             :  previous expression
#    []                      : matches any number inside the brackets
#                            :  can be used with a "-" to represent a
#                             :  range
#    ()                       : expression grouping
#    |                        : either or
#
#
#  If the dialled number doesn't match the dial plan then the call
#  is rejected.

sip digit timeout: 3       # set the inter-digit timeout in seconds

# Example dial plans
sip dial plan: "x+#|xx+*"   # this is the default dial string, note
                           # that is must be quoted since it contains
                            # a '#' character



#sip dial plan: [01]xxx|[2-8]xxxx|91xxxxxxxxxx
                            # accecpt any 4 digit number beginning
                            # with a 0 or 1, any 5 digit number
                           # beginning with a number between 2 and 8
                            # (inclusive) or a 12 digit number
                            # beginning with 91
```

```
#sip dial plan terminator: 1   # enable sending of the "#" symbol to
                               # to the proxy in the dial string

#-------------------------------------------------------------------


# General SIP Settings
# ====================

#sip session timer: 30      # enable support of RFC4028, the default
                            # value of 0 disables this functionality

#sip transport protocol: 0  # use UDP (1), TCP (2) or both (0) for
sip
                            # messaging

#sip use basic codecs: 1    # limit codecs to G711 and G729
#sip out-of-band dtmf: 0    # turn off support for RFC2833 (on by
                            # default)



# SIP User Settings
# ========================

sip screen name: Joe Smith     # the name display on the phone's
screen
sip user name: 4256            # the phone number
sip display name: Joseph Smith # the caller name sent out when making
                               # a call.
sip vmail: *78                 # the number to reach voicemail on

sip auth name: jsmith          # account used to authenticate user
sip password: 12345            # password for authentication account

sip mode: 0                    # line type:
                               #   0 - generic,
                               #   1 - BroadSoft SCA line
                               #   2 - Nortel line

sip proxy ip: proxy.aastra.com # IP address or FQDN of proxy
sip proxy port: 5060           # port used for SIP messages on the
                               # proxy.  Set to 0 to enable SRV
                               # lookups
```

```
sip registrar ip: aastra.com   # IP address or FQDN of registrar
sip registrar port: 0         # as proxy port, but for the registrar
sip registration period: 3600  # registration period in seconds



#------------------------------------------------------------------

# Programmable Key Settings
# =========================

# Programmable keys can be set either server wide or unique to each
phone.
# Setting programmable keys as line/call appearances should be done
in the
# "<mac>.cfg" file, since these are unique to each phone.

# Notes:
#
#  There are a maximum of 7 programmable keys that can be configured
#  on the 9133i phone, and only 2 on the 9112i phone. These can be
set
#  up through either of the 2 configuration files, depending on
#  whether this is to be server wide ("aastra.cfg") or phone specific
#  ("<mac>.cfg").  Each prgkey needs to be numbered from 1 - 7 (or 1
-
#  2 on the 9112i), for example "prgkey2 type: speeddial".
#  Programmable keys can be set up as speeddials or as additional
#  call/line appearances or as feature keys and have a type, value
and
#  line associated with it as seen here in the default programmable
#  settings.

#  PRGKEY TYPES:  "speeddial", "blf", "list", "dnd"
#  PRGKEY VALUE:  If prgkey type is a speeddial, any DTMFs (from
#                 0 - 9, *, "#") or a comma (,) for 500ms pause and
#                 'E' for On-hook can be set for the value.
#                 If prgkey type is blf it is the extension you want
#                 to monitor.

# Speed Dials
prgkey1 type: speeddial
prgkey1 value: *8
prgkey2 type: speeddial
prgkey2 value: *69

# DND Key
```

```
#prgkey1 type: dnd

# blf
#prgkey2 type: blf
#prgkey2 value: 4559

# list
#prgkey1 type: list
#prgkey2 type: list
```

9133i Sample Configuration File

```
# Sample Configuration File
#= = = = = = = = = = = = = = = = = = = = = = = = = = = =

# Date: October 26th, 2005

# Phone Model: 9133i

# Notes:
#
# The general format used here is similar to configuration files
# used by several UNIX-based programs. Any text following a number
# sign (#) is considered to be a comment, unless the number sign is
# contained within double-quotes ("#") where it is considered to be
# a pound. For Boolean fields, 0 = false, 1 = true.

