MODEL
53i, 55i, 57i, 57i CT

SIP IP PHONE

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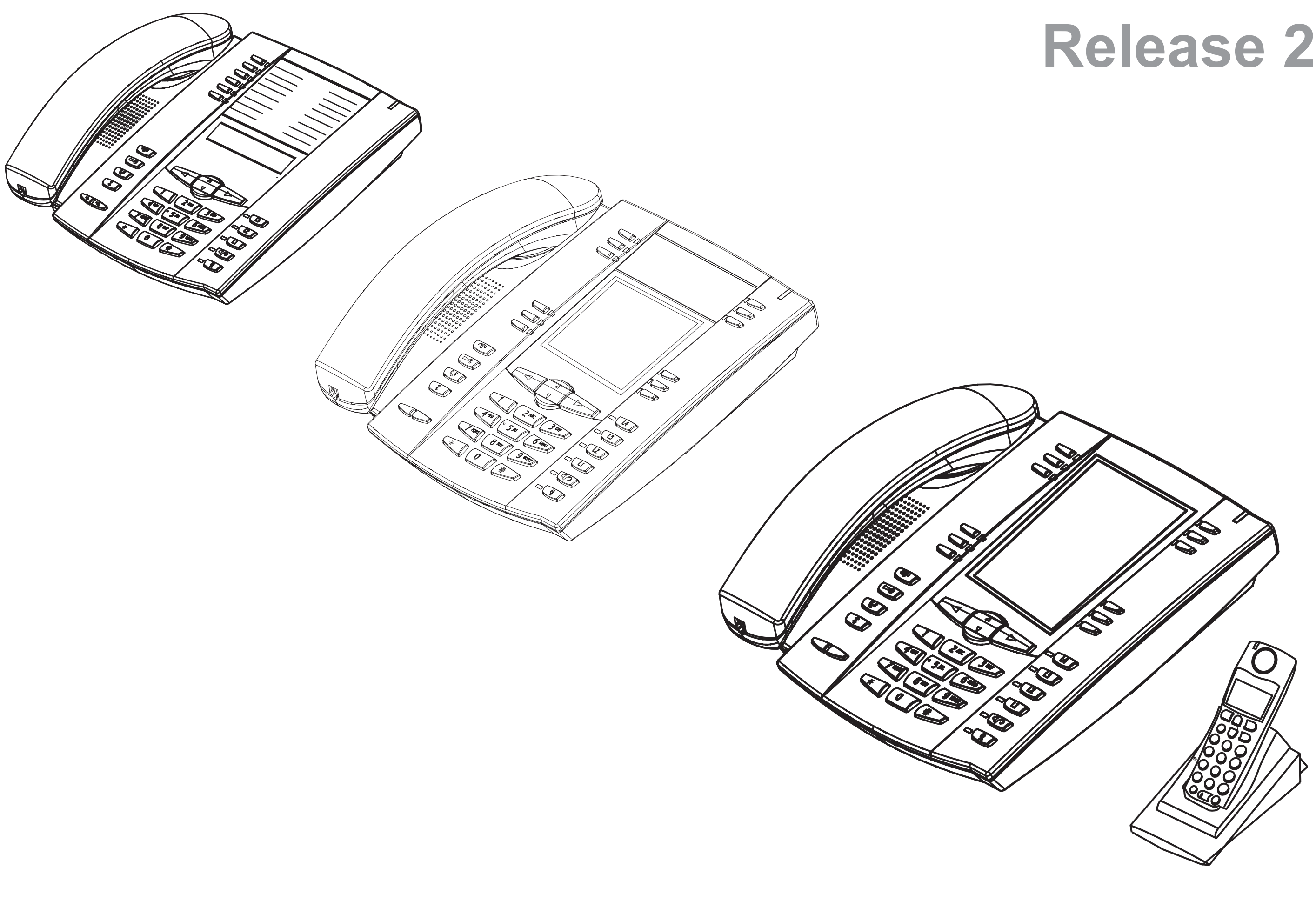
Administrator Guide
Release 2.1



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Preface

About this guide

Introduction

This *SIP IP Phone Administrator Guide* provides information on the basic network setup, operation, and maintenance of the IP phones, Models 53i, 55i, 57i, and 57i Cordless (57i CT). 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.

Audience

This guide is for network administrators, system administrators, developers and partners who need to understand how to operate and maintain 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.

Other Documentation

The IP phone documentation consists of:

- ***<Model-specific> SIP IP Phone Installation Guide*** – contains installation and set-up instructions, information on general features and functions, and basic options list customization. Included with the phone.
- ***Model 53i, 55i, 57i, 57i CT SIP IP Phone Administrator 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.
- ***<Model-specific> SIP IP Phone User Guides*** – explains the most commonly used features and functions for an end user.

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

Chapters and appendixes in this guide

This guide contains the following chapters and appendixes:

For	Go to
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Chapter 1
Overview

About this chapter

Introduction

This chapter briefly describes the IP Phone Models, and provides information about installing the IP phone firmware. It also describes the firmware and configuration files that the IP phone models use for operation.

Topics

This chapter covers the following topics:

IP Phone Models

Description

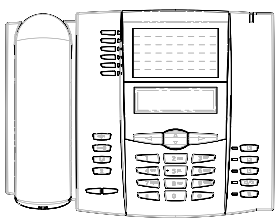
The IP Phone Models 53i, 55i, 57i, and 57i CT communicate over an IP network allowing you to receive and place calls in the same manner as a regular business telephone.

All phone models support the Session Initiation Protocol (SIP). The 57i CT offers the base phone along with a cordless extension.

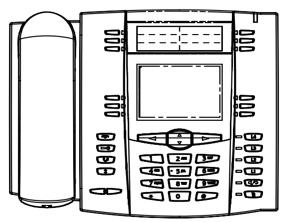
References

For more information about the features and installation requirements, see the *SIP IP Phone Installation Guide* for your specific model.

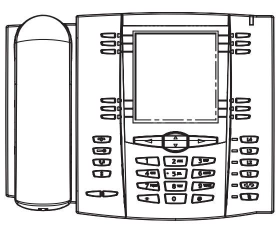
The following illustration shows the types of IP Phone Models.



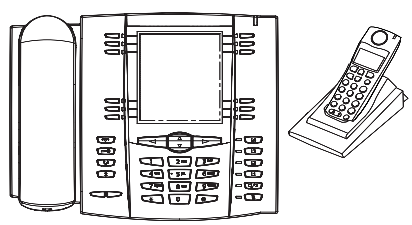
53i



55i



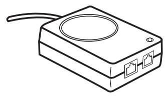
57i



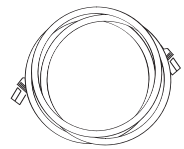
57i CT

Optional Accessories

The following are optional accessories for the IP Phones.



Power over Ethernet (PoE) Inline Power Injector



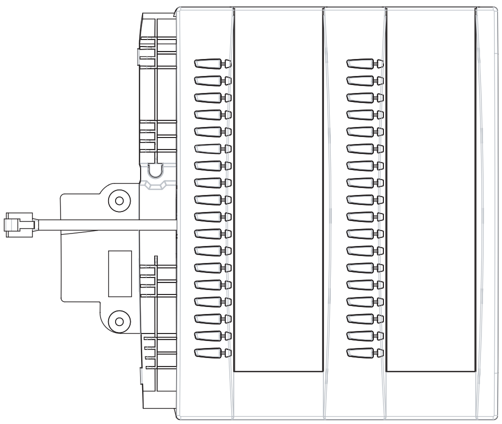
Additional Ethernet Cable (category 5/5e straight through cable)

A Power over Ethernet (PoE) inline power injector supplies 48V power to the IP phone through the Ethernet cable on pins 4 & 5 and 7 & 8.

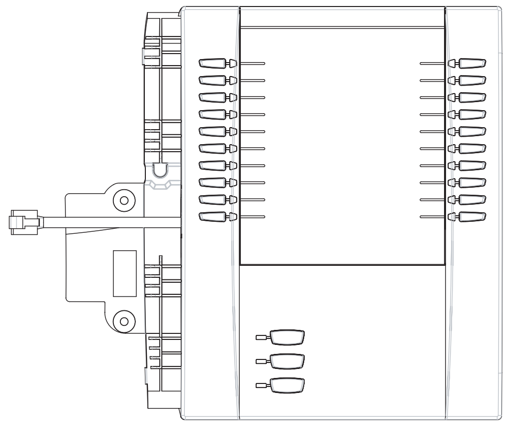
Warning: Do not use this inline PoE power injector to power other devices. See your phone-specific Installation Guide for more information.

Reference

For more information about installing the PoE and additional Ethernet cable, see your phone-specific Installation Guide.



536M Expansion Module for 53i, 55i, 57i, and 57i CT



560M Expansion Module for 55i, 57i, and 57i CT

The 536M module adds 36 additional softkeys to the IP phone models 53i, 55i, 57i, and 57i CT. The 536M provides paper labels for each softkey. Up to 3 modules can be piggy-backed to provide up to 108 additional softkeys for the phone.

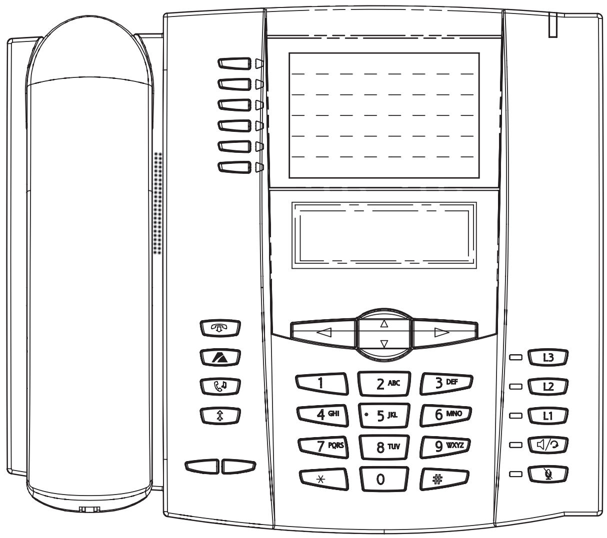
The 560M module adds 60 additional softkeys to the IP phone models 55i, 57i, and 57i CT (using the 3 function keys on the bottom right of the unit). The 560M module provides an LCD display for display softkey labels. Up to 3 modules can be piggy-backed to provide up to 180 additional softkeys for the phone.

Reference

For more information about installing and using the expansion modules, see your phone-specific Installation Guide and phone-specific User Guide.

Model 53i IP Phone

This section provides brief information about the Model 53i IP Phone. It includes a list of features, and describes the hard keys and default programmable keys on the 53i.

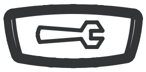
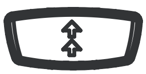


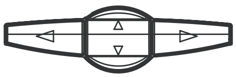
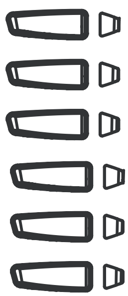
53i Phone Features

- 3-line LCD screen
- 6 top keys: 4 keys are programmable
- 3 call appearance lines with LEDs
- Supports up to 9 call lines
- Full-duplex speakerphone for handsfree calls
- Headset support (modular connector)
- Built-in two-port, 10/100 Ethernet ports - lets you share a connection with your computer
- AC power adapter (included)
- Enhanced busy lamp fields*
- Set paging*

*Availability of feature dependant on your phone system or service provider.

53i Key Descriptions*

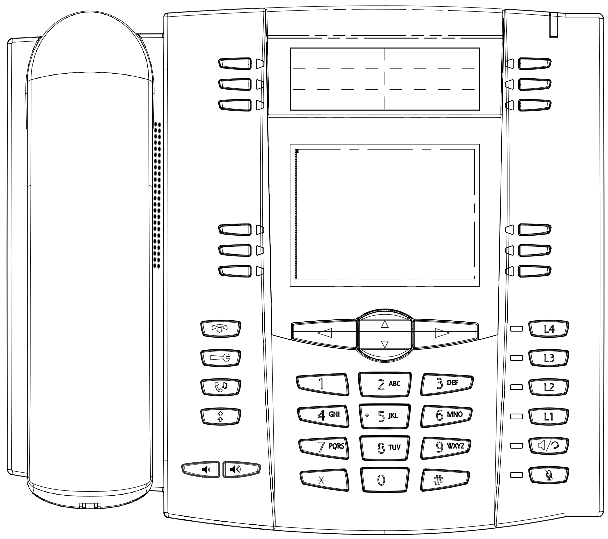
Keys	Key Description
	**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
	**Options key** - Accesses options to customize your phone. Your System Administrator may have already customized some of your settings. Check with your System Administrator before changing the administrator-only options.
	**Hold key** - Places an active call on hold. To retrieve a held call, press the call appearance button beside the light that is flashing.
	**Redial key** - Redials up to 100 previously dialed numbers. Pressing the Redial key twice simultaneously redials the last dialed number.
	**Volume control key** - Adjusts the volume for the handset, headset, ringer, and handsfree speaker.
  	**Line/Call Appearance key** - Connects you to a line or call. The Aastra 53i IP phone supports up to 3 line keys.
	**Handsfree key** - Activates Handsfree for making and receiving calls without lifting the handset. When the audio mode option is set, this key is used to switch between a headset and the handsfree speakerphone.
	**Mute key** - Mutes the microphone so that your caller cannot hear you (the light indicator flashes when the microphone is on mute).

Keys	Key Description
	**Navigation keys** - Pressing the UP and DOWN arrow keys lets you view different status and text messages on the LCD display (if there is more than 1 line of status/text messages). These buttons also let you scroll through menu selections, such as the Options List. Pressing the LEFT and RIGHT arrow keys lets you view the different line/ call appearances. While in the Options List, these keys allow you to exit or enter the current option. When you are editing entries on the display, pressing the LEFT arrow key erases the character on the left; pressing the RIGHT arrow key sets the option.
	**Programmable keys** - 6 Top Keys: 4 keys are programmable. Keys 1 and 2 are hardcoded as the SAVE and DELETE keys, respectively, and cannot be altered. The following are the default functions for the programmable keys on the 53i IP phone: **1** - **SAVE** (hardcoded) — Allows you to save numbers and/or names to the Directory. Using this key, you enter the number, name, and line (or speeddial key) to record in the Directory List. **2** - **DELETE** (hardcoded) — Allows you to delete a single entry or all entries from the Directory List and Callers List. **3** - **DIRECTORY** — Displays up to 200 names and phone numbers (stored in alphabetical order). **4** - **CALLERS LIST** — Accesses the last 200 calls received. **5** - **TRANSFER** — Transfers the active call to another number. **6** - **CONFERENCE** — Begins a conference call with the active call. **Note**: For more information about programming keys 3, 4, 5, and 6 to perform specific functions, see Chapter 5, “Configuring Operational Features” the section, “Softkeys/Programmable Keys/Feature Keys/ Expansion Module Keys” on page 5-93.

*See the Aastra 53i User Guide for more information about each of these keys.

Model 55i IP Phone

This section provides brief information about the Model 55i IP Phone. It includes a list of features, and describes the hard keys, default programmable keys, and default softkeys on the 55i.



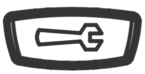
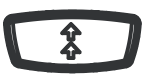
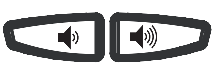
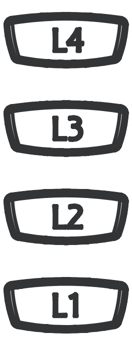
55i Phone Features

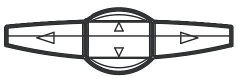
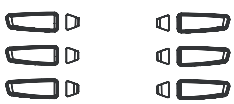
- 8 line graphical LCD screen (144 x 75 pixels) with white backlight
- 12 programmable keys
- 6 Top keys:Programmable hard keys (up to 6 programmable functions)
- 6 Bottom keys:Programmable state-based softkeys (up to 20 programmable functions)
- 4 call appearance lines with LEDs
- Supports up to 9 call lines
- Full-duplex speakerphone for handsfree calls
- Headset support (modular connector)
- Built-in-two-port, 10/100 Ethernet switch - lets you share a connection with your computer.

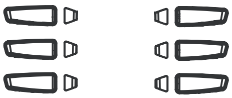
- Inline power support (based on 802.3af standard) which eliminates power adapters.
- AC power adapter (included)
- Enhanced busy lamp fields*
- Set paging*

*Availability of feature dependant on your phone system or service provider.

55i Key Descriptions*

Keys	Key Description
	**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
	**Options key** - Accesses options to customize your phone. Your System Administrator may have already customized some of your settings. Check with your System Administrator before changing the administrator-only options.
	**Hold key** - Places an active call on hold. To retrieve a held call, press the call appearance button beside the light that is flashing.
	**Redial key** - Redials up to 100 previously dialed numbers. Pressing the Redial key twice simultaneously redials the last dialed number.
	**Volume control key** - Adjusts the volume for the handset, headset, ringer, and handsfree speaker.
	**Line/Call Appearance key** - Connects you to a line or call. The Aastra 55i IP phone supports up to 4 line keys.

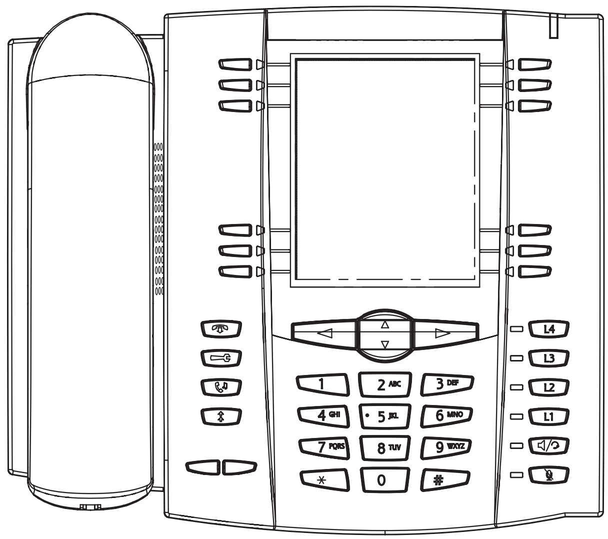
Keys	Key Description
	**Handsfree key** - Activates Handsfree for making and receiving calls without lifting the handset. When the audio mode option is set, this key is used to switch between a headset and the handsfree speakerphone.
	**Mute key** - Mutes the microphone so that your caller cannot hear you (the light indicator flashes when the microphone is on mute).
	**Navigation keys** - Pressing the UP and DOWN arrow keys lets you view different status and text messages on the LCD display (if there is more than 1 line of status/ text messages). These buttons also let you scroll through menu selections, such as the Options List. Pressing the LEFT and RIGHT arrow keys lets you view the different line/call appearances. While in the Options List, these keys allow you to exit or enter the current option. When you are editing entries on the display, pressing the LEFT arrow key erases the character on the left; pressing the RIGHT arrow key sets the option.
	**Programmable keys** - 6 Top keys: programmable hard keys (up to 6 programmable functions) By default, the top keys 1 through 4 are assigned as Services, Directory, Callers List, and Intercom, respectively. Keys 5 and 6 have no assigned functions. All 6 keys are programmable and can be assigned to perform specific functions. The following are the default functions for the programmable keys on the 55i IP phone: **1** - **SERVICES** Accesses enhanced features and services such as XML applications and voicemail, provided by third parties. **2** - **DIRECTORY** Displays up to 200 names and phone numbers (stored in alphabetical order). **3** - **CALLERS LIST** Accesses the last 200 calls received. **4** - **ICOM** Accesses another extension on the network. **5** - **NONE** No assigned function. **6 - NONE** No assigned function. **Note**: For more information about configuring the programmable keys 1 through 6 to perform specific functions, see Chapter 5, “Configuring Operational Features” the section, “Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys” on page 5-93.

Keys	Key Description
	**Softkeys** - 6 Bottom keys: programmable state-based softkeys (up to 20 programmable functions). By default, keys 1 through 6 have no assigned functions. You can configure all 6 bottom softkeys to perform specific functions on the 55i IP phone. However, after you lift the handset, there are specific static softkeys that display that cannot be changed. These are as follows: **1** - **DIAL** Allows you to dial out on the phone. **2** - **CONF** Begins a conference call with the active phone. **3** - **XFER** Transfers the active call to another number. **Note**: For more information about configuring softkeys 1 through 6 to perform specific functions, see Chapter 5, "Configuring Operational Features" the section, "Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys" on page 5-93.

*See the Aastra 55i User Guide for more information about each of these keys.

Model 57i and 57i CT IP Phone

This section provides brief information about the Model 57i IP Phone. It includes a list of features, and describes the hard keys and default softkeys on the 57i.



57i and 57i CT Phone Features

- 11 line graphical LCD screen (144 x 128 pixels) with white backlight
- 12 multi-functional softkeys

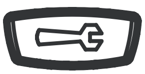
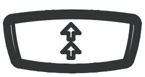
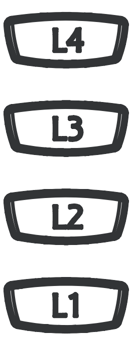
  - 6 Top Keys: programmable static softkeys (up to 10 programmable functions)

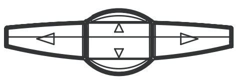
  - 6 Bottom Keys: programmable state-based softkeys (up to 20 programmable functions)
- 4 call appearance lines with LEDs
- Supports up to 9 call lines
- Full-duplex speakerphone for handsfree calls
- Headset support (modular connector)
- Built-in-two-port, 10/100 Ethernet switch - lets you share a connection with your computer.

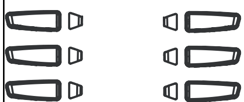
- Inline power support (based on 802.3af standard) which eliminates power adapters.
- AC power adapter (included)
- Enhanced busy lamp fields*
- Set paging*

*Availability of feature dependant on your phone system or service provider.

57i and 57i CT Key Descriptions*

Keys	Key Description
	**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
	**Options key** - Accesses options to customize your phone. Your System Administrator may have already customized some of your settings. Check with your System Administrator before changing the administrator-only options.
	**Hold key** - Places an active call on hold. To retrieve a held call, press the call appearance button beside the light that is flashing.
	**Redial key** - Redials up to 100 previously dialed numbers. Pressing the Redial key twice simultaneously redials the last dialed number.
	**Volume control key** - Adjusts the volume for the handset, headset, ringer, and handsfree speaker.
	**Line/Call Appearance key** - Connects you to a line or call. The Aastra 57i IP phone supports up to 4 line keys.

Keys	Key Description
	**Handsfree key** - Activates Handsfree for making and receiving calls without lifting the handset. When the audio mode option is set, this key is used to switch between a headset and the handsfree speakerphone.
	**Mute key** - Mutes the microphone so that your caller cannot hear you (the light indicator flashes when the microphone is on mute).
	**Navigation keys** - Pressing the UP and DOWN arrow keys lets you view different status and text messages on the LCD display (if there is more than 1 line of status/ text messages). These buttons also let you scroll through menu selections, such as the Options List. Pressing the LEFT and RIGHT arrow keys lets you view the different line/call appearances. While in the Options List, these keys allow you to exit or enter the current option. When you are editing entries on the display, pressing the LEFT arrow key erases the character on the left; pressing the RIGHT arrow key sets the option.

Keys	Key Description
	**Softkeys** - 12 softkeys on the 57i IP Phone. - 6 Top Keys: programmable static softkeys (up to 10 programmable functions) - 6 Bottom Keys: programmable state-based softkeys (up to 20 programmable functions) By default, the top softkeys 1 through 4 are assigned as Services, Directory, Callers List, and Intercom, respectively. Keys 5 and 6 have no assigned functions. All 6 keys are programmable and can be assigned to perform specific functions. The following are the default functions for the top softkeys on the 57i IP phone: **1** - **SERVICES** Accesses enhanced features and services such as XML applications and voicemail, provided by third parties. **2** - **DIRECTORY** Displays up to 200 names and phone numbers (stored in alphabetical order). **3** - **CALLERS LIST** Accesses the last 200 calls received. **4** - **ICOM** Accesses another extension on the network. **5** - **NONE** No assigned function. **6 - NONE** No assigned function. By default, the bottom softkeys 7 through 12 have no assigned functions. You can configure all 6 bottom softkeys to perform specific functions on the 57i IP phone. However, after you lift the handset, there are specific static softkeys that display that cannot be changed. These are as follows: **7**- **DIAL** Allows you to dial out on the phone. **8**- **CONF** Begins a conference call with the active phone. **9**- **XFER** Transfers the active call to another number. **Note**: For more information about programming the softkeys to perform specific functions, see Chapter 5, “Configuring Operational Features” the section, “Softkeys/ Programmable Keys/Feature Keys/Expansion Module Keys” on page 5-93.

*See the Aastra 57i or 57i CT User Guide for more information about each of these keys.

57i CT Cordless Handset Features

- 5 line backlit display screen
- 2 multi-functional softkeys
- Programmable function key supports up to 14 functions
- Vibration Alerter
- Headset Jack
- Desk charging stand

57i CT Cordless Handset Key Descriptions

Function #	Function Description
1	Receiver
2	Volume key During Ringing: Adjusts ringer volume During a call: Adjusts receiver volume During text mode (not in a call): Moves cursor right/left
3	Display
4	Features ƒ Key List Access key to the programmed Feature Key List Scrolls up when in the various lists Adds a space during editing

Function #	Function Description
5	Softkeys Activates feature or option shown on the display above the keys
6	Call key Used to obtain dial tone Also used as a Hold key
7	Dial Pad
8	Mute Key When used, prevents the caller from hearing you
9	Headset Jack
10	Status Light
11	Release key To end calls and go on hook Exits Menu and the various lists
12	Menu Key Access key to the different Options Scrolls down when in the various lists Used as Backspace during editing
13	Redial Key Displays the last 10 numbers dialed
14	Charging Jack
15	Charging Contacts
16	Microphone

Firmware Installation Information

Description

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 Chapter 4, the section, "Configuring Network Settings Manually" on page 4-8 for manually setting up an IP address.

To install the IP phone hardware and cabling, refer to the model-specific *SIP IP Phone Installation Guide*.

Installation Requirements

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

- SIP-based IP PBX system or network installed and running with a SIP account created for the 53i IP phone.
- Access to a Trivial File Transfer Protocol (TFTP), File Transfer Protocol (FTP), Hypertext Transfer Protocol (HTTP) server, or HyperText Transfer Protocol over Secure Sockets Layer (SSL) (HTTPS).
- Ethernet/Fast Ethernet LAN (10/100 Mb)
- Category 5/5e straight through cabling
- Power source
  - For Ethernet networks that supply in-line power to the phone (IEEE 802.3af):
    - For power, use the Ethernet cable (supplied) to connect from the phone directly to the network for power. (No 48v AC power adapter required.)
  - For Ethernet networks that DO NOT supply power to the phone:
    - For power, use the 48V AC Power Adapter (included) to connect from the DC power port on the phone to a power source.
      or
    - (optional) - For power, use a Power over Ethernet (PoE) power injector or a PoE switch. A PoE power injector is available as an optional accessory from Aastra Telecom. Contact your Administrator for more information.

Overview

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

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

Reference

To set the protocol for your configuration server, see Chapter 4, the section, "Configuring the Configuration Server Protocol" on page 4-79.

For setting up your configuration server, see Appendix B, "Configuration Server Setup."

Firmware and Configuration Files

Description

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
53i	53i.st
55i	55i.st
57i	57i.st
57i CT	57i Cordless.st

The IP Phone firmware file includes all the necessary files you need for your phone, including the language files. Loading the language files to your phone(s) is optional. For more information about loading languages files, see Chapter 5, the section, "Customizing the Display Columns on the 560M Expansion Module" on page 5-224.

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.

Installing the Firmware/Configuration Files

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

Step	Action
1	If DHCP is disabled, manually enter the configuration server's IP address. For details on manually setting DHCP, see Chapter 4, the section "DHCP" on page 4-4.
2	Copy the firmware file *<phone model>.st* to the root directory of the configuration server. The IP phone accepts the new firmware file 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., 53i.st, 55i.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*).
4	**Note:** Restart the IP phone as described in Chapter 3, "Restarting Your Phone" on page 3-13.

Chapter 2
Configuration Interface Methods

About this chapter

Introduction

This chapter describes the methods you, as an Administrator, can use to configure the IP phones.

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

Topics

This chapter covers the following topics:

Configuration Methods

Description

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

- IP phone UI
- Aastra Web UI
- Configuration files

Note: There are specific parameters you can configure using only the IP Phone UI, only and Aastra Web UI, only the configuration files, or a combination of any of these methods. For more information about configuring the phone, see Chapter 4, Chapter 5, and Chapter 6.

The following paragraphs describe each method of configuring the IP Phone.

IP Phone UI

The IP Phone User Interface (UI) provides an easy way to access features and functions for using and configuring the IP phone. An Administrator can configure all features and functions on the phone. A User can configure a subset of these features and functions. Users of the IP phones should see their Model-specific User's Guide for available features and functions to configure.

You use the phone's hard keys and keypad to configure specific features on the IP phone. By default, specific softkeys/programmable keys on each phone model can also access the Directory List and Callers List, and initiate transfers and conference calls.

Reference

Refer to Chapter 1, the section "IP Phone Models" on page 1-2 for keys specific to your phone model.

For more information about using the hard keys on each phone, see Chapter 5, the section, "Locking IP Phone Keys" on page 5-28.

For more information about the softkeys/programmable keys, see Chapter 5, the section, "Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys" on page 5-93.

Options Key 

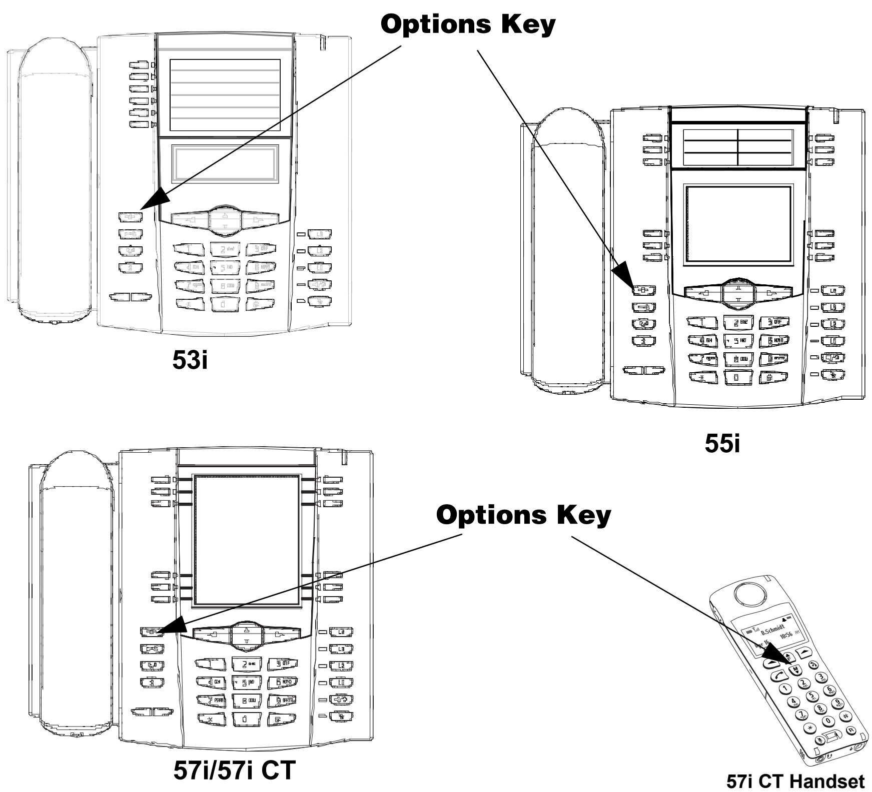
The Options key allows you to access the "Options List" on the IP phone. Accessible options in this list are for both user and Administrator use. An Administrator must enter a password for administrator options.

Note: An Administrator can apply a simplified options menu to the IP phones. An Administrator can also enable and disable the use of an Administrator password protection in the IP phone UI. These features are configurable using the configuration files only.

For more information about these features, see Chapter 3, the section, "Simplified IP Phone UI Options Menu" on page page 3-5, and Chapter 5, the section, "Administrator Passwords" on page 5-8.

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

The following illustration indicates the location of the Options Key on each phone model.



Configuration Interface Methods

Using the Options Key

From the 53i, 55i, or 57i/57i CT:

Step	Action
1	Press on the phone to enter the Options List.
2	Use the and to scroll through the list of options.
3	***On 53i:*** To select an option, press the **Enter** softkey, or select the number on the keypad that corresponds to the option in the Option List. ***On the 55i, 57i, 57i CT:*** To select an option, press the **Select** softkey, press , or select the number on the keypad that corresponds to the option in the Option List.
4	***On 53i:*** Use the **Set** softkey after making a change to an option, to save the change. ***On the 55i, 57i, 57i CT:*** 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 at any time to exit without saving changes.

From the 57i CT handset:

Step	Action
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.

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.

HTTP/HTTPS Support

The Aastra Web UI supports both Hypertext Transfer Protocol (HTTP) and Hypertext Transfer Protocol over Secure Socket Layer (HTTPS) client and server protocols.

HTTP is the set of rules for transferring files (text, graphic images, sound, video, and other multimedia files) over the Internet. When you open your Web browser, you are indirectly making use of HTTP. HTTP is an application protocol that runs on top of the TCP/IP suite of protocols (the foundation protocols for the Internet).

HTTPS is a Web protocol that encrypts and decrypts user page requests as well as the pages that are returned by the Web server. HTTPS uses Secure Socket Layer (SSL) or Transport Layer Security (TLS) as a sublayer under its regular HTTP application layering. SSL is a commonly-used protocol for managing the security of a message transmission on the Internet. It uses a 40-bit key size for the RC4 stream encryption algorithm, which is considered an adequate degree of encryption for commercial exchange. TLS is a protocol that ensures privacy between communicating applications and their users on the Internet. When a server and client communicate, TLS ensures that no third party may eavesdrop or tamper with any message. TLS is the successor to SSL.

Note: HTTPS uses port 443 instead of HTTP port 80 in its interactions with the TCP/IP lower layer.

HTTP/HTTPS Client and Server Support

The Aastra IP phones allow for HTTP request processing and associated data transfers to perform over a secure connection (HTTPS). The IP phones support the following:

- Transfer of firmware images, configuration files, script files, and web page content over a secure connection.
- Web browser phone configuration over a secure connection.
- TLS 1.0or SSL 3.0 methods for both client and server

HTTPS Client

When an HTTPS client opens and closes its TCP socket, the SSL software respectively handshakes upon opening and disconnects upon closing from the HTTPS server. The main HTTPS client functions are:

- Downloading of configuration files and firmware images.
- Downloading of script files based on an "HTTPS://" URL supplied by a softkey definition.

HTTPS Server

The HTTPS server provides HTTP functionality over secure connections. It coexists with the HTTP server but has its own set of tasks. The main HTTPS server functions are:

- Delivery of web page content to a browser client over a secure connection.
- Execution of HTTP GET and POST requests received over a secure connection.

Using HTTPS via the Aastra Web UI

HTTPS is enabled by default on the IP phones. When you open a browser window and enter an IP address or host name for a phone using HTTP, a server redirection occurs which automatically converts an HTTP connection to an HTTPS connection. After the redirection, a "Security Alert" certificate window displays alerting the user that information exchanged with the phone cannot be viewed or changed by others. Accepting the certificate then forwards you to the phone's Web UI.

Notes:
1. The private key and certificate generate outside the phone and embed in the phone firmware for use by the HTTPS server during the SSL handshake.

2. Using the configuration files, the IP phone UI, or the Aastra Web UI, you can configure the following regarding HTTPS:
- Specify HTTPS security client method to use (TLS 1.0 or SSL 3.0)
- Enable or disable HTTP to HTTPS server redirect function
- HTTPS server blocking of XML HTTP POSTS to the phone

Reference

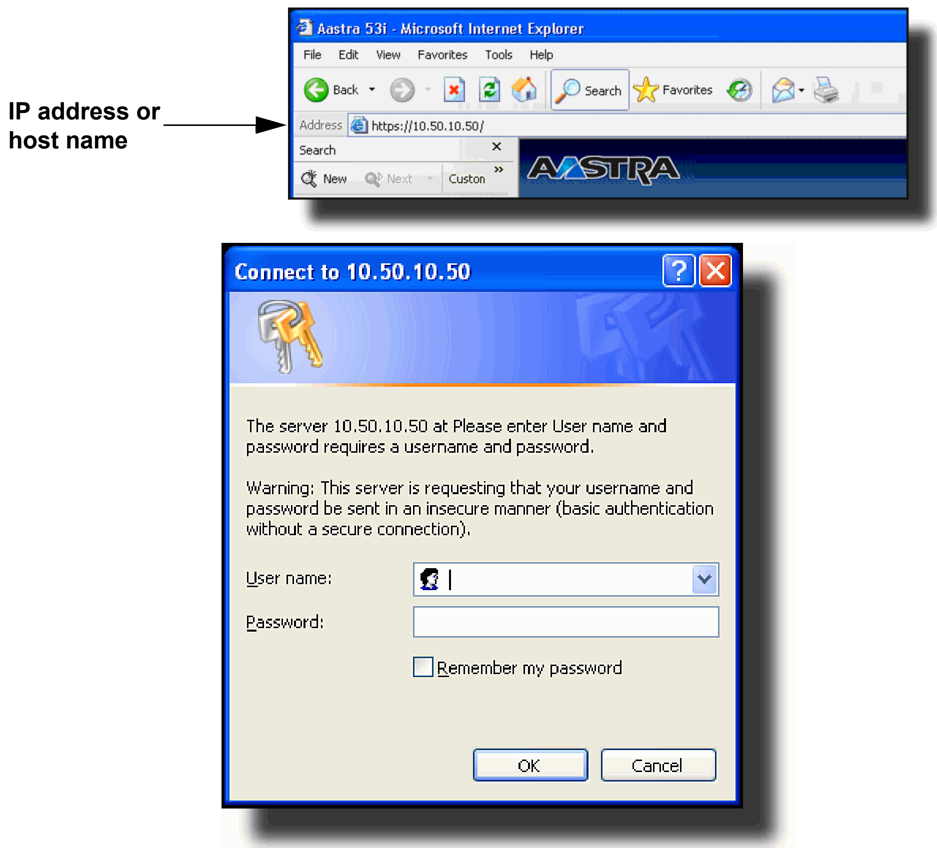
For more information on configuring the HTTPS protocol, see Chapter 4, "Configuring Network and Session Initiation Protocol (SIP) Features", the sections:

- "Configuring the Configuration Server Protocol" on page 4-79
- "HTTPS Client/Server Configuration" on page 4-23

Accessing the Aastra Web UI

Use the following procedure to access the Aastra Web UI.

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Open your web browser and enter the phone's IP address or host name into the address field. The following Login screen displays.



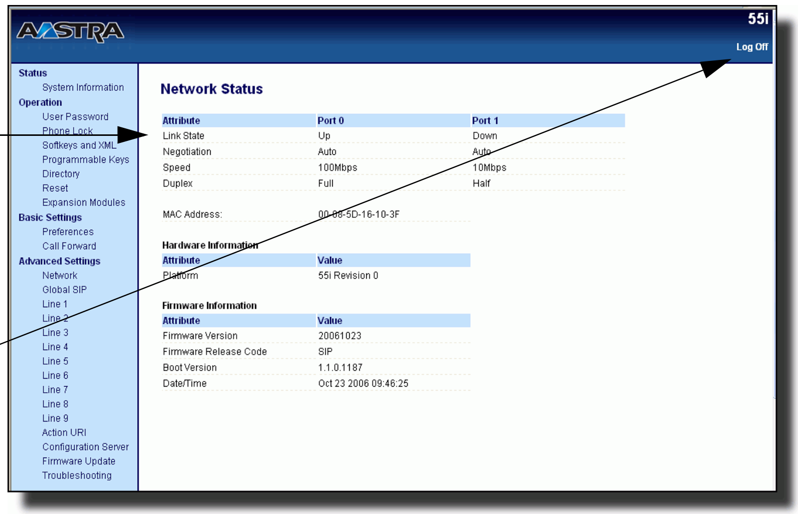
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>2</td><td>Enter your username and password and click OK.

Note: For an administrator, the default username is “admin” and the password is “22222”.
For a user, the default username is “user” and the password field is left blank.
The IP phones accept numeric passwords only.

The Network Status window displays for the IP phone you are accessing. The following illustration is an example of a Network Status screen for the 55i IP phone.

55i Network Status Window



Logout button</td></tr>
<tr><td>3</td><td>You can logout of the Aastra Web UI at any time by clicking LOGOFF.</td></tr>
</table>

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:

Heading	Description
User Password	Allows you to change user password. (Applicable to User and Administrator).
Phone Lock	Allows you to assign an emergency dial plan to the phone, lock the phone to prevent any changes to the phone and to prevent use of the phone, and reset the user password. **Note:** You can also configure a softkey to use for locking/ unlocking the phone. (Applicable to User and Administrator).
Programmable Keys	53i - 6 Top programmable keys (up to 4 programmable functions) 55i - 6 Top programmable hard keys (up to 6 programmable functions) (Applicable to User and Administrator).
Softkeys and XML	55i - 6 Bottom programmable state-based softkeys (up to 20 programmable functions) 57i/57i CT - 6 Top programmable, static softkeys (up to 10 programmable functions; and 6 bottom programmable state-based softkeys (up to 20 programmable functions) (Applicable to User and Administrator).

Heading	Description
Expansion Module <N>	The 536M has up to 36 configurable keys. The 560M has up to 60 configurable keys. You can have up to 3 expansion modules attached to a single phone allowing you to configure keys for Expansion Module 1, Expansion Module 2, and Expansion Module 3.See your *Model-specific User Guide* for applicable expansion modules for your model phone. (Applicable to User and Administrator).
Handset Keys (57i CT only)	Allows you to configure up to 15 softkeys on the handset. (Applicable to User and Administrator).
Directory	Allows you to copy the Callers List and Directory List from your IP phone to your PC. (Applicable to User and Administrator).
Reset	Allows you to restart the IP phone when required. (Applicable to User and Administrator). This setting also allows you to set the IP phone back to its factory default settings or remove the local configuration. (Applicable Administrator only).

Basic Settings

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

Heading	Description
Preferences	Allows you to set the following General specifications on the IP phone. • Local Dial Plan (Admin Only) • Send Dial Plan Terminator (Admin Only) • Digit Timeout (Admin Only) • Park Call (not available on 53i) • Pickup Parked Call (not available on 53i) • Suppress DTMF Playback • Display DTMF Digits • Call Waiting • Play Call Waiting Tone • Stuttered Dial Tone • XML Beep Support • Status Scroll Delay • Incoming Call Interrupts Dialing • Goodbye Key Cancels Incoming Call • UPnP Mapping Lines • Message Waiting Indicator Line This section also allows you to set: • Incoming Intercom Settings • Outgoing Intercom Settings (Admin Only; Administrator can enable these or a User if required) • Key Mapping (Admin Only) • Ring Tones • Priority Alert Settings (Admin Only) • Directed Call Pickup Settings (Admin Only) • Auto Call Distribution Settings (Admin Only) • Time and Date Settings • Language Settings (Admin only can specify the language pack names to load to the phone).
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:

Heading	Description
Network	Allows you to set Basic Network Settings, Advanced Network Settings, Type of Service DSCP, and VLAN settings. (Applicable to Administrator Only)
Global SIP	Allows you to set global Basic SIP Authentication Settings, Basic SIP Network Settings, Advanced SIP Settings, Real-time Transport Protocol (RTP) settings, and Autodial Settings that apply to all lines on the IP phone. (Applicable to Administrator Only)
Lines 1 through 9	Allows you to set per-line Basic SIP Authentication Settings, Basic SIP Network Settings, Advanced SIP Settings, Real-time Transport Protocol (RTP) settings, and Autodial Settings that apply to specific lines on the IP phone. (Applicable to Administrator Only)
Action URI	Allows an administrator to specify a uniform resource identifier (URI) that triggers a GET when certain events occur. (Applicable to Administrator Only)
Configuration Server	Allows you to set the protocol to use on the configuration server (TFTP (default), FTP, HTTP, or HTTPS), configure automatic firmware and configuration file updates, enable/disable auto-resync, and assign an XML push server list. (Applicable to Administrator Only)
Firmware Update	Allows you to manually perform a firmware update on the IP phone from the configuration server. (Applicable to Administrator Only)

Configuration Interface Methods

Heading	Description
TLS Support	Allows you to specify the SIP Root and Intermediate Certificate files to use when the phone uses the TLS transport protocol to setup a call. (Applicable to Administrator Only)
Troubleshooting	Allows you to perform troubleshooting tasks whereby the results can be forwarded to Aastra Technical Support for analyzing and troubleshooting. (Applicable to Administrator Only)

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.

To disable the Aastra Web UI:

Configuration Files

Step	Action
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.

Configuration Files (Administrator Only)

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

You can enter specific configuration parameters in either of the following configuration files:

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

References

For information about configuration file precedence, see Chapter 1, “Overview.”

For a description of each configuration file parameter, see Appendix A, “Configuration Parameters.”

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 Interface Methods

Configuration files

Step	Action
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.

Chapter 3
Administrator Options

About this chapter

Introduction

The IP phones provide specific options on the IP Phone that only an Administrator can access. These options are password protected and allow an Administrator to change or set features and configuration information as required. For all models, an Administrator can use the IP Phone UI, the Aastra Web UI, or the configuration files to enter and change values.

Note: Specific options are configurable only via the IP Phone UI, and/or Aastra Web UI, and/or configuration files.

This chapter provides information about the available Administrator options.

Topics

This chapter covers the following topics:

Administrator Level Options

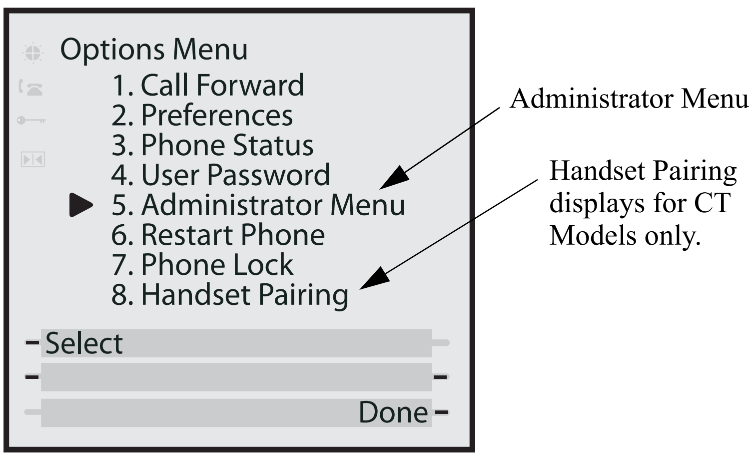
Description

There are options on the IP phone that both a User and Administrator can access. However, there are specific options that an Administrator can access only. These options allow the Administrator to configure and manage local and/or remote IP phones in a network.