# Comments:
#
# This file contains sample configurations for the "aastra.cfg" or
# "<mac>.cfg" file.  The settings included here are examples only.
# You should change/comment the values to suit your requirements.
#
# Not all possible paramters are shown, refer to the admin guide
# for the full list of supported parameters, their defaults and
# valid ranges.
#
# The Aastra 480i, 480iCT, 9112i and 9133i phones will download 2
# configuration files from the TFTP server while restarting, the
# "aastra.cfg" file and  the "<mac>.cfg" file. These two
# configuration files can be used to configure all of the settings
# of the phone with the exception of assigning a static IP address
# to a phone and line settings, which should only be set in the
"<mac>.cfg" file.
#
```

```
# The "aastra.cfg" file configures the settings server wide, while
# the "<mac>.cfg" file configures only the phone with the MAC
# address for which the file is named (for example,
# "00085d0304f4.cfg"). The settings in the "aastra.cfg" file will
# be overridden by settings which also appear in the "<mac>.cfg"
file.
#--------------------------------------------------------------------


# DHCP Setting
# ============

#dhcp: 1 # DHCP enabled.

# DHCP:
# 0 = false, means DHCP is disabled.
# 1 = true, means DHCP is enabled.
#
# Notes:
#
#  DHCP is normally set from the Options list on the phone or
#  the web interface
#
#  If DHCP is disabled, the following network settings will
#  have to be configured manually either through the configuration
#  files, the Options List in the phone, or the Web Client: IP
#  Address (of the phone), Subnet Mask, Gateway, DNS, and TFTP
#  Server.
#--------------------------------------------------------------------


# Network Settings
# = = = = = = = = = =

# Notes: If DHCP is enabled, you do not need to set these network
# settings.  Although depending on you DHCP server configuration
# you may still have to set the dns address.
```

```
#ip:     # This value is unique to each phone on a server
         # and should be set in the "<mac>.cfg" file if
         # setting this manually.

#subnet mask:
#default gateway:
#dns1:
#dns2:



# Time Server Settings
# ====================

#time server disabled: 1  # Time server disabled.
#time server1:            # Enable time server and enter at
#time server2:            # least one time server IP address or
#time server3:         # qualified domain name.

# Time Server Disabled:
#   0 = false, means the time server is not disabled.
#   1 = true, means the time server is disabled.




# NAT Settings
# = = = = = = =

# Option 1:
#
#  If you are connecting to a Nortel MCS call server and there is a
#  NAT device between the server and the phone, then you must set
#  the following two parameters for the phone to function
#  correctly.

#sip nortel nat support: 1      # 1 = enabled
#sip nortel nat timer: 60       # seconds between keep alive
messages
```

```
# Option 2:
#
#  If you are using a session border controller, you should set the
#  outbound proxy to the session border controller address

#sip outbound proxy: sbc.aastra.com
#sip outbound proxy port: 0              # a value of 0 enables SRV
                                        # lookups for the address of
                                         # the proxy.


# Option 3:
#
#  If you know the public IP address of your NAT device and and have
#  opened up a port for the SIP messages then you can statically
#  assign this information.

#sip nat ip: 67.123.122.90
#sip nat port: 5890


# Additional Network Settings
# ============================

#sip rtp port: 3000     # Eg. RTP packets are sent to port 3000.



#------------------------------------------------------------------



# Configuration Server Settings
# = = = = = = = = = = = = = = =

# Notes: This section defines which server the phone retrieves new
# firmware images and configuration files from.  Three protocols
# are supported TFTP, FTP and HTTP

download protocol: TFTP   # valid values are TFTP, FTP and HTTP
```

```
## TFTP server settings
tftp  server: 192.168.0.130
#alternative tftp server:
#use alternative tftp server: 1  # If your DHCP server assigns
                                 # a TFTP server address which
                                 # you do not use, you can use
                                 # the alternative tftp server.

## FTP server settings
#ftp server: 192.168.0.131   # can be IP or FQDN
#ftp username: aastra
#ftp password: 480iaastra


## HTTP server settings (for http://bogus.aastra.com/firmware/)
#http server: bogus.aastra.com   # can be IP or FQDN
#http path: firmware


#-----------------------------------------------------------------


# Dial Plan Settings
# ===================
#
# Notes:
#
#  As you dial a number on the phone, the phone will initiate a call
#  when one of the following conditions are meet:
#
#   (1) The entered number is an exact match in the dial plan
#   (2) The "#" symbol has been pressed
#   (3) A timeout occurs
#
#  The dial plan is a regular expression that supports the
#  following:
#  syntax:
#
```

```
#    0,1,2,3,4,5,6,7,8,9,*,#  : matches the keypad symbols
#    x                        : matches any digit (0...9)
#    +                       : matches 0 or more repetitions of the
#                             :  previous expression
#    []                     : matches any number inside the brackets
#                           :  can be used with a "-" to represent a
#                             :  range
#    ()                       : expression grouping
#    |                        : either or
#
#
#  If the dialled number doesn't match the dial plan then the call
#  is rejected.

sip digit timeout: 3       # set the inter-digit timeout in seconds

# Example dial plans
sip dial plan: "x+#|xx+*"   # this is the default dial string, note
                          # that is must be quoted since it contains
                            # a '#' character