An Administrator can access and manage these options using the IP Phone UI, the Aastra Web UI, or the configuration files.

IP Phone UI Options

Using the IP Phone UI, you can access the Administrator options at **Options->Administrator Menu** using the default password of "**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 Appendix A, the section "Password Settings" on page A-9.

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

- **Administrator Menu**
  - — Configuration Server
  - — Network Settings
  - — SIP Settings
  - — Factory Default
  - — Erase Local Config.

References

For information about all other user options in the "**Options Menu**", see your model-specific *SIP IP Phone User Guide*.

For procedures on configuring Administrator Options on the IP phone via the IP phone UI, see:

Chapter 4, "Configuring Network and Session Initiation Protocol (SIP) Features"

Chapter 5, "Configuring Operational Features"

Chapter 6, "Configuring Advanced Operational Features"

Simplified IP Phone UI Options Menu

An Administrator can replace the existing options menu on the Phone UI with a more simplified options menu. In the configuration files, the "**options simple menu**" parameter allows you to display either the full menu (if set to 0), or the simplified menu (if set to 1). The following table illustrates the differences between the full menu and the simplified menu.

When setting the "**options simple menu"** parameter, the menu changes in the Phone UI only. The Aastra Web UI is not affected.

Full Options Menu	Simplified Options Menu
Call Forward	**Call Forward**
Preferences	**Preferences**
Phone Status	**Phone Status**
User Password	Removed
Administrator Menu	Removed
Restart Phone	Removed
Phone Lock	**Phone Lock**
Handset Pairing (CT models only)	**Handset Pairing** (CT models only)

Warning: When using the simplified menu, you cannot change the Network settings from the IP Phone UI. If the network settings become misconfigured, you must "factory default" the phone and use the full menu to recover the network settings from the Phone UI **OR** use the Aastra Web UI to configure the network settings.

Configuring the Simplified IP Phone UI Options Menu

You can enable the simplified IP Phone UI Options menu using the configurations files only.

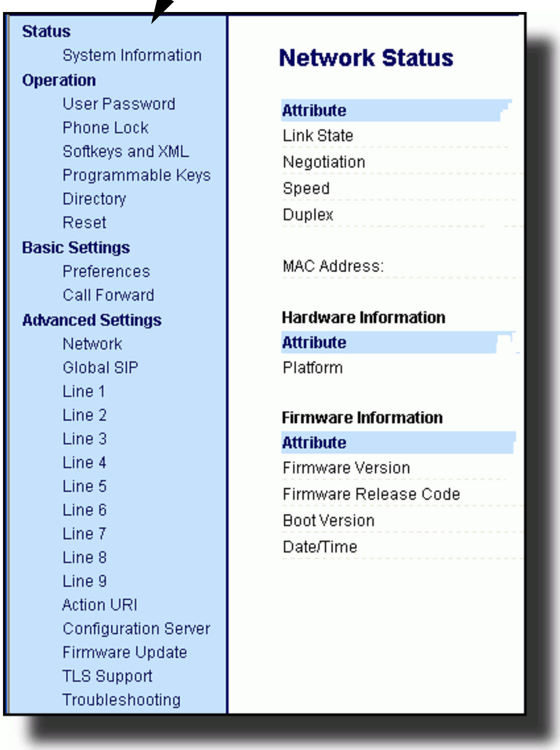
Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Simplified IP Phone UI Options Menu" on page A-5.

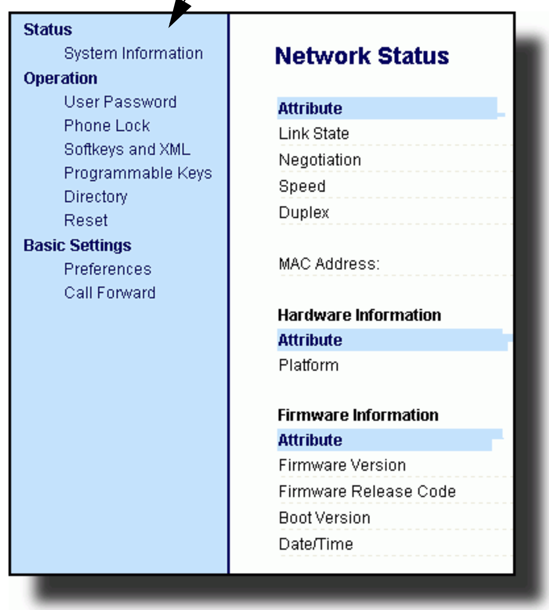
Aastra Web UI Options

An Administrator can configure specific options using the Aastra Web UI. These options display after an Administrator logs into the Web UI using a Web browser and entering the Admin username and password at the login prompt (The default username is "**admin**" and the default password is "**22222**". The IP phones accept numeric passwords only.) The column on the left side of the screen indicates the configurable options. A User has limited configuration options as shown in the following illustrations.

Administrator Web UI Menu



User Web UI Menu



Administrator Options

The following are options that an Administrator can configure in the Aastra Web UI (and are not available for the User to configure):

- **Operation->Reset**
  - Restore to Factory Defaults
  - Remove Local Configuration Settings
- **Basic Settings->Preferences->General**
  - Local Dial Plan
  - Send Dial Plan Terminator
  - Digit Timeout (seconds)
- **Basic Settings->Preferences->Outgoing Intercom Settings** (User can configure this via the Aastra Web UI if enabled by an Administrator)
- **Basic Settings->Preferences->Key Mapping**
- **Basic Settings->Preferences->Priority Alerting Settings**
- **Basic Settings->Preferences->Directed Call Pickup Settings**
- **Basic Settings->Preferences->Auto Call Distribution Settings**
- **Basic Settings->Preferences->Language Settings**
  - Language 1 (entering language pack filename)
  - Language 2 (entering language pack filename)
  - Language 3 (entering language pack filename)
  - Language 4 (entering language pack filename)
- **Advanced Settings**
  - Network
  - Global SIP
  - Line 1 through 9 Settings
  - Action URI
  - Configuration Server
  - Firmware Update
  - TLS Support
  - Troubleshooting

References

For information about options available to a User AND Administrator in the Aastra Web UI, see your Model-specific *User Guide*.

For procedures to Restart your phone or restore factory defaults, see "Restarting Your Phone" on page 3-13, and "Set Phone to Factory Defaults/Erase Local Configuration" on page 3-15.

For more information about Advanced Settings for the IP Phone, see Chapter 4, "Configuring Network and Session Initiation Protocol (SIP) Features."

For procedures on configuring the Basic Settings for the IP Phone, see Chapter 5, "Configuring Operational Features."

Configuration File Options

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

References

For a procedure on using the configuration files, see Chapter 2, the section, "Configuration Files (Administrator Only)" on page 2-1.

For a description of each parameter you can enter in the configuration files, see Appendix A, "Configuration Parameters."

Phone Status

The **Phone Status** on the IP Phone displays the network status and firmware version of the IP phone.

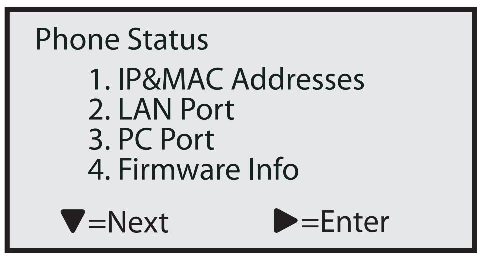
You can display phone status using the IP phone UI or the Aastra Web UI.

Phone Status via IP Phone UI

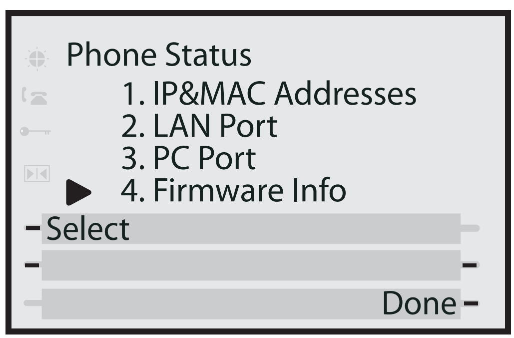
In the IP phone UI, the Phone Status options are available to the user and the administrator and do not require a password entry.

The following options display for phone status on the IP phone UI:

Phone Status Screen for 53i Phone



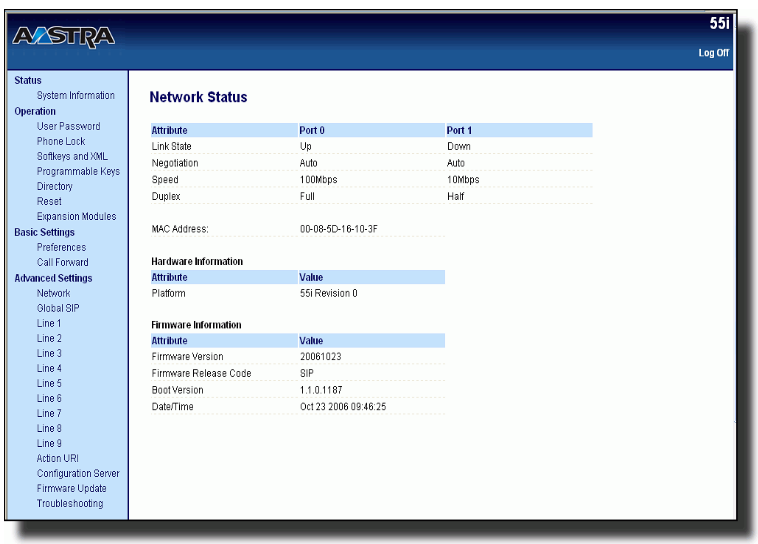
Phone Status Screen for 55i, 57i, and 57i CT Phones



- **IP&MAC Addresses**
  Displays the IP address and MAC address of the phone.
- **LAN Port**
  Displays the Link State, Negotiation Method, Speed, and Duplex Method that the phone uses on its LAN port.
- **PC Port**
  Displays the Link State, Negotiation Method, Speed, and Duplex Method that the phone uses on its PC Port.
- **Firmware Info**
  Displays information about the firmware that is currently installed on the IP phone.

Phone Status via Aastra Web UI

In the Aastra Web UI, the "**Network Attributes**", "**Hardware Information**", and "**Firmware Information**" options display on the Network Status screen.



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 Attributes**
  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.

Restarting Your Phone

As System Administrator, there may be times when you need to restart a phone. The Restart option allows you reboot the phone when required. A reset may be necessary when:

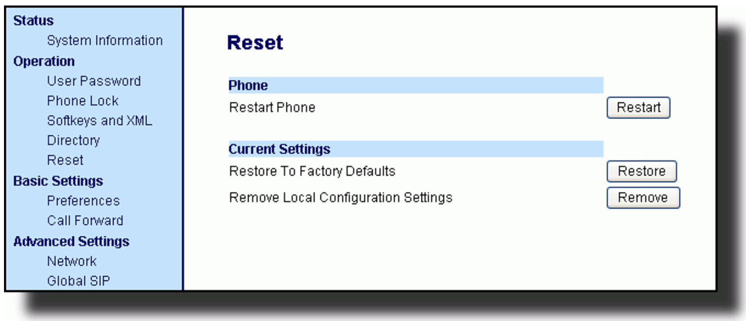
- There is a change in your network, **OR**
- To re-load modified configuration files, **OR**
- If the settings for the IP phone on the IP PBX system have been modified.

You can restart the phone using the IP Phone UI or the Aastra Web UI.

Restarting the Phone Using the IP Phone UI

IP Phone UI	
Step	**Action**
1	Press the Options key on the phone to enter the Options List.
2	Select **Restart Phone**.
3	***For 53i:*** Press **#** to confirm. **Note:** To cancel the Restart, press the ◀ key. ***For 55i, 57i, 57i CT:*** Press **Restart**. **Note:** To cancel the Restart, press **Cancel**.

Restarting the Phone Using the Aastra Web UI

	Aastra Web UI
1	Click on **Operation->Reset->Phone.** 
2	Click Restart to restart the phone.

Set Phone to Factory Defaults/Erase Local Configuration

You can set phones to their factory default setting or remove a local phone's configuration using the IP Phone UI or the Aastra Web UI.

Setting Factory Defaults on the Phone

Factory default settings are the settings that reside on the phone after it has left the factory. The factory default settings on the phone sets the factory defaults for all of the settings in the *aastra.cfg*, *<mac>.cfg,* and local configuration. Performing this action results in losing all user-modified settings.You can reset a phone to factory defaults using the IP Phone UI or the Aastra Web UI.

Setting Factory Defaults Using the IP Phone UI

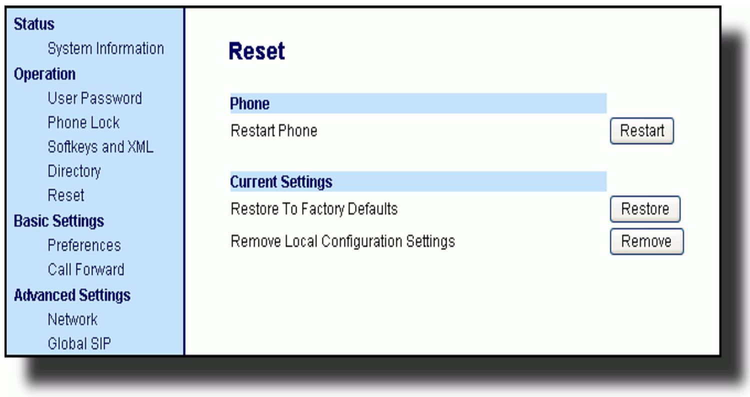
IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu** and enter your Administrator Password (default is **22222**).
3	Select **Factory Default.**
4	***For 53i:*** The "*Restore Defaults?*" prompt displays. Press **#** to confirm. ***For 55i/57i/57i CT:*** The "*Reset phone to factory defaults?*" prompt displays. Press **Default** to confirm.

Settings Factory Defaults Using the Aastra Web UI

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Click on Operation->Reset->Current Settings.



</td></tr>
<tr><td>2</td><td>In the "Restore to Factory Defaults" field, click  .

This restores all factory defaults, and removes any saved configuration and directory list files.</td></tr>
</table>

Erasing the Phone's Local Configuration

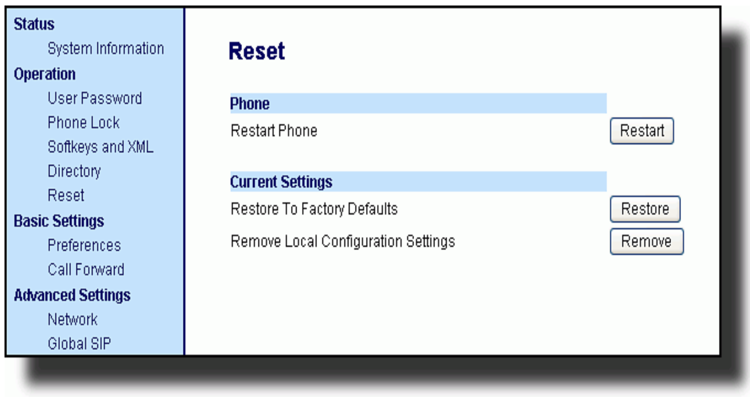
You can reset the IP Phone's local configuration if required. The local configuration is the last updated configuration you performed using the IP Phone UI or the Aastra Web UI. Performing this action results in losing all recently user-modified settings. For more information about local configuration, see Chapter 1, the section, "Configuration File Precedence" on page 1-22.

Erasing the Phone's Local Configuration Using the IP Phone UI

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu** and enter your Administrator Password (default is **22222**).
3	Select **Erase Local Config.**
4	***For 53i:*** The "*Erase local config?*" prompt displays. Press **#** to confirm. ***For 55i/57i/57i CT:*** The "*Erase local config?*" prompt displays. Press **Erase** to confirm.

Erasing the Phone's Local Configuration Using the Aastra Web UI

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Click on Operation->Reset->Current Settings.


</td></tr>
<tr><td>2</td><td>In the "Remove Local Configuration Settings" field, click Remove .

This removes the last customized configuration settings made on the phone.</td></tr>
</table>

Basic Settings

An Administrator has access to specific Basic Setting options to configure and manage the IP Phone in the network. The following sections identify the options available to an Administrator only, or where indicated, to a User and Administrator. These tables also identify whether you can configure them using the Aastra Web UI, IP Phone UI, or the configuration files.

General Settings

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Local Dial Plan	sip dial plan	A dial plan that describes the number and pattern of digits that a user dials to reach a particular telephone number. For more information on this feature, see "Local Dial Plan" on page 5-30.
Send Dial Plan Terminator	sip dial plan terminator	Allows you to enable or disable a dial plan terminator. When you configure the dial plan on the 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 after dialing the number on the keypad before making the call. For more information on this feature, see "SIP Dial Plan Terminator" on page 5-32.
Digit Timeout	sip digit timeout	Represents the time, in seconds, to configure the timeout between consecutive key presses. For more information on this feature, see. "Digit Timeout" on page 5-32.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Park Call **Note:** This option can be set by both Users and Administrators.	sprecode	The parking of a live call to a specific extension. **This feature on the Basic Preferences screen is available on the 55i, 57i, and 57i CT only.** To configure the Park feature on a global basis, see Chapter 5, the section, "Park Calls/Pick Up Parked Calls" on page 35. To configure the Park feature on a key, see Chapter 5, the section, "Park/Pick Up Key" on page 5-153.
Pick Up Parked Call **Note:** This option can be set by both Users and Administrators.	pickupsprecode	Picking up a parked call at the specified extension. **This feature on the Basic Preferences screen is available on the 55i, 57i, and 57i CT only.** To configure the Pickup feature on a global basis, see Chapter 5, the section, "Park Calls/Pick Up Parked Calls" on page 35. To configure the Pickup feature on a key, see Chapter 5, the section, "Park/Pick Up Key" on page 5-153.
Suppress DTMF Playback **Note:** This option can be set by both Users and Administrators.	suppress dtmf playback	Enables and disables suppression of DTMF playback when a number is dialed from the softkeys or programmable keys. For more information on this feature, see. "Suppressing DTMF Playback" on page 5-39.
Display DTMF Digits **Note:** This option can be set by both Users and Administrators.	display dtmf digits	Enables and disables the display of DTMF digits on the IP phone display during a connected state. For more information on this feature, see. "Display DTMF Digits" on page 5-41.
Call Waiting **Note:** This option can be set by both Users and Administrators.	call waiting	Enable or disables Call Waiting on the IP Phone. For more information on this feature, see. "Call Waiting/Call Waiting Tone" on page 5-43.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Play Call Waiting Tone **Note:** This option can be set by both Users and Administrators.	call waiting tone	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. For more information on this feature, see. "Call Waiting/Call Waiting Tone" on page 5-43.
Stuttered Dial Tone **Note:** This option can be set by both Users and Administrators.	stutter disabled	Enable or disables the playing of a stuttered dial tone when there is a message waiting on the IP phone. For more information on this feature, see. "Stuttered Dial Tone" on page 5-46.
XML Beep Support **Note:** This option can be set by both Users and Administrators.	xml beep notification	Enables or disables the playing of a beep to indicate a status on the phone. When the phone receives a status message, the BEEP notifies the user that the message is displaying. For more information on this feature, see "XML Beep Support" on page 5-48.
Status Scroll Delay (seconds) **Note:** This option can be set by both Users and Administrators.	xml status scroll delay	Allows you to set the time delay, in seconds, between the scrolling of each status message on the phone. For more information on this feature, see "Status Scroll Delay" on page 5-50.
Incoming Call Interrupts Dialing **Note:** This option can be set by both Users and Administrators.	incoming call interrupts dialing	Enable or disables how the phone handles incoming calls while the phone is dialing out. For more information on this feature, see "Incoming Call Interrupts Dialing" on page 5-52.
Goodbye Key Cancels Incoming Call **Note:** This option can be set by both Users and Administrators.	goodbye key cancels incoming call	Enable or disables the behavior of the Goodbye Key on the IP phone. For more information on this feature, see "Goodbye Key Cancels Incoming Call" on page 5-54.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
UPnP Mapping Lines **Note:** This option can be set by both Users and Administrators.	upnp mapping lines	Enables or disables the use of Universal Plug and Play (UpnP) on a specific line on the IP phone. For more information on this feature, see "UPnP Mapping Lines (for remote phones)" on page 5-56.
Message Waiting Indicator Line **Note:** This option can be set by both Users and Administrators.	mwi led line	Allows you to enable the Message Waiting Indicator (MWI) on a single line or on all lines on the phone. For example, if you set this parameter to 3, the LED illuminates if a voice mail is pending on line 3. If you set this parameter to 0, the LED illuminates if a voice mail is pending on any line on the phone (lines 1 through 9). For more information on this feature, see "Message Waiting Indicator Line" on page 5-58.

Incoming/Outgoing Intercom Calls

The Incoming/Outgoing Intercom Call settings on the IP Phone specify whether the IP phone or the server is responsible for notifying the recipient that an Intercom call is being placed. These settings also specify the prefix code for server-side Intercom calls, and specifies the configuration to use when making the Intercom call.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Incoming Intercom Settings (all models)		
Auto-Answer **Note:** This option can be set by both Users and Administrators.	sip allow auto answer	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. For more information on this feature, see "Incoming/ Outgoing Intercom with Auto-Answer and Barge In" on page 5-60.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Microphone Mute **Note:** This option can be set by both Users and Administrators.	sip intercom mute mic	Enables or disables the microphone on the IP phone for Intercom calls made by the originating caller. For more information on this feature, see "Incoming/ Outgoing Intercom with Auto-Answer and Barge In" on page 5-60.
Play Warning Tone **Note:** This option can be set by both Users and Administrators.	sip play warning tone	Enables or disables a warning tone to play when the phone receives an incoming intercom call on an active line. For more information on this feature, see "Incoming/ Outgoing Intercom with Auto-Answer and Barge In" on page 5-60.
Allow Barge In **Note:** This option can be set by both Users and Administrators.	sip intercom allow barge in	Enable or disables how the phone handles incoming intercom calls while the phone is on an active call. For more information on this feature, see "Incoming/ Outgoing Intercom with Auto-Answer and Barge In" on page 5-60.
Outgoing Intercom Settings (55i, 57i, and 57i CT only))		
Type	sip intercom type	Determines whether the IP phone or the server is responsible for notifying the recipient that an Intercom call is being placed. Applicable settings are Phone-Side, Server-Side, OFF. For more information on this feature, see "Incoming/ Outgoing Intercom with Auto-Answer and Barge In" on page 5-60.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Prefix Code	sip intercom prefix code	The prefix to add to the phone number for server-side outgoing Intercom calls. This parameter is required for all server-side Intercom calls. For more information on this feature, see “Incoming/ Outgoing Intercom with Auto-Answer and Barge In” on page 5-60.
Line	sip intercom line	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. For more information on this feature, see “Incoming/ Outgoing Intercom with Auto-Answer and Barge In” on page 5-60.

Key Mapping

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Map Redial Key To	map redial key to	Sets the **Redial** key as a speeddial 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 57i CT Base Station, the **Redial** key on the 57i CT handset retains its original functionality. The **Redial** key on the handset is not configured for speeddial. For more information on this feature, see "Key Mapping" on page 5-66.
Map Conf Key To	map conf key to	Sets the **Conf** key as a speeddial 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 57i CT Base Station, the **Conf** key on the 57i CT handset retains its original functionality. The **Conf** key on the handset is not configured for speeddial. For more information on this feature, see "Key Mapping" on page 5-66.

Ring Tones

Parameter in IP Phone UI	Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Tone Set **Note:** This option can be set by both Users and Administrators.	Tone Set	tone set	Globally sets a tone set for a specific country For more information on this feature, see "Ring Tones and Tone Sets" on page 5-70.
Ring Tone **Note:** This option can be set by both Users and Administrators.	Global Ring Tone	ring tone	Globally sets the type of ring tone on the IP phone. Ring tone can be set to one of six distinct rings. For more information on this feature, see "Ring Tones and Tone Sets" on page 5-70.
N/A	LineN **Note:** This option can be set by both Users and Administrators.	lineN ring tone	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. For more information on this feature, see "Ring Tones and Tone Sets" on page 5-70.

Priority Alerting Settings

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Enable Priority Alerting	priority alerting enabled	Enables and disables distinctive ringing on the IP phone for incoming calls and call-waiting calls. For more information on this feature, see "Priority Alerting" on page 5-75.
Group	alert group	When an "alert group" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone. For more information on this feature, see "Priority Alerting" on page 5-75.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
External	alert external	When an "alert external" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone. For more information on this feature, see "Priority Alerting" on page 5-75.
Internal	alert internal	When an "alert-internal" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone. For more information on this feature, see "Priority Alerting" on page 5-75.
Emergency	alert emergency	When an "alert emergency" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone. For more information on this feature, see "Priority Alerting" on page 5-75.
Priority	alert priority	When an "alert priority" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone. For more information on this feature, see "Priority Alerting" on page 5-75.
Auto Call Distribution	alert auto call distribution	When an "alert-acd" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone. For more information on this feature, see "Priority Alerting" on page 5-75.
Community 1 thru Community 4	alert community 1 alert community 2 alert community 3 alert community 4	When an "alert community-#" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone. Available Bellcore tones are: 0 Normal ringing (default) 1 Bellcore-dr2 2 Bellcore-dr3 3 Bellcore-dr4 4 Bellcore-dr5 5 Silent For more information on this feature, see "Priority Alerting" on page 5-75.

Administrator Options

Directed Call Pickup (DCP)

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Directed Call Pickup	directed call pickup	Enables or disables the use of "directed call pickup" feature. For more information on this feature, see "Directed Call Pickup (BLF or XML Call Interception)" on page 5-81.
Directed Call Pickup Prefix	directed call pickup prefix	Allows you to specify a prefix to use for "directed call pickup" that you can use with a BLF or BLF List softkey. For more information on this feature, see "Directed Call Pickup (BLF or XML Call Interception)" on page 5-81.
Play a Ring Splash	play a ring splash	Enables or disables the playing of a short "ring splash tone" when there is an incoming call on the BLF monitored extension. If the host tone is idle, the tone plays a "ring splash". For more information on this feature, see "Directed Call Pickup (BLF or XML Call Interception)" on page 5-81.

Auto Call Distribution (ACD) Settings

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Auto Available	acd auto available	Enables or disables the use of the ACD Auto-Available Timer. For more information on this feature, see "Automatic Call Distribution (ACD) (for Sylantro Servers)" on page 5-122.
Auto Available Timer	acd auto available timer	Specifies the length of time, in seconds, before the IP phone status switches back to "available." For more information on this feature, see "Automatic Call Distribution (ACD) (for Sylantro Servers)" on page 5-122.

Time and Date

Parameter in IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Time Format **Note:** This option can be set by both Users and Administrators.	Time Format	time format	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. For more information on this feature, see "Time and Date" on page 5-17.
Date Format **Note:** This option can be set by both Users and Administrators.	Date Format	date format	This parameter allows the user to change the date to various formats For more information on this feature, see "Time and Date" on page 5-17.

Language

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
WebPage Language **Note:** This option can be set by both Users and Administrators.	language	The language you want to display in the IP Phone UI and the Aastra Web UI. Valid values for 53i, 55i, 57i are: 0 (English) 1 (French - Canadian) 2 (Spanish - Mexican) 3 (German) 4 (Italian) Valid values for 57i CT are: 0 (English) 1 (French - Canadian) 2 (Spanish - Mexican) **Note:** All languages may not be available for selection. The available languages are dependant on the language packs currently loaded to the IP phone. For more information about loading language packs, see "Loading Language Packs" on page 5-21. For more information on this feature, see "Language" on page 5-21.
Language 1 thru Language	language N	The language pack you want to load to the IP phone. Valid values are: **lang_fr-ca.txt** **lang_es.txt** **lang_de.txt** **lang_it.txt** **Notes**: **1.** The languages packs you load are dependant on available language packs from the configuration server. **2.** You must reboot the phone to load a language pack. For more information about loading language packs, see "Loading Language Packs" on page 5-21. For more information on this feature, see "Language" on page 5-21.

Network Settings

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.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
DHCP	DHCP	dhcp	Enables or disables DHCP. Enabling DHCP populates the required network information. The DHCP server serves the network information that the IP phone requires. If the IP phone is unable to get any required information, then you must enter it manually. DHCP populates the following network information: IP Address, Subnet Mask, Gateway, Broadcast Address, Domain Name Servers (DNS), TFTP, HTTP HTTPS, and FTP servers, 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 4-4.
IP Address	IP Address	ip	IP address of the IP phone. To assign a static IP address, disable DHCP. For more information, see "Configuring Network Settings Manually" on page 4-8.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Subnet Mask	Subnet Mask	subnet mask	Subnet mask defines the IP address range local to the IP phone. To assign a static subnet mask, disable DHCP. For more information, see "Configuring Network Settings Manually" on page 4-8.
Gateway	Gateway	default gateway	The IP address of the network's gateway or default router IP address. To assign a static Gateway IP address, disable DHCP. For more information, see "Configuring Network Settings Manually" on page 4-8.
Primary DNS	Primary DNS	dns1	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. To assign static DNS addresses, disable DHCP. **Note:** If a host name is configured on the IP phone, you must also set a DNS. For more information, see "Configuring Network Settings Manually" on page 4-8.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Secondary DNS	Secondary DNS	dns2	A service that translates domain names into IP addresses. To assign static DNS addresses, disable DHCP. For more information, see "Configuring Network Settings Manually" on page 4-8.
LAN Port	LAN Port	ethernet port 0	The send (TX) and receive (RX) method to use on Ethernet port 0 (LAN Port) to transmit and receive data over the LAN. For more information, see Chapter 4, "Configuring LAN and PC Port Negotiation" on page 4-11.
PC Port	PC Port	ethernet port 1	The send (TX) and receive (RX) method to use on Ethernet port 1 (PC Port) to transmit and receive data to your PC. For more information, see Chapter 4, "Configuring LAN and PC Port Negotiation" on page 4-11.

Advanced Network Settings

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
UPnP	UPnP	upnp manager	Enables or disables Universal Plug and Play (UpnP) on the IP phone. If you set this parameter to "0", you can manually configure NAT on the IP phone and the UPnP manager will not start. For more information, see Chapter 4, "Universal Plug and Play (UPnP) (for remote phones)" on page 4-27.
N/A	N/A	upnp gateway	IP address or fully qualified Domain Name of the Internet gateway or router. This parameter stores the IP address of the gateway or router in the event that only non-default UPnP gateways get discovered on the network. The UPnP port mappings are saved to this IP address so even if the phone reboots, it will still have the correct port mappings. For more information, see Chapter 4, "Universal Plug and Play (UPnP) (for remote phones)" on page 4-27.
NAT IP	NAT IP	sip nat ip	IP address the network device that enforces NAT. For more information, see Chapter 4, "Configuring NAT Address and Port (optional)" on page 4-21.
NAT SIP Port	NAT SIP Port	sip nat port	Port number of the network device that enforces NAT. For more information, see Chapter 4, "Configuring NAT Address and Port (optional)" on page 4-21.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
NAT RTP Port	NAT RTP Port	sip nat rtp port	Indicates the port through which the RTP packets are sent. This value must specify the beginning of the RTP port range on the gateway or router. 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. For more information, see Chapter 4, "Configuring NAT Address and Port (optional)" on page 4-21.
Nortel NAT	Nortel NAT Traversal Enabled	sip nortel nat support	Enables or disables the phone to operate while connected to a network device that enforces NAT. For more information, see Chapter 4, "Configuring Nortel NAT (optional)" on page 4-18.
N/A	Nortel NAT Timer (seconds)	sip nortel nat timer	The interval, in seconds, that the phone sends SIP ping requests to the Nortel proxy. For more information, see Chapter 4, "Configuring Nortel NAT (optional)" on page 4-18.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	NTP Time Servers	time server disabled	Enables or disables the time server. This parameter affects the **time server1**, **time server2**, and **time server3** parameters. Setting this parameter to **0** allows the use of the configured Time Server(s). Setting this parameter to **1** prevents the use of the configured Time Server(s). For more information, see Chapter 4, "Network Time Servers" on page 4-42.
N/A	Time Servers 1, 2, and 3	time server1 time server2 time server3 time server4	The 1st, 2nd, 3rd, and 4th 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. For more information, see Chapter 4, "Network Time Servers" on page 4-42.
Client Method	HTTPS Client Method	https client method	Defines the security method that the client advertises to the server during the Secure Socket Layer (SSL) handshake. Available options are: **TLS 1.0** - Transport Layer Security version 1 (TLS 1.0) is a protocol that ensures privacy between communicating applications and their users on the Internet. TLS is the successor to SSL. **SSL 3.0** - Secure Socket Layer version 3 (SSL 3.0) is a commonly-used protocol for managing the security of a message transmission on the Internet. For more information, see Chapter 4, "HTTPS Client/Server Configuration" on page 4-23.

Administrator Options

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
HTTPS	HTTPS Server - Redirect HTTP to HTTPS	https redirect http get	Allows or disallows redirection from the HTTP server to the HTTPS server. For more information, see Chapter 4, "HTTPS Client/Server Configuration" on page 4-23.
XML HTTP POSTs	HTTPS Server - Block XML HTTP POSTs	https block http post xml	Enables or disables the blocking of XML scripts from HTTP POSTs. Some client applications use HTTP POSTs to transfer XML scripts. The phones's HTTP server accepts these POSTs even if server redirection is enabled, effectively bypassing the secure connection. When this parameter is enabled (blocking is enabled), receipt of an HTTP POST containing an XML parameter header results in the following response: "403 Forbidden". This forces the client to direct the POSTs to the HTTPS server through use of the "https://" URL. For more information, see Chapter 4, "HTTPS Client/Server Configuration" on page 4-23.

Type of Service (ToS), DSCP (53i)

Networks settings include Type of Service (ToS) and Differentiated Services Code Point (DSCP) for the 53i IP phone.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Type of Service SIPt	SIP	tos sip	The Differentiated Services Code Point (DSCP) for SIP packets. For more information, see Chapter 4, "Type of Service (ToS), Quality of Service (QoS), and DiffServ QoS" on page 4-31.
Type of Service RTP	RTP	tos rtp	The Differentiated Services Code Point (DSCP) for RTP packets. For more information, see Chapter 4, "Type of Service (ToS), Quality of Service (QoS), and DiffServ QoS" on page 4-31.
Type of Service RTCP	RTCP	tos rtcp	The Differentiated Services Code Point (DSCP) for RTCP packets. For more information, see Chapter 4, "Type of Service (ToS), Quality of Service (QoS), and DiffServ QoS" on page 4-31.

VLAN

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

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Global Settings			
VLAN Enable	VLAN Enable	tagging enabled	Enables or disables VLAN on the IP phones. For more information, see Chapter 4, "Virtual LAN (optional)" on page 4-31.
Other Priority	Priority, Non-IP Packet	priority non-ip	Specifies the priority value for non-IP packets. For more information, see Chapter 4, "Virtual LAN (optional)" on page 4-31.
LAN Port Settings (Port 0)			
Phone VLAN ID	VLAN ID	VLAN id	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. For more information, see Chapter 4, "Virtual LAN (optional)" on page 4-31.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
SIP Priority RTP Priority RTCP Priority	SIP Priority RTP Priority RTCP Priority	tos priority map	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. For more information, see Chapter 4, "Virtual LAN (optional)" on page 4-31.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
PC Port Settings (Port 1)			
PC Port VLAN ID	VLAN ID	VLAN id port 1	Specifies the VLAN ID used to pass packets through to a PC via Port 1. **Note:** If you set the PC Port VLAN ID (Port 1) to **4095**, all untagged packets are sent to this port. The following is an example of configuring the phone on a VLAN where all untagged packets are sent to the passthrough port. **<u>Example</u>** You enable tagging on the LAN Port (VLAN id) as normal but set the PC Port (VLAN id port 1) to 4095. The following example sets the phone to be on VLAN 3 on the LAN Port but the PC Port is configured as untagged. **`tagging enabled: 1`** **`VLAN id: 3`** **`VLAN id port 1: 4095`** For more information, see Chapter 4, "Virtual LAN (optional)" on page 4-31.
PC Port Priority	Priority	QoS eth port 1 priority	Specifies the priority value used for passing VLAN packets through to a PC via Port 1. For more information, see Chapter 4, "Virtual LAN (optional)" on page 4-31.

Administrator Options

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 on a global and per-line basis. You can also configure specific parameters using the IP Phone UI, the Aastra Web UI, or the configuration files. 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.

Basic SIP Authentication Settings

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Screen Name	Screen Name	sip screen name (global) sip lineN screen name (per-line)	Name that displays on the idle screen. Valid values are up to 20 alphanumeric characters. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
N/A	Screen Name 2	sip screen name 2 (global) sip lineN screen name 2 (per-line)	Custom text message that displays on the idle screen. Valid values are up to 20 alphanumeric characters. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
User Name	Phone Number	sip user name (global) sip lineN user name (per-line)	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. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
Display Name	Caller ID	sip display name (global) sip lineN display name (per-line)	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. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
Auth Name	Authentication Name	sip auth name (global) sip lineN auth name (per-line)	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. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
Password	Password	sip password (global) sip lineN password (per-line)	Password used to register the IP phone with the SIP proxy. Valid values are up to 20 numeric characters. Passwords are encrypted and display as asterisks when entering. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	BLA Number	sip bla number (global) sip lineN bla number (per-line)	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, see Chapter 4, "Basic Network Settings" on page 4-4. For more information about BLA, see Chapter 5, the section, "Bridged Line Appearance (BLA) (55i, 57i, 57i CT only)" on page 5-144.
N/A	Line Mode	sip mode (global) sip lineN mode (per-line)	The mode-type that you assign to the IP phone. Valid values are Generic (0), BroadSoft SCA (1), Nortel (2), or BLA (3). Default is Generic (0). For more information, see Chapter 4, "Basic Network Settings" on page 4-4.

Administrator Options

Basic SIP Network Settings

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Proxy Server	Proxy Server	sip proxy ip (global) sip lineN proxy ip (per-line)	IP address of the SIP proxy server. Up to 64 alphanumeric characters. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
Proxy Port	Proxy Port	sip proxy port (global) sip lineN proxy port (per-line)	SIP proxy server's port number. Default is 0. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
N/A	Backup Proxy Server	sip backup proxy server (global) sip lineN backup proxy server (per-line)	The IP address of the backup SIP proxy server for which the IP phone uses when the primary SIP proxy is unavailable. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
N/A	Backup Proxy Port	sip backup proxy port (global) sip lineN backup proxy port (per-line)	The port number of the backup SIP proxy server for which the IP phone uses when the primary SIP proxy port is unavailable. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	Outbound Proxy Server	sip outbound proxy server (global) sip lineN outbound proxy server (per-line)	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. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
N/A	Outbound Proxy Port	sip outbound proxy port (global) sip lineN outbound proxy port (per-line)	The proxy port on the proxy server to which the IP phone sends all SIP messages. Default is 0. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
Registrar Server	Registrar Server	sip registrar ip (global) sip lineN registrar ip (per-line)	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. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
Registrar Port	Registrar Port	sip registrar port (global) sip lineN registrar port (per-line)	SIP registrar's port number. Default is 0. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	Backup Registrar Server	sip backup registrar ip (global) sip lineN backup registrar ip (per-line)	The address of the backup registrar (typically, the backup SIP proxy) for which the IP phone uses to send *REGISTER* requests if the primary registrar is unavailable. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
N/A	Backup Registrar Port	sip backup registrar port (global) sip lineN backup registrar port	The backup registrar's (typically the backup SIP proxy) port number. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
N/A	Registration Period	sip registration period (global) sip lineN registration period (per-line)	The requested registration period, in seconds, from the registrar. For more information, see Chapter 4, "Basic Network Settings" on page 4-4.
N/A	Conference Server URI	sip centralized conf (global) sip lineN centralized conf (per-line)	Globally enables or disables SIP centralized conferencing for an IP phone. For more information, see Chapter 4, "Centralized Conferencing (for Sylantro and Broadsoft Servers)" on page 5-220.

Advanced SIP Settings

In addition to the basic SIP settings, you can also configure the following advanced SIP parameters. These parameters are configurable via the Aastra Web UI and the configuration files on a global basis only.

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Explicit MWI Subscription	sip 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. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.
Explicit MWI Subscription Period	sip explicit mwi subscription period	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.
Missed Call Summary Subscription	sip missed call summary subscription	Enables or disables the Missed Call Summary Subscription feature. This feature allows missed calls that have been redirected by the server, to be incremented in the missed calls indicator on the phone it was initially directed to. Default is disabled. For more information about this parameter, see Chapter 6, the section, "Missed Call Summary Subscription" on page 6-13.

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Missed Call Summary Subscription Period	sip missed call summary subscription period	Specifies the amount of time, in seconds, that the phone uses the Missed Calls Summary Subscription feature. This parameter is always enabled with a default value of 86400 seconds. When the phone reaches the limit set for this parameter, it sends the subscription again. For more information about this parameter, see Chapter 6, the section, "Missed Call Summary Subscription" on page 6-13.
Send MAC Address in REGISTER Message	sip send mac	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. For more information about this parameter, see Chapter 6, the section, "MAC Address/ Line Number in REGISTER Messages" on page 6-5.
Send Line Number in REGISTER Message	sip send line	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. For more information about this parameter, see Chapter 6, the section, "MAC Address/ Line Number in REGISTER Messages" on page 6-5.
Session Timer	sip 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. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Timer 1 and Timer 2	sip T1 timer sip T2 timer	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.
Transaction Timer	sip transaction timer	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.
Transport Protocol	sip transport protocol	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.
Registration Failed Retry Timer	sip registration retry timer	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.
Registration Timeout Retry Timer	sip registration timeout retry timer	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Registration Renewal Timer	sip registration renewal timer	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.
BLF Subscription Period	sip blf subscription period	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.
ACD Subscription Period	sip acd subscription period	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-58.

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Blacklist Duration	sip blacklist duration	Specifies the length of time, in seconds, that a failed server remains on the server blacklist. The IP phone avoids sending a SIP message to a failed server (if another server is available) for this amount of time. For more information about Blacklist Duration, see Chapter 6, the section, "Blacklist Duration (Broadsoft Servers)" on page 6-17.
Whitelist Proxy	sip whitelist	This parameter enables/disables the whitelist proxy feature, as follows: • Set to 0 to disable the feature. • Set to 1 to enable the feature. When this feature is enabled, an IP phone accepts call requests from a trusted proxy server *only*. The IP phone rejects any call requests from an untrusted proxy server. For more information about Whitelist Proxy see Chapter 6, the section, "Whitelist Proxy" on page 6-19.

RTP Settings

You can configure the following RTP settings.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
RTP Port Base	RTP Port	sip rtp port	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. Default is 3000. For more information, see Chapter 4, "Real-time Transport Protocol (RTP) Settings" on page 4-64.
N/A	Basic Codecs (G.711 u-Law, G.711 a-Law, G.729)	sip use basic codecs	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). For more information, see Chapter 4, "Real-time Transport Protocol (RTP) Settings" on page 4-64.
N/A	Force RFC2833 Out of Band DTMF	sip out-of-band dtmf	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). For more information, see Chapter 4, "Real-time Transport Protocol (RTP) Settings" on page 4-64.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	Customized Codec Preference List	sip customized codec	Specifies a customized Codec preference list which allows you to use the preferred Codecs for this IP phone. For more information, see Chapter 4, "Real-time Transport Protocol (RTP) Settings" on page 4-64.
N/A	DTMF Method	sip dtmf method (global) sip lineN dtmf method (per-line)	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). For more information, see Chapter 4, "Real-time Transport Protocol (RTP) Settings" on page 4-64.
N/A	RTP Encryption	sip srtp mode (global) sip lineN srtp mode (per-line)	This parameter determines if SRTP is enabled on this IP phone, as follows: • If set to 0, then disable SRTP. • If set to 1 then SRTP calls are preferred. • If set to 2, then SRTP calls only are generated/accepted. For more information, see Chapter 4, "Real-time Transport Protocol (RTP) Settings" on page 4-64.
N/A	Silence Suppression	sip silence suppression	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. For more information, see Chapter 4, "Real-time Transport Protocol (RTP) Settings" on page 4-64.

Administrator Options

Line Settings

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

IP Phone Model	Available Lines
53i	9
55i	9
57i	9
57i CT	9

On the IP Phones, you can configure the following on a per-line basis using the configuration files or the Aastra Web UI:

- **Basic SIP Authentication Settings**
- **Basic SIP Network Settings**
- **Advanced SIP Settings** (Missed Call Summary Subscription only)
- **RTP Settings** (DTMF Method and RTP Encryption only)
- **Autodial Settings** (You can enable/disable "**Use Global Settings**" on a per-line basis only)

References

For more information about configuring the features listed above on a per-line basis, see Chapter 4, the sections:

- "Basic SIP Settings" on page 4-46
- "Advanced SIP Settings (optional)" on page 4-58
- "Real-time Transport Protocol (RTP) Settings" on page 4-64
- "Autodial Settings" on page 4-74

Softkeys, Programmable Keys, Expansion Module Keys

A user or administrator can assign specific functions to softkeys, programmable keys, or expansion module keys. The available keys for configuration depend on the IP phone model as shown in the following table.

IP Phone Model	Softkeys	Expansion Module Keys	Programmable Keys
53i	-	36 to 108* (Model 536M)	4
55i	6	36 to 108* (Model 536M) 60 to 180** (Model 560M)	6
57i	12	36 to 108* (Model 536M) 60 to 180** (Model 560M)	-
57i CT	12	36 to 108* on Base Station (Model 536M) 60 to 180** on Base Station (Model 560M)	-

*The 536M expansion module consists of 36 softkeys. You can have up to 3 expansion modules on an IP phone totaling 108 softkeys. Valid for 53i, 55i, 57i, and 57i CT phones.

**The 560M expansion module consists of 60 softkeys. You can have up to 3 expansion modules on an IP phone totaling 180 softkeys. Valid for 55i, 57i, and 57i CT phones.

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

Reference

For more information about key functions see Appendix A, the section, "Softkey/ Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

For information about configuring softkeys, programmable keys, and expansion module keys, see Chapter 5, the section, "Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys" on page 5-93.

Administrator Options

Action URI

The IP phones have a feature that allows an administrator to specify a uniform resource identifier (URI) that triggers a GET when certain XML events occur. The IP phone XML events that support this feature are:

- Startup
- Successful registration
- Incoming call
- Outgoing call
- Offhook
- Onhook

You can set these parameters using the configuration files or the Aastra Web UI.

Reference

For more information about setting the Action URIs for XML applications, see "XML Action URIs" on page 5-209.

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:

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Download Protocol Settings			
Download Protocol	Download Protocol	download protocol	Protocol to use for downloading new versions of software to the IP phone. Valid values are: TFTP FTP HTTP HTTPS For DHCP to automatically populate the IP address or domain name for the download servers, your DHCP server must support Option 66. For more information, see Chapter 4, the section, "DHCP" on page 4-4. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.
Primary TFTP	TFTP Server	tftp server	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. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Alternate TFTP	Alternate TFTP	alternate tftp server	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. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.
Select TFTP	Use Alternate TFTP	use alternate tftp	Enables or disables the alternate TFTP server. Valid values are "0" disabled and "1" enabled. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.
FTP Server	FTP Server	ftp server	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. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.
FTP Username	FTP Username	ftp username	The username to enter for accessing the FTP server. This will become effective after this configuration file has been downloaded into the phone. **Note:** The IP Phones support usernames containing dots ("."). For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
FTP Password	FTP Password	ftp password	The password to enter for accessing the FTP server. This will become effective after this configuration file has been downloaded into the phone. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.
HTTP Server	HTTP Server	http server	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 next parameter (**http path**). For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.
HTTP Path	HTTP Path	http path	The HTTP path name to enter. If the IP phone's configuration and firmware 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. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Download Server	HTTPS Server	https server	The HTTPS server's IP address. This will become effective after this configuration file has been downloaded into the phone. **Optional**: You can also assign an HTTPS relative path to the HTTPS server. See the next parameter (**https path**). For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.
Download Path	HTTPS Path	https path	The HTTPS path name to enter. If the IP phone's configuration and firmware files are located in a sub-directory beneath the server's HTTPS root directory, the relative path to that sub-directory should be entered in this field. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-79.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Auto-Resync Settings			
N/A	Mode	auto resync mode	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. **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 the IP 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. For more information, see Chapter 8, the section, "Automatic Update (auto-resync)" on page 8-6.

Administrator Options

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	Time (24-hour)	auto resync time	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). For more information, see Chapter 8, the section, "Automatic Update (auto-resync)" on page 8-6.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
XML Push Server List (Approved IP Addresses)			
N/A	XML Push Server List (Approved IP Addresses)	xml application post list	The HTTP server that is pushing XML applications to the IP phone. For more information, see Chapter 5, the section, "XML Push Requests" on page 5-203.

Firmware Update Features

The IP phone uses a TFTP, FTP, HTTP, or HTTPS 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 Chapter 8, "Upgrading the Firmware."

Chapter 4
Configuring Network and Session Initiation Protocol (SIP) Features

About this chapter

Introduction

This chapter provides the information required to configure the Network and Global SIP features on the IP Phone. These features are password protected on the IP Phone UI and the Aastra Web UI. This chapter also includes procedures for configuring the Network and Global SIP features via the configuration files, the IP Phone UI, and the Aastra Web UI where applicable.

Note: The IP Phone User Interface (UI) procedures in the remainder of this Guide use the keys on the 55i, 57i, and/or 57i CT when configuring Administrator Options. For information on using the 53i keys to configure the Administrator Options, see Chapter 2, the section, "Using the Options Key" on page 2-4.

Topics

This chapter covers the following topics:

Overview

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 Appendix A, the section "Password Settings" on page A-9.

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 Appendix A, "Configuration Parameters."

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 username and password.

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

References

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

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

Configuring the IP Phones

Network Settings

This section describes the basic network settings on the IP phone which include configuring for:

- DHCP
- IP Address (of phone)
- Subnet Mask (of phone)
- Gateway
- Primary DNS
- Secondary DNS
- LAN Port
- PC Port

Basic Network Settings

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
- FTP Server

- HTTP Server
- HTTPS Server

Note: The IP Phones support download protocols according to RFC2131 and RFC1541 (TFTP, FTP, HTTP,. HTTPS) to support DHCP option 66. Option 66 is part of the DHCP Offer message that the DHCP server generates to tell the phone which configuration server it should use to download new firmware and configuration files.

For DHCP to automatically populate the IP address or domain name for the servers, your DHCP server must support Option 66. Option 66 is responsible for forwarding the server's 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.

Your DHCP server configuration file, such as the *dhcpd.conf* file, may include one of these lines to configure the configuration server protocol and the server details.

Protocol	Format	Examples
HTTP	http://<server>/<path>	option tftp-server-name "http://192.168.1.45"; option tftp-server-name "http://192.168.1.45/path"; option tftp-server-name "http://httpsvr.example.com/path";
HTTPS	https://<server>/<path>	option tftp-server-name "https://192.168.1.45"; option tftp-server-name "https://192.168.1.45/path"; option tftp-server-name "https://httpssvr.example.com/path";

Protocol	Format	Examples
FTP	ftp://user:password@ftpserver	option tftp-server-name “ftp://192.168.1.45”; option tftp-server-name “ftp://ftpsvr.example.com”; (for anonymous user) option tftp-server-name “ftp://userID:password@ ftpsvr.example.com”;
TFTP	tftp://tftpserver	option tftp-server-name “192.168.1.45”; option tftp-server-name “tftpsvr.example.com”; option tftp-server-name “tftp://tftpsvr.example.com”;

DNS Caching

The IP phones have the ability to cache DNS requests according to RFC1035 and RFC2181. The phone caches DNS lookups according to the TTL field, so that the phone only performs another lookup for an address when the TTL expires.

Configuring DHCP

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

Configuration Files

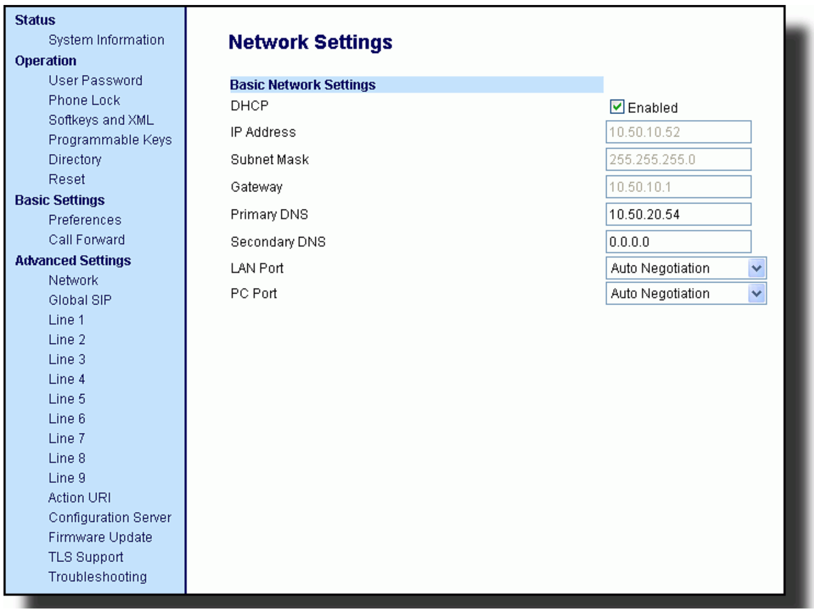
For specific parameters you can set in the configuration files, see Appendix A, the section, “Network Settings” on page A-6.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Network Settings**.
5	Select option **DHCP**.
6	Press **Change** to set "*Use DHCP?*" to "Yes" (enable) or "No" (disable).
7	Press **Done** to save the changes.

.

Aastra Web UI

Step	Action
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  to save your settings.

Configuring Network Settings Manually

If you disable DHCP on your phone, you need to configure the following network settings manually:

- IP Address
- Subnet Mask
- Gateway
- Primary DNS

- Secondary DNS

Note: If you disable DHCP on the phone, the phone uses the TFTP protocol as the default server protocol. If you want to specify a different protocol to use, see “Configuration Server Protocol” on page 4-79.

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 Appendix A, the section, “Network Settings” on page A-6.

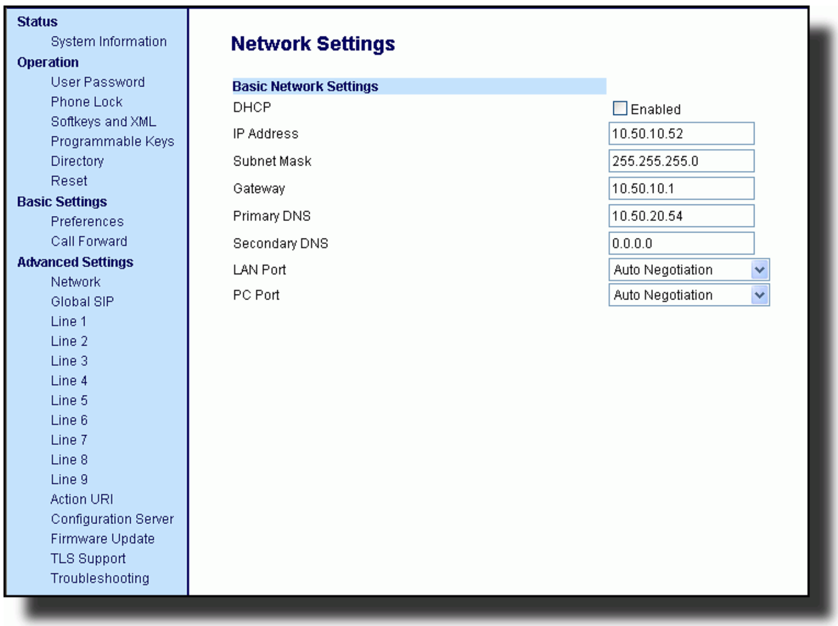
IP Phone UI

Step	Action
1	Press [options key] on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Network Settings**.
5	Select **IP Address** and enter the IP address of the phone.
6	Select **Subnet Mask** and enter the subnet mask.
7	Select **Gateway** and enter the gateway address.
8	Select **DNS** and enter a Primary and/or Secondary DNS server.
9	Press **Done** to save the changes. The IP phone is manually configured.

.

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Network->Basic Network Settings.



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

The IP phone is manually configured.</td></tr>
</table>

Configuring LAN and PC Port Negotiation

Ethernet is the computer networking technology for local area networks (LANs). You use the LAN Port to connect to a LAN using a twisted pair 10BASE-T cable to transmit 10BASE-T Ethernet. You use the PC Port to connect to the configuration server (your PC).

There are two Ethernet ports on the rear of the IP phones: LAN Port and PC Port. Using the Aastra Web UI, you can select the type of transmission you want these ports to use to communicate over the LAN. The IP phones support each of the following methods of transmission:

- Auto-negotiation
- Half-duplex (10Mbps or 100 Mbps)
- Full-duplex (10Mbps or 100Mbps)

Auto-negotiation

Auto-negotiation is when two connected devices choose common transmission parameters. In the auto-negotiation process, the connected devices share their speed and duplex capabilities and connect at the highest common denominator (HCD). Auto-negotiation can be used by devices that are capable of different transmission rates (such as 10Mbit/sec and 100Mbit/sec), different duplex modes (half duplex and full duplex) and/or different standards at the same speed. You can set the LAN and PC Ports on the IP phones to auto-negotiate during transmission.

Half-Duplex (10Mbps or 100Mbps)

Half-duplex data transmission means that data can be transmitted in both directions on a signal carrier, but not at the same time. For example, on a LAN using a technology that has half-duplex transmission, one device can send data on the line and then immediately receive data on the line from the same direction in which data was just transmitted. Half-duplex transmission implies a bidirectional line (one that can carry data in both directions). On the IP phones, you can set the half-duplex transmission to transmit in 10Mbps or in 100Mbps.

Full-Duplex (10Mbps or 100Mbps)

Full-duplex data transmission means that data can be transmitted in both directions on a signal carrier at the same time. For example, on a LAN with a technology that has full-duplex transmission, one device can be sending data on the line while another device is receiving data. Full-duplex transmission implies a bidirectional line (one that can move data in both directions). On the IP phones, you can set the full-duplex transmission to transmit in 10Mbps or in 100Mbps.

Configuring the LAN Port and PC Port

You can configure the Ethernet port transmission method to use on the IP phones 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 Appendix A, the section, "Network Settings" on page A-6.

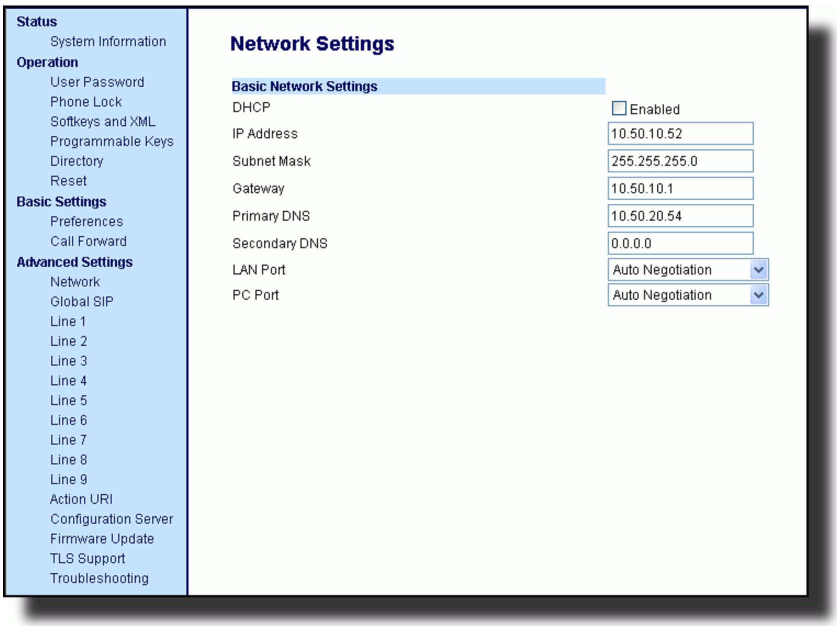
IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Network Settings**.
5	Select **Ethernet**.
6	Select **LAN Port Link.**

IP Phone UI

Step	Action
7	Select a negotiation method to use on port 0 and press **Done**. Valid values are: • AutoNegotiation • FullDuplex 10Mbps • FullDuplex 100Mbps • HalfDuplex 10Mbps • HalfDuplex 100Mbps Default is **AutoNegotiation.**
8	Select **PC Port Link**.
9	Select a negotiation method to use on port 1and press **Done**. Valid values are: • AutoNegotiation • FullDuplex 10Mbps • FullDuplex 100Mbps • HalfDuplex 10Mbps • HalfDuplex 100Mbps Default is **AutoNegotiation**.
10	Press **Done** (3 times) 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.
11	Select **Restart**.

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Network->Basic Network Settings.

</td></tr>
<tr><td>2</td><td>In the “LAN Port” field, select a negotiation method to use on port 0. Valid values are:
• Auto Negotiation
• Full Duplex, 10Mbps
• Full Duplex, 100Mbps
• Half Duplex, 10Mbps
• Half Duplex, 100Mbps

Default is Auto Negotiation.</td></tr>
</table>

Aastra Web UI

Step	Action
3	In the “**PC Port**” field, select a negotiation method to use on port 1. Valid values are: • Auto Negotiation • Full Duplex, 10Mbps • Full Duplex, 100Mbps • Half Duplex, 10Mbps • Half Duplex, 100Mbps Default is **Auto Negotiation**.
4	Click **Save Settings** to save your settings.

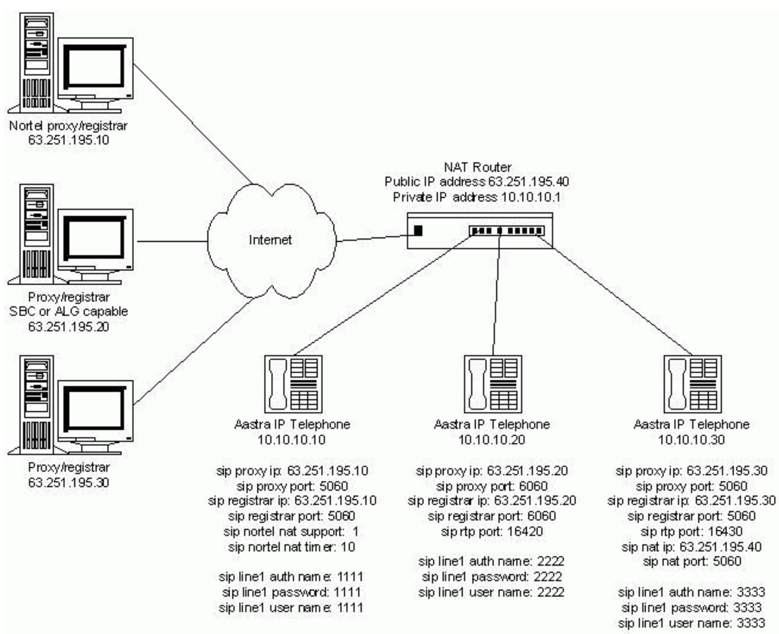
Advanced Network Settings

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), and Universal Plug and Play (UPnP) using the Aastra Web UI or the configuration files.

Note: The available advanced network parameters via the IP phone UI are NAT, Nortel NAT, UPnP, 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 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.

When you use NAT in your network, a network device (usually a router) provides a “firewall” division between the public network (usually the Internet) and the private network, to which the IP Phones are connected. The firewall protects the network by translating port numbers within packets between the public and private networks. When using NAT, and an RTP packet arrives at the public side of the firewall, it is expected to have the NAT RTP port within the packet. If the packet contains the proper port number, the firewall changes the NAT RTP port number in the packet to the RTP port number that the phone recognizes, and then forwards it onto the private network. Often the router/firewall automatically discovers these various port numbers and other information concerning public and private sides of the network with the use of the Universal Plug and Play (UPnP) Protocol between the phone and the router/firewall.

Configuring Nortel NAT (optional)

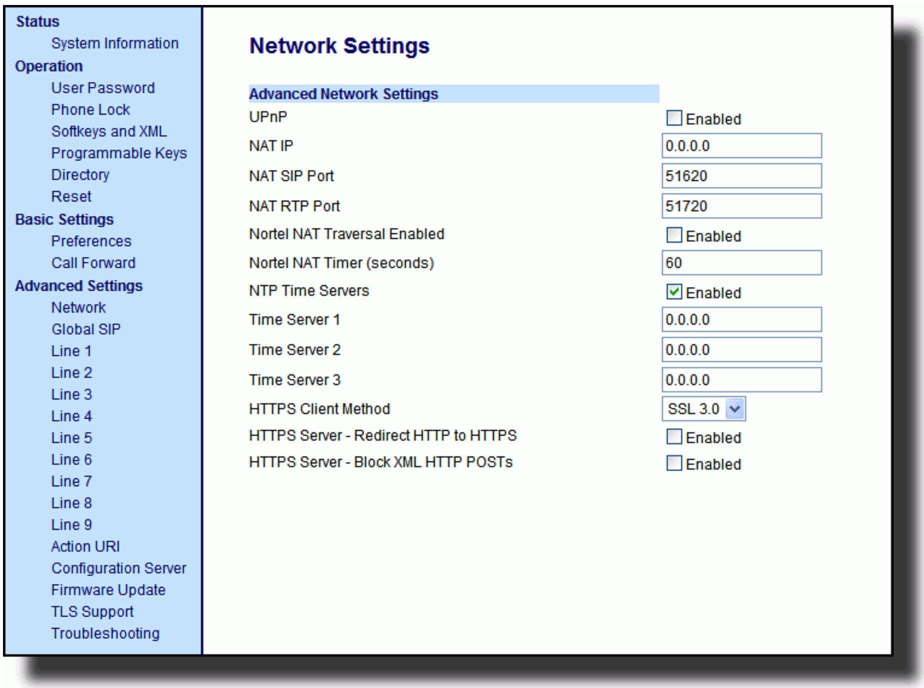
You can configure Nortel NAT using the configuration files, the IP phone UI, or the Aastra Web UI.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, “Network Address Translation (NAT) Settings” on page A-20.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Network Settings.**
5	Select **NAT Settings.**
6	Select **Nortel NAT.**
7	Press **Change** to set either "**Enabled**" or "**Disabled**".
8	Press **Done** (3 times) to finish. **Note:** The session prompts you to restart the IP phone to apply the configuration settings.
9	Select **Restart**.

Aastra Web UI

Step	Action
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 settings.

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

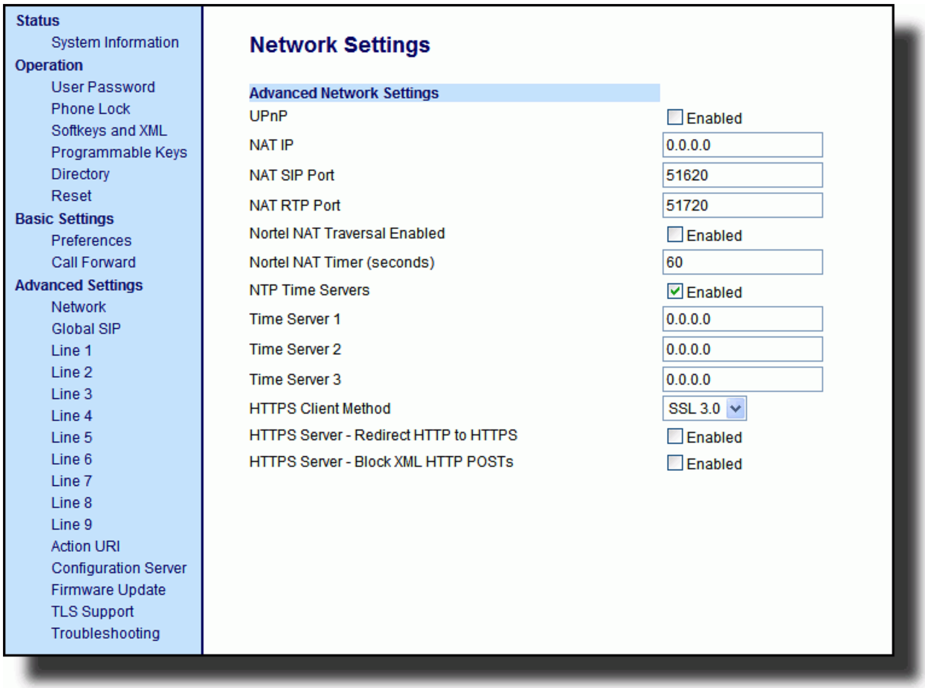
For specific parameters you can set in the configuration files, see Appendix A, the section, "Network Address Translation (NAT) Settings" on page A-20.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Network Settings.**
5	Select **Static NAT.**
6	Select **NAT IP.**
7	Enter a public IP address of your NAT device in dotted-decimal format.
8	Press **Done** to save the setting.
9	Select **NAT SIP Port**. Default is **51620**.
10	Enter the public SIP signalling port number of your NAT device.
11	Press **Done** to save the setting.
12	Select **NAT RTP Port.**
13	Enter the RTP Port number of your NAT device. Default is **51720**.
14	Press **Done** (4 times) to finish. **Note:** The session prompts you to restart the IP phone to apply the configuration settings.
15	Select **Restart**.

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Network->Advanced Network Settings.



</td></tr>
<tr><td>2</td><td>Enter a NAT IP address in the "NAT IP" field. The value must be entered in dotted decimal format. For example, 0.0.0.0. The “NAT IP” is the public IP address of your NAT device.</td></tr>
<tr><td>3</td><td>Enter a NAT port in the "NAT SIP Port" field. Default is 51620.
The “NAT SIP Port” is the public SIP signalling port number of your NAT device.</td></tr>
<tr><td>4</td><td>Enter a NAT port in the "NAT RTP Port" field. Default is 51720.
The “NAT RTP Port” is the RTP Port number of your NAT device.</td></tr>
<tr><td>5</td><td>Click Save Settings to save your settings.</td></tr>
</table>

HTTPS Client/Server Configuration

HTTPS is a Web protocol that encrypts and decrypts user page requests as well as the pages that are returned by the Web server. HTTPS uses Secure Socket Layer (SSL) or Transport Layer Security (TLS) as a sublayer under its regular HTTP application layering. SSL is a commonly-used protocol for managing the security of a message transmission on the Internet. It uses a 40-bit key size for the RC4 stream encryption algorithm, which is considered an adequate degree of encryption for commercial exchange. TLS is a protocol that ensures privacy between communicating applications and their users on the Internet. When a server and client communicate, TLS ensures that no third party may eavesdrop or tamper with any message. TLS is the successor to SSL.

Note: HTTPS uses port 443 instead of HTTP port 80 in its interactions with the TCP/IP lower layer.

When an HTTPS client opens and closes its TCP socket, the SSL software respectively handshakes upon opening and disconnects upon closing from the HTTPS server. The main HTTPS client functions are:

- Downloading of configuration files and firmware images.
- Downloading of script files based on an “HTTPS://” URL supplied by a softkey definition.

The HTTPS server provides HTTP functionality over secure connections. It coexists with the HTTP server but has its own set of tasks. The main HTTPS server functions are:

- Delivery of web page content to a browser client over a secure connection.
- Execution of HTTP GET and POST requests received over a secure connection.

Using the configuration files, the IP phone UI, or the Aastra Web UI, you can configure the following regarding HTTPS:

- Specify HTTPS security client method to use (TSLv1 or SSLv3)
- Enable or disable HTTP to HTTPS server redirect function
- HTTPS server blocking of XML HTTP POSTS to the phone

Configuring HTTPS Client and Server Settings

Use the following procedures to configure the HTTPS client and server for the IP phones.

Note: To enable or disable the IP phones to use the HTTPS protocol as the configuration server, see the section, "Configuring the Configuration Server Protocol" on page 4-79.

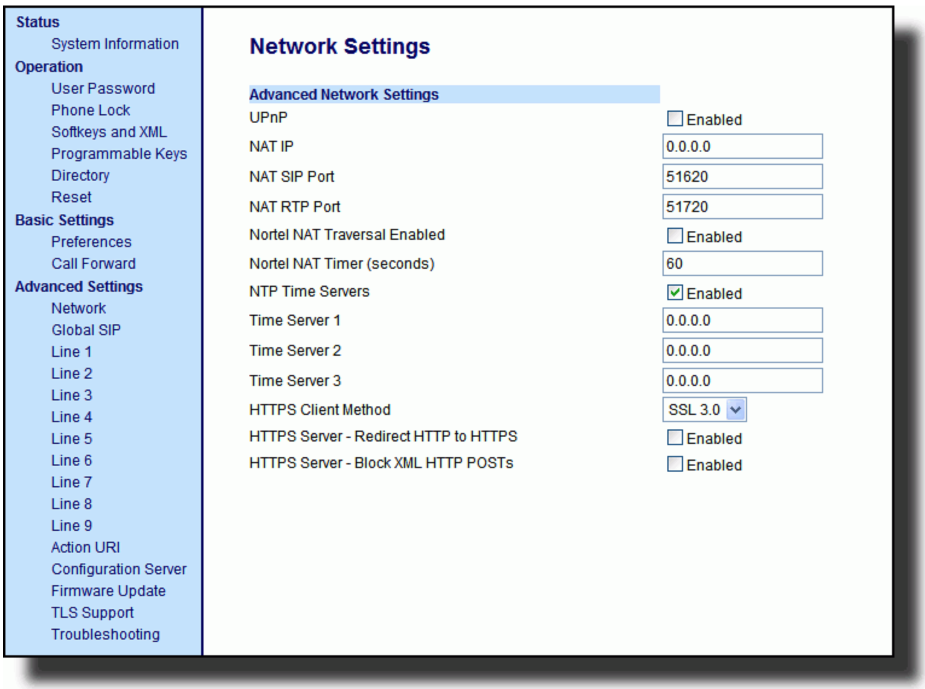
Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "HTTPS Client and Server Settings" on page A-23.

IP Phone UI

Step	Action
1	Press [Options key] on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Configuration Server.**
5	Select **HTTPS Settings.**
Configure HTTPS Client	
6	Select **HTTPS Client.**
7	Select **Client Method.**
8	Press **Change** to select a client method to use for HTTPS. Valid values are: • **SSL 3.0 (default)** • **TLS 1.0**
9	Press **Done** to save the changes.
Configure HTTPS Server	
10	Select **HTTPS Server.**
11	Select **HTTP->HTTPS.**
12	Press **Change** to select "**Yes**" or "**No**". Default is "**No**". Enabling this feature redirects the HTTP protocol to HTTPS.
13	Press **Done** to save the changes.
14	Select **XML HTTP POSTs.**
15	Press **Change** to select "**Yes**" or "**No**". Default is "**No**". Enabling this feature blocks XML HTTP POSTs from the IP Phone.
16	Press **Done** (4 times) to finish. **Note:** The session prompts you to restart the IP phone to apply the configuration settings.
17	Select **Restart**.

Aastra Web UI

Step	Action
1	Click on **Advanced Settings->Network->Advanced Network Settings.** 
2	Select an HTTPS client method to use from the **HTTPS Client Method** field. Valid values are: **SSL 3.0** (default) **TSL 1.0**
3	Enable HTTP to HTTPS redirect by checking the **HTTPS Server - Redirect HTTP to HTTPS** field check box. (Disable this field by unchecking the check box). Default is disabled.
4	Enable the blocking of XML HTTP POSTs by the HTTPS server by checking the **HTTPS Server - Block XML HTTP POSTs** field check box. (Disable this field by unchecking the check box). Default is disabled.
5	Click Save Settings to save your settings.

Universal Plug and Play (UPnP) (for remote phones)

UPnP is a standard that uses Internet protocols to enable devices to be plugged into a network and automatically know about each other. With UPnP, when a user plugs a device into the network, the device configures itself, acquires a TCP IP address, and uses a discovery protocol based on the Internet's HTTP or HTTPS URL to announce its presence on the network to other devices.

This method of device discovery on a network is called "Universal Plug and Play" or UPnP. If you enable UPnP, and the phone is discovered on the network, port mappings are set up between the phone and the Internet Gateway Device (IGD). The phone controls the opening, closing, and polling of ports on the IGD. HTTP and SIP use a single port each. RTP/RTCP uses a range of ports.

The UPnP manager performs its functions when you set the phone to remote mode. When you switch the phone back to local mode, the UPnP manager removes any open port mappings and shuts itself down. If you boot the phone in remote mode, the UPnP manager initializes after the phone obtains an IP address and before a SIP registration is sent out. If you want to manually configure your NAT, you must disable UPnP on the remote mode phone.

Note: Enabling UPnP allows the IP phones to access the Internet even if a firewall has been set on the IGD. This allows the phone to send and receive SIP calls and XML pushes without interruption. UPnP does not work with multiple firewalls.

You can enable UPnP on remote IP phones using the configuration files, the IP Phone UI, or the Aastra Web UI. Using the configuration files, you can enable UPnP using the following parameters:

- **upnp manager**
- **upnp gateway**
- **sip nat rtp port**

The "**upnp manager**" parameter enables or disables UPnP. The "**upnp gateway**" parameter is the IP address or qualified domain name of the Internet gateway or router that stores the port mappings. In the event a phone using UPnP is rebooted, it will still have the previously set port mappings on the gateway. The "**sip nat rtp port**" parameter specifies the RTP port range on the gateway.

A User or Administrator can specify UPnP on specific lines using the configuration files (using the "**upnp mapping lines**" parameter) or the Aastra Web UI (at the path **Basic Settings->Preferences->UPnP Mapping Lines**).

Reference

For more information about enabling/disabling UPnP Mapping on specific lines, see Chapter 5, the section, "UPnP Mapping Lines (for remote phones)" on page 5-56.

Configuring UPnP (optional)

Use the following procedures to configure UPnP on the IP phones.

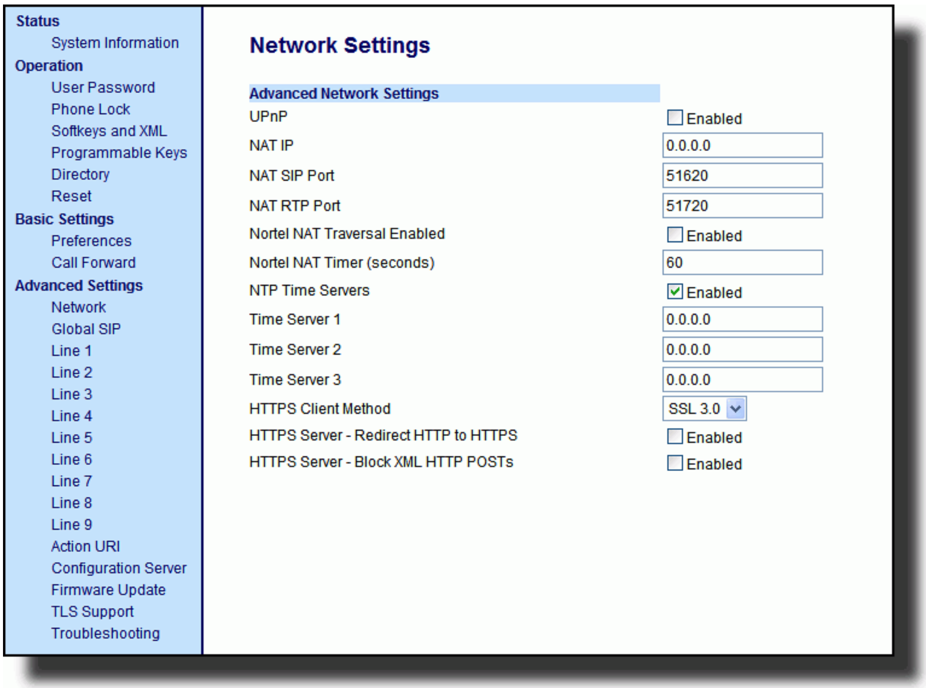
Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "UPnP Settings" on page A-25.

IP Phone UI

Step	Action
1	Press  on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Network Settings.**
5	Select **NAT Settings.**
6	Select **UPnP.**
7	Press **Change** to select "**Yes**" or "**No**". Default is "**No**". This field enables or disables the use of UPnP on the IP Phone.
8	Press **Done** (3 times) to save the changes. **Note:** The session prompts you to restart the IP phone to apply the configuration settings
9	Select **Restart.**

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Network->Advanced Network Settings.



</td></tr>
<tr><td>2</td><td>The “UPnP” field is disabled by default. To enable UPnP, place a check in the “Enabled” box. Uncheck the box to disable this field. This field enables and disable UPnP on the IP phone.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your settings.</td></tr>
</table>

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 LAN Port connected to the network. Port 1 is the Ethernet PC Port used for passthrough to a PC.

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:

Parameters in Configuration Files	Parameters in Aastra Web UI
Global	
tagging enabled	VLAN enable
priority non-ip	Priority, Non-IP Packet
LAN Port	
vlan id	VLAN ID
tos priority map	SIP Priority
tos priority map	RTP Priority
tos priority map	RTCP Priority
PC Port	
vlan id port 1	VLAN ID
QoS eth port 1 priority	Priority

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. When the LAN Port (**vlan id**) and the PC Port (**vlan id port 1**) parameters have the same value, VLAN functionality is compatible with earlier IP phone software releases.

If you set the PC Port (**vlan id port 1**) to **4095**, all untagged packets are sent to this port. For configuring this feature via the Phone UI and the Aastra Web UI, see "Configuring VLAN (optional)" on page 4-37. For configuring this feature using the configuration files, see Appendix A, the section, "Virtual Local Area Network (VLAN) Settings" on page A-26.

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 4-33).

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 priority 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 Appendix A, the section, "Type of Service (ToS)/DSCP Settings" on page A-30.

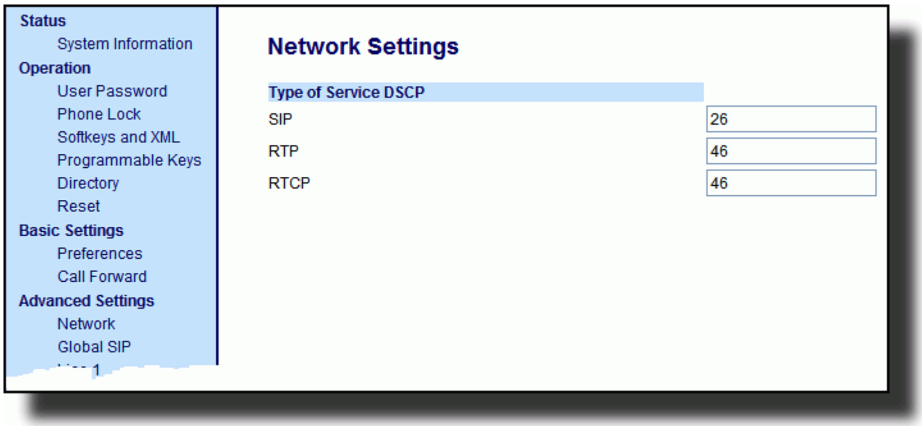
IP Phone UI

Step	Action
1	Press [Options key] on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Network Settings.**
5	Select **Type of Service DSCP.**

IP Phone UI

Step	Action
6	Select **Type of Service SIP.** **or** Select **Type of Service RTP.** **or** Select **Type of Service RTCP**.
7	Enter a value for "**Type of Service SIP**". Default is **26**. **or** Enter a value for "**Type of Service RTP**". Default is **46**. **or** Enter a value for "**Type of Service RTCP**". Default is **46**. Valid values are **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 4-33 For more information, see the section "Configuring VLAN (optional)" on page 4-37.
8	Press **Done** (3 times) to save the changes. **Note:** The session prompts you to restart the IP phone to apply the configuration settings
9	Select **Restart.**

Aastra Web UI

Step	Action
1	Click on **Advanced Settings->Network->Type of Service,DSCP.** 
2	Select a Protocol field: • **"SIP"** **or** • **"RTP"** **or** • **"RTCP"**
3	Enter a value from **0** to **63**. Default values are as follows: SIP = 26 RTP = 46 RTCP = 46 **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 4-37 For more information, see the section "Configuring VLAN (optional)" on page 4-37.
4	Click Save Settings to save your settings.

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 procedure.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Virtual Local Area Network (VLAN) Settings" on page A-26.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Network Settings.**
5	Select **VLAN Settings.**
***To globally enable/disable VLAN and set priority for non-IP packets*:**	
6	Select **VLAN Enable**.
7	Press **Change** to set VLAN Enable to "**Yes"** to enable or "**No"** to disable. Default is "**No**".
8	Press **Done** to save the changes.
9	Select **Phone VLAN**.
10	Select **VLAN Priority.**
11	Select **Other** and enter a non-IP priority value from **0** to **7** for non-IP packets. Default for this field is **5**.
12	Press **Done** (3 times) to return to the VLAN Settings menu.

Configuring the IP Phones

IP Phone UI

Step	Action
To set VLAN ID and priority for LAN Port (Port 0):	
13	Select **Phone VLAN.**
14	Select **Phone VLAN ID** and enter a value from **1** to **4094** to specify the VLAN ID for the LAN Port. Default is **1**.
15	Press **Done** to save the change.
16	Select **VLAN Priority**.
17	Select one of the following VLAN Protocols: • **SIP Priority** • **RTP Priority** • **RTCP Priority**
18	Enter a VLAN priority value from **0** to **7** for the associated Protocol. Default values for each Protocol are: SIP Priority = 3 RTP Priority = 5 RTCP Priority = 5
19	Press **Done** (3 times) to return to the VLAN Settings menu.
To set VLAN ID and priority for PC Port (Port 1):	
20	Select **PC Port VLAN**.
21	Select **PC Port VLAN ID**.

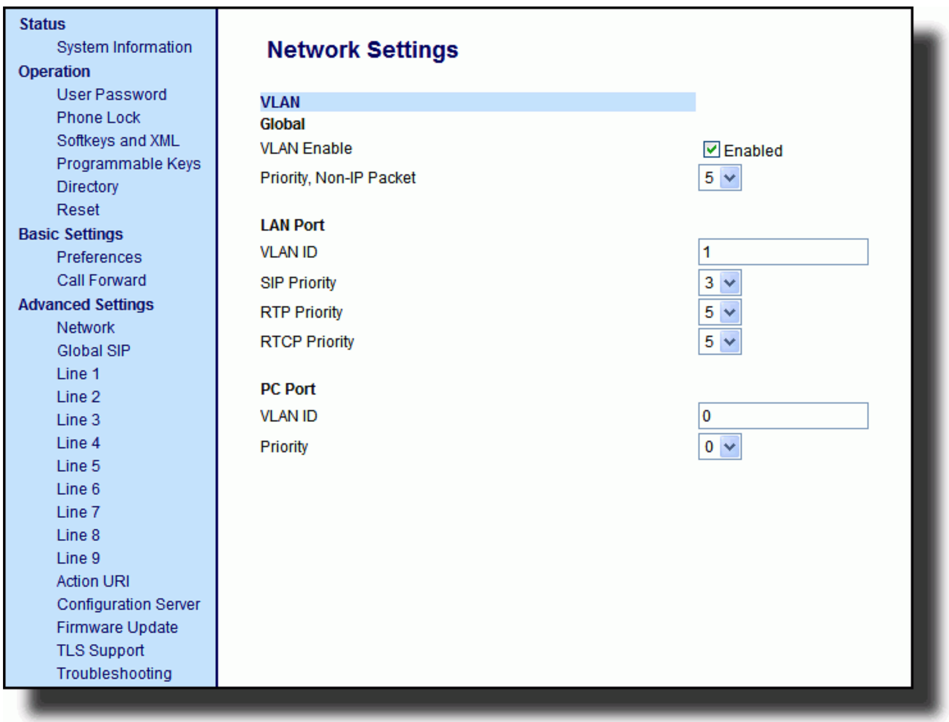
22	Enter a value from **1** to **4095** to specify the VLAN ID for the PC Port.
	Default is **1**.
	
	Note: If you set the PC Port VLAN ID (Port 1) to **4095**, all untagged packets are sent to this port. The following is an example of configuring the phone on a VLAN where all untagged packets are sent to the PC Port (passthrough port).
	
	Example
	You enable tagging on the phone port as normal but set the passthrough port (PC Port) to 4095. The following example sets the phone to be on VLAN 3 but the passthrough port is configured as untagged.
	
	`VLAN Settings->VLAN Enable: Yes`
	`VLAN Settings->Phone VLAN->Phone VLAN ID: 3`
	`VLAN Settings->PC Port VLAN->PC Port VLAN ID: 4095`

IP Phone UI

Step	Action
23	Press **Done** to save the change.
24	Select **PC Port Priority**.
25	Select a PC Port VLAN priority value from **0** to **7** for the PC Port. Default is **0**..
26	Press **Done** (4 times) to save the changes. **Note:** The session prompts you to restart the IP phone to apply the configuration settings
27	Select **Restart.**

Aastra Web UI

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



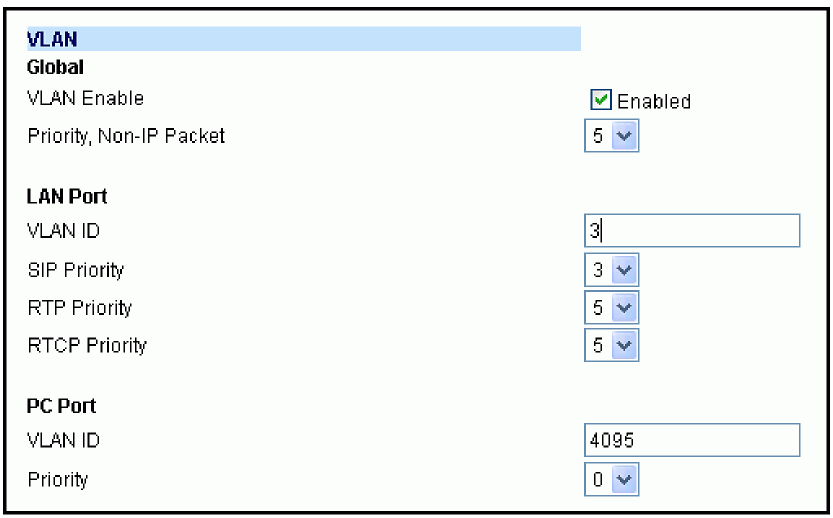
Step	Action
***To globally enable/disable VLAN and set priority for non-IP packets*:**	
2	Enable VLAN by checking the **VLAN Enable** field check box. (Disable this field by unchecking the check box).
3	With VLAN enabled, select the priority (**0** to **7**) for non-IP packets in the **Priority, Non-IP Packet** field.
***To set VLAN ID and priority for the LAN Port (Port 0)*:**	
4	Enter a VLAN ID value from **1** to **4094** in the **VLAN ID** field. Default is **1**.
5	Choose a VLAN Protocol (**SIP Priority**, **RTP Priority**, and/or **RTCP Priority**), and select a priority for the associated Protocol. Valid values are **0** to **7**, Defaults are as follows: SIP Priority = 3 RTP Priority = 5 RTCP Priority = 5

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td colspan="2">To set VLAN ID and priority for the PC Port (Port 1):</td></tr>
<tr><td>6</td><td>Enter a VLAN ID value from 1 to 4095 in the VLAN ID field. Default is 0.

Note: If you set the PC Port VLAN ID (Port 1) to 4095, all untagged packets are sent to this PC Port. The following is an example of configuring the phone on a VLAN where all untagged packets are sent to the passthrough port.

<u>Example</u>
You enable tagging as normal, enter a value for the LAN Port VLAN ID, and then set the PC Port VLAN ID to 4095. The following example sets the phone to be on VLAN 3 but the PC Port VLAN ID is configured as untagged..