#sip dial plan: [01]xxx|[2-8]xxxx|91xxxxxxxxxx
                            # accecpt any 4 digit number beginning
                            # with a 0 or 1, any 5 digit number
                           # beginning with a number between 2 and 8
                            # (inclusive) or a 12 digit number
                            # beginning with 91



#sip dial plan terminator: 1   # enable sending of the "#" symbol to
                               # to the proxy in the dial string

#-----------------------------------------------------------------
```

```
# General SIP Settings
# ====================

#sip session timer: 30     # enable support of RFC4028, the default
                           # value of 0 disables this functionality

#sip transport protocol: 0  # use UDP (1), TCP (2) or both (0) for
                            # sip messaging

#sip use basic codecs: 1    # limit codecs to G711 and G729
#sip out-of-band dtmf: 0    # turn off support for RFC2833 (on by
                            # default)



# Global SIP User Settings
# ========================
#
# Notes:
#   These settings are used as the default configuration for the
#   hard key lines on the phone.  That is:
#
#     L1 to L4 on the 480i and 480iCT
#     L1 to L3 on the 9133i
#     L1 on the 9112i
#
#   These can be over-ridden on a per-line basis using the per-line
#   settings.
#
#   See the Admin Guide for a detailed explaination of how this
#   works

sip screen name: Joe Smith # the name display on the phone's screen
sip user name: 4256        # the phone number
sip display name: Joseph Smith # the caller name sent out when
making
                           # a call.
sip vmail: *78             # the number to reach voicemail on

sip auth name: jsmith      # account used to authenticate user
```

```
sip password: 12345          # password for authentication account

sip mode: 0                   # line type:
                              #   0 - generic,
                              #   1 - BroadSoft SCA line
                              #   2 - Nortel line

sip proxy ip: proxy.aastra.com # IP address or FQDN of proxy
sip proxy port: 5060          # port used for SIP messages on the
                              # proxy.  Set to 0 to enable SRV
                              # lookups

sip registrar ip: aastra.com  # IP address or FQDN of registrar
sip registrar port: 0         # as proxy port, but for the
registrar
sip registration period: 3600 # registration period in seconds


# Per-line SIP Settings
# =====================

# configure line 3 as the support Broadsoft SCA line
#   - the proxy and registrar settings are taken from the global
#     settings above

sip line3 screen name: Support
sip line3 user name: 4000
sip line3 display name: Aastra Support
sip line3 auth name: support
sip line3 password: 54321
sip line3 mode: 1
sip line3 vmail: *78


# configure line 5 (a soft key line) as an ordinary line
# of a test server

sip line5 screen name: Test 1
sip line5 user name: 5551001
sip line5 display name: Test 1
```

```
sip line5 auth name: 5551001
sip line5 password: 5551001
sip line5 mode: 0
sip line5 proxy ip: 10.50.10.102
sip line5 proxy port: 5060
sip line5 registrar ip: 10.50.10.102
sip line5 registrar port: 5060
sip line5 registration period: 60



#-------------------------------------------------------------------


# Programmable Key Settings
# ===========================


# Programmable keys can be set either server wide or unique to each
# phone.
# Setting programmable keys as line/call appearances should be done
# in the "<mac>.cfg" file, since these are unique to each phone.

# Notes:
#
#  There are a maximum of 7 programmable keys that can be configured
#  on the 9133i phone, and only 2 on the 9112i phone. These can be
#  set up through either of the 2 configuration files, depending on
#  whether this is to be server wide ("aastra.cfg") or phone
#  specific ("<mac>.cfg").  Each prgkey needs to be numbered from
#  1 - 7 (or 1 -2 on the 9112i), for example "prgkey2 type:
# speeddial". Programmable keys can be set up as speeddials or as
# additional call/line appearances or as feature keys and have a
# type, value and line associated with it as seen here in the
# default programmable settings.

#  PRGKEY TYPES:  "line", "speeddial", "blf", "list", "dnd"
#  PRGKEY VALUE:  If prgkey type is a speeddial, any DTMFs (from
#                 0 - 9, *, "#") or a comma (,) for 500ms pause and
#                 'E' for On-hook can be set for the value.
#                If prgkey type is blf it is the extension you want
#                 to monitor.
```

```
#  PRGKEY LINE:   This is line associated with the prgkey.  For line
#                 prgkeys the value must be between 4 and 9 (1 - 3
#                 are already hardcoded as the L1, L2 and L3 hard
#                 key line/call appearances).
#                 This parameter is not used for the 9112i

# Speed Dials
prgkey1 type: speeddial
prgkey1 value: *8
prgkey2 type: speeddial
prgkey2 value: *69

# DND Key
prgkey3 type: dnd

# Line appearance
prgkey4 type: line
prgkey4 line: 5

# blf
prgkey5 type: blf
prgkey5 value: 4559
prgkey5 line: 1

# list
prgkey6 type: list
prgkey7 type: list
```

Appendix D: Sample BLF Softkey Settings

Asterisk BLF

The following are sample softkey and programmable key configurations to enable Asterisk BLF support on Aastra IP phones.

480i and 480i CT Configuration Parameters for Asterisk BLF