</td></tr>
<tr><td>7</td><td>Select a VLAN priority value from 0 to 7 for the PC Port in the Priority field. Default is 0.</td></tr>
<tr><td>8</td><td>Click Save Settings to save your settings.</td></tr>
</table>

Configuring the IP Phones

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 three time servers in your network.

Note: The IP phones support NTP version 1.

Configuring NTP Servers (optional)

Use the following procedure to enable/disable and configure the NTP servers using the configuration files.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Time Server Settings" on page A-31.

Use the following procedure to enable/disable the NTP server using the IP Phone UI.

IP Phone UI

Step	Action
1	Press [Options key] on the phone to enter the Options List.
2	Select **Preferences.**
3	Select **Time and Date.**
4	Select **Time Server.**
5	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
6	Select **Timer Server 1**, **Timer Server 2**, or **Time Server 3**. **Note:** The Time Servers are disabled by default.
7	To set a Time Server, press **Enable**. (Press **Disable** to disable a Time Server.)

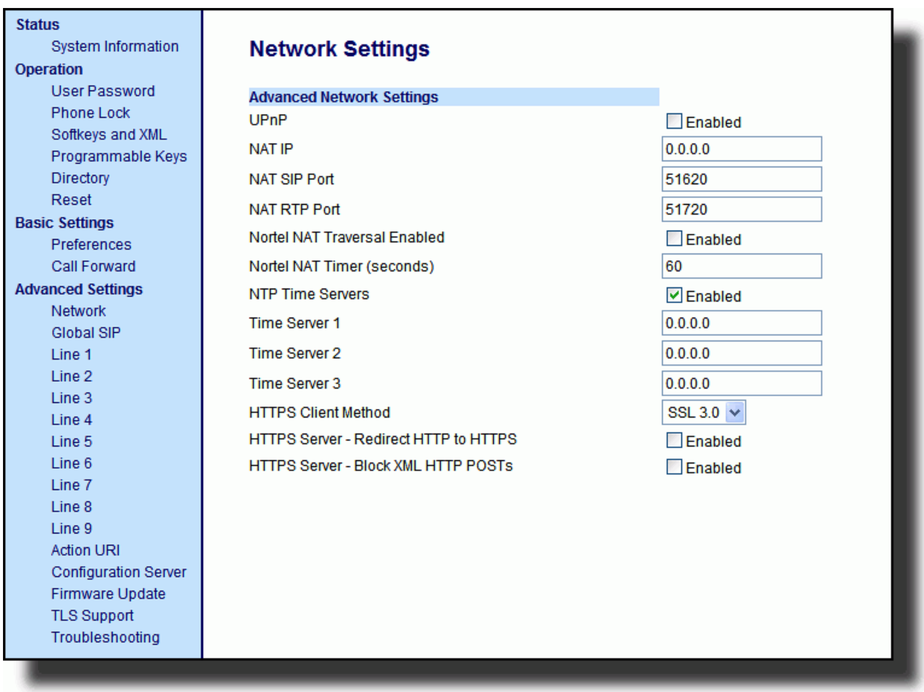
IP Phone UI

Step	Action
8	Enter the IP Address (in dotted decimal) or qualified domain name for the Time Server.
9	Press **Done** to save the change.

Use the following procedure to enable/disable and configure the NTP Servers using the Aastra Web UI.

Aastra Web UI

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



Step	Action
2	Enable the "**NTP Time Servers**" field by checking the check box. (Disable this field by unchecking the box). Default is disabled.

Configuring the IP Phones

Aastra Web UI

Step	Action
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.

Global SIP Settings

Description

The IP phone uses the information in the Global 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.

On the IP Phones, you can configure Basic and Advanced SIP Settings. The Basic SIP Settings include authentication and network settings. The Advanced SIP Settings include other features you can configure on the IP Phone.

Reference

For more information about Basic SIP Settings (for authentication and network), see "Basic SIP Settings" on page 4-46.

For more information bout Advanced SIP Settings, see "Advanced SIP Settings (optional)" on page 4-58.

Basic SIP Settings

The following tables identify the SIP global and per-line, authentication and network parameters on the IP phones.

SIP Global Parameters

IP Phone UI Parameters	Aastra Web UI Parameters	Configuration File Parameters
SIP Global Authentication Parameters		
• Screen Name • N/A • User Name • Display Name • Auth Name • Password • N/A • N/A	• Screen Name • Screen Name 2 • Phone Number • Caller ID • Authentication Name • Password • BLA Number • Line Mode	• sip screen name • sip screen name 2 • sip user name • sip display name • sip auth name • sip password • sip bla number • sip mode • sip vmail
SIP Global Network Parameters		
• Proxy Server • Proxy Port • N/A • N/A • N/A • N/A • Registrar Server • Registrar Port • N/A • N/A • N/A • N/A	• Proxy Server • Proxy Port • Backup Proxy Server • Backup Proxy Port • Outbound Proxy Server • Outbound Proxy Port • Registrar Server • Registrar Port • Backup Registrar Server • Backup Registrar Port • Registration Period • Conference Server URI (see Chapter 5)	• sip proxy ip • sip proxy port • sip backup proxy ip • sip backup proxy port • sip outbound proxy • sip outbound proxy port • sip registrar ip • sip registrar port • sip backup registrar ip • sip backup registrar port • sip registration period • sip centralized conf • (see Chapter 5)

Reference

For more information about centralized conferencing, see Chapter 5, the section, "Centralized Conferencing (for Sylantro and Broadsoft Servers)" on page 5-220.

SIP Per-Line Parameters

IP Phone UI Parameters	Aastra Web UI Parameters	Configuration File Parameters
SIP Per-Line Authentication Parameters		
• Screen Name • N/A • User Name • Display Name • Auth Name • Password • N/A • N/A	• Screen Name • Screen Name 2 • Phone Number • Caller ID • Authentication Name • Password • BLA Number • Line Mode	• sip lineN screen name • sip lineN screen name 2 • sip lineN user name • sip lineN display name • sip lineN auth name • sip lineN password • sip lineN bla number • sip lineN mode • sip lineN vmail
SIP Per-Line Network Parameters		
• Proxy Server • Proxy Port • N/A • N/A • N/A • N/A • Registrar Server • Registrar Port • N/A • N/A • N/A • N/A	• Proxy Server • Proxy Port • Backup Proxy Server • Backup Proxy Port • Outbound Proxy Server • Outbound Proxy Port • Registrar Server • Registrar Port • Backup Registrar Server • Backup Registrar Port • Registration Period • Conference Server URI • (see Chapter 5)	• sip lineN proxy ip • sip lineN proxy port • sip lineN backup proxy ip • sip lineN backup proxy port • sip lineN outbound proxy • sip lineN outbound proxy port • sip lineN registrar ip • sip lineN registrar port • sip lineN backup registrar ip • sip lineN backup registrar port • sip lineN registration period • sip lineN centralized conf • (see Chapter 5)

Reference

For more information about centralized conferencing, see Chapter 5, the section, "Centralized Conferencing (for Sylantro and Broadsoft Servers)" on page 5-220.

Note: The "sip vmail" and "sip lineN vmail" parameters are configurable using the configuration files only. To configure voicemail see Chapter 5, the section, "Voicemail (55i, 57i, and 57i CT only)" on page 5-193.

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 by a per-line configuration.

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

Backup Proxy/Registrar Support

The IP phones support a backup SIP proxy and backup SIP registrar feature. If the primary server is unavailable, the phone automatically switches to the backup server allowing the user's phone to remain in service.

How it Works

All SIP registration messages are sent to the primary registrar first. If the server is unavailable, then a new registration request is sent to the backup registrar. This also applies to registration renewal messages, which try the primary server before the backup.

Similarly, any outgoing calls attempt to use the primary proxy first, then the backup if necessary. In addition, subscriptions for BLF, BLA, and explicit MWI can also use the backup proxy when the primary fails. Outgoing calls and the previously mentioned subscriptions behave the same as registrations, where the primary proxy is tried before the backup.

You can configure the backup SIP proxy on a global or per-line basis via the configuration files or the Aastra Web UI.

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 multiple 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 Network Settings (optional)

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.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "SIP Basic, Global Settings" on page A-41 or "SIP Basic, Per-Line Settings" on page A-50

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

IP Phone UI

Step	Action
1	Press [Options key] on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **SIP Settings.**
Configuring Proxy IP and Proxy Port	
5	Select **Proxy IP/Port.**
6	Enter an IP address or fully qualified host name in the **Proxy Server** field. Default is **0.0.0.0**.
7	Enter a Proxy Port number in the **Proxy Port** field for accessing the SIP proxy server. Default is **0**.
8	Press **Done** to save the changes.
Configuring Registrar IP and Registrar Port	
9	Select **Registrar IP/Port.**
10	Enter an IP address or fully qualified host name in the **Registrar Server** field. Default is **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. 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.
11	Enter a Registrar Port number in the **Registrar Port** field for accessing the SIP registrar server. Default is **0**.
12	Press **Done** to save the changes.
Enabling/Disabling the Use of the Registrar Server	
13	Select **SIP Register**.
14	Press **Change** to set Register to "**Yes**" (enable) or "**No**" (disable). Default is “**Yes**”. This parameter enables/disables the IP phone to register on the network.
15	Press **Done** to save the changes.

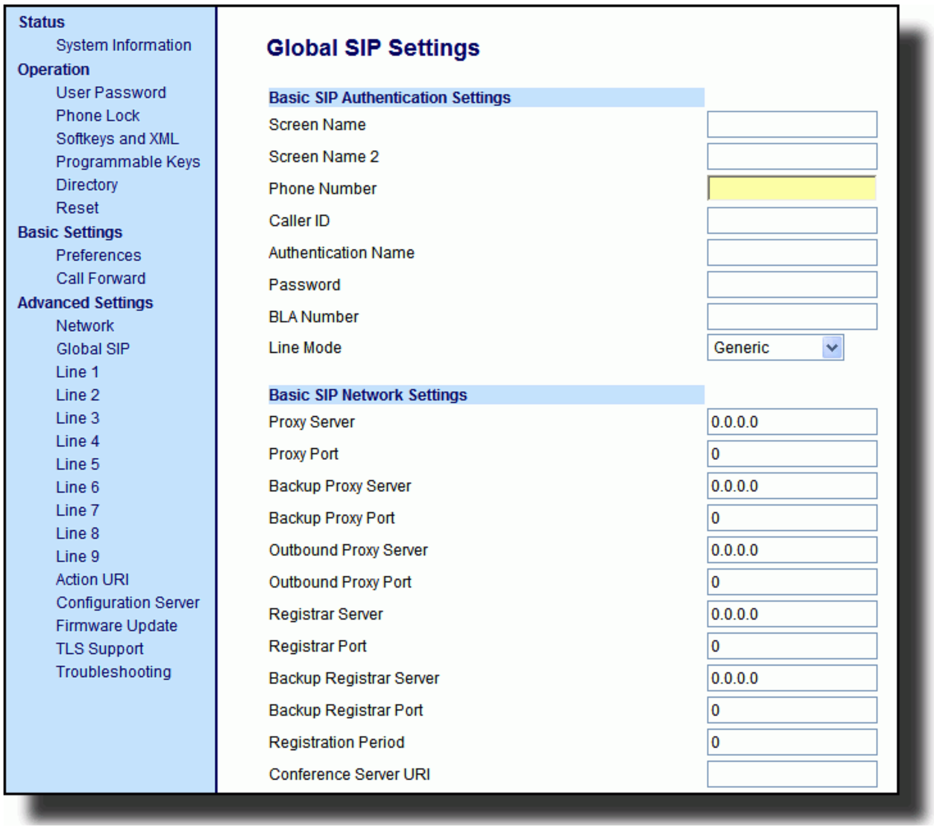
IP Phone UI

Step	Action
16	Select **User Name** to enter the username in the name field of the SIP URI for the IP phone, and for registering the phone at the registrar. **Note:** The IP phones allow usernames containing dots (“.”).
17	Press **Done** to save the changes.
18	Select **Display Name** to enter the name used in the display name field of the "From SIP" header field.
19	Press **Done** to save the changes.
20	Select **Screen Name** and enter the name to display on the idle screen.
21	Press **Done** to save the changes.
22	Select **Authentication Name** to enter the authorization name used in the username field of the Authorization header field of the SIP REGISTER request.
23	Press **Done** to save the changes.
24	Select **Password** to enter the password used to register the IP phone with the SIP proxy. **Note:** The IP phones accept numeric passwords only.
25	Press **Done** (3 times) to save the changes. **Note:** The session prompts you to restart the IP phone to apply the configuration settings
26	Select **Restart.**

.

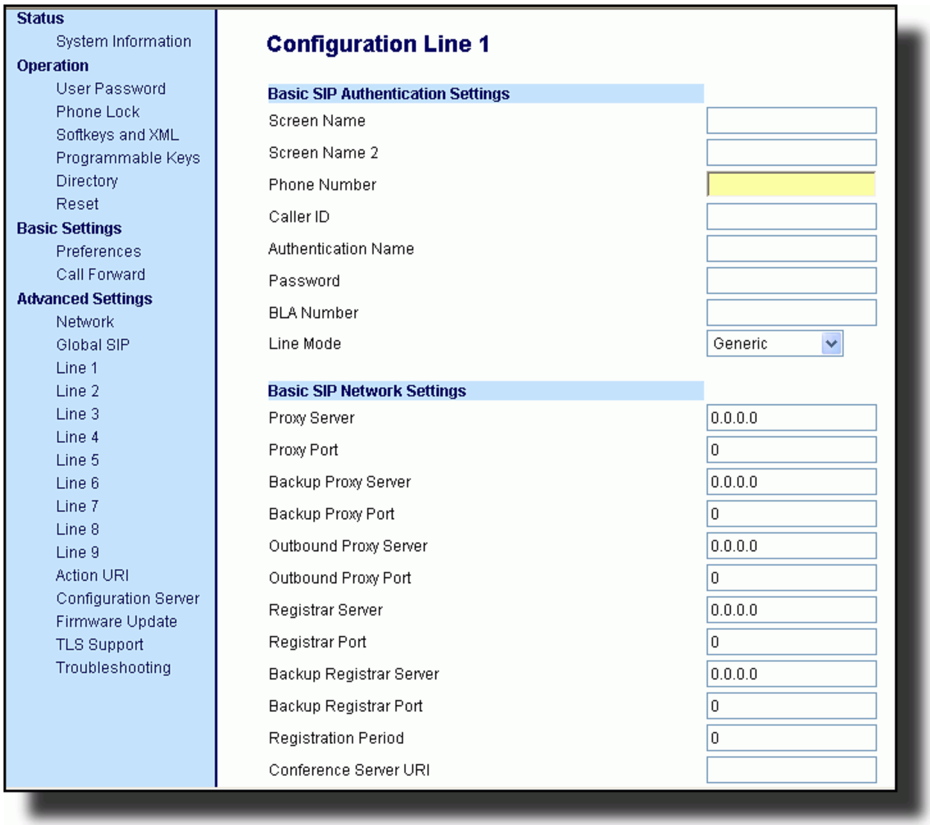
Aastra Web UI

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



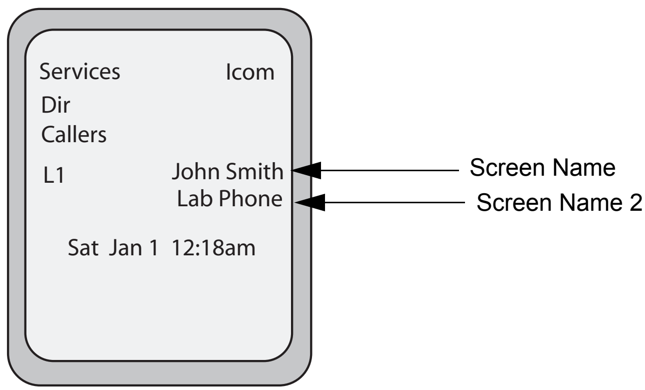
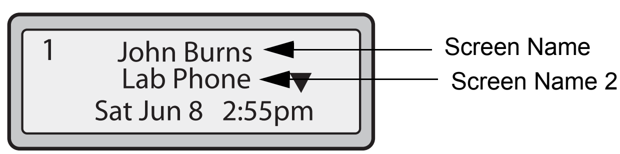
Aastra Web UI

Step	Action
	Or, for per-line configuration, click on **Advanced Settings->Line *N* (1-9).**



To configure SIP authentication settings:	
2	In the "**Screen Name**" field, enter the screen name that displays on the idle screen.

Aastra Web UI

Step	Action
3	In the "**Screen Name 2**" field, enter the text you want to display on the phone under the “Screen Name” on the idle screen. **Notes**: **1.** If other status messages display on the phone, such as “Network Disconnected”, the Screen Name 2 value does not display. **2.** Symbol characters are allowed (such as “#”). **3.** If the text is longer than the display width, than the display truncates the text to fit the display. **Figure 1** **57i**  **Figure 2** **53i** 
4	In the "**Phone Number**" field, enter the phone number of the IP phone.
5	In the "**Caller ID**" field, enter the phone number of the IP phone.
6	In the "**Authentication Name**" field, enter the name used in the username field of the Authorization header of the SIP REGISTER request.
7	In the "**Password**" field, enter the password used to register the IP phone with the SIP proxy. **Note:** The IP phones accept numeric passwords only.
8	In the "**BLA Number**" field, enter the Bridge Line Appearance (BLA) number to be shared across all IP phones.

Aastra Web UI

Step	Action
9	In the "**Line Mode**" field, select "Generic" for normal mode, "BroadSoft SCA" for a BroadWorks network, or "Nortel" for a Nortel network.
To configure SIP network settings:	
10	In the "**Proxy Server**" field, enter an IP address or fully qualified host name of the SIP proxy server.
11	In the "**Proxy Port**" field, enter a port number for accessing the SIP proxy server.
12	In the "**Backup Proxy Server**" field, enter an IP address or fully qualified host name for the backup proxy server.
13	In the "**Backup Proxy Port**" field, enter a port number for accessing the backup proxy server.
14	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.
15	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.
16	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.
17	In the "**Registrar Port**" field, enter the port number associated with the Registrar.
18	In the "**Backup Registrar Server**" field, enter an IP address or fully qualified host name for the backup 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 Backup 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.
19	In the "**Backup Registrar Port**" field, enter the port number associated with the backup registrar.
20	In the "**Registration Period**" field, enter the requested registration period, in seconds, from the registrar.

Aastra Web UI

Step	Action
21	To enter a value in the "**Conference Server URI**" field, see Chapter 5, the section, "Centralized Conferencing (for Sylantro and Broadsoft Servers)" on page 5-220.
22	Click Save Settings to save your changes.

Configuring the IP Phones

Advanced SIP Settings (optional)

Advanced SIP Settings on the IP Phone allow you to configure specific features on the phone. The following table provides a list of Advanced SIP Settings that you can configure using the Aastra Web UI or the configuration files.

Aastra Web UI Parameters	Configuration File Parameters
Explicit MWI Subscription Explicit MWI Subscription Period	sip explicit mwi subscription sip explicit mwi subscription period
Missed Call Summary Subscription (see Chapter 6) (global and per-line settings)	missed call summary subscription (see Chapter 6) (global and per-line settings)
Missed Call Summary Subscription Period (see Chapter 6)	missed call summary subscription period (see Chapter 6)
Send MAC Address in REGISTER Message (see Chapter 6)	sip send mac (see Chapter 6)
Send Line Number in REGISTER Message (see Chapter 6)	sip send line (see Chapter 6)
Session Timer	sip session timer
T1 Timer	sip T1 timer
T2 Timer	sip T2 timer
Transaction Timer	sip transaction timer
Transport Protocol	sip transport protocol
Registration Failed Retry Timer	sip registration retry timer
Registration Timeout Retry Timer	sip registration timeout retry timer
Registration Renewal Timer	sip registration renewal timer
BLF Subscription Period (see Chapter 5)	sip blf subscription period (see Chapter 5)
ACD Subscription Period (see Chapter 5)	sip acd subscription period (see Chapter 5)
Blacklist Duration (see Chapter 6)	sip blacklist duration (see Chapter 6)
Whitelist Proxy (see Chapter 6)	sip whitelist (see Chapter 6)

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

Reference

Refer to Appendix A,"Advanced SIP Settings" on page A-62 for a description of each of the above parameters.

For more information about Blacklist Duration and Whitelist Proxy, see Chapter 6, "Configuring Advanced Operational Features."

Configuring Advanced SIP Settings

Use the following procedures to configure the advanced SIP settings on the IP phone.

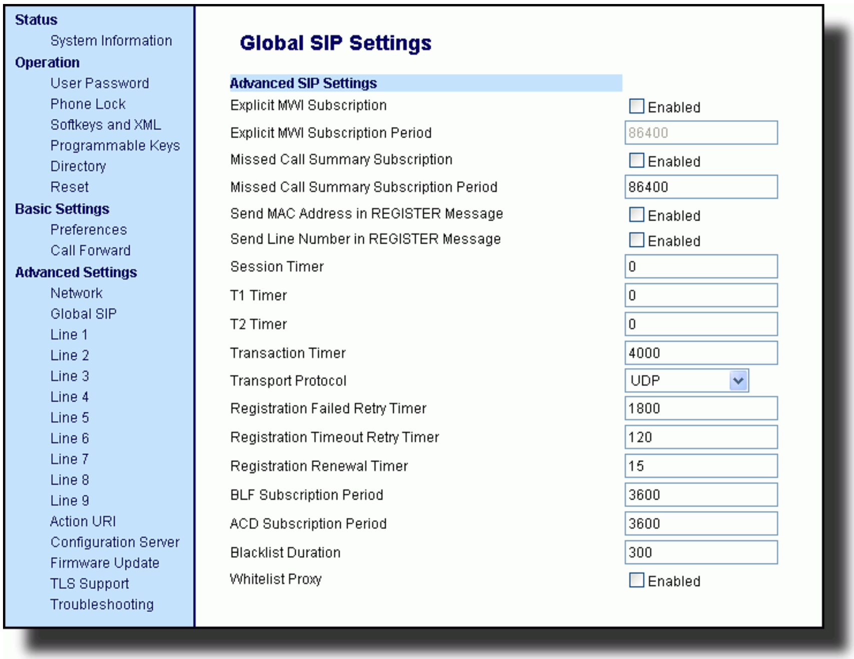
Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Advanced SIP Settings" on page A-62.

Aastra Web UI

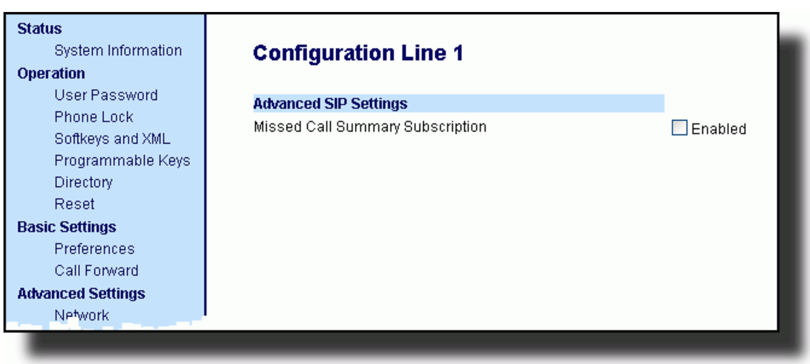
<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td></td><td>

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



</td></tr>
<tr><td>1</td><td>

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



</td></tr>
</table>

Aastra Web UI

Step	Action
2	Enable the "**Explicit MWI Subscription**" field by checking the check box. (Disable this field by unchecking the check box. Default is disabled). 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	If you enable the "**Explicit MWI Subscription**" field, then in the "**Explicit MWI Subscription Period**" field, enter the requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends. Default is 86400.
4	Enable the "**Missed Call Summary Subscription**" field by checking the check box. (Disable this field by unchecking the check box. Default is disabled). This feature allows missed calls that have been redirected by the server, to be incremented in the missed calls indicator on the phone it was initially directed to. For more information about this feature, see Chapter 6, "Configuring Advanced Operational Features.". **Note:** The "Missed Call Summary Subscription" feature is configurable on a global or per-line basis.
5	If you enable the "**Missed Call Summary Subscription**" field, then in the "**Missed Call Summary Subscription Period**" field, enter the amount of time, in seconds, that the phone uses the Missed Calls Summary Subscription feature. Default is 86400. For more information about this feature, see Chapter 6, "Configuring Advanced Operational Features." **Note:** The "Missed Call Summary Subscription Period" is configurable on a global basis only.
6	Enable the "**Send MAC Address in REGISTER Message**" and the "**Send Line Number in REGISTER Message**" fields by checking the check boxes. (Disable these fields by unchecking the check boxes. Default is disabled for both fields). For more information about these message features, see Chapter 6, "Configuring Advanced Operational Features.".
7	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.
8	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.

Aastra Web UI

Step	Action
9	In the "**Transaction Timer**" field, enter the amount of time, in milliseconds, that the phone allows the call server (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.
10	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) and Transmission Control Protocol (TCP), UDP, TCP, Transport Layer Security (TLS) or Persistent TLS. The value "**UDP**" is the default. For more information about TLS, see "RTP Encryption" on page 4-68 and Chapter 5, the section, "Transport Layer Security (TLS)" on page 6-21.
11	In the " **Registration Failed Retry Timer"** field, enter the amount of time, in seconds, that the phone waits between registration attempts when a registration is rejected by the registrar. Valid values are 30 to 1800. Default is 1800.
12	In the " **Registration Timeout Retry Timer"** field, enter the amount of time, in seconds, that the phone waits until it re-attempts to register after a REGISTER message times out. Valid values are 30 to 214748364. Default is 120.
13	In the " **Registration Renewal Timer"** field, enter the length of time, in seconds, prior to expiration, that the phone renews registrations. For example, if the value is set to 20, then 20 seconds before the registration is due to expire, a new REGISTER message is sent to the registrar to renew the registration. Valid values are 0 to 214748364. Default is 15.
14	The "**BLF Subscription Period**" field is enabled by default with a value of 3600 seconds. This feature sets 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. For information about setting the "BLF Subscription Period", see Chapter 5, the section, "BLF Subscription Period" on page 5-120.
15	(For Sylantro Servers) The "**ACD Subscription Period**" field is enabled by default with a value of 3600 seconds. This feature sets 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. For information about setting the "ACD Subscription Period", see Chapter 5, the section, "ACD Subscription Period" on page 5-133.

Aastra Web UI

Step	Action
16	(For Broadsoft Servers) The "**Blacklist Duration**" field is enabled by default with a value of 300 seconds (5 minutes). Valid values are 0 to 9999999. This feature sets 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. **Note:** The value of "**0**" disables the blacklist feature. For information about setting the "Blacklist Duration", see Chapter 6, the section, "Blacklist Duration (Broadsoft Servers)" on page 6-17.
17	Enable the "**Whitelist Proxy**" field by checking the check box. (Disable this field by unchecking the check box. Default is disabled). When this feature is enabled, an IP phone accepts call requests from a trusted proxy server *only*. The IP phone rejects any call requests from an untrusted proxy server. For information about setting the "Whitelist Proxy", see Chapter 6, the section, "Whitelist Proxy" on page 6-19.
18	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.

You can set the following parameters for RTP on the IP Phones:

Aastra Web UI Parameters	Configuration File Parameters
RTP Port	sip rtp port
Basic Codecs (G.711 u-Law, G.711 a-Law, G.729)	sip use basic codecs
Force RFC2833 Out-of-Band DTMF	sip out-of-band dtmf
Customized Codec Preference List	sip customized codec
DTMF Method (global and per-line settings)	sip dtmf method (global and per-line settings)
RTP Encryption (global and per-line settings)	sip srtp mode (global and per-line settings)
Silence Suppression	sip silence suppression

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.

The RTP port is assigned to the first line on the phone, and is then incremented for each subsequent line available within the phone to provided each line a unique RTP port for its own use.

On the 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 (G.711 u-Law, G.711 a-Law, G.729)

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 parameters 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 measurement 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 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.

RTP Encryption

The IP Phones include support for Secure Real-time Transfer Protocol (SRTP), using Session Description Protocol Security (SDES) key negotiation, for encryption and authentication of RTP/RTCP messages sent and received by the Aastra IP phones on your network.

As administrator, you specify the global SRTP setting for all lines on the IP phone. You can choose among three levels of SRTP encryption, as follows:

- **SRTP Disabled (defaul**t): IP phone generates and receives nonsecured RTP calls. If the IP phone gets called from SRTP enabled phone, it ignores SRTP tries to answer the call using RTP. If the receiving phone has SRTP only enabled, the call fails; however, if it has SRTP preferred enabled, it will accept RTP call.
- **SRTP Preferre**d: IP phone generates RTP secured calls, and accepts both secured and non-secured RTP calls. If the receiving phone is not SRTP enabled, it sends non-secured RTP calls instead.
- **SRTP Onl**y: IP phone generates and accepts RTP secured calls only; all other calls are rejected (fail).

You can override the global setting as necessary, configuring SRTP support on a per-line basis. This allows IP phone users to have both secured and unsecured lines operating on the same phone.

If an SRTP enabled IP phone initiates a call, and the receiving phone is also SRTP enabled, the IP Phone UI displays a “lock” icon, indicating that the call is secure. If the receiving phone does not support SRTP, the IP phone will send unsecured RTP messages instead of SRTP encrypted messages. However in this case, the IP Phone UI does not display the lock icon - indicating a non-secure call.

Note: If you enable SRTP, then you should also enable Transport Layer Security (TLS). This prevents capture of the key used for SRTP encryption. To enable TLC, set the **Transport Protocol** parameter (located on the Global SIP Settings menu) to **TLS**.

You can configure SRTP 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 silence suppression on a global-basis using the configuration files or the Aastra Web UI.

Configuring RTP Features

Use the following procedures to configure the RTP features on the IP phone.

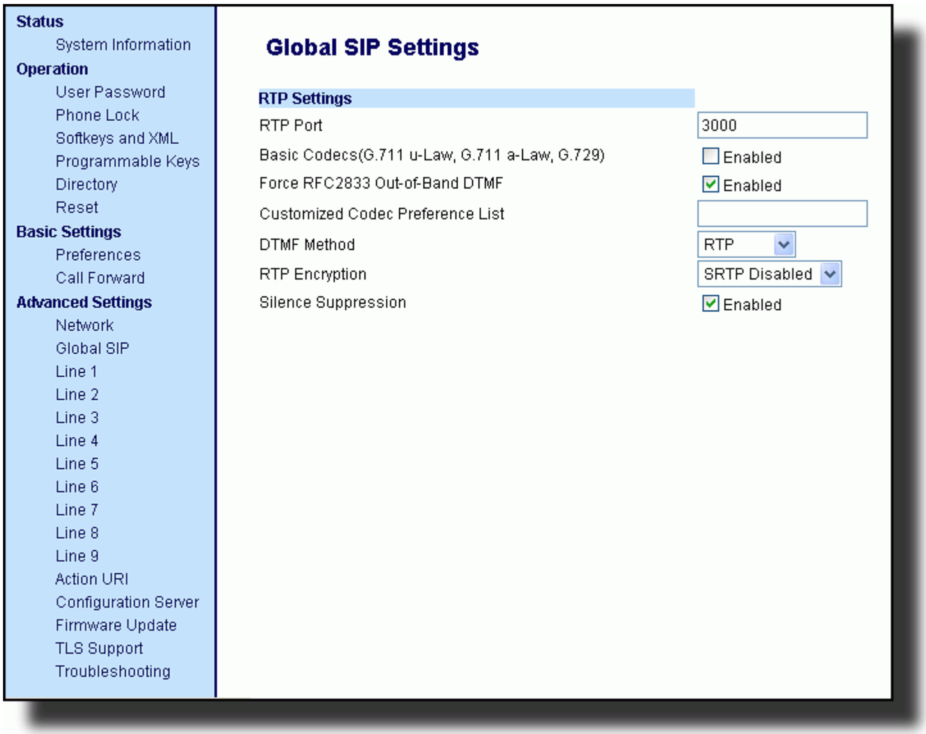
Configuration Files

For specific parameters you can set for RTP features in the configuration files, see Appendix A, the section, "RTP, Codec, DTMF Global Settings" on page A-77.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **SIP Settings.**
5	Select **RTP Port Base** to change the RTP port base setting. Default is **3000**.
6	Press **Done** (2 times) to save the change. **Note:** The session prompts you to restart the IP phone to apply the configuration settings
7	Select **Restart.**

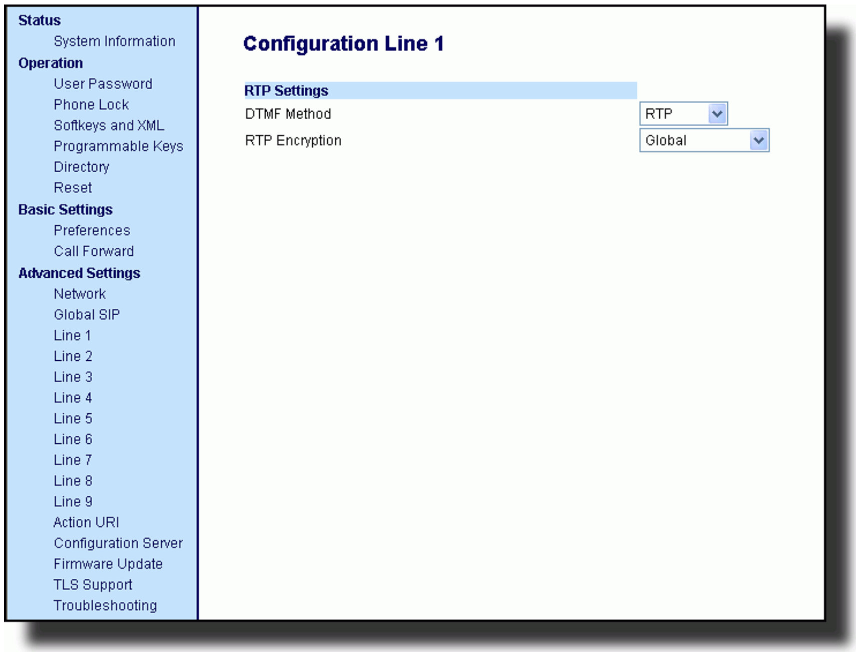
Aastra Web UI

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

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td></td><td>

Click on **Advanced Settings->Line <N>->RTP Settings.**
***Per-Line Settings*.**



</td></tr>
<tr><td>2</td><td>

Enter an RTP Port Base in the **RTP Port** field. Default is **3000**.

The RTP Port indicates the port through which the RTP packets are sent. This value must specify the beginning of the RTP port range on the gateway or router. 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.

</td></tr>
<tr><td>3</td><td>

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. Default is disabled).

Enabling this parameter allows the IP phone to use the basic Codecs when sending/receiving RTP packets.

</td></tr>
<tr><td>4</td><td>

The "**Force RFC2833 Out-of-Band DTMF**" field is enabled by default. Disable this field by unchecking the box.

Enabling this parameter forces the IP phone to use out-of-band DTMF according to RFC2833.

</td></tr>
</table>

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>5</td><td>Enter a "Customized Codec Preference List". For example,

<code>payload=8;ptime=10;
silsupp=on;
payload=0;ptime=10;
silsupp=off</code>

Valid values are:
<table>
<tr><th>Attribute</th><th>Value</th></tr>
<tr><td>payload</td><td>0 for G.711 u-Law
8 for G.711 a-Law
18 for G.729a</td></tr>
<tr><td>ptime (in milliseconds)</td><td>5, 10, 15, 20.....90</td></tr>
<tr><td>silsupp</td><td>on
off</td></tr>
</table>

For this parameter, you specify a customized codec list which allows you to use the preferred Codecs for this IP phone. Default for the “Customized Codec Preference List” is blank.</td></tr>
<tr><td>6</td><td>Select a method to use from the “DTMF Method” list box. Valid values are RTP, SIP Info, Both. Default is RTP.

Note: You can configure the DTMF Method on a global or per-line basis.</td></tr>
<tr><td>7</td><td>Select the type of RTP encryption to use from the “RTP Encryption” list box. Valid values are SRTP Disabled, SRTP Preferred, or SRTP Only. Default is SRTP Disabled.

Note: You can configure RTP Encryption on a global or per-line basis.</td></tr>
<tr><td>8</td><td>The “Silence Suppression” field is enabled by default. Disable this field by unchecking the check box.

When enabled, the phone negotiates whether or not to use silence suppression. Disabling this feature forces the phone to ignore any negotiated value.</td></tr>
<tr><td>9</td><td>Click [Save Settings] to save your changes.</td></tr>
</table>

Autodial Settings

The IP phones include a feature called "Autodial". When you configure Autodial on an IP phone, the phone automatically dials a preconfigured number whenever it is off-hook. Depending on the configuration you specify, the Autodial functions as either a "**hotline**", or as a "**warmline**," as follows:

- **Hotline (default)**: The IP phone immediately dials a preconfigured number when you lift the handset.
- **Warmline**: The IP phone waits for a specified amount of time after you lift the handset before dialing a preconfigured number. If you do not dial a number within the time allotted, then the IP phone begins to dial the number.

By default, the Autodial feature functions as a hotline. If you want Autodial to function as a warmline, you can use the Autodial "time-out" parameter to specify the length of time (in seconds) the IP phone waits before dialing a preconfigured number.

As administrator, you configure Autodial globally, or on a per-line basis, for an IP phone. The line setting overrides the global setting. For example, you can disable Autodial on a specific line simply by setting the line's autodial number parameter to empty (blank).

Note: IMPORTANT INFORMATION before configuring Autodial on your IP phone:

- Any speeddial numbers that you configure on an IP phone are not affected by autodial settings.
- If you configure autodial on your IP phone, any lines that function as hotlines do not accept conference calls, transferred calls, and/or intercom calls.

Configuring AutoDial Using the Configuration Files

You use the following parameters to configure Autodial using the configuration files:

Global Configuration

- **`sip autodial number`**
- **`sip autodial timeout`**

Per-Line Configuration

- **`sip lineN autodial number`**
- **`sip lineN autodial timeout`**

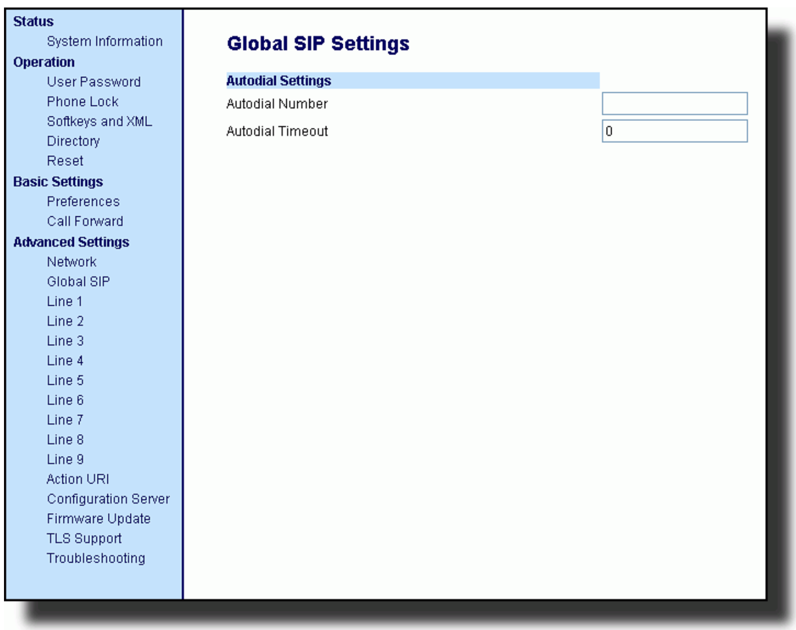
Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Autodial Settings" on page A-81.

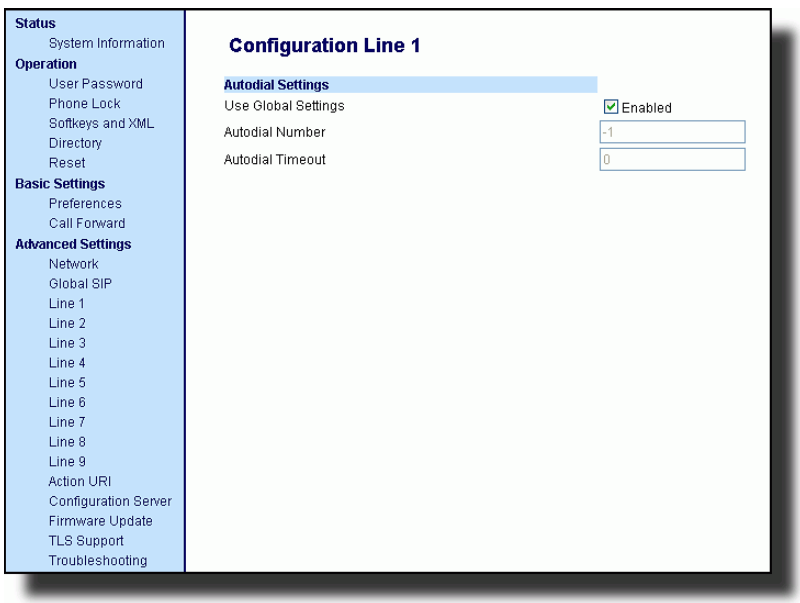
Configuring Autodial Using the Aastra Web UI

Use the following procedure to configure Autodial using the Aastra Web UI.

By default, your IP phone uses the global settings you specify for Autodial for all lines on your IP phone. However, you can also configure Autodial on a per-line basis.

Aastra Web UI	
Global Configuration	
1	Click on **Advanced Settings->Global SIP->Autodial Settings.** 
2	In the "**Autodial Number**" field, specify the SIP number that the IP phone dials whenever the IP phone is off-hook. An empty (blank) value disables autodial on the phone. For example: **8500**

	Aastra Web UI
3	In the **"Autodial Timeout"** field, specify a value, in seconds, for the timer as follows: • If you want the IP phone to autodial the number immediately (hotline) whenever the IP phone is off-hook, accept the default value of **0.** • If you want to specify a length of time for the IP phone to wait before dialing the number, enter the length of time (in seconds). For example: **30** Valid values are 0 to 120.
4	Click Save Settings to save your changes.
Per-Line Configuration	
1	Click on **Advanced Settings->Line <1 - 9>->Autodial Settings.**



	Aastra Web UI
2	Do one of the following actions: • To allow this line to use the global autodial settings, click on the **Use Global Settings** parameter to enable it, then click Save Settings to save your changes. • To specify a different autodial configuration for this specific line, disable the **Use Global Settings** parameter. Then proceed to step 3.
3	In the "**Autodial Number**" field, specify the SIP number for this line that the IP phone dials whenever the IP phone is off-hook as follows: • If set to -1, then the global autodial settings for this IP phone to this line. • If set to empty (blank), then disable Autodial on this line. • If set to a valid SIP number, dial the SIP number specified for this line. For example: **8500**
4	In the **"Autodial Timeout"** field, specify a value, in seconds, for the timer for this line as follows: • If you want the IP phone to autodial the number immediately (hotline) whenever the IP phone is off-hook, accept the default value of **0.** • If you want to specify a length of time for the IP phone to wait before dialing the number, enter the length of time (in seconds). For example: **30** Valid values are 0 to 120.
5	Click Save Settings to save your changes.

Configuration Server Protocol

You can download new versions of firmware and configuration files from the configuration server to the IP phone using any of the following types of protocols: TFTP, FTP, HTTP, and HTTPS. 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, FTP, HTTP, or HTTPS server, your DHCP server must support download protocol according to RFC2131 and RFC1541 for Option 66. For more information, see this chapter, the section, "DHCP" on page 4-4.

Configuring the Configuration Server Protocol

Use the following procedure to configure the configuration server protocol.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Configuration Server Settings" on page A-13.

IP Phone UI

Step	Action
1	Press [Options key] on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Configuration Server**.
5	Select **Download Protocol.**

Configuring the IP Phones

IP Phone UI

Step	Action
6	Select from the following: • **Use TFTP** • **Use FTP** • **Use HTTP** • **Use HTTPS** Default is "**Use TFTP**". The IP phone uses the protocol you select to download new firmware and configuration files from the configuration server.
7	Press **Done** to save the changes.
8	From the Configuration Server menu, select from the following. This selection is dependent on the Download Protocol you selected in step 6. **TFTP Settings** **FTP Settings** **HTTP Settings** **HTTPS Settings**

IP Phone UI

Step	Action
9	Enter the IP address of the protocol server (in dotted decimal format). Use the following table to configure the applicable server.

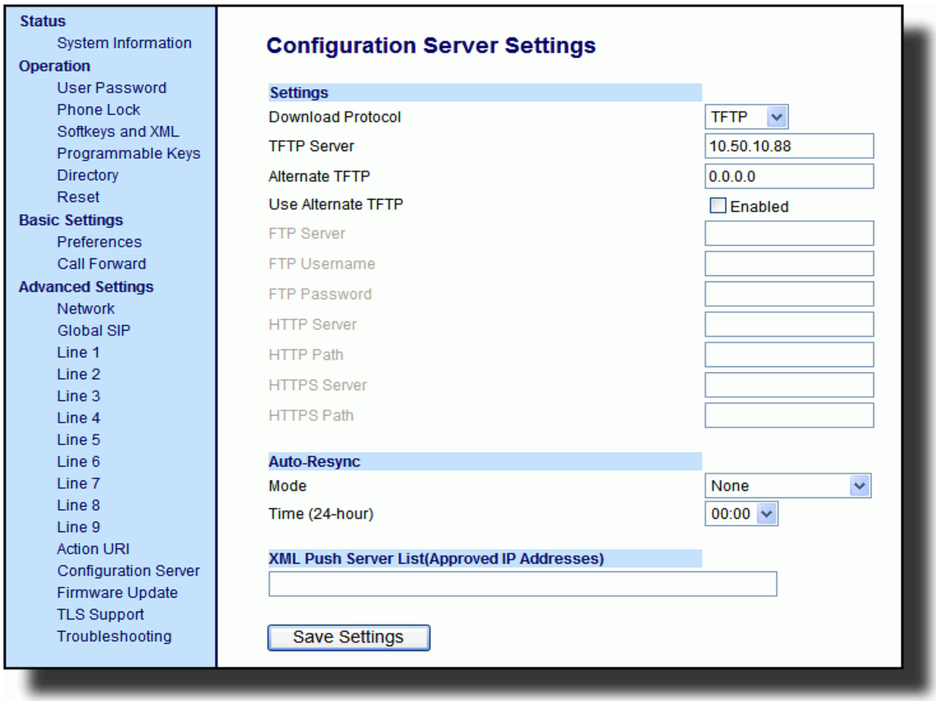
TFTP Settings
- Select **Primary TFTP** - Enter the IP address or qualified domain name of the primary TFTP server. - Press **Done** to save the change. **Optional**: You can also configure an Alternate TFTP server if required by selecting the "Alternate TFTP" selection and configuring an IP address or qualified domain name for the alternate TFTP server. - From the TFTP Settings menu, select "**Select TFTP"**. - Press **Change** to select the "**Primary**" or "**Alternate**" TFTP server to use for the phone. - Press **Done** to save the change.
FTP Settings
- 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 Settings
- 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**.

IP Phone UI

Step	Action
9 (Cont'd)	**HTTPS Settings** - Select **HTTP Client.** - Select **Download Server**. - Enter the IP address of the HTTPS server. - Press **Done**. - Select **Download Path**. - Enter the HTTPS path name. If the IP phone's configuration files and firmware files are located in a sub-directory beneath the server's HTTPS root directory, the relative path to that sub-directory should be entered in this field. - Press **Done**. - Select **HTTPS Server.** - Select **HTTP->HTTPS**. - Press **Change** to select "**Yes**" or "**No**". Default is "**No**". Enabling this feature redirects the HTTP protocol to HTTPS. - Press **Done**. - Select **XML HTTP POSTs**. - Press **Change** to select "**Yes**" or "**No**". Default is "**No**". Enabling this feature blocks XML HTTP POSTs from the IP Phone. **Reference**: For more information on configuring the HTTPS security method, HTTP to HTTPS redirect, and HTTPS server blocking for HTTP XML POSTs, see the section, "HTTPS Client/ Server Configuration" on page 4-23.
10	Press **Done** (4 times) to save the changes. **Note:** The session prompts you to restart the IP phone to apply the configuration settings
11	Select **Restart.**

Aastra Web UI

Step	Action
1	Click on **Advanced Settings->Configuration Server.**



Aastra Web UI

Step	Action
2	Select the protocol from the "**Download Protocol**" list box. Valid values are **TFTP**, **FTP**, **HTTP**, and **HTTPS**. Default is **TFTP**. 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 username 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 **"XML Push Server List (Approved IP Addresses)"** field. **HTTPS** - Enter an IP address in the **"HTTPS Server"** field. - Enter a root directory path for the HTTPS 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 **"XML Push Server List (Approved IP Addresses)"** field. **Reference**: For more information on configuring the HTTPS security method, HTTP to HTTPS redirect, and HTTPS server blocking for HTTP XML POSTs, see the section, "HTTPS Client/Server Configuration" on page 4-23.

Aastra Web UI	
Step	**Action**
3	Click Save Settings to save your settings. **Note:** The session prompts you to restart the IP phone to apply the configuration settings.
4	Select **Operation->Reset** and click Restart.

Chapter 5
Configuring Operational Features

About this chapter

Introduction

The IP phones have specific operational features you can configure to customize your IP phone. This chapter describes each feature and provides procedures for configuring your phone to use these features.

Topics

This chapter covers the following topics:

Operational IP Phone Features

Operational Features

Description

This section describes the operational features managed and configured by a System Administrator.

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.

Note: The IP phones support numeric characters only in passwords. If you enter a password with alpha characters, the phone uses the default password instead.

Configuring a User Password

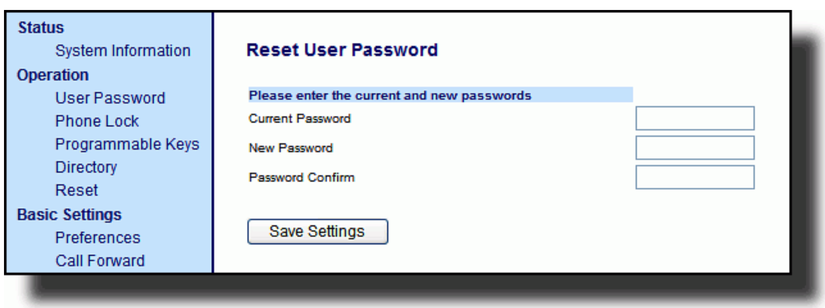
Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Password Settings" on page A-9.

IP Phone UI

1	Press on the phone to enter the Options List.
2	Select **User Password**.
3	Enter the current user password.
4	Press **Enter**.

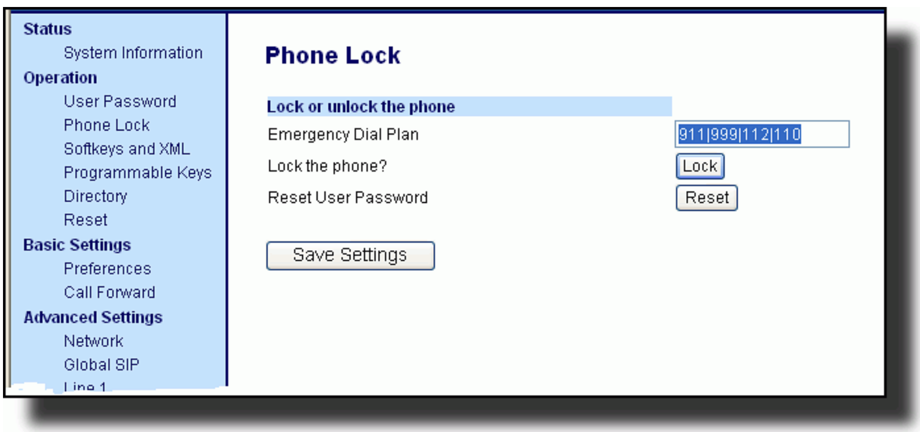
IP Phone UI	
5	Enter the new user password. **Note:** The IP phones support numeric characters only in passwords. If you enter a password with alpha characters, the phone uses the default password instead.
6	Press **Enter**.
7	Re-enter the new user password.
8	Press **Enter**. A message, "Password Changed" displays on the screen.

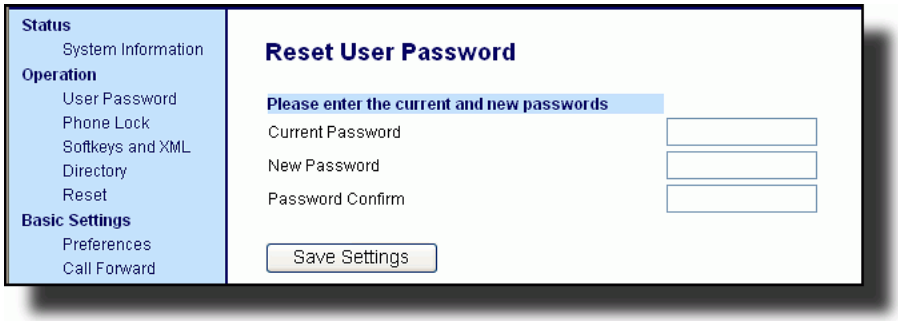
Aastra Web UI	
1	Click on **Operation->User Password**. 
2	In the "**Current Password**" field, enter the current user password.
3	In the "**New Password**" field, enter the new user password. **Note:** The IP phones support numeric characters only in passwords. If you enter a password with alpha characters, the phone uses the default password instead.
4	In the "**Password Confirm**" field, enter the new user password again.
5	Click Save Settings to save your changes.

Resetting a User Password

If a user forgets his password, either the user or an administrator can reset it so a new password can be entered. The reset user password feature resets the password to the factory default which is blank (no password).

You can reset a user password using the Aastra Web UI only at the path *Operation->Phone Lock*. Use the following procedure to reset a user password.

Aastra Web UI	
1	Click on **Operation->Phone Lock**. 

	Aastra Web UI
2	In the "**Reset User Password**" field, click Reset . The following screen displays: 
3	In the "**Current Password**" field, leave this blank.
4	In the "**New Password**" field, enter a new password for the user. **Note:** The IP phones support numeric characters only in passwords. If you enter a password with alpha characters, the phone uses the default password instead.
5	In the "**Password Confirm**" field, re-enter the new user password.
6	Click Save Settings to save the new user password and perform the next procedure.

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 Menu is denied and the IP phone returns to the idle screen.

Procedure

Use the following procedure to change the administrator password.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Password Settings" on page A-9.

Locking/Unlocking the Phone

A user or administrator can lock a phone to prevent it from being used or configured. Once the phone is locked, the user or administrator can enter their password to unlock the phone.

You can lock/unlock a phone using the configuration files, the IP Phone UI, or the Aastra Web UI.

You can use any of the following methods to lock/unlock a phone:

- Using the IP Phone UI via the “Phone Lock” option in the Options Menu.
- Using the Aastra Web UI via the path *Operation->Phone Lock*.
- Using the configuration files to configure a key as “phonelock”, and then pressing the key to lock/unlock the phone.
- Using the Aastra Web UI to configure a key as “Phone Lock”, and then pressing the key to lock/unlock the phone.

Note: All of the methods above configure locking/unlocking of the phone dynamically. Once configured, the feature takes affect immediately. To unlock the phone, a user or administrator must enter their password.

Locking/Unlocking the Phone Using the IP Phone UI

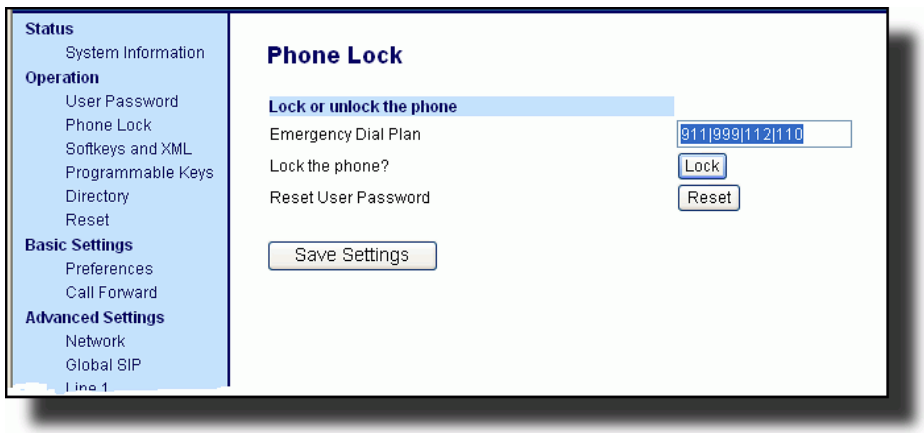
Use the following IP Phone UI procedure to lock/unlock an IP phone and prevent it from being used or configured.

IP Phone UI

Step	Action
Lock the phone:	
1	Press the Options key on the phone to enter the Options List.
2	Select **Phone Lock.** The prompt, "*Lock the phone*?" displays.
3	Press **Lock** to lock the phone.
Unlock the phone:	
1	Press the Options key on the phone to enter the Options List. The prompt, "*To unlock the phone...Password:*"
2	Enter the user or administrator password and press **Enter**. The phone unlocks.

Locking/Unlocking the Phone Using the Aastra Web UI

Use the following Aastra Web UI procedure to lock/unlock an IP phone and prevent it from being used or configured.

	Aastra Web UI
1	Click on **Operation->Phone Lock**. 
Lock the phone:	
2	In the "**Lock the Phone?**" field, click Lock . The phone locks dynamically and displays the following message: "*Phone is locked*".
Unlock the phone:	
3	Click on **Operation->Phone Lock**.
4	In the "**Unlock the Phone?**" field, click Unlock . The phone unlocks dynamically and displays the following message: "*Phone is unlocked*".

Configuring a Lock/Unlock Key Using the Configuration Files

Using the configuration files, you can configure a key on the phone (softkey, programmable key, or expansion module key) to use as a lock/unlock key. In the configuration files, you assign the function of the key as "**phonelock**".

Use the following procedure to configure a key as a lock/unlock key using the configuration files.

Configuration Files
To configure a softkey/programmable key as a lock/unlock key using the configuration files, see Appendix A, the section, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

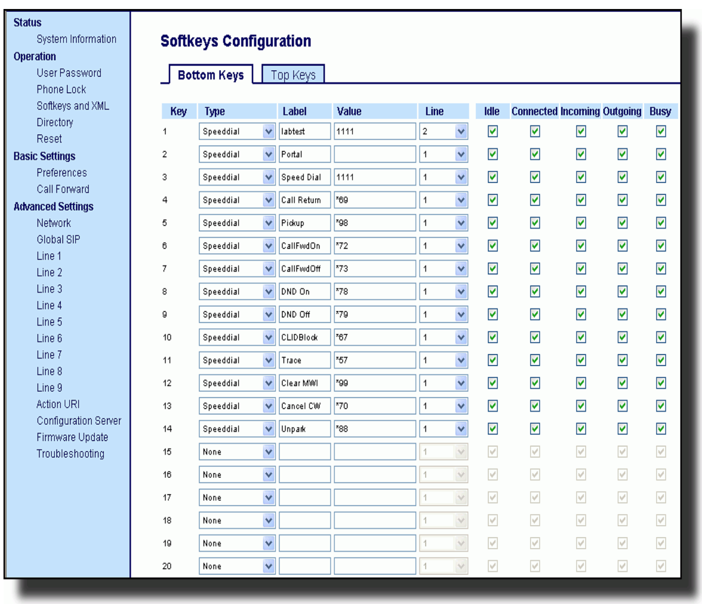
Reference

To use the lock/unlock softkey or programmable key, see "Using the Configured Lock/Unlock Key on the IP Phone" on page 5-14.

Configuring a Lock/Unlock Key using the Aastra Web UI

Using the Aastra Web UI, you can configure a key on the phone (softkey, programmable key, expansion module key) to use as a lock/unlock key. In the Aastra Web UI, you assign the function of the as “**Phone Lock**”.

Use the following procedure to configure a key as a lock/unlock key using the Aastra Web UI.

	Aastra Web UI
1	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** **Note:** Depending on your phone-model, the key configuration screen displays. The 57i Key Screen is shown as an example. 
2	Select a key you want to configure for locking/unlocking the phone.

Aastra Web UI	
3	In the "**Type**" field, select **Phone Lock** from the list of options.
4	Click Save Settings to save your changes.

Using the Configured Lock/Unlock Key on the IP Phone

After configuring a key as a lock/unlock key, refer to the following procedure to use the key on the IP phone.

IP Phone UI

Step	**Action**
Lock the phone:	
1	Press the **LOCK** softkey. The phone locks. The LED for the softkey AND the Message Waiting Lamp illuminate steady ON. An "**Unlock**" label appears next to the softkey you just pressed.
Unlock the phone:	
1	Press the **UNLOCK** softkey. A password prompt displays.
2	Enter the user or administrator password and press ENTER. The phone unlocks. The LED for the softkey AND the Message Waiting Lamp go OFF. The "**Lock**" label appears next to the softkey you just pressed.

Defining an Emergency Dial Plan

Public telephone networks in countries around the world have a single emergency telephone number (emergency services number), that allows a caller to contact local emergency services for assistance when required. The emergency telephone number may differ from country to country. It is typically a three-digit number so that it can be easily remembered and dialed quickly. Some countries have a different emergency number for each of the different emergency services.

You can specify the digits to dial on the IP phone for contacting emergency services. Once you specify the emergency number(s) on the phone, you can dial those numbers directly on the dial pad when required and the phone automatically dials to those emergency services.

➡ **Note:** Contact your local phone service provider for available emergency numbers in your area.

The following table describes the default emergency numbers on the IP phones.

Emergency Number	Description
911	A United States emergency number
999	A United Kingdom emergency number
112	An international emergency telephone number for **GSM mobile phone** networks. In all European Union countries it is also the emergency telephone number for both mobile and fixed-line telephones.
110	A police and/or fire emergency number in Asia, Europe, Middle East, and South America.

You can set the emergency dial plan via the configuration files or the Aastra Web UI.

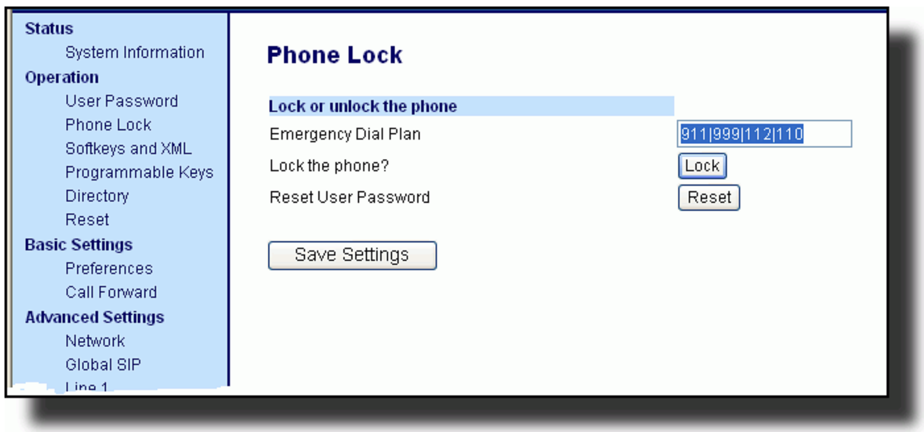
Configuring an Emergency Dial Plan

Use the following procedures to specify the numbers to use on your phone for dialing emergency services in your area.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, “Emergency Dial Plan Settings” on page A-11.

Aastra Web UI

1	Click on **Operation->Phone Lock**. 
2	In the “**Emergency Dial Plan**” field, enter the 3-digit number used in your local area to contact emergency services. For multiple numbers, enter a “**|**” between each emergency number. For example: **911|110**. Default for this field is **911|999|112|110**.
3	Click **Save Settings** to save the emergency dial plan to your phone.

Time and Date

In addition to enabling/disabling the time server, you can also set the time and date, set the time and date format, set the time zone, and set daylight savings time on the IP phones. You configure these features using the configuration files, the IP Phone UI, or the Aastra Web UI. The following table identifies which method of configuration applies to each feature.

Feature	Method of configuration
Set Time	IP Phone UI
Set Time Format	Configuration Files IP Phone UI Aastra Web UI
Set Date	IP Phone UI
Set Date Format	Configuration Files IP Phone UI Aastra Web UI
Set Time Zone	IP Phone UI Configuration Files
Set Daylight Savings Time	IP Phone UI Configuration Files

Daylight Savings Time (DST) Information

The Aastra IP Phones incorporate the federally mandated DST observance change. This change became affective starting in 2007.

The US has made a change to its daylight savings time observance starting in 2007. The Energy Policy Act of 2005 mandates that DST will now begin at 2:00 A.M. on the second Sunday in March and revert to Standard time on the first Sunday in November.

Note: In previous years, the DST began on the first Sunday of April and ended on the first Sunday of October.

The changes to daylight savings time applies to the U.S. and Canada, but may impact other countries outside North America.

Use the following procedures to set a time and date, time and date format, time zone, and daylight savings time on the IP phones.

Configuration Files

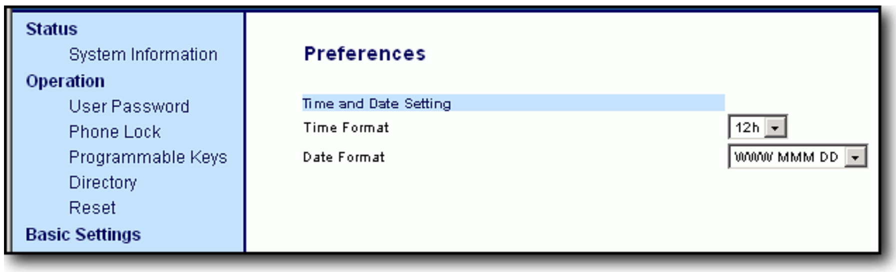
For specific parameters you can set in the configuration files, see Appendix A, the section, "Time and Date Settings" on page A-33.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
Set Time and Time Format:	
2	Select **Time and Date.**
3	Select **Set Time.**
4	Using the keys on the keypad, enter a time to set on the IP phone.
5	Press **Done** to save the time you entered.
6	Select **Time Format.** Valid values are **12hr** and **24hr**. **Note:** The default Time Format is **12hr**.
7	Press **Change** to toggle between 24hr and 12hr format.
8	Press **Done** to save the Time Format you selected.
Set Date and Date Format:	
9	Select **Set Date.**
10	Using the keys on the keypad, enter a date to set on the IP phone.
11	Press **Done** to save the date you entered.
12	Select **Date Format.**

IP Phone UI

Step	Action
13	Select a date format from the list of options. Valid values are: • WWW MMM DD (default) • DD-MMM-YY • YYYY-MM-DD • DD/MM/YYYY • DD/MM/YY • DD-MM-YY • MM/DD/YY • MMM DD **Note:** The default Date Format is **WWW MMM DD** (Day of Week, Month, Day).
14	Press **Done** to save the Date Format.
Set Time Zone:	
15	Select **Time Zone**.
16	***For 53i:*** Press * to display a list of Time Zone options.
17	Select a Time Zone from the list of options. For valid values, see Appendix A, the section, "Time and Date Settings" on page A-33. **Note:** The default Time Zone is **US-Eastern**.
18	Press **Done** to save the Time Zone you selected.
Set Daylight Savings Time:	
19	Select **Daylight Savings**.
20	Select a Daylight Savings time from the list of options. Valid values are: • OFF • 30 min summertime • 1 hr summertime • automatic **Note:** The default for Daylight Savings is **Automatic**.
21	Press **Done** to save the Daylight Savings value you selected.

	Aastra Web UI
1	Click on **Basic Settings->Preferences->Time and Date Setting**. 
2	In the **"Time Format"** field, select the time format you want to use on your phone. Valid values are: • **12h** (12 hour format) (default) • **24h** (24 hour format). **Note:** The time displays on the phone's idle screen in the format you select for this field.
3	In the **"Date Format"** field, select the date format you want to use on your phone. Valid values are: • WWW MMM DD (default) • DD-MMM-YY • YYYY-MM-DD • DD/MM/YYYY • DD/MM/YY • DD-MM-YY • MM/DD/YY • MMM DD **Note:** The date displays on the phone's idle screen in the format you select for this field.
4	Click Save Settings to save your changes.

Language

The IP phones support several different languages. You can have the IP phone UI and the Aastra Web UI display in a specific language as required. When you set the language to use, all of the display screens (menus, services, options, configuration parameters, etc.) display in that language. The IP phones support the following languages:

- English (default)
- French
- Spanish
- German (not applicable to 57i CT cordless handset)
- Italian (not applicable to 57i CT cordless handset)

Loading Language Packs

You make languages available to use on the phone by loading language packs from the configuration server to the local *<mac>.cfg* configuration file. You can use the configuration files or the Aastra Web UI to perform the download. Each language pack consists of the IP Phone UI and Aastra Web UI translated in a specific language.

Loading Language Packs via the Configuration File (<mac>.cfg)

Using the configuration files, you specify a language pack to load in the following format:

```
lang_<ISO 639>-<ISO 3166>.txt
or
lang_<ISO 639>.txt
```

where <ISO 639> is the language code specified in Standard ISO 639 (see Appendix A, the section, Language Codes (from Standard ISO 639) on page A-111) and <ISO 3166> is the country code specified in Standard ISO 3166 (see Country Codes (from Standard ISO 3166) on page A-111).
The <ISO 3166> attribute is optional.

Note: Adding/changing language packs can only be done at bootup of the IP phone. The default language (English) cannot be changed or removed.

Example

The following is an example of the parameters you would enter in the *<mac>.cfg* file to load a French, Italian, German, and Spanish language pack to the IP phone.

```
language 1: lang_fr_ca.txt
language 2: lang_it.txt
language 3: lang_de.txt
language 4: lang_es_mx.txt
```

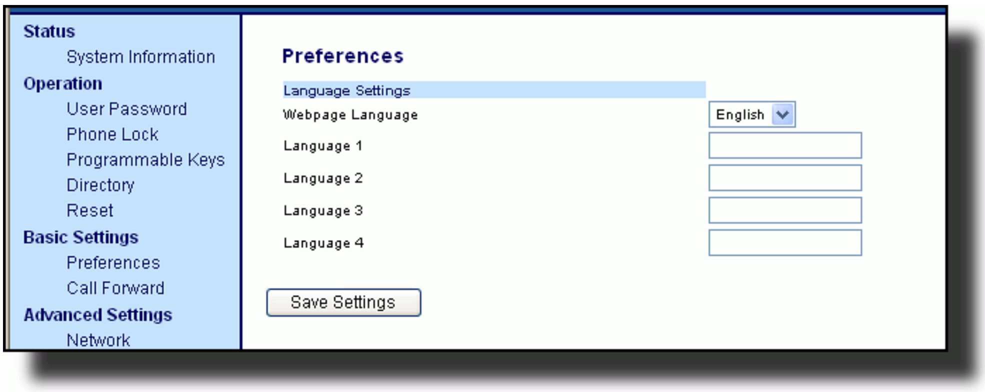
The above entries in the *<mac>.cfg* file tells the phone which language packs to load. When the language pack(s) have loaded, you must then use the configuration files IP Phone UI to specify which language to display on the IP phone. You must use the Aastra Web UI to specify the language to use in the Web UI.

For more information about specifying the language to use, see the section, "Specifying the Language to Use" on page 5-24.

For more information about language codes and country codes, see Appendix A, the section, "Language Pack Settings" on page A-110.

Loading Language Packs via the Aastra Web UI

Using the Aastra Web UI, you can specify a language pack to load using the parameters at *Basic Settings->Preferences->Language Settings*.



You use the following fields in the Aastra Web UI to specify which language packs to load:

Once the language pack is loaded to the phone, it is available for selection from either the configuration files, the IP Phone UI or the Aastra Web UI.

Specifying the Language to Use

Once the language pack(s) have loaded, you must then specify which language to use on the phone. After the phone has booted up, you can specify which language(s) to use. You can use the configuration files and the IP Phone UI to specify the language for the IP Phone UI. You can use the Aastra Web UI to specify the files for the Aastra Web UI.

Use the following procedures to specify the language to use on the IP phone.

Configuration Files

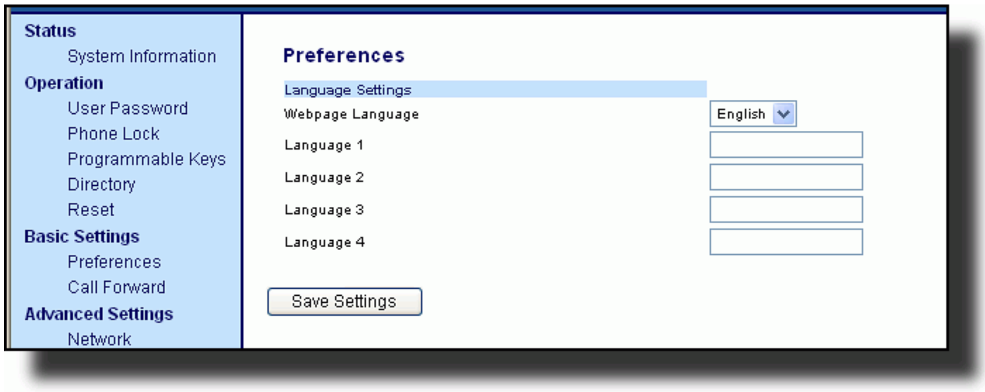
For specific parameters you can set in the configuration files, see Appendix A, the section, "Language Settings" on page A-109 and "Language Pack Settings" on page A-110.

Notes:
1. If you specify the language to use on the phone via the configuration files, you must reboot the phone for the changes to take affect.
2. All languages may not be available for selection. The available languages are dependant on the language packs currently loaded to the IP phone.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Language**.
3	Select **English** (English)**, Français** (French)**, Español** (Spanish)**, Deutsch** (German)**, Italiano** (Italian). **Notes**: **1**. Valid values for the 57i CT are English, French, and Spanish only. **2**. All languages may not be available for selection. The available languages are dependant on the language packs currently loaded to the IP phone. English is the default language and cannot be changed or removed. For more information about loading language packs, see "Loading Language Packs" on page 5-21.
4	Press **Done** to save the changes. The change is dynamic. The IP phone UI displays all menu items in the language you chose.

Aastra Web UI

1	Click on **Basic Settings->Preferences->Language Settings.** 
Loading the Language Pack	

2	In the "**Language N**" fields, enter the file name of the language pack you want to use to display a specific language in the Aastra Web UI. For example, you could enter the following in the "Language 1", "Language 2", "Language 3", and "Language 4" fields to display the Aastra Web UI in French, Spanish, German, and Italian:
	
	`lang_fr-ca.txt`
	`lang_es.txt`
	`lang_de.txt`
	`lang_it.txt`
	
	Note: You must have the language pack(s) already loaded to your phone in order to use them. For more information about loading language packs, see "Loading Language Packs" on page 5-21. For more information about language codes and country codes, see Appendix A, the section, "Language Pack Settings" on page A-110.

3	Click  to save your changes.
Specifying the Language to Use in the Aastra Web UI	
4	After restarting your phone, log back in using the Aastra Web UI.
5	Click on **Basic Settings->Preferences->Language Settings.**

	Aastra Web UI
6	In the “**Webpage Language**” field, select a language to apply to the Aastra Web UI. The IP phone supports the following languages: • English (default) • French (Canadian) • Spanish (Mexican) • German • Italian **Notes**: **1**. Valid values for the 57i CT are English, French, and Spanish only. **2**. All languages may not be available for selection. The available languages are dependant on the language packs currently loaded to the IP phone. English is the default language and cannot be changed or removed.
7	Click **Save Settings** to save your changes. The Aastra Web UI displays all screens in the language you chose.

Locking IP Phone Keys

The IP phones allow you to lock or unlock programmable keys, softkeys, hard keys, cordless handset keys, and expansion keys (for expansion modules). When key locking is enabled, the phone uses the server settings and ignores any previous local configuration. A user cannot override the configuration of a locked key.

You can lock and unlock keys using the configuration files, the IP Phone UI, or the Aastra Web UI. When viewing the locked key via the Aastra Web UI, the key is grayed out (disabled) and cannot be changed. Locking is dynamic for XML pushes.

You use the following “locking” parameters in the configuration files to lock the softkeys and programmable keys on the 53i, 55i, 57i, and 57i CT. The locking parameters impact existing softkey and programmable key parameters as shown in the table below.

Locking Parameter	Impacted Parameters	Phone Model Affected
softkeyN locked	softkeyN type softkeyN label softkeyN value softkeyN line softkeyN states	55i 57i 57i CT
topsoftkeyN locked	topsoftkeyN type topsoftkeyN label topsoftkeyN value topsoftkeyN line	57i 57i CT
prgkeyN locked	prgkeyN type prgkeyN value prgkeyN line	53i 55i
featurekeyN locked	featurekeyN type featurekeyN label	57i CT
expmodX keyN locked	expmodX keyN type expmodX keyN value expmodX keyN line	5-series expansion modules

Note: The 53i IP phone prevents users from setting a speed dial key via the Phone UI on a key that has been locked.

Locking the IP Phone Keys

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

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Locking Softkeys and Programmable Keys" on page A-163.

Reference

For more information about locking/unlocking the phones using the Phone UI and Aastra Web UI, see your applicable phone-specific *User Guide*.

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|91XXXXXX
XXXX|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 '*'.

Prefix Dialing

The IP phones support a prefix dialing feature for outgoing calls.

You can manually dial a number or dial a number from a list. The phone automatically maps the pre-configured prepended digit in the configuration, to the outgoing number. When a match is found, the prepended digits are added to the beginning of the dial string and the call is dialed.

Note: The prepend digits are also added if the dialing times-out on a partial match.

You can enable this feature by adding a prepend digit(s) to the end of the **Local Dial Plan** parameter string in the configuration files or the Aastra Web UI at **Basic Settings->Preferences->General**.

For example, if you add a prepend map of “[2-9]XXXXXXXXX,91”, the IP phone adds the digits “91” to any 10-digit number beginning with any digit from 2 to 9 that is dialed out. Other examples of prepend mappings are:

- **1X+#,9** (Prepends 9 to any digit string beginning with “1” and terminated with “#”.)
- **6XXX,579** (Prepends “579” to any 4-digit string starting with “6”.)
- **[4-6]XXXXXX,78** (Prepends “78” to any 7-digit string starting with “4”, “5”, or “6”.)

Note: You can configure a local dial plan via the configuration files or the Aastra Web UI.

Example

If you enter the following dial string for a local dial plan:

`sip dial plan: 1+#,9`

where "9" is the prepended digit, and you dial the following number:

`15551212`

the IP phone automatically adds the "**9**" digit to the beginning of the dialed number before the number is forwarded as **915551212**.

➡ **Note:** You can configure a local dial plan via the configuration files or the Aastra Web UI.

SIP Dial Plan Terminator

The IP phone allows you to enable or disable the use of the "dial plan terminator". When you configure the phone's dial plan 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 after you finish dialing the numbers on the keypad before making the call.

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

Digit Timeout

The IP phone allows you to configure a "**Digit Timeout**" feature on the IP phone. The Digit Timeout is the time, in seconds, between consecutive key presses on the IP phone's keypad. The default for this parameter is 4 seconds. If you press a key on the phone and wait 4 seconds before pressing the next key, the key times out and cancels the digit selection. You must press consecutive keys before the timeout occurs.

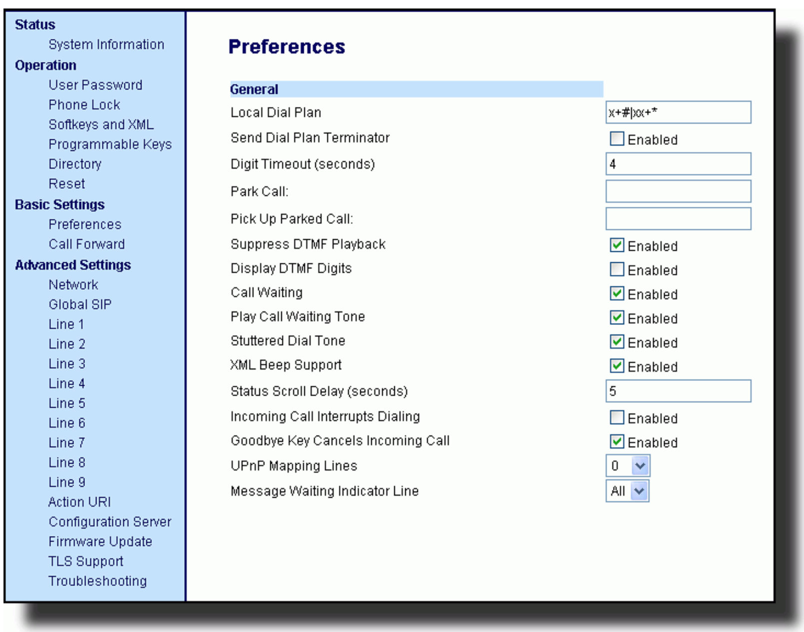
Configuring the SIP Local Dial Plan, Dial Plan Terminator, and Digit Timeout

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 Appendix A, the section, “SIP Local Dial Plan Settings” on page A-39.

Aastra Web UI

1	Click on **Basic Settings->Preferences**. 
2	In the "**Local Dial Plan**" field, enter a valid local dial plan (up to 127 alphanumeric characters) for the IP phone. Default is X+#|XX+*. Enter prepended digits as required.
3	Enable the "**Send Dial Plan Terminator**" field by checking the check box. (Disable this field by unchecking the box). Default is disabled.

Aastra Web UI	
4	In the "**Digit Timeout (in seconds)**" field, enter a timeout value. This is the length of time, in seconds, he phone waits before dialing. Default is 4 seconds.
5	Click Save Settings to save your changes.

Park Calls/Pick Up Parked Calls

The IP phones (including the 57i 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 administrator can configure this feature:

- Using a static configuration (globally configures park and pickup)
- 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 paragraph describes the park and pickup static configuration on the IP phones.

Reference

For information on configuring the park and pickup programmable configuration (using a key), see “Park/Pick Up Key” on page 5-153.

Park/Pickup Static Configuration (55i, 57i, 57i CT only)

Using the static method of configuring park and pickup configures these features on a global basis for all IP phones connected in the network. You can use the configuration files or the Aastra Web UI to configure a park/pickup static configuration.

In the configuration files, you use the following parameters to statically configure park/pickup:

- **sprecode:**
- **pickupsprecode:**

In the Aastra Web UI, you use the following fields at **Basic Settings-> Preferences** to configure park/pickup statically:

- **Park Call**
- **Pickup Call**

How It displays on the Phone

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 first screen of the Phone UI.
- After pressing the "**Park**" softkey to park the call, the default label of "**Pickup**" displays on the first screen of the phone UI.

Note: On the 57i CT handset, pressing displays the "Park" and "Pickup" labels.

The values you enter for the Park/Pickup feature are dependant on your type of server. The following table provides the values you enter for the "**sprecode"** and "**pickupsprecode"** parameters (configuration files), or "**Park Call**" and "**Pickup Parked Call**" fields (Aastra Web UI).

Park/Pickup Call Server Configuration Values

Server	Park Values*	Pickup Values*
Aasterisk	70	70
Sylantro	*98	*99
BroadWorks	*68	*88
ININ PBX	callpark	pickup

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

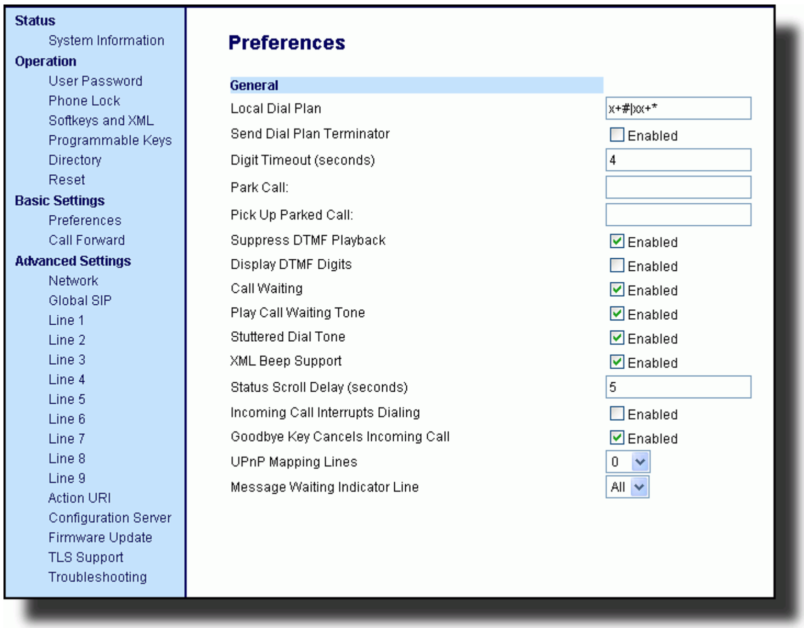
Configuring Park /Pickup using Static Configuration
(55i, 57i, 57i CT only)

Use the following procedures 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.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Park and Pickup Global Settings (57i/57i CT only)" on page A-132.

	Aastra Web UI
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 5-36.
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 5-36.
4	Click  to save your changes.

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.

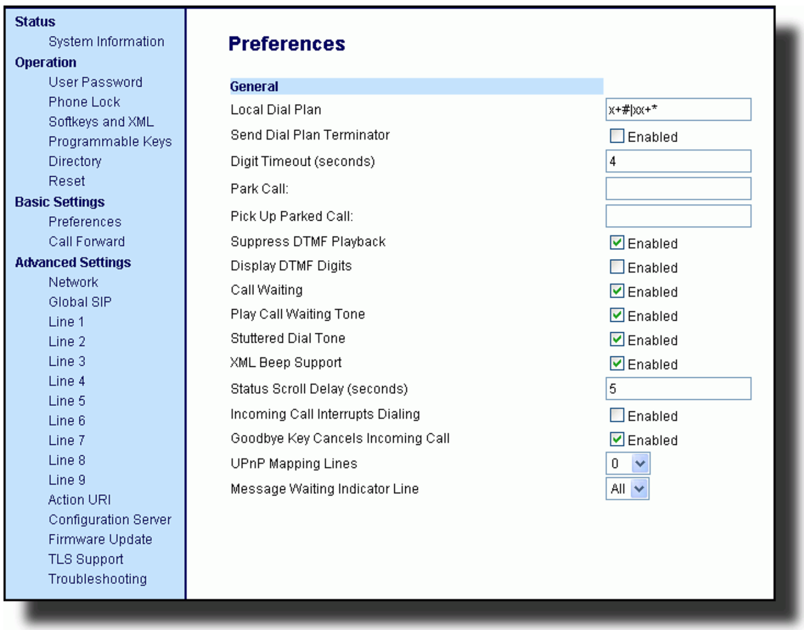
Configuring Suppression of DTMF Playback

Use the following procedures to configure the suppression of DTMF playback on the IP phone.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Suppress DTMF Playback Setting" on page A-119.

Aastra Web UI

1	Click on **Basic Settings->Preferences->General.** 
2	Enable the "**Suppress DTMF Playback**" field by checking the check box. (Disable this field by unchecking the box). Default is disabled.
3	Click Save Settings to save your settings. 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 changes.

Display DTMF Digits

A feature on the IP phones allows users and administrators to enable or disable DTMF (dual-tone multi-frequency) digits to display to the IP phone when using the keypad to dial, or when dialing from a softkey or programmable key.

DTMF is the signal sent from the phone to the network that you generate when you press the phone's touch keys. This is also known as "touchtone" dialing. Each key you press on your phone generates two tones of specific frequencies. One tone is generated from a high-frequency group of tones and the other from a low frequency group.

If you enable the Display DTMF Digits parameter, the digits you are dialing from the keypad or from a softkey or programmable key display to the IP phone's LCD display. This parameter is disabled by default (no digits display when dialing).

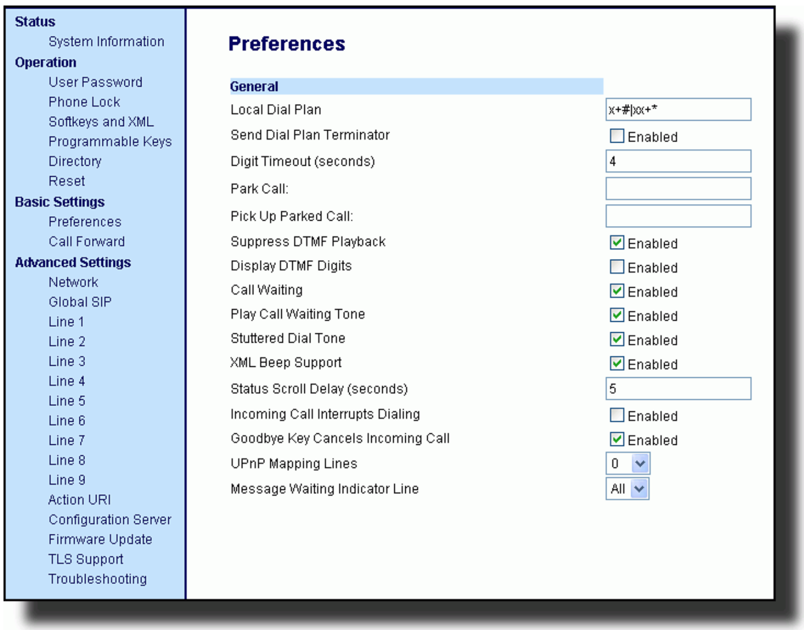
You can enable the "Display DTMF Digits" parameter using the configuration files or the Aastra Web UI.

Configuring Display DTMF Digits

Use the following procedures to configure the suppression of DTMF playback on the IP phone.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Display DTMF Digits Setting" on page A-120.

Aastra Web UI	
1	Click on **Basic Settings->Preferences->General.** 
2	Enable the "**Display DTMF Digits**" field by checking the check box. (Disable this field by unchecking the box). Default is disabled.
3	Click **Save Settings** to save your settings. 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 changes.

Call Waiting/Call Waiting Tone

A call waiting feature notifies the user currently on the phone, of a new incoming call. You can disable this call waiting feature, so that the new incoming call is automatically rejected by the phone with a busy message. A User or Administrator can configure this feature.

If you disable call waiting on the phone, and the user is on a call, any further incoming calls will receive busy unless "Call Forward Busy" or "Call Forward No Answer and Busy" is configured on the phone. It will then forward the call according to the rule configured. The phone can only:

- transfer the currently active call

  or

- accept transferred calls if there is no active calls.

If call waiting is disabled:

- on the 57i CT bases, and the handset is currently on a call, all additional incoming calls are rejected on the handset.
- intercom calls are treated as regular incoming calls and are rejected.
- pre-dialing with live dial pad disabled still accepts incoming calls.
- the "Incoming Call Cancels Dialing" parameter is ignored because the incoming call is automatically rejected.
- the Missed Calls List does not get updated with details of calls.
- the Blind Transfer feature on the phone may not work if two calls are made to the phone at one time.

You can enable/disable call waiting using the configuration files or the Aastra Web UI.

Call Waiting Tone

You can also enable or disable the playing of a short "Call Waiting Tone" when there is an incoming call on your phone. This feature is enabled by default. If you have Call Waiting enabled, and a call comes into the line for which you are on an active call, a tone is audible to notify you of that incoming call. The tone is also audible to the caller to indicate to that caller you are currently on another call.

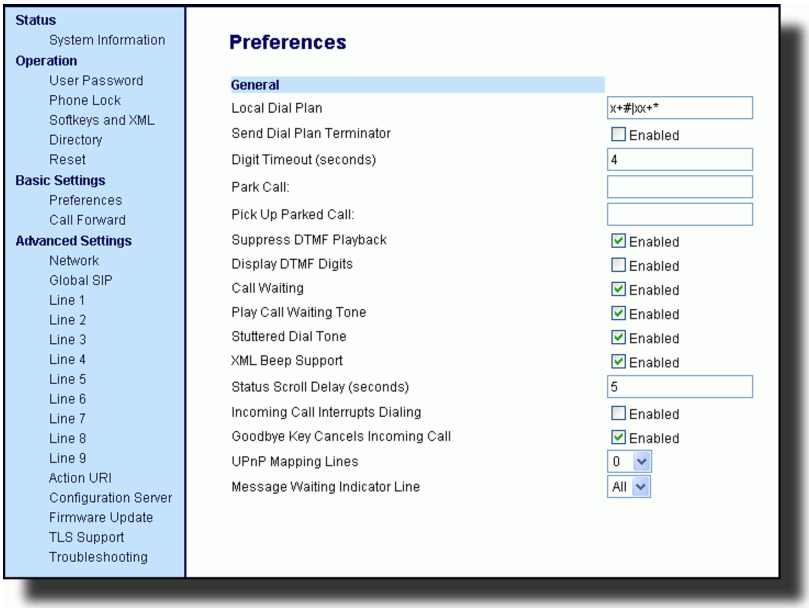
Note: The Call Waiting Tone feature works only if Call Waiting is enabled.