```
softkey1 type: blf
softkey1 value: 9995551212
softkey1 label: John
softkey1 line: 1
```

9133i Configuration Parameters for Asterisk BLF

```
prgkey1 type: blf
prgkey1 value: 9995551212
prgkey1 label: John
prgkey1 line: 1

prgkey7 type: blf
prgkey7 value: 9995551313
prgkey7 label: Jane
prgkey7 line: 1
```

BroadSoft BroadWorks BLF

The following are sample softkey and programmable key configurations to enable Broadsoft BroadWorks Busy Lamp Field support on Aastra IP phones.

480i and 480i CT Configuration Parameters for Broadsoft BroadWorks BLF

Note: One softkey must be defined of type "list" for EACH monitored user. So if there are 2 users being monitored, 2 softkeys must be defined of type list.

```
softkey1 type: list
softkey1 label:
softkey1 value:
softkey1 line: 1
softkey2 type: list
softkey2 label:
softkey2 value:
softkey2 line: 1
list uri: sip:my480i-blf-list@as.broadsoft.com
```

9133i Configuration Parameters for Broadsoft BroadWorks BLF

Note: One prgkey must be defined of type "list" for each monitored user. So if there are 2 users being monitored, 2 prgkeys must be defined of type list.

```
prgkey6 type: list
prgkey6 label:
prgkey6 value: 1

prgkey7 type: list
prgkey7 label:
prgkey7 value: 1

list uri: sip:my9133i-blf-list@as.broadsoft.com
```

Appendix E: Sample Multiple Proxy Server Configuration

Multiple proxy servers can be configured in the *aastra.cfg* file or the *<mac>.cfg* file. In the example below, the default proxy setting is used if no specific setting is specified in the line configuration. Line2 and line3 are used for the global proxy configurations, while line1 and line4 use their own specific settings.

#sip settings
sip proxy ip: #.#.#.#
sip proxy port: 5060
sip registrar ip: #.#.#.#
sip registrar port: 5060
— sip registration period:3600
sip nortel nat support:0
sip nortel nat timer:0
sip broadsoft talk:0
sip broadsoft hold:0
sip broadsoft conference:0

sip dial plan: "x+#""

#line info
 # Fill in all necessary information below carefully. Populate all lines even if there is only
 # one account
#line 1
sip line1 auth name:
sip line1 password:
sip line1 mode: 0
sip line1 user name:
sip line1 display name:
sip line1 screen name:
sip line1 proxy ip: &.&.&.&
sip line1 proxy port: 5060
sip line1 registrar ip: #.#.#.#
sip line1 registrar port: 5060
sip registration period:600
sip nortel nat support:1
sip nortel nat timer:120
sip broadsoft talk:0
sip broadsoft hold:0
sip broadsoft conference:0

Continued.....

#line 2
sip line2 auth name:
sip line2 password:
sip line2 mode: 0
sip line2 user name:
sip line2 display name:
sip line2 screen name:

#line 3
sip line3 auth name:
sip line3 password:
sip line3 mode: 0
sip line3 user name:
sip line3 display name:
sip line3 screen name:

#line 4
sip line4 auth name:
sip line4 password:
sip line4 mode: 0
sip line4 user name:
sip line4 display name:
sip line4 screen name:
sip line4 proxy ip: %.%.%.%
sip line4 proxy port: 5060
sip line4 registrar ip: %.%.%.%
sip line4 registrar port: 5060
sip registration period:500
sip nortel nat support:0
sip nortel nat timer:0
sip broadsoft talk:1
sip broadsoft hold:1
sip broadsoft conference:1

Appendix F: How to Create an XML Application

This Appendix describes how to create an XML application for your IP phones. Sections in this appendix include:

- Creating XML Objects
- XML Schema File

XML format

The text in the Aastra XML objects must be compliant with XML recommendations and special characters must be escape encoded. The default character set for the XML API is ISO-8859-1.

Character	Description	Escape Sequence
&	Ampersand	&
"	Quote	"
'	Apostrophe	'
<	Left angle bracket	<
>	Right angle bracket	>

Creating XML Objects

This section describes how to create XML objects.

The Aastra IP phone XML API supports four proprietary objects that allow the creation of menu screens, message screens, input screens, and directory screens:

- "Text Menu Object (Menu Screens)"
- "Text Screen Object (Message Screens)"
- "UserInput Object (User Input Screens)"
- "Directory Object (Directory List Screen) (480i only)"

Creating Custom Softkeys

Developers can link arbitrary URIs to softkeys in the XML screens and can invoke softkey behavior to each XML screen type (Text Menu, Text Screen, User Input, Directory). A developer can define up to six softkeys before the closing tag of any object on the 480i/480i CT.

The following softkey functionality is available to the developer for the purpose of reordering or preserving the default functionality of the XML screens. The "Dial" function is available to screens that allow input. The dial string for the "Dial" function is taken from the menu items URI on the Menu Screen, and from the editor field input on the Input Screen.

Existing Action Keys	Text Screen	Menu Screen	Input Screen
Select		X	
Exit	X	X	X
Dial		X	X
Submit			X
Backspace			X
Nextspace			X
Dot			X
ChangeMode			X

Text Menu Object (Menu Screens)

The Text Menu object allows application developers to create a numerical list of menu items on the IP phones. The go-to line support, arrow indicator, and scroll key support are built into these objects, along with the "**Select**" and "**Done**" soft keys. The Text Menu object allows users to navigate the application, by linking HTTP requests to menu items.

Text Menu Object Implementation

The following is how you would implement the Text Menu object.

Note: For all available parameters you can use for the Text Menu object, and for an explanation of each parameter, see Aastra Telecom's "XML Developer's Guide".

Default Softkeys:

- 1=Select
- 6=Done

XML Description:

```
<AastraIPPhoneTextMenu
    defaultIndex = "some integer"
    destroyOnExit = "yes/no"
    >
       <Title>Menu Title</Title>
       <MenuItem base ="http://base/">
       <Prompt>First Choice</Prompt>
       <URI>http://somepage.xml</URI>
       <Selection></Selection>
       </MenuItem>
       <!-Additional Menu Items may be added -->
       <!-Additional Softkey Items may be added -->
</AastraIPPhoneTextMenu>
```

XML example:

```
<AastraIPPhoneTextMenu>
       <Title>Phone Services</Title>
       <MenuItem base = "http://10.50.10.53/">
           <Prompt>Traffic Reports</Prompt>
           <URI> rss_to_xml.pl</URI>
       </MenuItem>
       <MenuItem>
           <Prompt>Employee List</Prompt>
           <URI>employees.xml</URI>
       </MenuItem>
<MenuItem base ="">
           <Prompt>Weather</Prompt>
           <URI>http://10.50.10.52/weather.pl</URI>
       </MenuItem>
</AastraIPPhoneTextMenu>
```

XML Screen Example:

Note: The maximum number of items to be included in a Text Menu object is 15.

Text Screen Object (Message Screens)

The screen object can be used to display text. The screen word wraps appropriately and can scroll to display a message longer then four lines.

Note: For all available parameters you can use for the Text Screen object, and for an explanation of each parameter, see Aastra Telecom's "XML Developer's Guide".

Text Screen Object Implementation

The following is how you would implement the Text Screen object.

Softkey:

- 6=Done

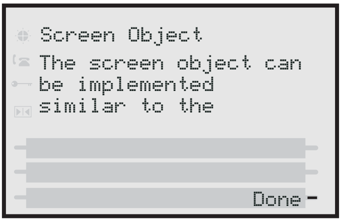
XML Description:

```
<AastraIPPhoneTextScreen
    destroyOnExit = "yes/no"
    >
<Title>Screen Title</Title>
<Text>The screen text goes here</Text>
</AastraIPPhoneTextScreen>
```

XML example:

```
<AastraIPPhoneTextScreen>
       <Title>Screen Object</Title>
       <Text>The screen object can be implemented similar to the
firmware info screen. Note that white space is preserved in XML
so the display should word-wrap appropriately. Only three lines
can display at a time.</Text>
</AastraIPPhoneTextScreen>
```

XML Screen Example:



UserInput Object (User Input Screens)

The UserInput object allows application developers to be able to input text on the phone screen where applicable. (Line 1 is a title, Line 4 is an input prompt, and Line 5 is an input field). The IP phones support three parameter types: IP Addresses, Numbers (integers), and Strings. Each parameter has a URL tag that is used to send information back to the HTTP server. The label in the parameter tag is appended to the address in the URL tag and sent via HTTP GET.

UserInput Object Implementation (IP Addresss)

The following is how you would implement the UserInput object using an IP Address.

Note: For all available parameters you can use for the UserInput object, and for an explanation of each parameter, see Aastra Telecom's "XML Developer's Guide".

Softkeys:

- 1=Backspace,
- 2=Dot,
- 3=ChangeCase,
- 4=Numeric/Alpha,
- 5=Cancel,
- 6=Done

XML Description:

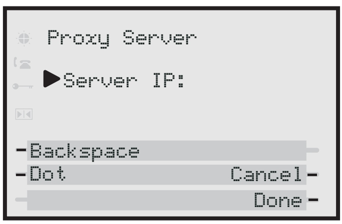
```
<AastraIPPhoneInputScreen type = "IP/string/number" password = "yes/no"
destroyOnExit = "yes/no">
<!-password attribute is optional and set to "no" by defaultà
<!-destroyOnExit is optional and "no" by default à
       <Title>Title string, usually same as menu title</Title>
       <Prompt>Enter IP address or host name</Prompt>
       <URL>Target receiving the input</URL>
       <Parameter>parameter added to URL</Parameter>
       <Default />
       <SoftKey index = "1">
           <Label> Backspace </Label>
           <URI>SoftKey:Exit</URI>
       </Softkey>
       <SoftKey index = "2">
           <Label> Dot </Label>
           <URI>SoftKey:Exit</URI>
       </Softkey>
       <SoftKey index = "3">
           <Label> ChangeCase </Label>
           <URI>SoftKey:Exit</URI>
       </Softkey>
       <SoftKey index = "4">
           <Label> Numeric/Alpha </Label>
           <URI>SoftKey:Exit</URI>
```