A User or Administrator can configure this feature. An Administrator can configure this feature using the configuration files or the Aastra Web UI.

Configuring Call Waiting/Call Waiting Tone

Use the following procedures to configure the Call Waiting/Call Waiting Tone features on the IP phone.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Call Waiting Settings" on page A-101.

	Aastra Web UI
1	Click on **Basic Settings-> Preferences->General.** 
2	The "**Call Waiting**" field is enabled by default. To disable this field, uncheck the box. This feature allows you to enable or disable the call waiting feature on the IP phone.
3	The "**Play Call Waiting Tone**" field is enabled by default. To disable this field, uncheck the box. This feature allows you to enable or disable the call waiting tone on the IP phone. **Note:** The Call Waiting Tone feature works only if the "**Call Waiting**" tone field is enabled.
4	Click Save Settings to save your changes.

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.

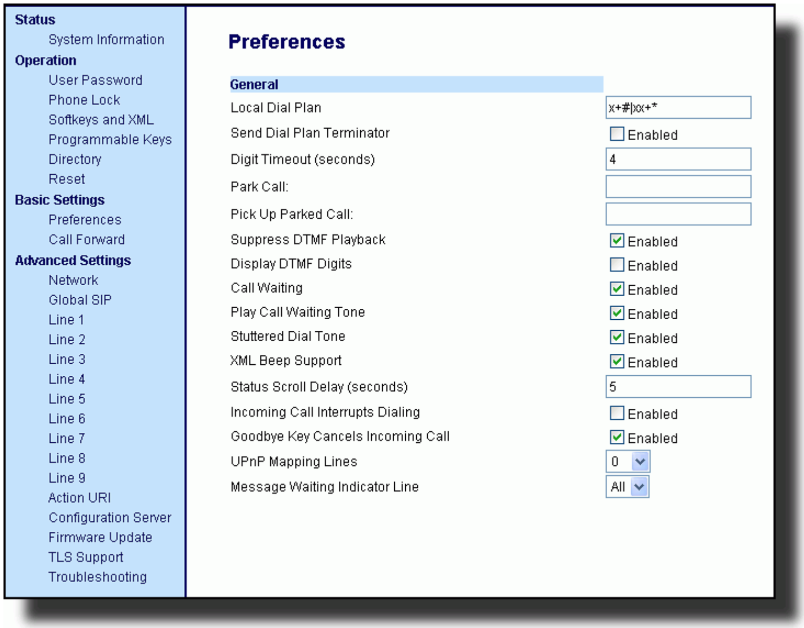
You can configure this feature using the configuration files and the Aastra Web UI.

Configuring Stuttered Dial Tone

Use the following 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 Appendix A, the section, "Stuttered Dial Tone Setting" on page A-100.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Basic Settings->Preferences->General.



</td></tr>
<tr><td>2</td><td>The "Stuttered Dial Tone" field is enabled by default. To disable this field, uncheck the box.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your changes.</td></tr>
</table>

XML Beep Support

The IP phones have a feature that allows you to enable or disable a beep on the phone with it receives a status message from an XML application. This beep can be turned ON or OFF using the Aastra Web UI, the configuration files, or in an XML script.

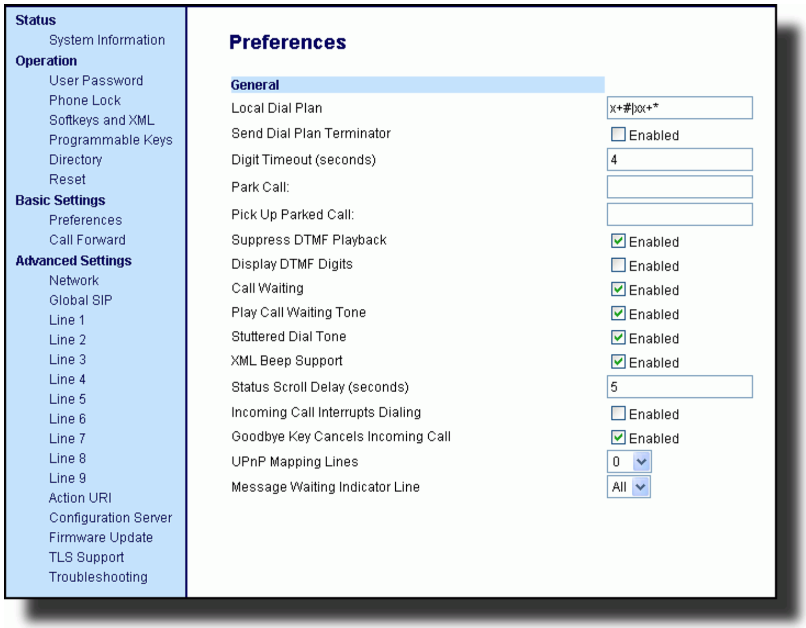
Reference

For more information about enabling/disabling the XML Beep Support in an XML script, see "XML Customized Services" on page 5-196.

Configuring XML Beep Support

Use the following procedures to enable/disable XML Beep Support.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "XML Settings" on page A-89.

	Aastra Web UI
1	Click on **Basic Settings->Preferences->General.** 
2	The “**XML Beep Support**” field is enabled by default. To disable this field, uncheck the box.
3	Click Save Settings to save your changes.

Status Scroll Delay

The IP phones have a feature that allows you to specify the time delay, in seconds, between the scrolling of each status message (including XML status messages) on the phone. The default time is 5 seconds for each message to display before scrolling to the next message. You can configure this option via the configuration files or the Aastra Web UI.

Reference

For more information about configuring the status scroll delay for XML status messages, see "XML Customized Services" on page 5-196.

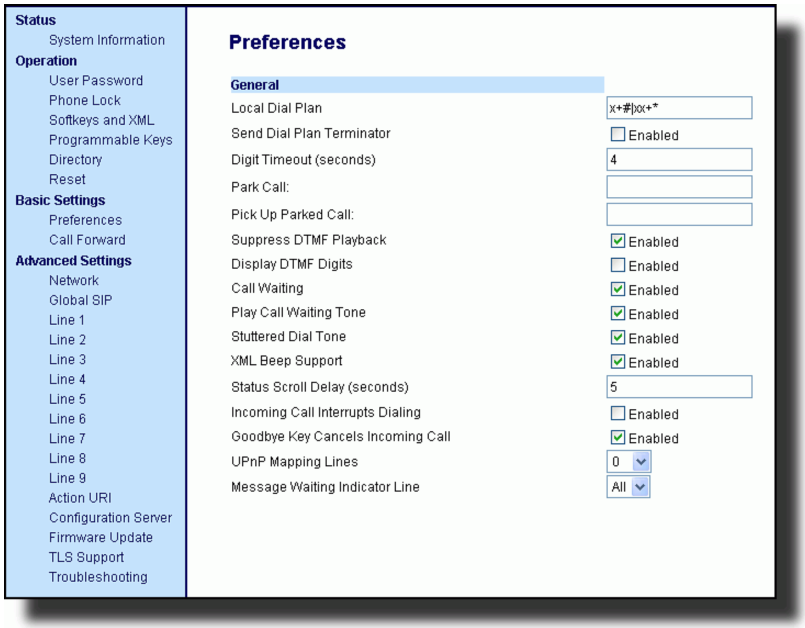
Configuring Status Scroll Delay

Use the following procedures to configure Status Scroll Delay.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "XML Settings" on page A-89.

Aastra Web UI

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



2 In the “**Status Scroll Delay (seconds)**” field, enter a value, in seconds, that each XML status message displays on the phone. Default is 5 seconds. Valid values are 1 to 25.

3 Click **Save Settings** to save your changes.

Operational IP Phone Features

Incoming Call Interrupts Dialing

You can configure whether or not an incoming call interrupts an outgoing call that is dialing. The **incoming call interrupts dialing** parameter controls this feature.

How it Works

When you enable this parameter (1 = enable), an incoming call interrupts the outgoing call during dialing and allows the phone to ring for the user to answer the incoming call.

When you disable the "**incoming call interrupts dialing**" parameter (0 = disable), which is the default, the phone does not interrupt the outgoing call during dialing and instead rings the incoming call on another free line (or sends busy signal if all remaining lines are busy). You have a choice to ignore the incoming call, or answer the incoming call on another line, via the **Ignore** and **Answer** softkeys that display. If you choose to answer the incoming call, you can answer the call, finish the call, and then hang up. You can still go back to the original outgoing call and finish dialing out.

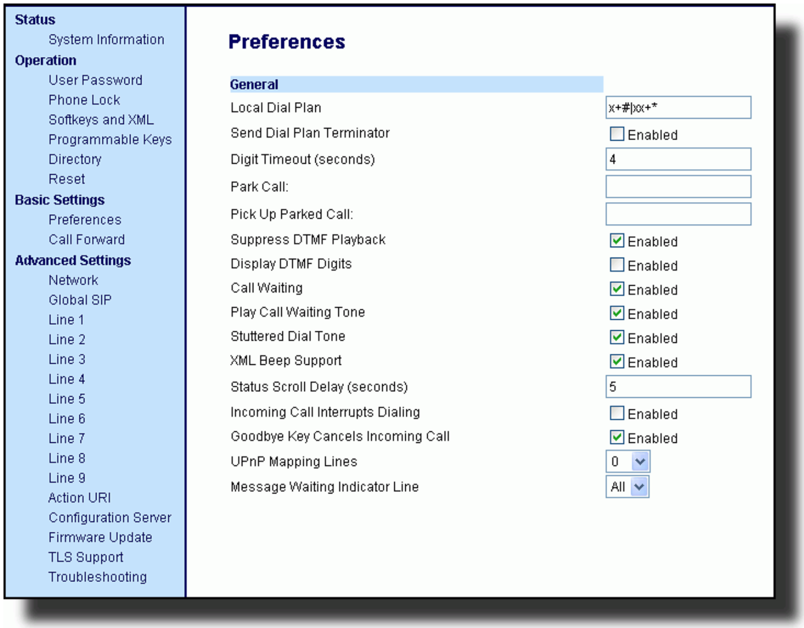
Notes:
1. On a 53i, you must use the down arrow key to ignore the call. To answer the call you must press the line key where the call is coming in.
2. For all models, if you disable this parameter (0=disable), and the phone receives an incoming call while you are dialing an outgoing call, you can pick up the call and perform transfer or conference as required.

Configuring Incoming Call Interrupts Dialing

Use the following procedures to configure how the IP phone handles incoming calls that interrupt outgoing dialing.

Configuration Files

For specific parameters you can set in the configuration files for enabling/disabling how the IP phones handle incoming calls that interrupt outgoing dialing, see Appendix A, the section, "Incoming Call Interrupts Dialing Setting" on page A-98.

	Aastra Web UI
1	Click on **Basic Settings->Preferences->General.** 
2	Incoming Call Interrupts Dialing is disabled by default. If required, enable the "**Incoming Call Interrupts Dialing**" field by checking the check box.
3	Click **Save Settings** to save your changes.

Goodbye Key Cancels Incoming Call

You can configure the Goodbye key to drop active calls or ignore incoming calls using the "**goodbye cancels incoming call**" parameter. This parameter controls the behavior of the goodbye key when the phone is on an active call and a second call is presented to the phone.

How it Works

When you enable this parameter (1 = enable in the configuration files), which is the default, the Goodbye key rejects the incoming call. When you disable this parameter (0 = disable in the configuration files), the Goodbye key hangs up the active call.

For the 55i, 57i, and 57i CT:

If you enable this parameter, and the phone receives another call when an active call is already present, the phone displays softkey 1 as "**answer**" and softkey 2 as "**ignore**". You can press the required softkey as applicable.

For the 53i:

If you enable this parameter, and the phone receives another call when an active call is already present, the "**ignore**" option *only* displays in the LCD window. The phone ignores the incoming call. If you press the **DOWN** arrow key, the phone answers the incoming call.

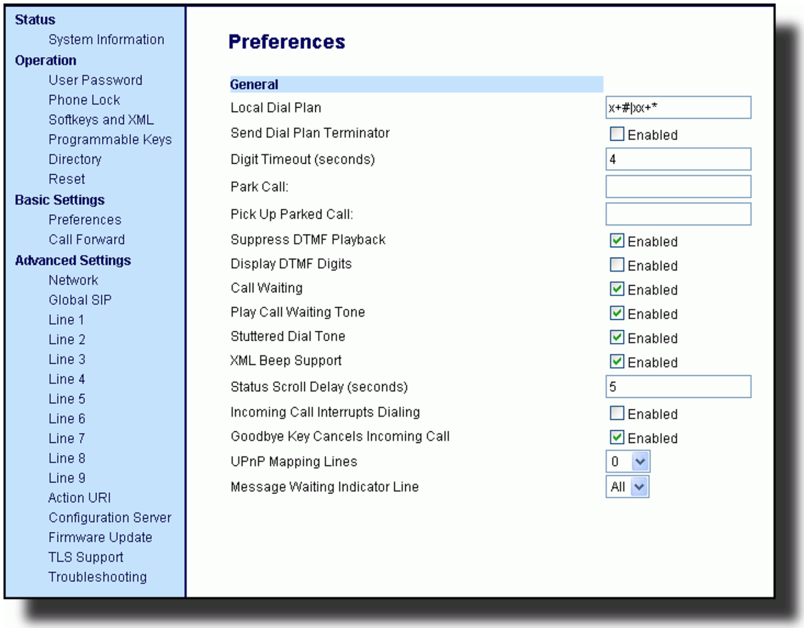
You can set this parameter using the configuration files or the Aastra Web UI.

Configuring the Goodbye Key to Cancel Incoming Calls

Use the following procedures to configure the behavior of the Goodbye Key on the IP phone.

Configuration Files
For specific parameters you can set in the configuration files for enabling/disabling the behavior of the Goodbye Key, see Appendix A, the section, "Incoming Call Interrupts Dialing Setting" on page A-98.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Basic Settings->Preferences->General.



</td></tr>
<tr><td>2</td><td>Incoming Call Interrupts Dialing is disabled by default.
If required, enable the "Incoming Call Interrupts Dialing" field by checking the check box.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your changes.</td></tr>
</table>

UPnP Mapping Lines (for remote phones)

Universal Plug and Play (UPnP) is a standard that uses Internet protocols to enable devices to be plugged into a network and automatically know about each other. With UPnP, when a user plugs a device into the network, the device configures itself, acquires a TCP IP address, and uses a discovery protocol based on the Internet's HTTP or HTTPS URL to announce its presence on the network to other devices.

This method of device discovery on a network is called "Universal Plug and Play" or UPnP. If you enable UPnP, and the phone is discovered on the network, port mappings are set up between the phone and the Internet Gateway Device (IGD) in your network. The phone controls the opening, closing, and polling of ports on the IGD. HTTP and SIP use a single port each. RTP/RTCP uses a range of ports.

You can enable the UPnP mappings to specific lines on your phone. You set this configuration using the Aastra Web UI at **Basic Settings->Preferences->UPnP Mapping Lines.**

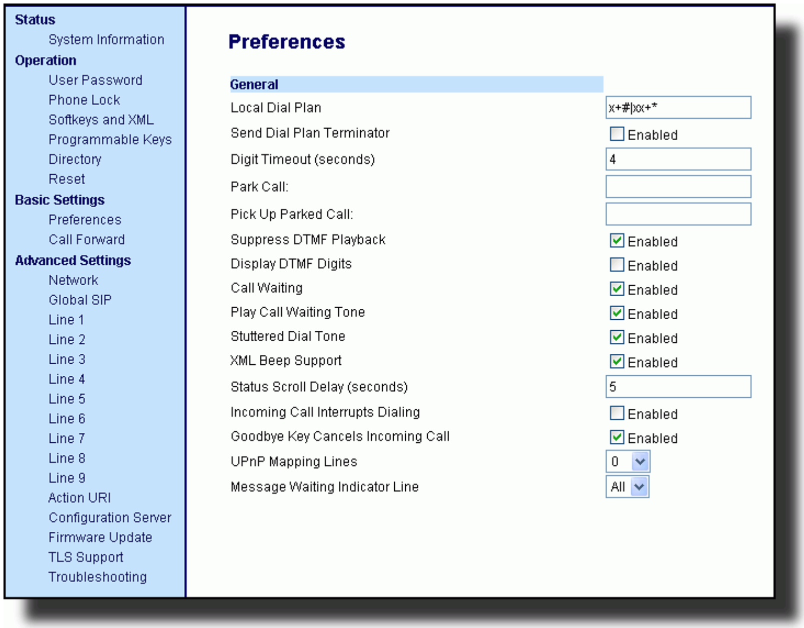
Note: UPnP must be enabled on your remote phone before you can configure the UPnP mapping lines. For information on enabling/disabling UPnP see Chapter 4, the section, "Universal Plug and Play (UPnP) (for remote phones)" on page 4-27.

Configuring UPnP Mapping Lines

Use the following procedures to configure UPnP mapping on specific lines on the IP phone.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "UPnP Settings" on page A-25.

	Aastra Web UI
1	Click on **Basic Settings->Preferences->General.** 
2	The "**UPnP Mapping Lines**" field is set to **0** by default. If required, change the setting to a specific line by selecting a line from the list. Valid values are **0** through **10**. The line you select is the line that your remote phone uses to perform Universal Plug and Play on the network you are connecting to.
3	Click Save Settings to save your changes.

Message Waiting Indicator Line

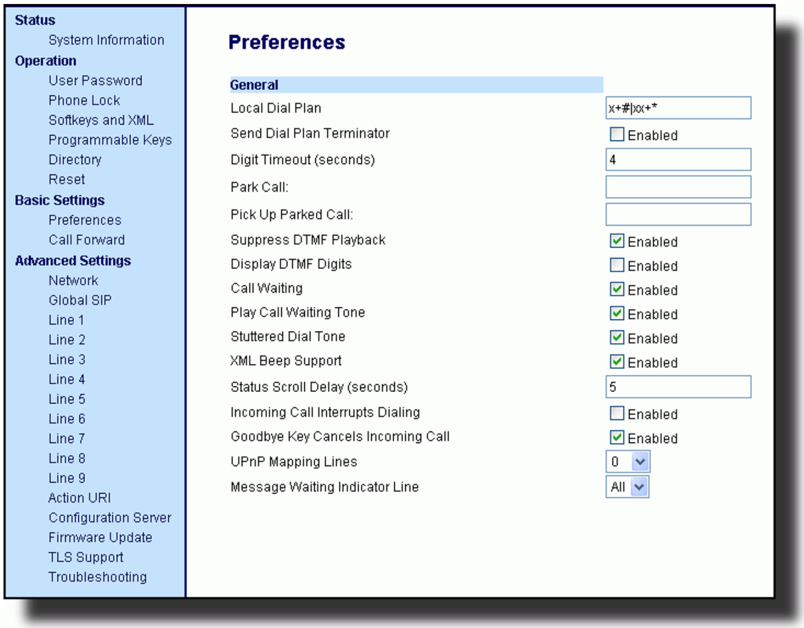
A User or Administrator can configure the Message Waiting Indicator (MWI) to illuminate for a specific line or for all lines. For example, if you configure the MWI LED on line 3 only, the LED illuminates if a voice mail is pending on line 3. If you configure the MWI LED for all lines, the LED illuminates if a voice mail is pending on any line on the phone (lines 1 through 9).

A User can configure the MWI using the Aastra Web UI only. An Administrator can configure the MWI on single or all lines using the configuration files or the Aastra Web UI.

Configuring Message Waiting Indicator (MWI)

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

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Message Waiting Indicator Settings" on page A-102.

	Aastra Web UI
1	Click on **Basic Settings-> Preferences->General.** 
2	In the “**Message Waiting Indicator Line**” field, select a single line from 1 to 9, or select “All” for all lines. If you select a single line, the MWI illuminates when a voice mail message is pending on that line. If you select all lines, the MWI illuminates when a voice mail message is pending on any line from 1 to 9.
3	Click Save Settings to save your changes.

Incoming/Outgoing Intercom with Auto-Answer and Barge In

The Intercom feature allows you to press the configured Intercom button on the IP phone 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).

You can configure incoming and outgoing intercom calls on all phone models. A User can configure incoming intercom calls 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. Only an Administrator can configure outgoing intercom calls.

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.

For outgoing intercom calls, an administrator can configure the following parameters:

Configuration File Parameters	Web UI Parameters
• sip intercom type	• Type)
• sip intercom prefix code	• Prefix Code
• sip intercom line	• Line

Note: To configure outgoing intercom calls using these parameters, see “Configuring Intercom Calls Settings” on page 5-62.

Incoming Intercom Calls

You can configure how the phone handles incoming intercom calls. You can receive incoming intercom calls whether or not there are active calls on the phone. The way the phone handles the call depends on the incoming intercom call configuration. The following paragraphs describe the configuration parameters for incoming intercom calls.

Microphone Mute

You can mute or unmute the microphone on the IP phone for intercom calls made by the originating caller. If you want to mute the intercom call, you enable this feature. If you want to unmute (or hear the intercom call), you disable this feature.

Auto-Answer/Play Warning Tone

The auto-answer feature on the IP phone allows you to enable or disable automatic answering for an Intercom call. If “Auto-Answer” is enabled, the phone automatically answers an incoming intercom call. If “Play Warning Tone” is also enabled, 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.

“Delay” before Auto-Answer

The IP Phones include support for the "delay" parameter (in the Alert-Info header, used in conjunction with info=alert-autoanswer) in order to facilitate auto-answer functionality. When present, the value of the "delay" parameter specifies the length of time in seconds an IP phone rings before a call is auto-answered. If this value of the "delay" parameter set to 0 (delay=0), then an incoming call is immediately auto-answered. The absence of the parameter is considered as ring forever.

In order for the delay functionality to operate, you must first enable Auto-Answer on the IP Phone.

Allow Barge In

You can configure whether or not the IP phone allows an incoming intercom call to interrupt an active call. The "**sip intercom allow barge in"** parameter controls this feature. When you enable the **sip intercom allow barge in** parameter (1 = enable in the configuration files), which is the default value, an incoming intercom call takes precedence over any active call, by placing the active call on hold and automatically answering the intercom call. When you disable this parameter (0 = disable in the configuration files), and there is an active call, the phone treats an incoming intercom call like a normal call and plays the call warning tone. You can set this parameter using the configuration files or the Aastra Web UI.

For incoming intercom calls, an administrator or user can configure the following parameters:

Configuration File Parameters	Web UI Parameters
• sip allow auto answer	• Auto-Answer
• sip intercom mute mic	• Microphone Mute
• sip play warning tone	• Play Warning Tone
• sip intercom allow barge in	• Barge In

Note: To configure incoming intercom calls using these parameters, see "Configuring Intercom Calls Settings" on page 5-62.

Configuring Intercom Calls Settings

You can configure the Intercom feature 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 on the IP phone.

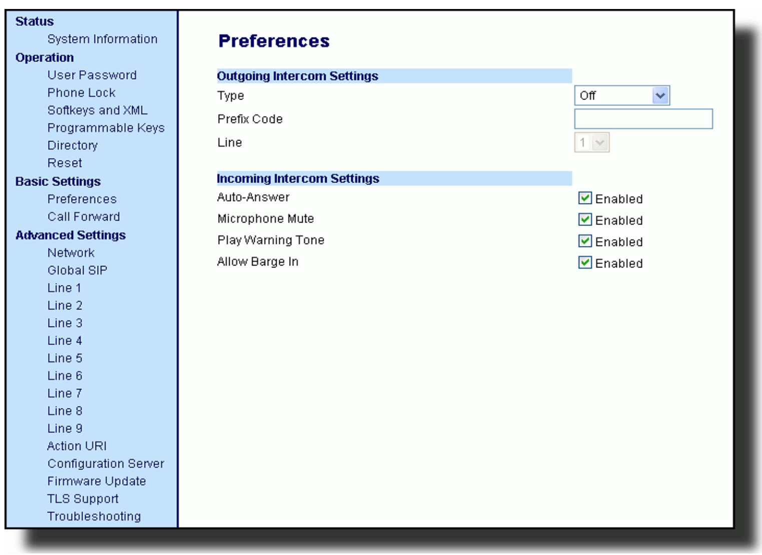
Configuration Files

For specific parameters you can set in the configuration files for outgoing Intercom, see Appendix A, the section, "Outgoing Intercom Settings" on page A-121.

For specific parameters you can set in the configuration files for incoming Intercom, see Appendix A, the section, "Incoming Intercom Settings" on page A-123.

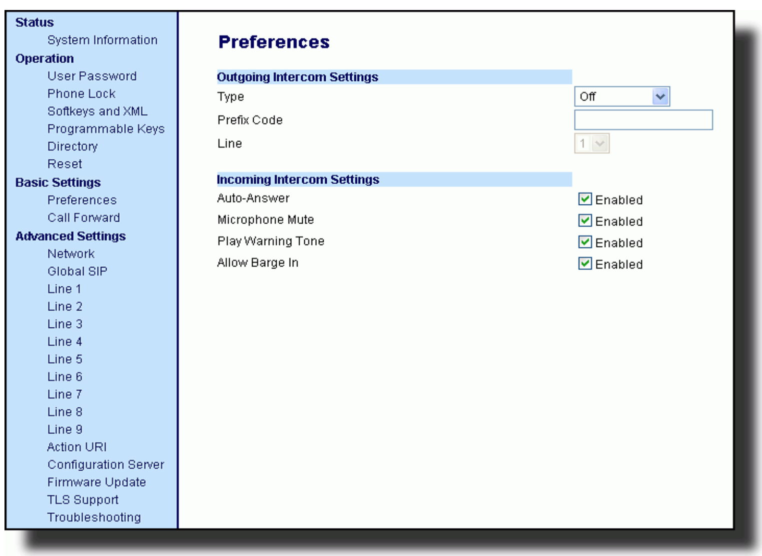
Aastra Web UI

Outgoing intercom settings (55i, 57i, and 57i CT only):

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.

Aastra Web UI	
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.

Incoming intercom settings:	
1	Click on **Basic Settings->Preferences->Incoming Intercom Settings**.



2	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 (disabled), the phone rejects the incoming intercom call and sends a busy signal to the caller.

	Aastra Web UI
3	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.
4	The "**Play Warning Tone**" field is enabled by default. If “Auto-Answer” is enabled, the phone plays a warning tone when it receives in incoming intercom call. To disable this field, uncheck the box.
5	The "**Allow Barge In**" field is enabled by default. If an active line on the phone receives an incoming intercom call, the active call is put on hold and the phone automatically answers the incoming intercom call. To disable this field, uncheck the box.
6	Click Save Settings to save your changes.

Key Mapping

There are hard keys on your phone, such as **Hold, Redial, Xfer,** and **Conf** that are configured by default for specific call-handling features. (See the product-specific User Guide for more information about these key functions.

Notes:

1. On the 55i and 57i, the Xfer and Conf keys are hard-coded by default on keys 5 and 6 to the left of the LCD display and cannot be reassigned. The Xfer and Conf labels display when you lift the handset. To disable these keys, see the next paragraph.

2. On the 53i, the Xfer and Conf keys are assigned by default to keys 5 and 6, respectively. These keys are programmable keys and can be reassigned if applicable. To disable these keys, see the next paragraph.

Enabling/Disabling Redial, Xfer, and Conf Keys

You can enable or disable the **Redial**, **Xfer**, and **Conf** 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.

This feature is configurable via the configuration files only.

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 Appendix A, the section, "Mapping Key Parameters" on page A-133.

Mapping Redial and Conf Keys as Speeddials

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

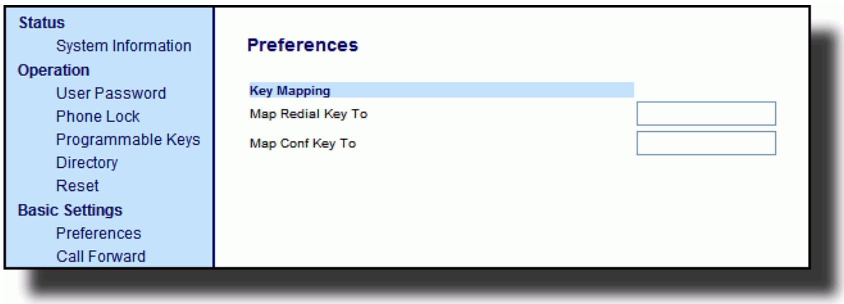
You can configure this feature using the configuration files or the Aastra Web UI.

Note: If you configure the **Redial** and **Conference** keys for speeddialing on the 57i CT Base Station, the **Redial** and **Conference** keys on the 57i CT handset retain their original functionality. The **Redial** and **Conference** keys on the handset are not configured for speeddial.

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

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Mapping Key Parameters" on page A-133.

	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 speeddial when the Redial key is pressed.
3	Enter a number in the "**Map Conf Key To**" field, that the IP phone will use to speeddial when the Conf Key is pressed.
4	Click Save Settings to save your changes.

Using Redial Key for "Last Number Redial"

The IP phones have an enhanced redial user interface that allows a user to quickly redial the last number that was dialed out from the phone. You can:

- Press the REDIAL key twice to redial the last number dialed.
- Press the REDIAL key once, scroll the list of numbers, then press the REDIAL button again to dial the number that displays on the screen.

The "last number redial" feature for the Redial key is static and is not configurable.

Note: You can use the Redial key during active calls.

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 each 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	Global: Tone 1

Configuration Method	Valid Values	Default Value
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**)**
- **Australia**
- **Europe (generic tones)**
- **France**
- **Germany**
- **Italy**
- **Mexico**
- **United Kingdom (UK)**

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

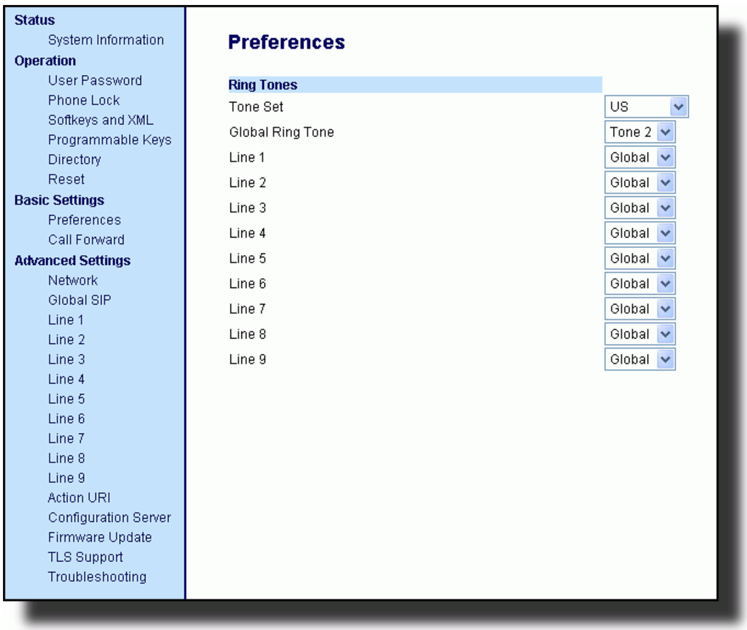
For specific parameters you can set in the configuration files for ring tones, see Appendix A, the section, "Ring Tone and Tone Set Global Settings" on page A-94 or "Ring Tone Per-Line Settings" on page A-97.

IP Phone UI

Step	Action
For global configuration only:	
1	Press [Options key] on the phone to enter the Options List.
2	Select **Preferences**.
3	Select **Tones**.
4	Select **Set Ring Tone**.
5	Select the type of ring tone (**Tone 1** through **Tone 5**, or **Silent**).
6	Press **Done** to save the change.
7	Select **Tone Set**.
8	Select the country for which you want to apply the tone set. Valid values are **Australia**, **Europe, France**, **Germany**, **Italy**, **Mexico, UK**, and **US**. Default is **US**.
9	Press **Done** to save the change. The ring tone and tone set you select is immediately applied to the IP phone.

Operational IP Phone Features

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. Valid values are **Australia**, **Europe, France**, **Germany**, **Italy**, **Mexico, UK**, and **US**. Default is **US**.
3	Select a value from the "**Global Ring Tone**" field. **Note:** See the "Ring Tone Settings Table" on page 5-70 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 5-70 for valid values.
6	Click  to save your changes.

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, Broadwords, 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 Server 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 Server Ring Tone Keywords
alert-acd (auto call distribution) alert-community-1 alert-community-2 alert-community-3 alert-community-4 alert-emergency alert-external alert-group alert-internal 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 for each call criteria
alert-acd (auto call distribution) alert-community-1 alert-community-2 alert-community-3 alert-community-4 alert-emergency alert-external alert-group alert-internal 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	90 90	100 100	110 110
Bellcore-dr3 **CallWaitingTone3**	3	Tone On Tone Off Tone On Tone Off	90 90 90 90	100 100 100 100	110 110 110 110
Bellcore-dr4 **CallWaitingTone4**	4	Tone On Tone Off Tone On Tone Off	90 90 270 90	100 100 300 100	110 110 330 110

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 76.

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 77.

Configuring Priority Alerting

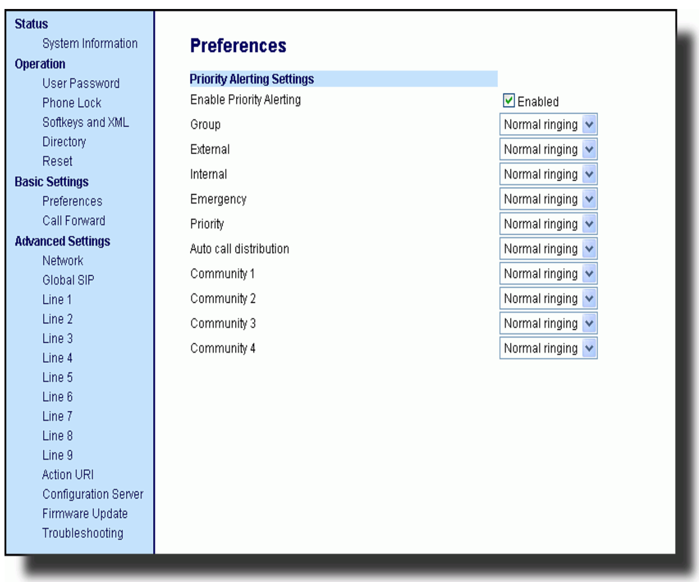
Use the following procedures to configure priority alerting on the IP phones.

Configuration Files

For specific parameters you can set in the configuration files for priority alerting, see Appendix A, the section, "Priority Alert Settings" on page A-103.

Aastra Web UI

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



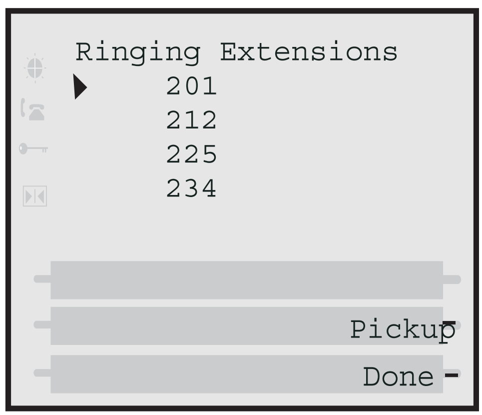
2 In the "Priority Alerting Settings" section, enable the "**Enable Priority Alerting**" field by checking the check box. (Disable this field by unchecking the box).

Aastra Web UI	
For Sylantro Servers:	
3	Select a ring tone pattern for each of the following fields: • **Group** • **Internal** • **External** • **Emergency** • **Priority** • **auto call distribution** • **community-1** • **community-2** • **community-3** • **community-4**
4	Click Save Settings to save your changes.

Directed Call Pickup (BLF or XML Call Interception)

Directed call pickup is a feature on the phones that allows a user to intercept a call on a ringing phone which is part of the same interception group. You can use the Directed call pickup feature on the phone in two ways:

- With the existing BLF feature on Asterisk, a user can dial "*76" followed by the extension to pick up a ringing call on another phone. (For more information about BLF, see "Busy Lamp Field (BLF)" on page 114
- Using XML, a user can intercept a call by selecting an extension from a list and then pressing a "Pickup" softkey/programmable key. To use the Directed call pickup feature from an XML application, you must list all ringing extensions using the **AastraIPPhoneTextMenu** XML object in an XML script. This allows the user to select the ringing extension from a text menu without having to dial. The following illustration shows an example of how this feature displays to the LCD from an XML application.:



(For more information about using the **AastraIPPhoneTextMenu** object, see Appendix G, the section, "Text Menu Object (Menu Screens)" on page G-6.

BLF and XML softkeys/programmable keys 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 or XML 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.

Directed Call Pickup Prefix (optional)

The optional “directed call pickup prefix” allows you to enter a specific prefix string (depending on what is available on your server), that the phone automatically dials when dialing the Directed Call Pickup number. For example, for Broadsoft servers, you can enter a value of *98 for the “directed call pickup prefix”. When the phone performs the Directed Call Pickup after pressing a BLF or BLF/List softkey, the phone prepends the *98 value to the designated extension of the BLF or BLF/List softkey when dialing out.

How this feature works when Directed Call Pickup is enabled with BLF or BLF/List

1. Phone A monitors Phone B via BLF/List.
2. Phone C calls Phone B; Phone B rings.
3. If you press the BLF/List softkey on Phone A, it picks up the ringing line on Phone B.
4. Phone C connects to Phone A.

How this feature works when Directed Call Pickup is disabled with BLF or BLF/List

1. Phone A monitors Phone B via BLF/List.
2. Phone C calls Phone B; Phone B rings.
3. If you press the BLF/List softkey on Phone A, it performs a speeddial to Phone B.
4. Phone C and Phone A are ringing Phone B on separate lines (if available).

Notes:
1. The default method for the phone to use is Directed Call Pickup over BLF if the server provides applicable information. If the Directed Call Pickup over BLF information is missing in the messages to the server, the Directed Call Pickup by Prefix method is used if a value for the prefix code exists in the configuration.

2. You can define only one prefix at a time for the entire BLF/List.

3. The phone that picks up displays the prefix code + the extension number (for example, *981234 where prefix key = *98, extension = 1234).

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

Note: The "Directed Call Pickup" feature is disabled by default.

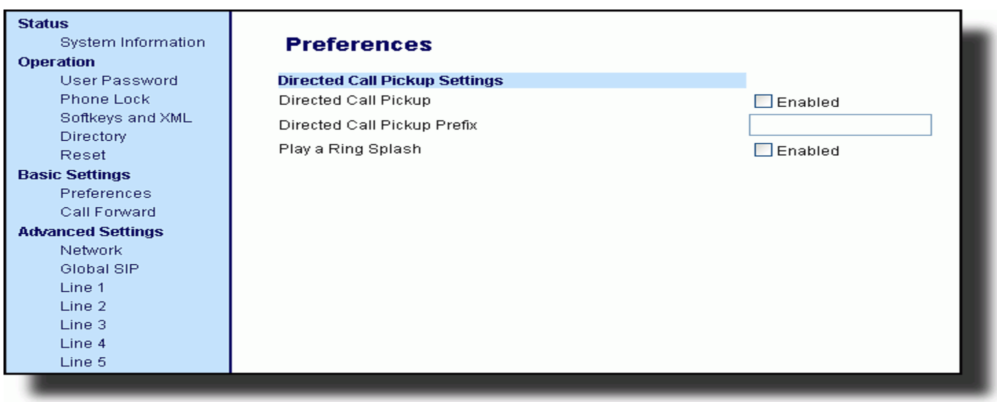
Enabling/Disabling Directed Call Pickup

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 using the configuration files, see Appendix A, the section, "Directed Call Pickup (BLF or XML Call Interception) Settings" on page A-128.

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	(optional) Enter a prefix in the "**Directed Call Pickup Prefix**" field. For example, *98. This prefix is appended to the beginning of the Directed Call Pickup number when dialed from the BLF or BLF/List softkey.

Aastra Web UI	
4	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 "ring splash" 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".
5	Click **Save Settings** to save your changes.

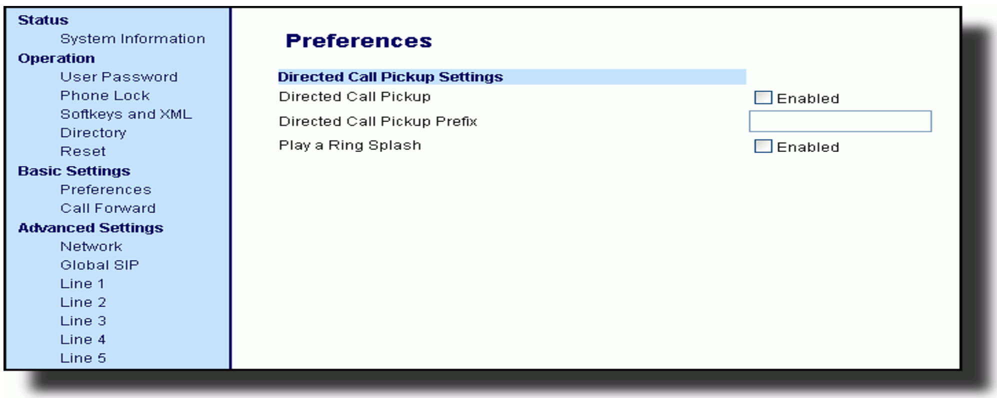
Configuring BLF/BLF List for Directed Call Pickup

Use the following procedure to configure BLF/BLF List for Directed Call Pickup in the configuration files.

Note: You must enable Directed Call Pickup before performing these procedures. See "Enabling/Disabling Directed Call Pickup" on page 5-84.

Configuration Files
To set BLF or BLF\List in the configuration files for Directed Call Pickup, see Appendix A, the section, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

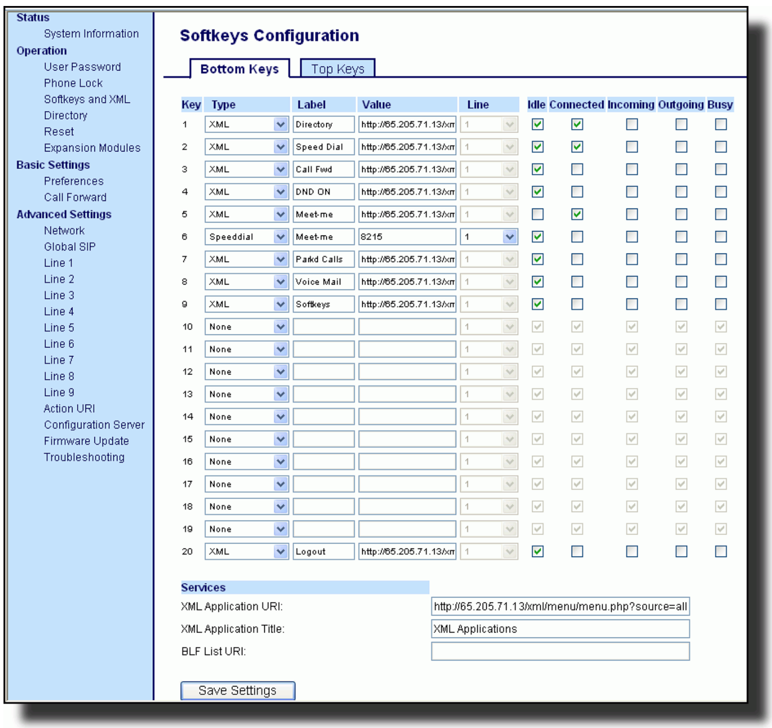
Use the following procedure to configure BLF or BLF/List for Directed Call Pickup in the Aastra Web UI.

	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.
3	(optional) Enter a prefix in the "**Directed Call Pickup Prefix**" field. For example, *98. This prefix is appended to the beginning of the Directed Call Pickup number when dialed from the BLF or BLF/List softkey.
4	Enable the "**Play a Ring Splash**" field by checking the check box. The IP phone plays a short "ring splash" 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".

Aastra Web UI

5 Click on **Operation->Softkeys and XML**
or
Click on **Operation->Programmable Keys**
or
Click on **Operation->Expansion Module <N>.**

Note: Depending on your phone-model, the key configuration screen displays. The 57i Key Screen is shown as an example.



6 Select a softkey or programmable key to configure.

7 In the "**Type**" field, select "**BLF**" (Asterisk), "**BLF\List**" (BroadSoft BroadWorks).

8 **For the 55i, 57i, and 57i CT softkeys**:

In the "**Label**" field, enter the name of the person who's extension you are monitoring (if "Type" is BLF).

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.

	Aastra Web UI
9	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.
10	Click Save Settings to save your changes.
11	In the "**Line**" field, select a line number that is actively registered to the appropriate SIP proxy you are using.
12	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, my57i-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 57i screen and the feature is disabled.
13	Select the line state (idle, connected, incoming, outgoing, busy) that you want to apply to the BLF softkey or programmable key.
14	Click Save Settings to save your changes.

Configuring XML for Directed Call Pickup

Use the following procedure to configure XML for Directed Call Pickup in the configuration files.

Notes:

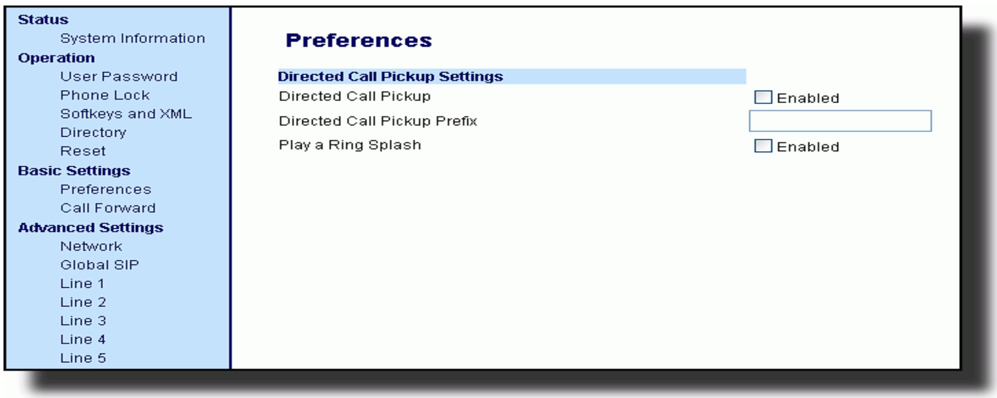
1. Before implementing this procedure, you must create an XML application that the phone uses when the XML softkey or programmable key is pressed. This XML application must be entered as a URI in the "Value" field of the XML key. For information about creating an XML script, see Appendix G, the section "Text Menu Object (Menu Screens)" on page G-6.

2. You must enable Directed Call Pickup before performing these procedures. See "Enabling/Disabling Directed Call Pickup" on page 5-84.

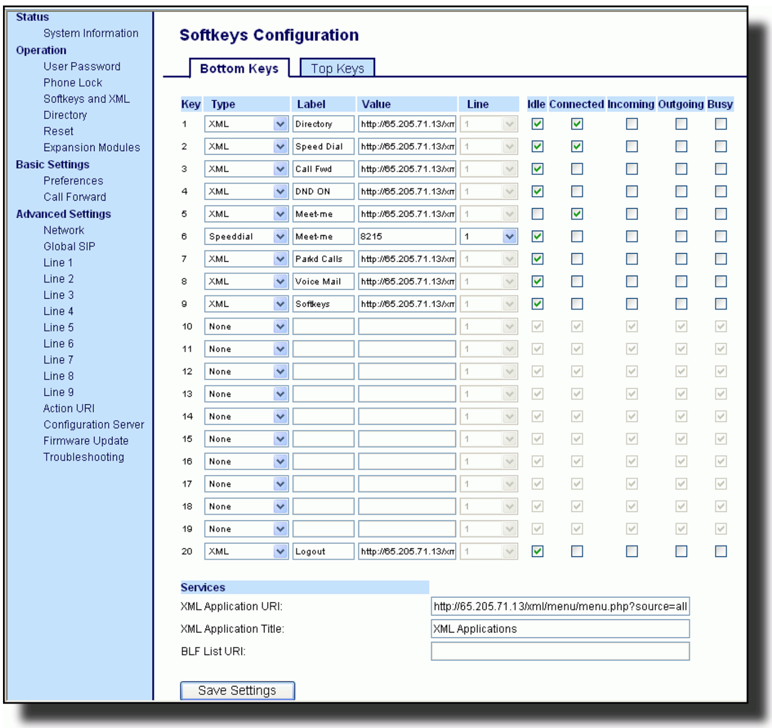
Configuration Files

To set XML in the configuration files for Directed Call Pickup, see Appendix A, the section, "Softkey/ Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

Use the following procedure to configure XML for Directed Call Pickup in the Aastra Web UI.

	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.
3	(optional) Enter a prefix in the "**Directed Call Pickup Prefix**" field. For example, *98. This prefix is appended to the beginning of the Directed Call Pickup number when dialed from the BLF or BLF/List softkey.
4	Enable the "**Play a Ring Splash**" field by checking the check box. 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".

Aastra Web UI

5	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** **Note:** Depending on your phone-model, the key configuration screen displays. The 57i Key Screen is shown as an example. 
6	Select a softkey or programmable key to configure.
7	In the "**Type**" field, select "**XML**".
8	**For the 55i, 57i, and 57i CT softkeys**: In the "**Label**" field, enter the name of the person who's extension you are monitoring.

	Aastra Web UI
9	In the "**Value**" field, enter the URI that the phone uses to display the XML application to the LCD. For example, http://65.205.71.13/xml/startup/key.php?user=$$SIPREMOTENUMBER$$. **Note:** For more information about creating an XML script to use with Directed Call Pickup, see Appendix G, the section "Text Menu Object (Menu Screens)" on page G-6.
10	Select the line state (idle, connected, incoming, outgoing, busy) that you want to apply to the XML softkey or programmable key.
11	Click Save Settings to save your changes.

Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys

You can configure the softkeys, programmable keys (53i has programmable keys only), feature keys, and expansion module keys to perform specific functions on the IP phones.

Note: When entering definitions for softkeys in the configuration files, the "#" sign must be enclosed in quotes.

Softkeys (55i, 57i, 57i CT)

The 55i IP phone has 6 softkeys you can configure to perform specific functions, The 57i and 57i CT IP phones have 12 softkeys you can configure. With up to 3 Expansion Modules attached to the phone, you can get an additional 72 softkeys to configure (not all functions apply to Expansion Module softkeys). The following table provides the number of softkeys you can configure, and the number of lines available for each type of phone.

IP Phone Model	Softkeys	Expansion Module Keys	Programmable Keys	Lines Available	Handset Keys Available
55i	6	36 to 108* (Model 536M) 60 to 180** (Model 560M)	6	9	-
57i	12	36 to 108* (Model 536M) 60 to 180** (Model 560M)	-	9	-
57i CT	12	36 to 108* on Base Station (Model 536M) 60 to 180** on Base Station (Model 560M)	-	9	15

*The 536M expansion module consists of 36 softkeys. You can have up to 3 expansion modules on an IP phone totaling 108 softkeys. Valid for 55i, 57i, and 57i CT phones.

**The 560M expansion module consists of 60 softkeys. You can have up to 3 expansion modules on an IP phone totaling 180 softkeys. Valid for 57i and 57i CT phones only.

State-Based Softkeys (55i, 57i, 57i CT only)

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 current line 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.
- **busy** - The current line is busy because the line is in use or the line is set as "Do Not Disturb".

The following table identifies the applicable default states for each type of softkey you can configure on the IP phone.

Softkey Type	Default States
None	All states disabled.
Line	idle, connected, incoming, outgoing, busy
Do Not Disturb (DND)	idle, connected, incoming, outgoing, busy
Speeddial	idle, connected, incoming, outgoing, busy
Busy Lamp Field (BLF)	idle, connected, incoming, outgoing, busy
BLF List	idle, connected, incoming, outgoing, busy
Auto Call Distribution (ACD)	idle
Directed Call Pickup (DCP)/ Group Call Pickup (GCP)	idle, connected, incoming, outgoing, busy
XML	idle, connected, incoming, outgoing, busy
Flash	All states disabled.
Sprecode	connected
Park	connected
Pickup	idle, outgoing

Softkey Type	Default States
Last Call Return (lcr)	idle, connected, incoming, outgoing, busy
Directory	idle, connected, incoming, outgoing, busy
Callers List	idle, connected, incoming, outgoing, busy
Intercom	idle, connected, incoming, outgoing, busy
Services	idle, connected, incoming, outgoing, busy
Empty	idle, connected, incoming, outgoing, busy

You can enable or disable the softkey states using the configuration files or the Aastra Web UI. In the Aastra Web UI, you disable a state by unchecking 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, busy**) 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: 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 Appendix A, the section, "Softkey Settings for 55i, 57i, 57i CT" on page A-137.

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.

Programmable Keys (53i, 55i)

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	Expansion Module Keys	Programmable Keys	Lines Available	Handset Keys Available
53i	-	36 to 108* (Model 536M)	4	9	-
55i	6	36 to 108* (Model 536M) 60 to 180** (Model 560M)	6	9	-

*The 536M expansion module consists of 36 softkeys. You can have up to 3 expansion modules on an IP phone totaling 108 softkeys.

Softkey/Programmable Key/Expansion Module Key Functions

You can configure the softkeys, programmable keys, and any attached expansion module keys on the 53i, 55i, 57i, and 57i CT to perform specific functions using the configuration files or the Aastra Web UI. The following table identifies the available functions of the softkeys, programmable keys, and expansion module keys on the IP phones. Available functions may vary on each model phone.

The following **Key Functions Table** lists the available functions for the keys on the IP Phones and Expansion Modules.

Note: These functions apply to the 53i, 55i, 57i, 57i CT and Expansion Modules unless specifically stated otherwise.

Key Functions Table

Softkey/ Programmable Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
None	none	None	Indicates not setting for the key.
Line	line	Line	Indicates the key is configured for line use.
Speeddial	speeddial	Speeddial	Indicates the key is configured for speeddial use. You can configure a softkey to speeddial a specific number by pressing that softkey. Optionally, you can also configure a speeddial key to dial prefix numbers. With this option, the prefix numbers automatically dial when you press the softkey, and the phone waits for you to enter the remaining numbers to dial out. For more information about speeddial prefixes, see "Speeddial Prefixes" on page 5-113. You can also create speeddial keys using the IP Phone keypad. For more information about speeddial keys, see your Model-specific *User Guide* for more information.
Busy Lamp Field (BLF)	blf	BLF	Indicates the key is configured for Busy Lamp Field (BLF) use. A user can dial out on a BLF configured key. You can also set a BLF subscription period. For more information about BLF, see the section "Busy Lamp Field (BLF)" on page 5-114. For more information about BLF Subscription Period, see "BLF Subscription Period" on page 5-120.
Busy Lamp Field List	list	BLF/List	Indicates the key is configured for BLF list use. A user can dial out on a BLF\List configured key. For more information on BLF, see the section "Busy Lamp Field (BLF)" on page 5-114.

Softkey/ Programmable Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
Auto Call Distribution (ACD)	acd	Auto Call Distribution	(For Sylantro Servers) Indicates the key is configured for automatic call distribution (ACD). ACD allows the Sylantro Server to distribute calls from a queue to registered IP Phones (agents). You can also set an ACD subscription period. For more information about ACD, see the section "Automatic Call Distribution (ACD) (for Sylantro Servers)" on page 5-122. For more information about ACD subscription period, see "ACD Subscription Period" on page 5-133.
Directed Call Pickup (DCP)/ Group Call Pickup (GCP)	dcp	Directed Call Pickup	(For Sylantro Servers) Indicates the key is configured for either Directed Call Pickup or Group Call Pickup. The DCP/GCP feature allows you to intercept - or pickup - a call on a monitored extension(s). For more information about DCP/GCP, see the section "Directed Call Pickup/Group Call Pickup (for Sylantro Servers)" on page 5-135.
Do Not Disturb (DND)	dnd	Do Not Disturb	Indicates key is configured for "do not disturb" use. For more information on DND, see the section "Do Not Disturb (DND)" on page 5-142.
Extensible Markup Language) (XML)	xml	XML	Indicates the key is configured to accept an XML application for accessing customized XML services. You can also specify a URL for an XML key. For more information on XML, see the section "XML Customized Services" on page 5-196.

Softkey/ Programmable Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
Flash (Not applicable on Expansion Modules)	flash	Flash	Indicates the key is set to generate a flash event when it is pressed, or when a feature key is pressed on the 57i CT handset. 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). For more information about the Flash key, see your Model-specific *User Guide*.
Sprecode (Not applicable on Expansion Modules)	sprecode	Sprecode	Indicates the key is set to automatically activate specific services offered by the server. For example, if the sprecode value of *82 is configured, then by pressing the key, *82 automatically activates a service provided by the server. The value you enter for this field is dependent on the services provided by the server. For more information about the Flash key, see your Model-specific *User Guide*.
Park (Not applicable on Expansion Modules)	park	Park	Indicates the key is set to be used as a park key to park an incoming call. For more information on park, see the section "Park/Pick Up Key" on page 5-153.
Pickup (Not applicable on Expansion Modules)	pickup	Pickup	Indicates the key is set to be used as a pickup key to pick up a parked call. For more information on pickup, see the section "Park/Pick Up Key" on page 5-153.
Last Call Return (LCR)	lcr	Last Call Return	(For Sylantro Servers) Indicates the key is set to be used to dial the last call that came in on that line. For more information on lcr, see the section "Last Call Return (lcr) (For Sylantro Servers" on page 5-164.

Softkey/ Programmable Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
Callers List	callers list	Callers List	Indicates the key is set for accessing the Callers List. For more information on the Callers List, see the section "Callers List" on page 174.
Directory	directory	Directory	Indicates the key is set for accessing the Directory List. For more information about the Directory List, see the section "Directory List" on page 182.
Icom	icom	Intercom	Indicates the key is set to be used as the Intercom key. For more information about using the Intercom key, see your model-specific Aastra IP Phone User's Guide. For information about other Intercom features, see "Incoming/Outgoing Intercom with Auto-Answer and Barge In" on page 5-60.
Conference (Not applicable on Expansion Modules)	conf	Conference	Indicates the key is configured as a conference key (for local conferencing). (For Sylantro and Broadsoft Servers) An Administrator can also enable centralized conferencing on the IP Phones. For more information about using the Conference key, see your Model-specific *User's Guide*. For information about enabling centralizing conferencing, see "Centralized Conferencing (for Sylantro and Broadsoft Servers)" on page 5-220.
Transfer (Not applicable on Expansion Modules)	xfer	Transfer	Indicates the key is configured as a transfer key for transferring calls. For more information about using the Xfer key, see your Model-specific *User's Guide.*

Softkey/ Programmable Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
Services (Not applicable to the 53i)	services	Services	Indicates the key is set to access Services, such as, Directory List, Callers List, Voicemail, and any other XML applications configured on the phone. For more information about using the Services key, see your Model-specific *User's Guide*.
Phone Lock (Not applicable to the 57i CT cordless handset)	phone lock	Phone Lock	Indicates the key is configured as a phone lock key, allowing you to press this key to lock/ unlock the phone. For more information about the lock/unlock key, see "Locking IP Phone Keys" on page 5-28.
Empty (Not applicable to the 53i)	empty	Empty	Indicates the key is configured to force a blank entry on the IP phone display for a specific key. If a particular key is not defined, it is ignored. For more information about empty keys, see your Model-specific *User's Guide*.

Reference

For more information about key functions for your model phone, see your Model-specific *User's Guide*.

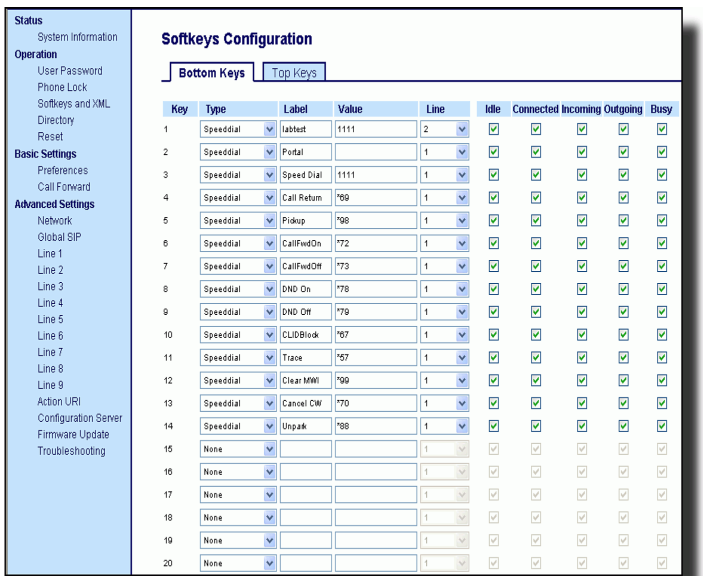
Configuring Softkeys and Programmable Keys

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 Appendix A, the sections, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

Aastra Web UI

1	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** Depending on your phone-model, the key configuration screen displays. The 57i Key Screen is shown as an example. 
2	Select a key to configure.
For Softkeys and Expansion Module Keys:	
3	In the "**Type**" field, select the type of softkey you want to configure. **Reference**: For available type values on each IP phone model, see Appendix A, the section, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.
4	If applicable, enter a label in the "**Label**" field.

Aastra Web UI	
5	If applicable, in the "**Value**" field, enter a value to associate with the softkey. For example, for a speeddial value, you can enter a number you want to use for the speeddial key, or 12345+ as a speeddial prefix.
6	If applicable, in the "**Line**" field, select the line for which you want to associate the softkey.
7	Some softkey types allow you to configure specific operational states. Operational states display to the IP phone when a softkey is used. To enable/disable an operational state, click the "**Idle**", "**Connected**", "**Incoming**", or "**Outgoing**" fields to check or uncheck the box. **Note:** Operational states are not applicable to expansion modules.
8	Click Save Settings to save your changes.
For programmable keys:	
9	In the "**Hard Key**" field, select the programmable key type you want to configure. **Reference**: For available type values on each IP phone model, see Appendix A, the section, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.
10	In the "**Value**" field, enter a value to associate with the programmable key. For example, for a speeddial value, you can enter a number you want to use for the speeddial key, or 12345+ as a speeddial prefix.
11	In the "**Line**" field, select the line for which you want to associate the programmable key.
12	Click Save Settings to save your changes.

57i Cordless (CT) Feature Keys

In addition to the softkeys on the 57i 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 57i 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.

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

Handset Feature Key Functions

You can configure the features keys on the 57i CT handset to perform specific functions using the configuration files or the Aastra Web UI.The following table identifies the available functions for the feature keys on the 57i CT handset.

The following **Handset Key Functions Table** lists the available functions for the keys on the 57i CT IP Phone.

Handset Key Functions Table

Feature Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
None	none	None	Indicates the key is disabled. This option is available from Web UI only.
Line (Lines 1 through 9 are available for selection)	line	Line	Indicates the key is configured for line use.
Icom	icom	Icom	Indicates the key is set to be used as the Intercom key. For more information about the Icom key, see your *Aastra IP Phone 57i CT User's Guide.* For information about other Intercom features, see "Incoming/Outgoing Intercom with Auto-Answer and Barge In" on page 5-60.
Directory	dir	Dir	Indicates the key is set for accessing the Directory List. For more information about the Directory List, see the section "Directory List" on page 182.
Callers List	callers	Callers	Indicates the key is set for accessing the Callers List. For more information on the Callers List, see the section "Callers List" on page 174.

Feature Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
Transfer	xfer	Xfer	Indicates the key is configured as a transfer key for transferring calls. For more information about the Xfer key, see your *Aastra IP Phone 57i CT User's Guide.*
Park	park	Park	Indicates the key is set to be used as a park key to park an incoming call. For more information on park, see the section "Park/Pick Up Key" on page 5-153.
Pickup	pickup	PickUp	Indicates the key is set to be used as a pickup key to pick up a parked call. For more information on pickup, see the section "Park/Pick Up Key" on page 5-153.
Conference	conf	Conf	Indicates the key is configured as a conference key (for local conferencing). (For Sylantro and Broadsoft Servers) An Administrator can also enable centralized conferencing on the IP Phones. For more information about using the Conference key, see your *Aastra IP Phone 57i CT User's Guide.* For information about enabling centralizing conferencing, see "Centralized Conferencing (for Sylantro and Broadsoft Servers)" on page 5-220.

Feature Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
Public	public	Public	Indicates the key is configured to toggle from public to private mode. A public and private key can be used when at a line item in the Directory List. 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 57i CT accepts a maximum of 50 entries with the public attribute. For more information about the public/private keys, see your *Aastra IP Phone 57i CT User's Guide.*
Flash	flash	Flash	Indicates the key is set to generate a flash event when it is pressed, or when a feature key is pressed on the 57i CT cordless handset. 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). For more information about the Flash key, see your *Aastra IP Phone 57i CT User's Guide.*

Reference

For more information about features key functions for your 57i CT, see your *Aastra IP Phone 57i CT User's Guide*.

Configuring Handset Feature Keys

You can program up to 15 feature keys on the 57i CT IP phone using the configuration files or the Aastra Web UI. Use the following procedure to program the feature keys on your 57i CT Base Station and all paired handsets.

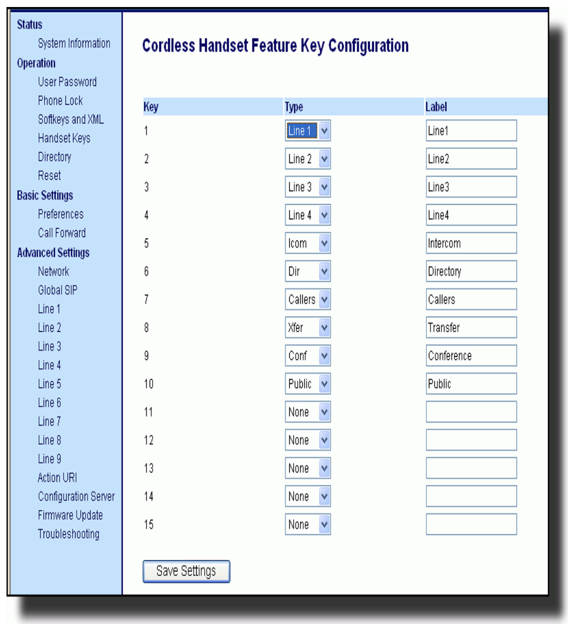
Use the following procedures to configure the IP phone handset feature keys.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Handset Feature Key Settings for the 57i CT" on page A-156.

Aastra Web UI

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



2	Select the handset key you want to program.
3	Select the function for that handset key from the "**Key Function**" field.
4	Click Save Settings to save the function you selected to the handset key. The key programming information is sent to the 57i 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.

Speeddial Prefixes

The normal function of the **speeddial** option allows you to configure a specific key on the phone to dial a number quickly by pressing the configured key. For example, if you had the following speeddial configuration in the configuration files:

```
softkey1 type: speeddial
softkey1 label: Office
softkey1 value: 5552345
softkey1 line: 1
```

after you press softkey1 on the phone, it dials the Office number at 555-2345 on line 1.

A new feature for the speeddial option allows you to configure a preset string of numbers followed by a "+". This feature allows the phone to speeddial a prefix number and then pause to let you enter the remaining phone number. You can use this feature for numbers that contain long prefixes. For example, if you had the following speeddial configuration in the configuration files:

```
softkey2 type: speeddial
softkey2 label: Europe Office
softkey2 value: 1234567+
softkey2 line: 2
```

after you press softkey2 on the phone, it dials the prefix number automatically and pauses for you to enter the remaining number using the keypad on the phone.

You can configure the speeddial prefix using the configuration files or the Aastra Web UI.

Busy Lamp Field (BLF)

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 55i, 57i, and 57i CT, the busy and idle indicators show on the IP phone screen display next to the softkey or programmable key configured for BLF functionality. When the monitored user is idle, an icon with the handset on-hook shows next to the BLF softkey or programmable key. When the monitored user is on an active call, a small telephone icon is shown with the handset off-hook.

On the 53i, 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.

Note: You can configure a maximum of 50 BLFs on the 536M and 560M Expansion Modules.

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 53i 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 57i 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 57i screen next to a softkey button. The softkey icons on the screen change depending on the state of the extensions.

On the 53i 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" are dynamic and the Aastra IP phones are automatically updated without requiring a restart.

Reference

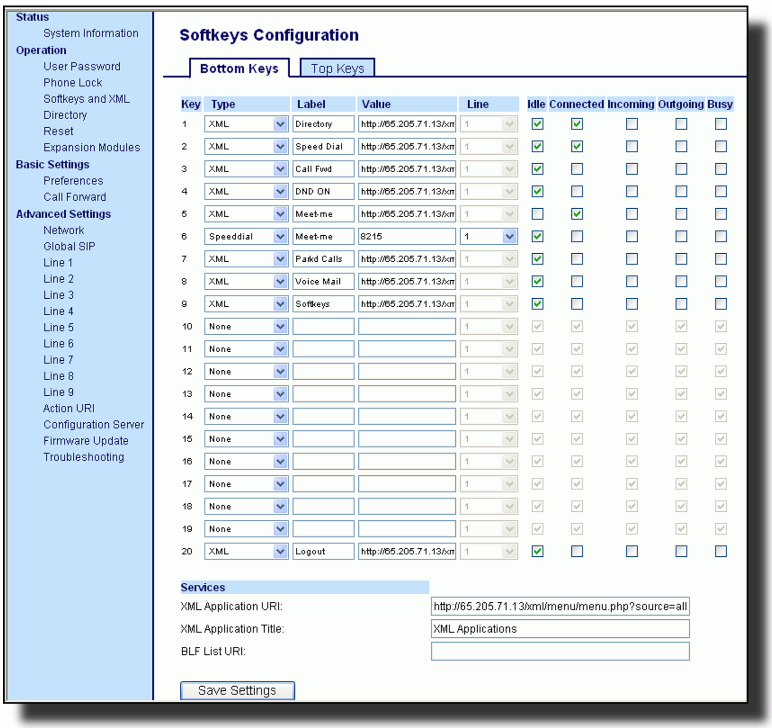
For sample BLF configurations, see Appendix E, "Sample BLF Softkey Settings."

Configuring BLFs

Use the following procedures to configure BLF and BLF\List on the IP phone.

Configuration Files

To set BLF or BLF\List in the configuration files, see Appendix A, the section, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

	Aastra Web UI
1	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** Depending on your phone-model, the key configuration screen displays. The 57i Key Screen is shown as an example. 
2	Select a softkey, programmable, or expansion module key to configure.
3	In the "**Type**" field, select "BLF" (Asterisk), "BLF\List" (BroadSoft BroadWorks).

Aastra Web UI	
4	**For the 55i, 57i, and 57i CT softkeys**: In the "**Label**" field, enter the name of the person who's extension you are monitoring (if "Type" is BLF). **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.
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, my57i-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 57i screen and the feature is disabled.
9	Click Save Settings to save your changes.

BLF Subscription Period

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.

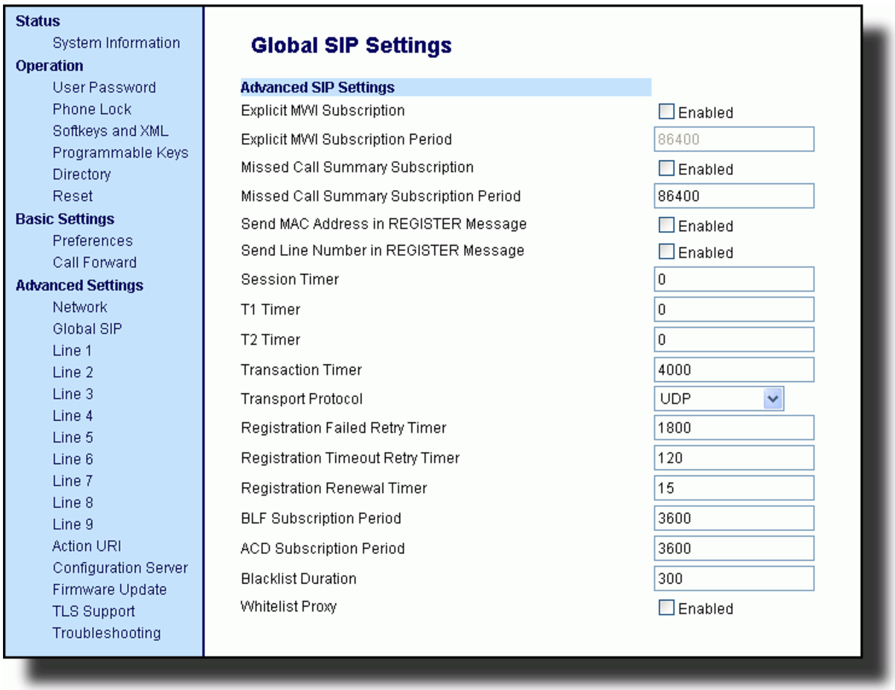
Configuring BLF Subscription Period

Use the following procedures 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 Appendix A, the section, “Advanced SIP Settings” on page A-62.

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.

Automatic Call Distribution (ACD) (for Sylantro Servers)

The IP phones support an Automatic Call Distribution (ACD) feature for Sylantro servers. The ACD feature allows the Sylantro server to distribute calls from a queue to registered IP phone users (agents).

To use the ACD feature on an IP phone, the administrator must first configure an an ACD softkey or programmable key. When an IP phone user wants to subscribe to a queue (in order to receive incoming calls), the user presses the ACD key. The IP phone UI prompts the user to specify the following information:

- **User ID**: the phone number(s) used to login into the queue.
- **Password**: the password used to login to the queue.
- **Available/unavailable**: Shows the current status of the IP phone. Specifies if the IP phone user is available/unavailable to receive a call from the queue. This parameter is set to "unavailable" by default.

When the IP phone user is ready to receive calls from the server, the user logs into a queue. Depending on the server configuration, the IP phone is either in an "unavailable" or "available" state. If the phone is set to "available" then the server begins to distribute calls to this phone immediately. If the phone is set to unavailable, then server waits until the IP phone user manually changes the phone status to "available" (using the IP phone UI) before distributing calls.

Whenever the IP phone user answers a call, or misses a call, the server automatically changes the phone's status to unavailable. The server updates it database with this new information and no longer distributes calls to this phone. The IP phone will remain in this state until:

— the IP phone user makes himself "available" again.

— the ACD auto-availability timer expires. This occurs only if the administrator has configured an ACD auto-availability timer as described in "ACD Auto-Available Timer" on page 5-123.

The IP phone user can also choose to manually change the phone status to unavailable, using the IP Phone UI.

➡ **Note:** It is recommended you configure no more than a single ACD softkey or programmable key per IP phone.

ACD Auto-Available Timer

Whenever the IP phone user answers a call, or misses a call, the server automatically changes the phone's status to unavailable. The administrator can control how long the IP phone remains in the unavailable state by configuring an auto-available timer. When the timer expires, the IP phone status is automatically changed to available. The default setting for the timer is 60 seconds.

You use the following parameters to configure an ACD Auto-Available Tmer in the configuration files:

- **`acd auto available`**
- **`acd auto available timer`**

Configuring an Automatic Call Distribution (ACD) Key

You can configure an ACD key on softkeys, programmable keys, and extension module keys. The following table illustrates examples of configuring an ACD key on the phone.

Softkey Examples	Top Softkey Examples	Programmable Key Examples	Extension Module Examples
softkey1 type: acd softkey1 label: sales softkey1 line: 1 softkey1 states: idle	topsoftkey1 type: acd topsoftkey1 label: sales topsoftkey1 line: 1 topsoftkey1 states: idle	prgkey1 type: acd prgkey1 line: 1:	expmod1 key1 type: acd expmod1 key1 label: sales expmod1 key 1 line: 1

Use the following procedures to configure an ACD key n the IP phone.

Configuration Files

To configure an ACD key using the configuration files, see Appendix A, the section, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

Operational IP Phone Features

Configuring the ACD Auto-Available Timer.

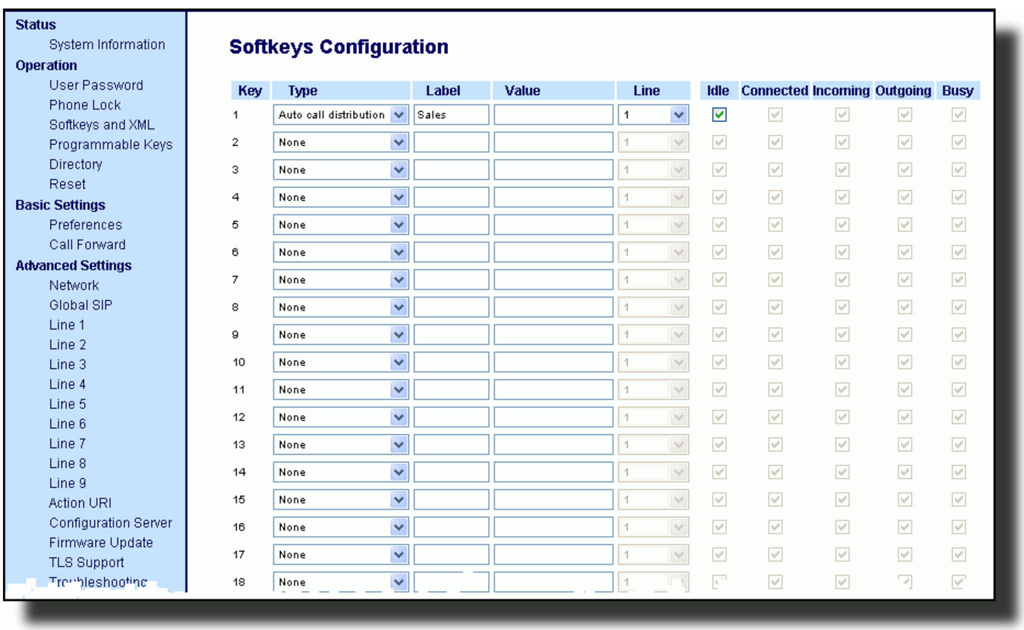
Configuration Files

To configure the ACD Auto-Available Timer using the configuration files, see Appendix A, the section, "ACD Auto-Available Timer Settings" on page A-131.

Configuring an ACD Key Using the Aastra Web UI

Use the following procedure to configure an ACD softkey, programmable key, or expansion module key using the Aastra Web UI. This procedure uses the 55i IP phone as an example.

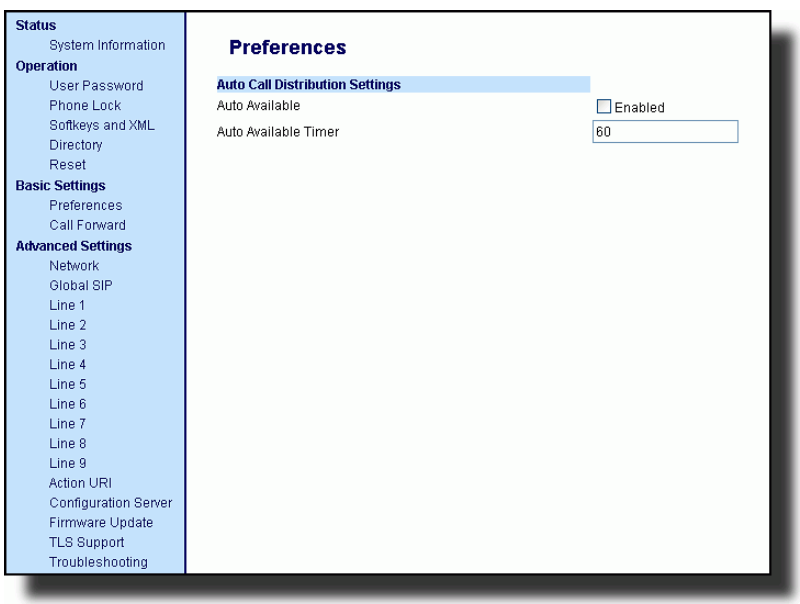
Aastra Web UI

1	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** Depending on your phone-model, the key configuration screen displays. The 57i Key Screen is shown as an example. 
2	In the "**Type**" field, select **Auto Call Distribution**.
3	In the **"Label"** field, specify a name for this ACD softkey. The Label helps identify which queue you are subscribing to when you login. (This field does not apply to the 53i.) For example: **Sales**

	Aastra Web UI
4	In the **“Line”** field, select the line which the IP phone uses to subscribe to the queue. For example: **Line 1**
5	Click Save Settings to save your changes.

Configuring the ACD Auto-Available Timer Using the Aastra Web UI

Use the following procedure to configure an ACD auto-available timer using the Aastra Web UI.

	Aastra Web UI
1	Click on **Basic Settings->Preferences->Auto Call Distribution Settings**. 
2	In the “**Auto Available**” check-box, click **Enabled**.

Aastra Web UI	
3	In the **“Auto Available Timer”** field, specify the length of time (in seconds) before the IP phone state is automatically reset to “available.” Valid values are 0 to 120 seconds. Default is 60. For example: **60**
4	Click **Save Settings** to save your changes.

Using the ACD Feature on your IP Phone

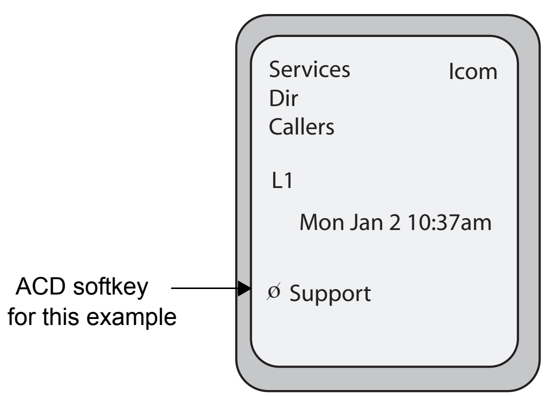
The ACD feature allows you to login to a phone queue in order to receive distributed calls on your IP phone. To login to a phone queue, your system administrator must preconfigure an ACD softkey or programmable key on your Aastra IP phone.

For models 55i, 57i, 57i CT, the ACD softkey is labeled according to your network requirements. Check with your administrator to verify the label assigned to the ACD softkey on your IP phone. The label usually describes which phone queue you are accessing when you press the ACD softkey.

For example, suppose the administrator wants to configure an ACD softkey to allow an IP phone user to log into the Customer Support phone queue. The administrator assigns the label "Support" to the softkey, so it is easily recognizable to the IP phone user. When the IP phone user wants to subscribe to the Customer Support queue, the user presses the Support key and can log in.

Once logged in to the queue, you can make himself "available" or "unavailable" to take calls by pressing the Available/Unavailable key on the phone UI. The server monitors your IP phone status. When you set the IP phone to "available," the server begins distributing calls to your phone. When you set the IP phone to "unavailable," the server temporarily stops distributing calls to your phone.

The icon that appears next to the ACD softkey or programmable key on the IP Phone UI reflects your current status. In the example shown below, the ∅ icon shows the current status of this IP phone user as "logged off."



This icon changes when you log on to the phone queue and are available to take calls. The icon changes again when you are busy with an active call. The table below describes the meaning of the LED, and each icon, as they may appear on your IP phone:

Phone Model	Status: Logged In and Available	Status: Unavailable	Logged Out
53i	Solid Red LED	Blinking red LED	No LED
55i, 57i, 57i CT	Solid Red LED √ icon	Blinking Red LED Blinking √ icon	No LED Ø icon

Logging In to a Phone Queue (55i, 57i, 57i CT)

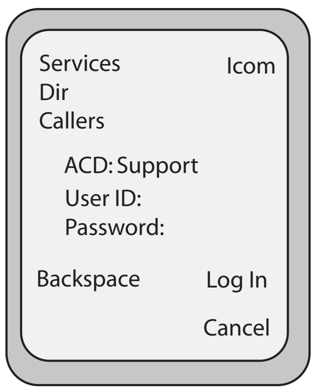
Use the following procedure to log into a phone queue from your Aastra IP phone.

Aastra IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Press the ACD softkey on your IP phone.

Note: Check with your administrator to verify the label assigned to the ACD softkey on your IP phone.

The login screen (see below) appears. In this example, the ACD softkey accesses the Customer Support phone queue and is labelled “Support.”


</td></tr>
<tr><td>2</td><td>To log into the phone queue, use your IP phone keypad to enter the following information:
User ID: The phone number you use to access the queue.
Password: The password you use to access this queue.</td></tr>
</table>

Aastra IP Phone UI

Step	Action
3	Press the **Log In** softkey. You are logged into the phone queue. Once you log in, examine the IP Phone UI, and note the following information: • If your IP phone status is set to “available” then the server will begin to distribute phone calls from this queue to your IP phone. • If your IP phone status remains “unavailable” after you log in, then you must manually change the state to “available” in order to start receiving calls. • To temporarily stop receiving calls, you can switch the IP phone status to “unavailable.” While you are on a call (or miss a call that has been distributed to your IP phone), your IP phone status automatically switches to “unavailable.” Your IP phone remains in the unavailable state until one of the following things occur: • You use the IP Phone UI to manually switch the IP phone state back to available, or • The availability “timer” for your IP phone expires. This only occurs if your administrator has configured an auto-availability timer on your IP phone.
4	To Log out of the queue, press the **Log Out** softkey. The server no longer distributes phone calls to your IP phone.

Logging In To a Phone Queue (53i)

Use the following procedure to log into a phone queue from your Aastra IP phone.

Aastra IP Phone UI	
Step	**Action**
1	Press the ACD programmable key on your IP phone.
2	To login to the phone queue, use your IP phone keypad to enter the following information: **User ID:** The phone number you use to access the queue. **Password:** The password you use to access this queue.
3	Select **Login**. You are logged into the phone queue. Once you log in, examine the IP Phone UI, and note the following information: • If your IP phone status is set to “available” then the server will begin to distribute phone calls from this queue to your IP phone. • If your IP phone status remains “unavailable” after you log in, then you must manually change the state to “available” in order to start receiving calls. • To temporarily stop receiving calls, you can switch the IP phone status to “unavailable.” While you are on a call (or miss a call that has been distributed to your IP phone), your IP phone status automatically switches to “unavailable.” Your IP phone remains in the unavailable state until one of the following things occur: • You use the IP Phone UI to manually switch the IP phone state back to available, or • The availability “timer” for your IP phone expires. This only occurs if your administrator has configured an auto-availability timer on your IP phone.
4	To Log out of the queue, select **Logout**. The server no longer distributes phone calls to your IP phone.