```
        </Softkey>
        <SoftKey index = "5">
            <Label> Cancel </Label>
            <URI>SoftKey:Exit</URI>
        </Softkey>
        <SoftKey index = "6">
            <Label> Done </Label>
            <URI>SoftKey:Exit</URI>
        </Softkey>
</AastraIPPhoneInputScreen>
```

XML Example:

```
<AastraIPPhoneInputScreen type = "IP">
        <Title>Proxy Server</Title>
        <Prompt>Server IP:</Prompt>
        <URL>http://10.50.10.53/script.pl</URL>
        <Parameter>proxy</Parameter>
        <Default></Default>
<AastraIPPhoneInputScreen>
```

XML Screen Example:



UserInput Object Implementation (Number)

The following is how you would implement the UserInput object using Numbers.

Softkeys:

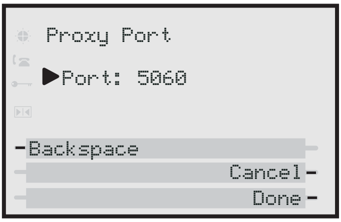
- 1=Backspace,
- 5=Cancel,
- 6=Done

XML Example:

```
<AastraIPPhoneInputScreen type = "number">
```

```
        <Title>Proxy Port</Title>
        <Prompt>Port:</Prompt>
        <URL>http://10.50.10.53/script.pl</URL>
        <Parameter>port</Parameter>
        <Default>5060</Default>
<AastraIPPhoneInputScreen>
```

__XML Screen Example__:



__UserInput Object Implementation__ __(String)__

The following is how you would implement the UserInput object using Strings in XML.

__Implementation__ __(String)__

The following is how you would implement the UserInput object using String.

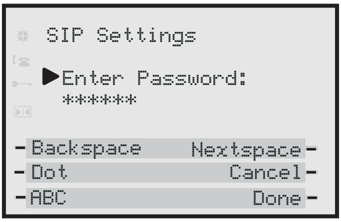
__Softkeys__:

- 1=Backspace,
- 2=Dot,
- 3=Tri-Mode key,
- 4=Nextspace,
- 5=Cancel,
- 6=Done

__XML Example:__

```
<AastraIPPhoneInputScreen type = "string" password = "yes">
        <Title>SIP Settings</Title>
        <Prompt>Enter something</Prompt>
        <URL>http://10.50.10.53/script.pl</URL>
        <Parameter>passwd</Parameter>
        <Default></Default>
<AastraIPPhoneInputScreen>
```

__XML Screen Example:__



Note: In the above example, if user entered 12345, then the URL sent back to the server is http://10.50.10.53/script.pl?passwd=12345.

Directory Object (Directory List Screen) *(480i only)*

The Directory object allows you to browse an online directory in real time. It displays an automatically numbered list of contacts. By selecting a contact with the cursor, the contact can be dialed directly by pressing the "Dial" softkey or picking up the receiver. The Directory object has the optional softkeys of "Previous" and "Next" which can be linked to other XML objects.

Directory Object Implementation

The following is how you would implement the Directory object in XML.

Notes: 1. For all available parameters you can use for the Directory object, and for an explanation of each parameter, see Aastra Telecom's "XML Developer's Guide".

2.If the URI entry contains a "?" the phone appends an "&" instead.

Softkeys:

- 1=Dial,
- 6= Done,
- 2=Previous (optional),
- 5=Next (optional)

XML Description:

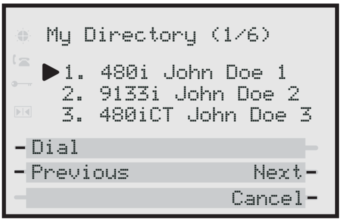
```
<AastraIPPhoneDirectory destroyOnExit="no" next="uri" previous="uri">
<!-Attributes are optionalà
        <Title>Directory Title</Title>
        <Menu Item>
            <Prompt>Contact Name</Prompt>
            <URI>number</URI>
        </Menu Item>
        <!-Additional Menu Items may be added -->
</AastraIPPhoneDirectory>
```

XML Example:

```
<AastraIPPhoneDirectory next="more.xml" previous="back.xml">
        <Title>My Directory</Title>
        <MenuItem>
            <Prompt>480i - John Doe 1</Prompt>
            <URI> 10.50.10.49</URI>
        </MenuItem>
        <MenuItem>
            <Prompt>9133i - John Doe 2</Prompt>
            <URI>4326</URI>
        </MenuItem>
```

```
<MenuItem>
                <Prompt>480i CT - John Doe 3</Prompt>
                <URI>9982691234</URI>
          </MenuItem>
</AastraIPPhoneDirectory
```

XML Screen Example:



Note: The maximum number of items to be included in a Directory object is 15 per page. In this example, there are six pages.

HTTP Post

In addition to initiating a request to an XML application from the Services menu, an HTTP server can push an XML object to the phone via HTTP Post. The phone parses this object immediately upon receipt and displays the information to the screen.

The HTTP post packet must contain an "xml=" line in the message body. The string to parse is located after the equals sign in the message. HTML forms that post objects to the phone must use a field named "xml" to send their data. See the following examples (Example 1 and Example 2) for a sample HTTP post packet and php source code.

Example 1:

```
POST / HTTP/1.1
Accept: image/gif, image/x-xbitmap, image/jpeg, image/pjpeg,
      application/vnd.ms-powerpoint,
      application/vnd.ms-excel, application/msword,
      application/x-shockwave-flash, */*
Referer: http://10.50.10.53
Accept-Language: en-us..Content-Type: application/x-www-form-urlencoded
Accept-Encoding: gzip, deflate..User-Agent: Mozilla/4.0
            (compatible;MSIE 6.0;
            Windows NT 5.0; .NET CLR 1.1.4322)
Host: 10.50.10.49
Content-Length: 194..Connection: Keep-Alive
Cache-Control: no-cache..Authorization: Basic YWRtaW46MjIyMjI=
xml=%3CAastraIPPhoneTextScreen%3E%
   %3CTitle%3E480i+Tester%3C%2FTitle%3E

%3CText%3EMessage+to+go+on+phone.++Limit+to+512+bytes.%3C%2FText%3E
   %2FAastraIPPhoneTextScreen%3E%
```

Note: The XML object cannot be larger than 2150 bytes. Any posts larger than this limit are denied.

Example 2:

Below is a sample php source code which sends an XML object to an Aastra phone.

```
<?php
#
function push2phone($server,$phone,$data)
{
# url-encode the xml object
$xml = "xml=".urlencode($data);

$post = "POST / HTTP/1.1\r\n";
$post .= "Host: $phone\r\n";
$post .= "Referer: $server\r\n";
$post .= "Connection: Keep-Alive\r\n";
$post .= "Content-Type: application/x-www-form-urlencoded\r\n";
$post .= "Content-Length: ".strlen($xml)."\r\n\r\n";

$fp = @fsockopen ( $phone, 80, $errno, $errstr, 5);
if($fp)
{
@fputs($fp, $post.$xml);
flush();
fclose($fp);
}
}

##############################

$xml = "<AastraIPPhoneTextScreen>\n";
$xml .= "<Title>Push test</Title>\n";
$xml .= "<Text>This is a test for pushing a screen to a phone </Text>\n";
$xml .= "</AastraIPPhoneTextScreen>\n";
push2phone("172.16.96.63',"172.16.96.75",$xml);
?>
```

XML Schema File

After creating an XML application for your IP phone, you can validate the XML objects using the Schema file provided in this section. This helps you find any parsing errors that may exist, and verify that your XML objects conform to the Aastra API.

Note: Aastra IP phonees do not contain validating XML parsers. There are many free XML
validators available on the Web (i.e., http://apps.gotdotnet.com/xmltools/xsd-validator/Default.aspx) that can perform validation using the schema file.

XML Schema

```
<?xml version="1.0" encoding="ISO-8859-1" ?>
<xs:schema xmlns:xs="http://www.w3.org/2001/XMLSchema">


<xs:element name="AastraIPPhoneTextScreen">
 <xs:complexType>
  <xs:sequence>
   <xs:element name="Title" type="xs:string" />
   <xs:element name="Text">
    <xs:simpleType>
     <xs:restriction base="xs:string">
      <xs:minLength value="1" />
      <xs:maxLength value="1000" />
     </xs:restriction>
    </xs:simpleType>
   </xs:element>
  </xs:sequence>
  </xs:complexType>
 </xs:element>

<xs:element name="AastraIPPhoneTextMenu">
 <xs:complexType>
  <xs:sequence>
   <xs:element name="Title" type="xs:string" />
   <xs:element name="MenuItem" minOccurs="1" maxOccurs="15">
    <xs:complexType>
     <xs:sequence>
      <xs:element name="Prompt" type="xs:string" />
      <xs:element name="URI" type="xs:string" />
     </xs:sequence>
     <xs:attribute name="base" type="xs:string" />
    </xs:complexType>
   </xs:element>
  </xs:sequence>
  <xs:attribute name="destroyOnExit" default="no">
   <xs:simpleType>
    <xs:restriction base="xs:string">
     <xs:pattern value="yes|no" />
    </xs:restriction>
   </xs:simpleType>
  </xs:attribute>
 </xs:complexType>
</xs:element>

<xs:element name="AastraIPPhoneInputScreen">
 <xs:complexType>
```

```
<xs:sequence>
  <xs:element name="Title" />
  <xs:element name="Prompt" />
  <xs:element name="URL" />
  <xs:element name="Parameter" />
  <xs:element name="Default" />
 </xs:sequence>
 <xs:attribute name="type">
  <xs:simpleType>
   <xs:restriction base="xs:string">
    <xs:pattern value="IP|string|number" />
   </xs:restriction>
  </xs:simpleType>
 </xs:attribute>
 <xs:attribute name="password" default="no">
  <xs:simpleType>
   <xs:restriction base="xs:string">
    <xs:pattern value="yes|no" />
   </xs:restriction>
  </xs:simpleType>
 </xs:attribute>
 <xs:attribute name="destroyOnExit" default="no">
  <xs:simpleType>
   <xs:restriction base="xs:string">
    <xs:pattern value="yes|no" />
   </xs:restriction>
  </xs:simpleType>
 </xs:attribute>
 </xs:complexType>
</xs:element>

<xs:element name="AastraIPPhoneDirectory">
 <xs:complexType>
  <xs:sequence>
   <xs:element name="Title" type="xs:string" />
   <xs:element name="MenuItem" minOccurs="1" maxOccurs="15">
    <xs:complexType>
     <xs:sequence>
      <xs:element name="Prompt" type="xs:string" />
      <xs:element name="URI" type="xs:string" />
     </xs:sequence>
    </xs:complexType>
   </xs:element>
  </xs:sequence>
  <xs:attribute name="destroyOnExit" default="no">
   <xs:simpleType>
    <xs:restriction base="xs:string">
     <xs:pattern value="yes|no" />
    </xs:restriction>
   </xs:simpleType>
  </xs:attribute>
  <xs:attribute name="next" type="xs:string" />
  <xs:attribute name="previous" type="xs:string" />
 </xs:complexType>
</xs:element>

</xs:schema>
```

Limited Warranty

Aastra Telecom warrants this product against defects and malfunctions during a one (1) year period from the date of original purchase. If there is a defect or malfunction, Aastra Telecom shall, at its option, and as the exclusive remedy, either repair or replace the telephone set at no charge, if returned within the warranty period.

If replacement parts are used in making repairs, these parts may be refurbished, or may contain refurbished materials.

If it is necessary to replace the telephone set, it may be replaced with a refurbished telephone of the same design and color. If it should become necessary to repair or replace a defective or malfunctioning telephone set under this warranty, the provisions of this warranty shall apply to the repaired or replaced telephone set until the expiration of ninety (90) days from the date of pick up, or the date of shipment to you, of the repaired or replacement set, or until the end of the original warranty period, whichever is later. Proof of the original purchase date is to be provided with all telephone sets returned for warranty repairs.

Exclusions

Aastra Telecom does not warrant its telephone sets to be compatible with the equipment of any particular telephone company. This warranty does not extend to damage to products resulting from improper installation or operation, alteration, accident, neglect, abuse, misuse, fire or natural causes such as storms or floods, after the telephone is in your possession.

Aastra Telecom shall not be liable for any incidental or consequential damages, including, but not limited to, loss, damage or expense directly or indirectly arising from the customers use of or inability to use this telephone, either separately or in combination with other equipment. This paragraph, however, shall not apply to consequential damages for injury to the person in the case of telephones used or bought for use primarily for personal, family or household purposes.

This warranty sets forth the entire liability and obligations of Aastra Telecom with respect to breach of warranty, and the warranties set forth or limited herein are the sole warranties and are in lieu of all other warranties, expressed or implied, including warranties or fitness for particular purpose and merchantability.

Warranty Repair Services

Should the set fail during the warranty period;

In North America, please call 1-800-574-1611 for further information.

Outside North America, contact your sales representative for return instructions.

You will be responsible for shipping charges, if any. When you return this telephone for warranty service, you must present proof of purchase.

After Warranty Service

Aastra Telecom offers ongoing repair and support for this product. This service provides repair or replacement of your Aastra Telecom product, at Aastra Telecom's option, for a fixed charge. You are responsible for all shipping charges. For further information and shipping instructions;

In North America, contact our service information number: 1-800-574-1611.
Outside North America, contact your sales representative.

Note: Repairs to this product may be made only by the manufacturer and its authorized agents, or by others who are legally authorized. This restriction applies during and after the warranty period. Unauthorized repair will void the warranty.

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