ACD Subscription Period

On the IP phones, you can set the time period, in seconds, that the IP phone resubscribes the ACD 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 ACD subscription period:

```
sip acd 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 ACD 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.

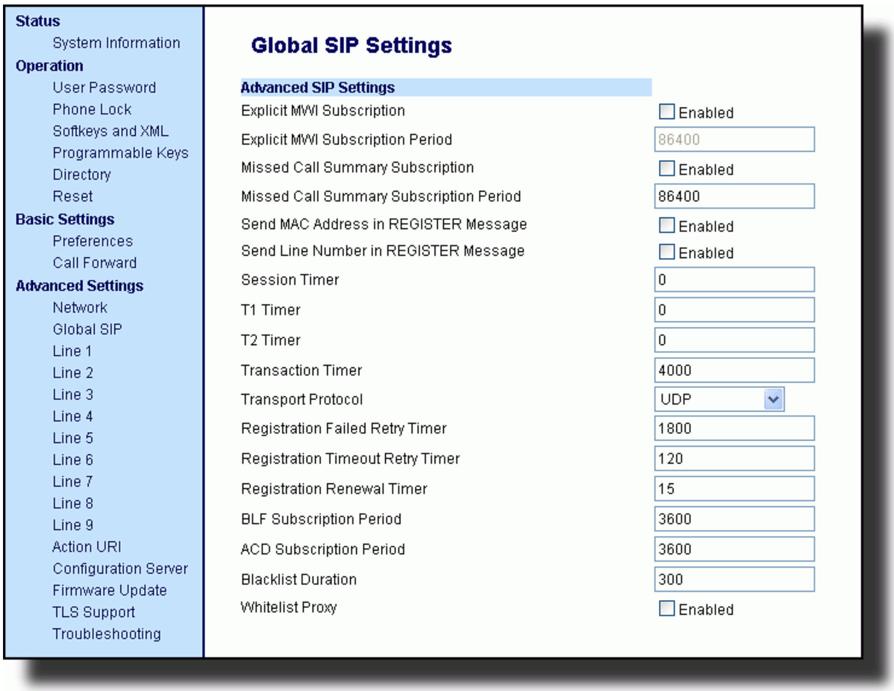
You can configure this feature using the configuration files or the Aastra Web UI.

Configuring ACD Subscription Period

Use the following procedures to configure the ACD subscription period on the IP phone.

Configuration Files
To configure the ACD subscription period on the IP phones using the configuration files, see Appendix A, the section, "Advanced SIP Settings" on page A-62.

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 "**ACD Subscription Period**" field.
3	Click Save Settings to save your changes.

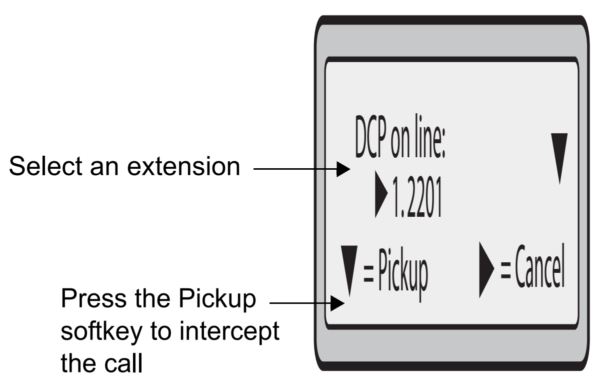
Directed Call Pickup/Group Call Pickup (for Sylantro Servers)

Aastra IP phones and any attached Expansion Modules support the Directed Call Pickup (DCP) and Group Call Pickup (GCP) features.

The Directed Call Pickup/Group Call Pickup feature allows you to intercept - or pickup - a call on a monitored extension. An Administrator or User can configure this feature using the Aastra Web UI to create a DCP or GCP softkey on the IP phone. When you configure a DCP softkey, you specify the extension that you want to monitor. Then, when the monitored extension receives a call, you press the DCP softkey to “pickup” (intercept) it. If the monitored extension receives multiple incoming calls simultaneously, the IP Phone UI displays a list of incoming calls. You select a call from this list, and are connected to the call.

When you configure a GCP softkey, you specify the ring group that you want to monitor for incoming calls. For example, suppose an Operator configures a GCP softkey to monitor incoming calls for a specific ring group (extensions 2200-2210). When an incoming call is received on any of these extensions, the Operator presses the GCP softkey and is connected to the call. If multiple incoming calls are received simultaneously, the Operator does the following actions:

- Presses the GCP softkey. The Operator Phone UI displays the current list of incoming calls (see below).
- Selects an extension to “pickup” first.
- Presses the Pickup softkey. The Operator is connected to the incoming call.



Configuring DCP/GCP Using the Configuration Files (for Sylantro Servers)

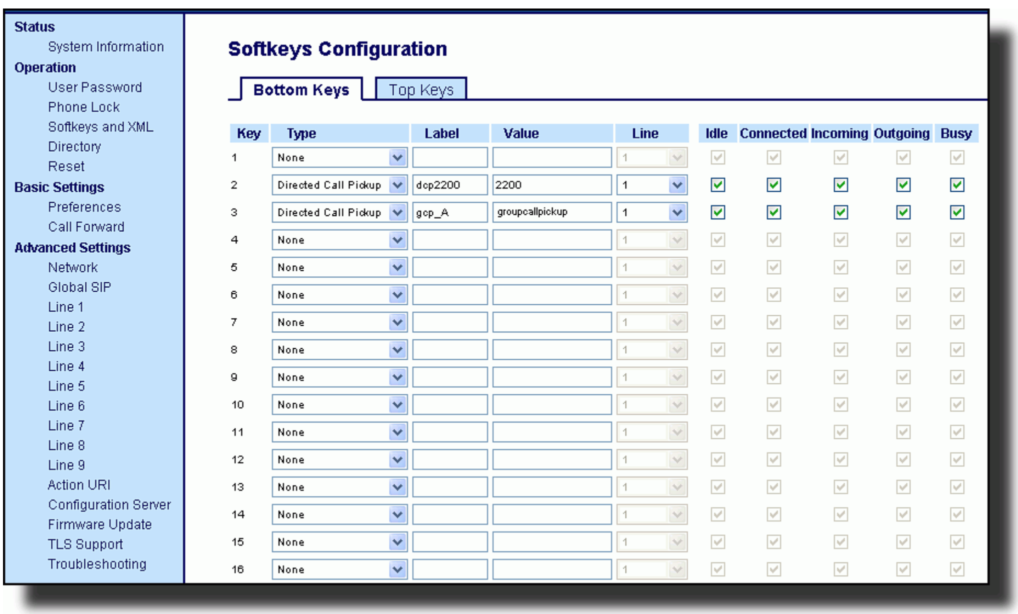
Use the following procedures to configure DCP/GCP using the configuration files.

Configuration Files
To set DCP/GCP in the configuration files, see Appendix A, the section, "Softkey/Programmable Key/ Feature Key/Expansion Module Key Parameters" on page A-136.

Configuring Directed Call Pickup (DCP) Using the Aastra Web UI (for Sylantro Servers)

Use the following procedure to configure Directed Call Pickup using the Aastra Web UI. This procedure uses the 57i IP Phone as an example.

Aastra Web UI

1	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** Depending on your phone-model, the key configuration screen displays. The 57i Key Screen is shown as an example. 
2	Select a key for which to configure Directed Call Pickup.
3	In the "**Type**" field, select **Directed Call Pickup**.
4	In the **"Label"** field, specify a name for this Directed Call Pickup softkey. For example: **DCP2200**

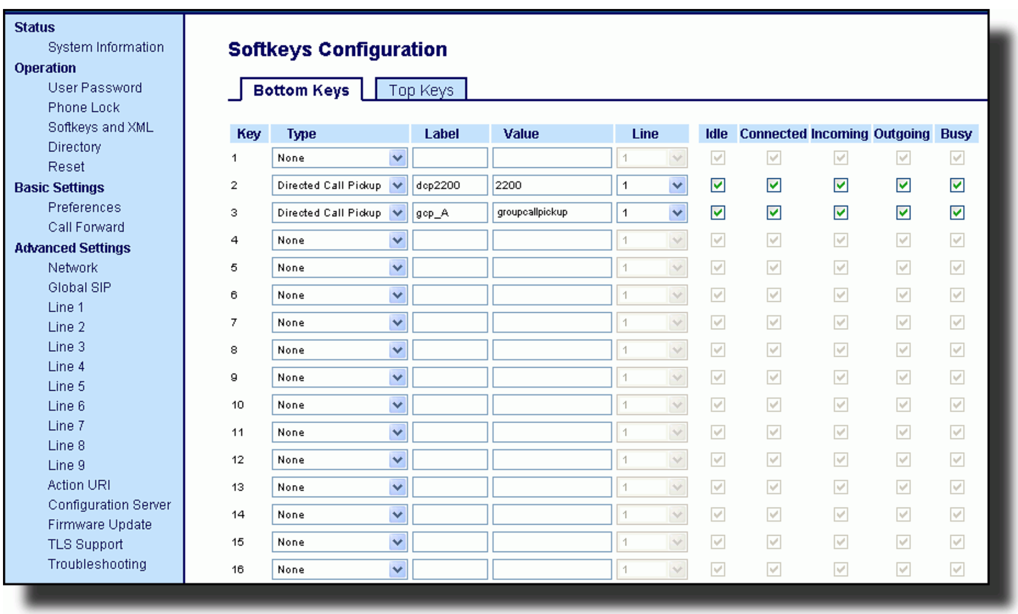
Aastra Web UI	
5	In the **"Value"** field, specify the extension you want to intercept when you press this softkey. For example: **2200**
6	Click **Save Settings** to save your changes.

Configuring Group Call Pickup (GCP) Using the Aastra Web UI (for Sylantro Servers)

Use the following procedure to configure Group Call Pickup using the Aastra Web UI.

Note: A ring group must be configured on the Sylantro Server in order for a GCP softkey to function.

Aastra Web UI

1	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** Depending on your phone-model, the key configuration screen displays. The 57i Key Screen is shown as an example. 
2	Select a key for which to configure Group Call Pickup.
3	In the “**Type**” field, select **Directed Call Pickup**.

Aastra Web UI	
4	In the **"Label"** field, specify a name for this Directed Call Pickup softkey. For example: **GCP_A**
5	In the **"Value"** field, enter **groupcallpickup.**
6	Click Save Settings to save your changes.

Using Directed Call Pickup/Group Call Pickup

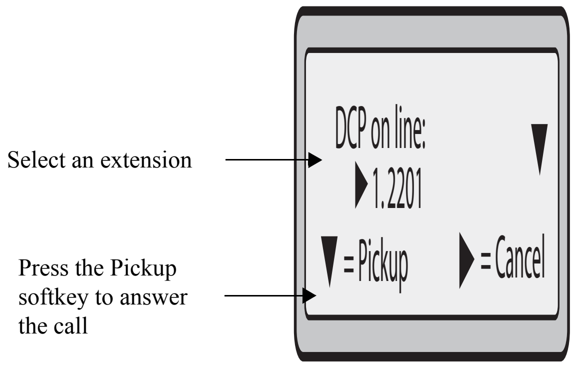
Use the following procedure for the DCP/GCP on your phone.

Note: Before using the DCP/GCP feature on your phone, you must first configure the DCP or GCP key. You must identify the extension(s) or phone number(s) you want to monitor when configuring the key.

IP Phone UI

Step	Action
Using Directed Call Pickup (DCP)	
1	When the monitored extension receives a call, press the **DCP** softkey to pick up the call.
2	If the monitored extension receives multiple incoming calls simultaneously, the phone displays a list of incoming calls.
3	Select a call from this list using the **UP** and **DOWN** arrow keys. The call is answered.
Using Group Call Pickup (GCP)	
1	When any of the monitored group of extensions receives a call, press the **GCP** softkey to pick up the call. The call is answered.

IP Phone UI

Step	Action
2	If the monitored extension receives multiple incoming calls simultaneously, the phone displays a list of incoming calls.
3	Select a call from this list using the **UP** and **DOWN** arrow keys. Press the "**Pickup**" softkey to answer the call you selected. 

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.

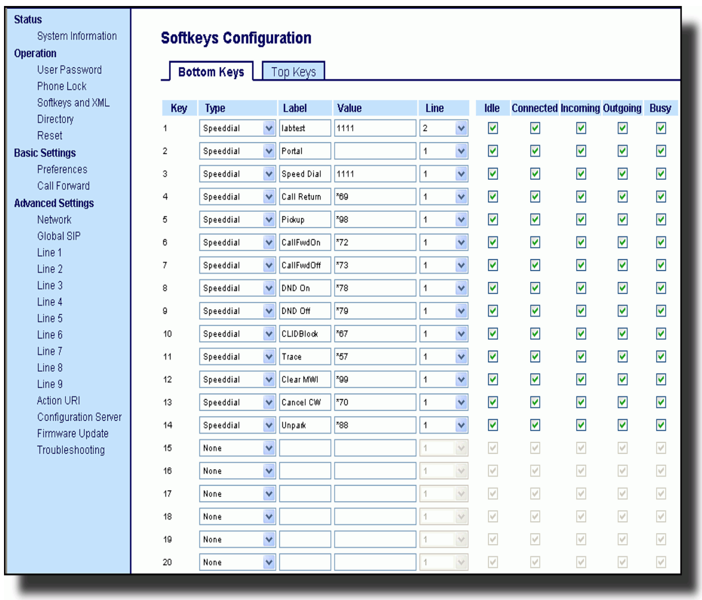
Configuring DND

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

Configuration Files

For specific softkey and programmable key parameters you can set in the configuration files, see Appendix A, the section, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

Aastra Web UI

1	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** Depending on your phone-model, the key configuration screen displays. The 57i Key Screen is shown as an example 
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.

Bridged Line Appearance (BLA) (55i, 57i, 57i CT 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.

Notes:

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
57i	9
57i CT	9
53i	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-line 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 procedures to configure BLA on the IP phone.

Configuring Global BLA

Configuration Files

For specific **global** parameters you can set in the configuration files, see Appendix A, the section, "SIP Basic, Global Settings" on page A-41.

Configuring Per-Line BLA

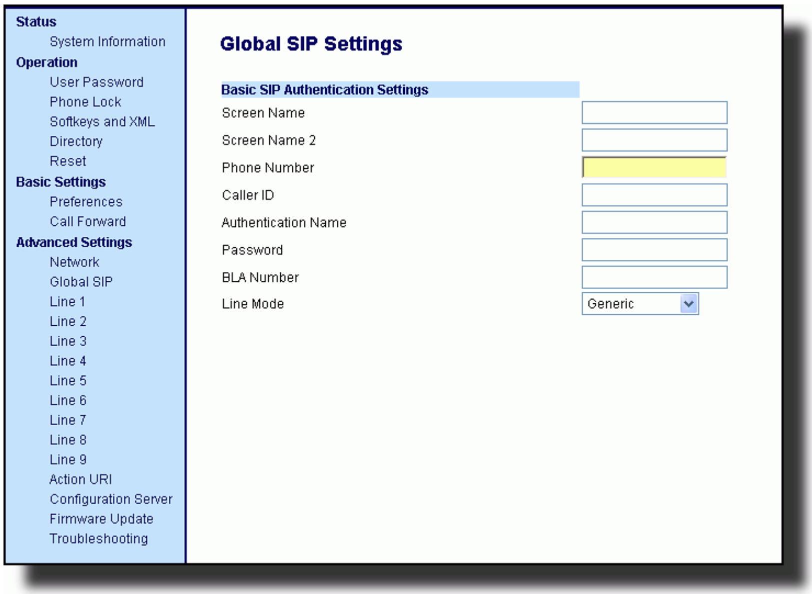
Configuration Files

For specific **per-line** parameters you can set in the configuration files, see Appendix A, the section, "SIP Basic, Per-Line Settings" on page A-50.

Aastra Web UI

1 ***For global configuration of BLA:***

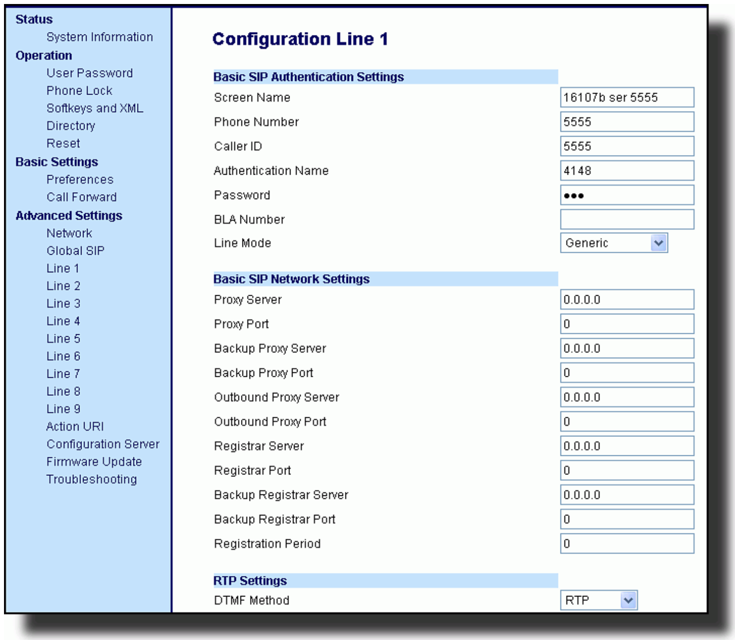
Click on **Advanced Settings->Global SIP->Basic SIP Authentication Settings.**



Aastra Web UI

For per-line configuration of BLA:

Click on **Advanced Settings->Line *N* (1-9).**



	In the "**Line Mode**" field, select the **BLA** option.
2	In the "**Phone Number**" field, enter the phone number of the IP phone.
3	***For global configuration of BLA:*** In the "**BLA Number**" field, enter the Bridge Line Appearance (BLA) number to be shared across all IP phones. ***For per-line configuration of BLA:*** In the "**BLA Number**" field, enter the Bridge Line Appearance (BLA) number to be shared on a specific line.
4	Click Save Settings to save your changes.

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.

BLA Support for Third Party Registration

BLA allows an Address Of Record (AOR) to be assigned onto different line appearances for a group of SIP user agents (IP phones). When a call is made to this BLA number, the call is offered to all user agents that have mapping to this BLA. To support this, the IP phones need to support third party registration for the BLA along with the registration for its own primary appearance number. If the IP phone has the primary appearance as a BLA, then there is no need for third party registration.

When configuring the BLA feature on a per-line basis for third party registration and subscription, the third party name must be configured using the "*sip lineN bla number*" parameter. For third party registration to work effectively, one of the lines should register as generic with its own username.

For example, Bob has Alice's appearance on his phone. Bob's configuration is as follows:

```
#line 1 Bob

sip line1 auth name:4082272203
sip line1 password:
sip line1 mode: 0
sip line1 user name:4082272203
sip line1 display name:Bob
sip line1 screen name:Bob


#line 2 Alice

sip line2 auth name:4082272203
sip line2 password:


#BLA mode 3

sip line2 mode: 3
sip line2 user name:4082272203
```

```
#Alice phone number

sip line2 bla number:4085582868
sip line2 display name:Alice
sip line2 screen name:Alice
```

Alice's configuration is as follows:

```
#line 1

sip line1 auth name:4085582868
sip line1 password:
sip line1 mode: 3
sip line1 user name:4085582868
sip line1 display name: Alice
sip line1 screen name: Alice
```

Park/Pick Up Key

The IP phones (including the 57i 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 administrator can configure this feature:

- Using a static configuration (globally configures park and pickup)
- Using a programmable configuration (using a key)

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 paragraph describes the configuration of a park and pickup key on the IP phone. For information about configuring the park and pickup static configuration method see "Park Calls/Pick Up Parked Calls" on page 5-35.

Park/Pickup Programmable Configuration (using a key)

The programmable method of configuration creates park and pickup keys (softkeys, programmable keys, extension module keys) that you can configure on the IP phones.

For the 55i, 57i, and 57i CT you can set a key 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 53i, you can set a programmable key as "Park" or "Pickup" and then:

- specify a value
- specify a line to use

On 57i/57i 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 57i CT, the customized labels apply to the base unit only. On the 57i CT handset, pressing Ï displays the default labels of "Park" and "Pickup".

Notes:

1. On the 57i CT, the customized labels apply to the base unit only. On the 57i CT handset, pressing displays the default labels of "Park" and "Pickup".

2. On the 57i/57i 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 53i

- 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 key configuration using the configuration files or the Aastra Web UI.

Configuring Park/Pickup Key Using Configuration Files

In the configuration files, you configure Park/Pickup using the key parameters. You must specify the "softkeyN value", "prgkeyN value", “topsoftkeyN value“, or “expmodX keyN value”. The following examples show Park/Pickup configurations using specific servers.

Model 57i/57i CT Examples

Server	Park Configuration	Pickup Configuration
Asterisk	softkeyN type: **park** softkeyN label: **parkCall** softkeyN value: **asterisk;70** softkeyN line: **1** softkeyN states: **connected***	softkeyN type: **pickup** softkeyN label: **pickupCall** softkeyN value: **asterisk;70** 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 53i Examples

Server	Park Configuration	Pickup Configuration
Asterisk	prgkeyN type: **park** prgkeyN value: **asterisk;70** prgkeyN line: **1**	prgkeyN type: **pickup** prgkeyN value: **asterisk;70** 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**

Note: The 53i does not allow for the configuration of labels and states.

Use the following procedure to configure a Park/Pickup key using the configuration files.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Softkey Settings for 55i, 57i, 57i CT" on page A-137 and "Programmable Key Settings for 53i and 55i" on page A-145.

Configuring a Park/Pickup Key Using Aastra Web UI

On the 57i/57i 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 57i 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 57i handset uses the first programmable configuration.

For example, if line 1 and line 6 are configured for park, the 57i CT handset uses the configuration set for line 1 to park a call.

On the 53i, you configure a Park and/or Pickup key at **Operation->Programmable Keys**, and then enter the appropriate value and line.

Note: Applicable values depend on the server in your network (Asterisk, BroadWorks, Sylantro, ININ PBX. See the table below for applicable values.

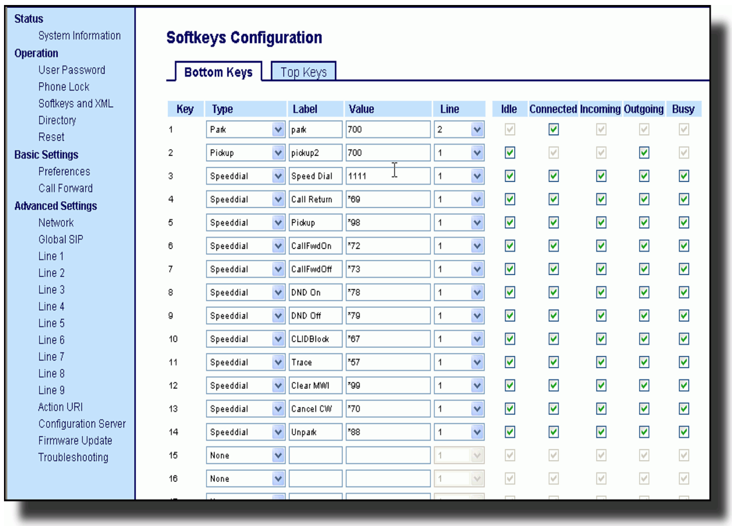
Park/Pickup Call Server Configuration Values

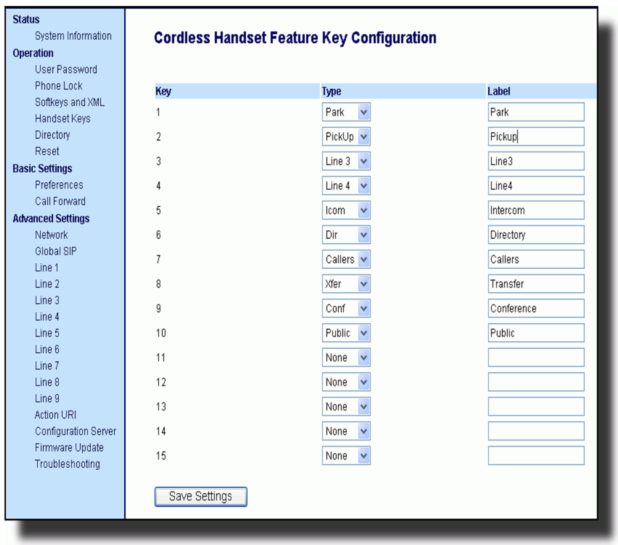
Server	Park Values*	Pickup Values*
Aasterisk	70	70
Sylantro	*98	*99
BroadWorks	*68	*88
ININ PBX	callpark	pickup

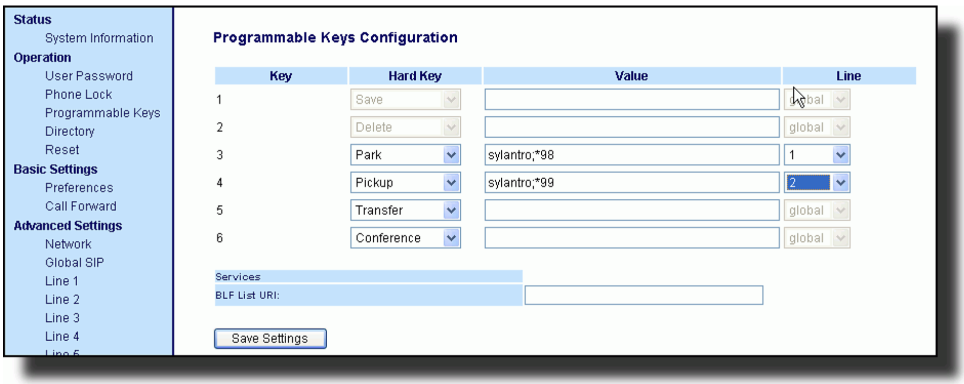
*Leave "value" fields blank to disable the park and pickup feature.

Use the following procedure to configure the Park/Pickup call feature using the programmable configuration method and the Aastra Web UI.

Aastra Web UI

For the 57i/57i 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 appropriate 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 5-157.
6	In the "**Line**" field, select a line for which to apply the Park configuration.
7	The park softkey has a default state of "**Connected**". Leave this state enabled or to disable, uncheck the check box.
8	Pick a softkey to configure for Picking up a call.
9	In the "**Type**" field, select **Pickup**.
10	In the "**Label**" field, enter a label for the Pickup softkey.

	Aastra Web UI
11	In the "**Value**" field, enter the appropriate 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 5-157.
12	In the "**Line**" field, select a line for which to apply the Pickup configuration.
13	The pickup softkey has a default state of “**Idle**” and “**Outgoing**”. Leave these states enabled or to disable, uncheck the check boxes.
14	Click Save Settings to save your changes.
For the 57i 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 for picking up a call.
5	In the "**Key Function**" field, select **Pickup**.
6	Click Save Settings to save your changes.

	Aastra Web UI
For the 53i:	
1	Click on **Operation->Programmable Keys.** ***53i Screen*** 
2	Pick a hard key (from keys 3 through 6) to configure for Parking a call.
3	In the "**Hard Key**" field, select **Park**.
4	In the "**Value**" field, enter the appropriate 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 5-157.
5	***For the 53i:*** In the "**Line**" field, select a line for which to apply the Park configuration.
6	Pick a hard key to configure for Picking up a call.
7	In the "**Hard Key**" field, select **Pickup**.
8	In the "**Value**" field, enter the appropriate 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 5-157.
9	***For the 53i:*** In the "**Line**" field, select a line for which to apply the Pickup configuration.

Aastra Web UI	
10	Click [Save Settings] to save your changes. 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 apply the changes.

Using the Park Call/Pickup Parked Call Feature

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

IP Phone UI

Step	Action
Parking 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 Sylantro 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.
Picking up a Parked Call	
4	Pick up the handset on the phone.

IP Phone UI	
Step	**Action**
5	Enter the extension number where the call was parked.
6	Press the "**Pickup**" softkey. If the call pick up is successful, you are connected with the parked call.

Last Call Return (lcr) (For Sylantro Servers

Last call return (lcr) allows an administrator or user to configure a "last call return" function on a softkey or programmable key. This feature is for Sylantro servers only.

You can configure the "lcr" softkey feature via the configuration files or the Aastra Web UI.

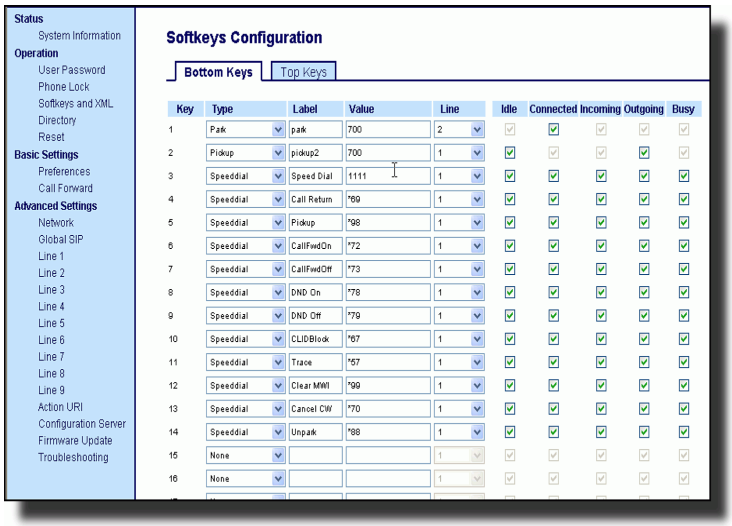
How it works

If you configure "lcr" on a softkey or programmable key, and a call comes into your phone, after you are finished with the call and hangup, you can press the key configured for "**lcr**" and the phone dials the last call you received. When you configure an "lcr" softkey, the label "LCR" displays next to that softkey on the IP phone. When the Sylantro server detects an "lcr" request, it translates this request and routes the call to the last caller.

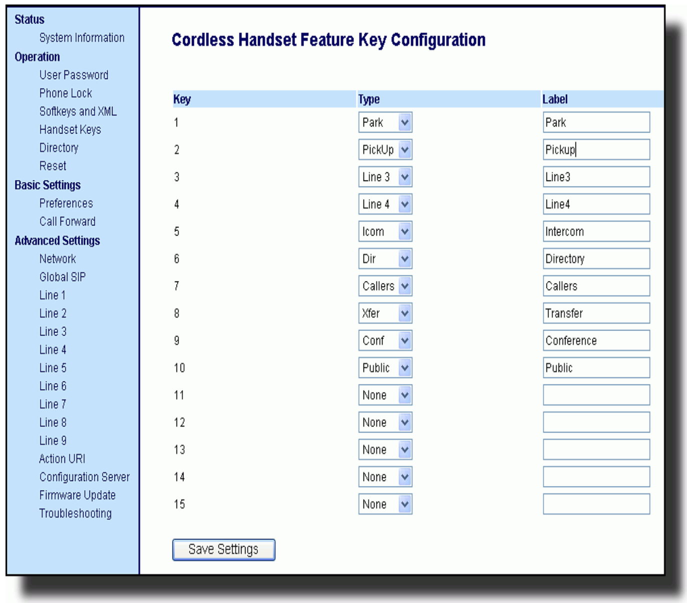
Configuring Last Call Return

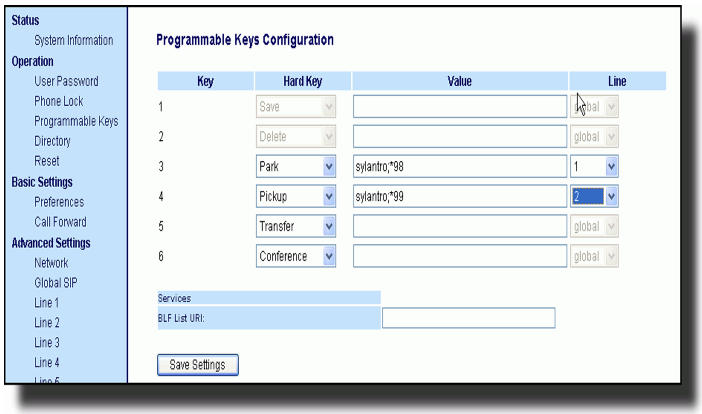
Use the following procedures to configure LCR on the IP phones.

Configuration Files
For specific last call return (lcr) parameters you can set in the configuration files, see Appendix A, the section, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

Aastra Web UI	
For the 57i/57i CT:	
1	Click on **Operation->Softkeys and XML.** 
2	Pick a softkey to configure for Last Call Return.
3	In the "**Type**" field, select **lcr**.
4	In the "**Line**" field, select a line for which to apply the last call return configuration.
5	The lcr softkey has a default state of "**Idle**", "**Connected**", "**Incoming**", "**Outgoing**". Leave these states enabled or to disable, uncheck the check boxes as required.
6	Click **Save Settings** to save your changes.

Aastra Web UI

For the 57i CT Handset:	
1	Click on **Operation->Handset Keys.** 
2	Pick a handset key to configure for Last Call Return.
3	In the "**Key Function**" field, select **lcr**.
4	Click  to save your changes.

Aastra Web UI	
For the 53i	
1	Click on **Operation->Programmable Keys.** ***53i Screen*** 
2	Pick a hard key (from keys 3 through 6) to configure for Last Call Return.
3	In the "**Type**" field, select **lcr**.
4	***For the 53i:*** In the "**Line**" field, select a line for which to apply the lcr configuration.
5	Click Save Settings to save your changes.

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 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 *(per-line only)*	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
53i	9
55i	9
57i	9
57i CT	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 Forwarding via the Aastra Web UI and the IP Phone UI using the "Call Forward" 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 Forwarding.

Use the following procedures to enable/disable Call Forwarding on the IP phones.

Configuration Files

For specific parameters you can set in the configuration files for enabling/disabling Call Forwarding, see Appendix A, the section, “Call Forward Settings” on page A-87.

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 55i, 57i, and 57i CT.

Configuring Call Forwarding

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

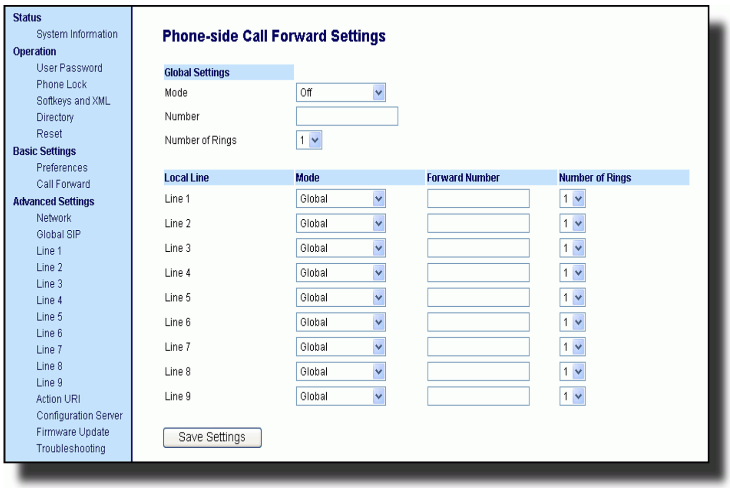
IP Phone UI

Step	Action
For global configuration of call forwarding:	
1	Press [Options key] on the phone to enter the Options List.
2	Select **Call Forward.**
3	***For 53i:*** In the "**Cfwd Number**" field, enter the destination number for which you want your incoming calls to be forwarded. ***For 55i/57i/57i CT:*** In the "**Number**" field, enter the destination number for which you want your incoming calls to be forwarded. **Note:** Leaving the number field blank disables call forwarding.

IP Phone UI

Step	Action
4	***For 53i:*** In the “**Cfwd Mode**” field, enter the mode that you want to set on your phones. ***For 55i/57i/57i CT:*** In the “**Mode**” field, enter the mode that you want to set on your phones. Valid modes are: • Off • All • Busy • No Answer • Busy No Answer
5	In the “**No. Rings**” field, enter the number of rings you want the phone to ring before the call is forwarded. Valid values are 1 to 9. **Note:** "**No. Rings**" field applies to **No Answer** and **Busy No Answer** modes only.
6	***For 53i:*** Press **Set** to save the changes. **For 55i/57i/57i CT**: Press **Done** to save the changes.

Aastra Web UI

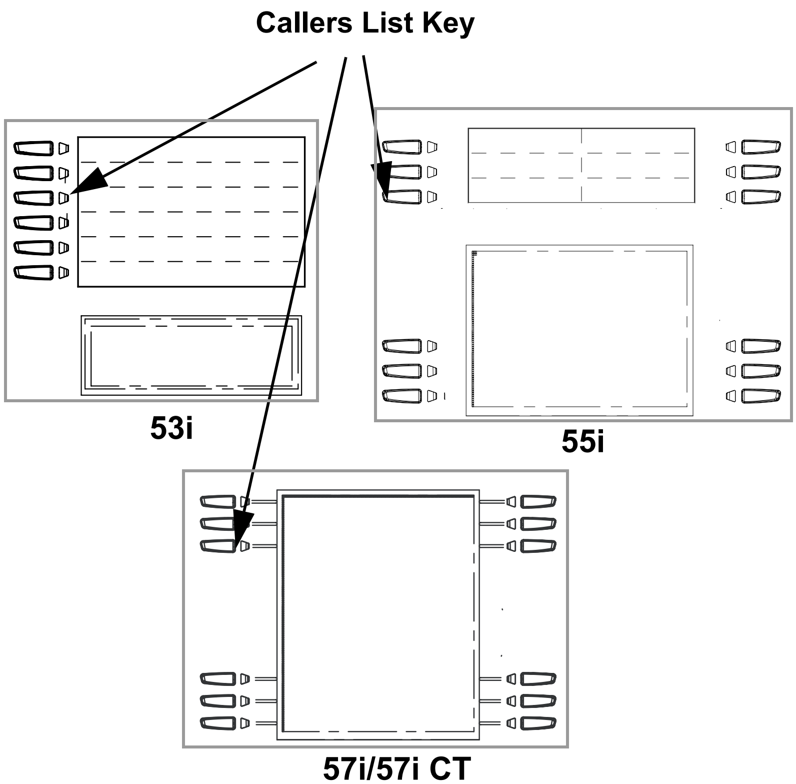
1	Click on **Basic Settings->Call Forward.** 
For global configuration of call forwarding:	
2	In the "**Mode**" field, select the mode you want to set on your phone. Valid modes are: • Off • All • Busy • No Answer • Busy No Answer **Note:** To disable call forwarding in the Aastra Web UI, set the mode to **OFF** and remove the phone number in the **"Number"** field.
3	In the "**Number**" field, enter the destination number for which you want your calls to be call forwarded.

	Aastra Web UI
4	In the "**Number of Rings**" field, enter the number of rings you want your phone to ring 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.
5	Click Save Settings to save your changes.
For per-line configuration of call forwarding	
6	Select a line to configure Call Forwarding on.
7	In the "**Mode**" field, select the mode you want to set on your phone. Valid modes are: • Off • All • Busy • No Answer • Busy No Answer • Global **Notes**: **1.** To disable call forwarding in the Aastra Web UI, set the mode to **OFF** and remove the phone number in the "**Number"** field. **2.** To force a line to use the global settings, set the "**Mode**" field to **Global**.
8	In the "**Forward Number**" field, enter the destination number for which you want your calls on this line to be call forwarded.
9	In the "**Number of Rings**" field, select the number of rings you want this line to ring 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.
10	Click Save Settings to save your changes.

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.

The following illustrating shows the default location of the Callers List Key on each type of phone model.

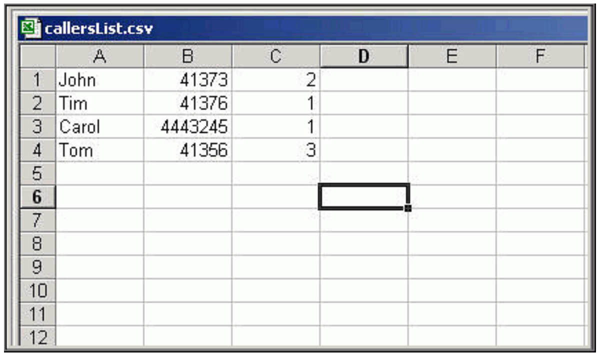


You can enable and disable the Callers List feature using the configuration files. When disabled, the Callers List does not display on the IP phone UI and the Caller List key is ignored when pressed.

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 57i and 57i 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.

Use the following procedures to enable/disable the Callers List on the IP phones.

Configuration Files

For specific parameters you can set in the configuration files for enabling/disabling the Callers List, see Appendix A, the section, "Callers List Settings" on page A-86.

Using the Callers List

Use the following procedure to access and use the Callers List.

IP Phone UI

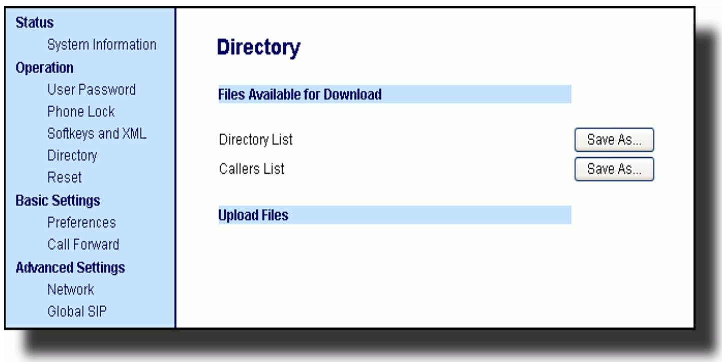
Step	Action
1	Press the **"Callers List"** key on the phone to enter the Callers List.
2	Use the ▲ and ▼ to scroll through the line items in the Callers List. To the left of a line item, a ☎ icon displays with the handset ON or OFF the receiver. The ON receiver indicates the call came in as a missed 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.

IP Phone UI

Step	Action
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 "**Callers List**" key again or the "**Goodbye"** 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 Save .
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.

Customizable Callers List and Services Keys

The IP phones currently have a Callers List key (all 5i Series phones) and a Services key (55i, 57i, and 57i CT). An Administrator can specify URI overrides for these keys using the following parameters in the configuration files:

- **`services script`**
- **`callers list script`**

Specifying URIs for these parameters cause the creation of an XML custom application instead of the standard function of the Callers List and Services keys.

An Administrator can configure these parameters using the configuration files only.

Creating Customizable Callers List and Services Keys

Use the following procedure to create customized Callers List and Services keys on the IP Phone using the configuration files.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Customize Callers List and Services Key" on page A-86.

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.

Use the following procedures to enable/disable the Missed Calls Indicator on the IP phones.

Configuration Files

For specific parameters you can set in the configuration files for enabling/disabling the Missed Calls Indicator, see Appendix A, the section, "Missed Calls Indicator Settings" on page A-88.

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.

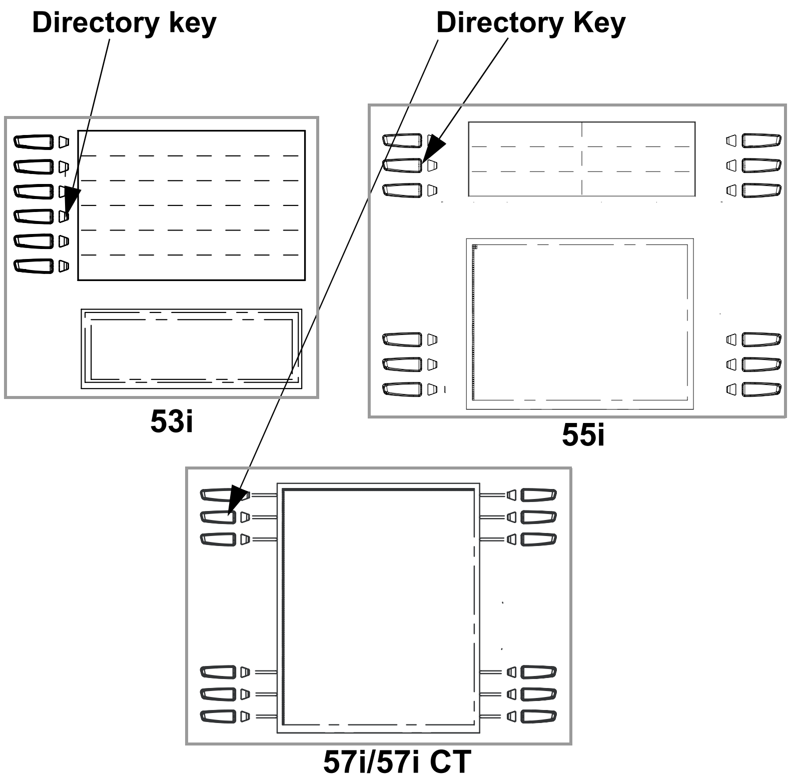
IP Phone UI

Step	Action
1	Press the **"Callers List"** key 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 ◄, **Clear,** or **Delete** key (depending on your phone model). **Note:** To cancel a delete function, press the ▲ or the ▼ **Scroll** keys. The line item deletes from the Callers List.

Directory List

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

The following illustrating shows the default location of the Directory Key on each type of phone model.



Directory List Capabilities

In the Directory List a user or administrator can store a maximum of 7 numbers associated with a unique name. You can apply pre-defined labels to the entry which include, **Office**, **Home**, **Cell**, and **Pager**, or create your own labels. Labels can be up to 14 characters in length.

You can also sort multiple numbers according to preference and perform 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.The quick-search feature in the Directory List works only when the Directory is first accessed.

Reference

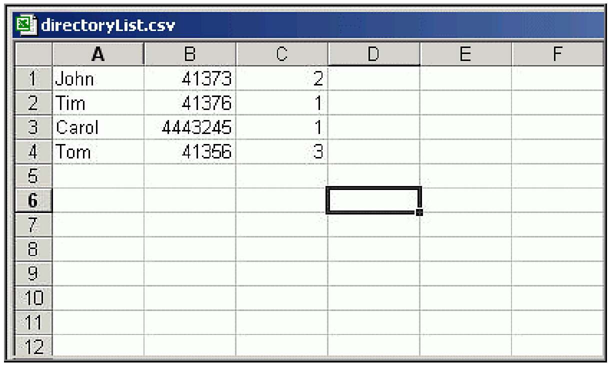
For more detailed information about the Directory Key on your IP phone, and the Directory List, see your model-specific *User Guide.*

Administrator/User Functions for Directory List

You can perform the following pertaining to the Directory List:

- 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 softkey is ignored when pressed. This is an administrator function only.
- 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. This is an administrator and user function.
- A public and private softkey can be used when at a line item in the Directory List. 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 57i CT accepts a maximum of 50 entries with the public attribute. This is an administrator and user function.
- 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. This is an administrator and user function.

- 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. This is an administrator and user function.



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(s), and line number(s) 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 53i, the "Directory" option is also removed from the "Services" menu.

Use the following procedures to enable/disable the Directory List on the IP phones.

Configuration Files

For specific parameters you can set in the configuration files for enabling/disabling the Directory List, see Appendix A, the section, "Directory Settings" on page A-85.

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 (57i CT only)	50 entries

Using the Directory List

Use the following procedures to access and use the Directory List.

Note: In the following procedure, the location of keys (hard keys, softkeys, and programmable keys) on the phone are dependant on your specific phone model. See Chapter 1, Overview, for the keys that are specific to your phone model.

IP Phone UI

Step	Action
1	Press the **DIRECTORY** key to enter the Directory List. **Note:** After entering the Directory List, 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.
To dial from an entry in the Directory List:	
3	At a line item in the Directory List, pick up the handset, press the / key, or press a line key. The phone automatically dials the Directory List number for you.
To add a new entry to the Directory List:	
4	a Press the **SAVE** key or **ADD NEW** softkey (depending on your model phone) at the Directory List header screen and perform step 4. or Press the **SAVE** key or **ADD NEW** softkey at a line item and press the **DIRECTORY** key again. b Enter a phone number, name, and line number and press the **SAVE** key after each field entry.

IP Phone UI

Step	Action
To edit an entry in the Directory List:	
5	a At a line item in the Directory List, press the ◀ key. **Note:** Use the **SAVE** key to scroll between the number, name and line entries. b Edit the phone number if required and press **SAVE**. c Edit the name if required and press **SAVE**. d Edit the line if required and press **SAVE**. e Press **SAVE** to save the changes and exit the editing function.
To delete an entry from the Directory List:	
6	a At a line item in the Directory List, press **DELETE**. The following prompt displays: "*DELETE again to erase this item*". b Press DELETE again to delete the entry from the Directory List. **Note:** To cancel a delete function, press the ▲ or the ▼ scroll keys.
To delete all entries from the Directory List:	
7	a At the Directory List header, press **DELETE** or **DELETE LIST** (depending on your phone model). The following prompt displays: "*DELETE again to erase all items*". b Press DELETE again to delete all entries from the Directory List. **Note:** To cancel a delete function, press the ▲ or the ▼ scroll keys.
To copy an entry from the Directory List to a programmable speeddial key:	
8	At a line item in the Directory List, press the **COPY** key and enter the line number at the "Save to?" prompt that is already configured for speeddialing at a programmable key. **Note:** You must have a speeddial key previously configured on your phone to use this feature. To configure a speeddial key, see your Model-specific *User's Guide*.

IP Phone UI

Step	Action
9	To exit the Directory List, press the **DIRECTORY** key again, the **GOODBYE** key, or the **QUIT** key (depending on your specific phone model).
From the 57i CT handset:	
10	a Press the **Public/Private** softkeys to toggle between making the new entry public or private. **Note:** The entry is set to **Private** by default. If the entry is made **Public**, the entry is sent to the handsets. A 57i CT accepts a maximum of 50 entries with the public attribute. b 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. c To dial a displayed entry from the Directory List, pick up the handset, press the / handsfree key, or press the **Dial** 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.

Use the following procedure to configure the download.

Configuration Files

For specific parameters you can set in the configuration files for downloading the Directory List, see Appendix A, the section, "Directory Settings" on page A-85.

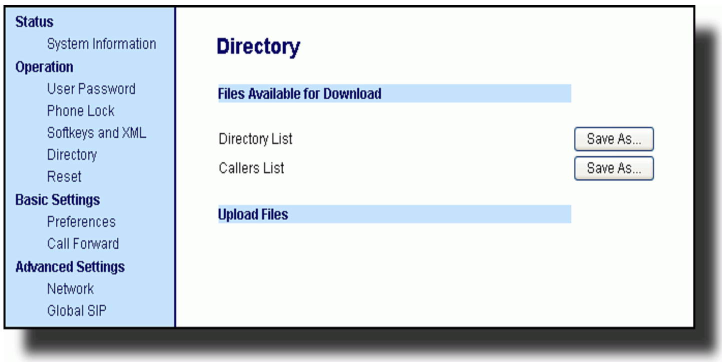
Downloading from the IP Phone to the Server

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.

Use the following procedure to configure the download.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Operation->Directory.



</td></tr>
<tr><td>2</td><td>In the Directory List field, click on Save As... .

A “File Download” message displays.</td></tr>
<tr><td>3</td><td>Click Save .</td></tr>
<tr><td>4</td><td>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.</td></tr>
<tr><td>5</td><td>Use a spreadsheet application to open and view the Directory List.</td></tr>
</table>

Voicemail (55i, 57i, and 57i CT only)

The Voicemail feature on the 55i, 57i and 57i 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 55i, 57i, and 57i 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 (55i, 57i, and 57i 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 57i/57i 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.

Use the following procedure to configure voicemail using the configuration files.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Voicemail Settings" on page A-84.

Using Voicemail (57i57i CT only)

Use the following procedure to access and use voicemail.

IP Phone UI

Step	Action
For the 57i/57i CT:	
1	Press **Services** on the phone to display the **Services** menu.
2	From the **Services** menu, 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

Creating Customized XML Services on 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 the following proprietary objects that allow for the customization of the IP phone's display.

XML Object	Description
AastraIPPhoneTextMenu (for Menu screens)	Creates a numerical list of menu items on the IP phones.
AastraIPPhoneTextScreen (for Text screens)	Creates a screen of text that wraps appropriately.
AastraIPPhoneFormattedTextScreen (for Text screens)	Creates a formatted screen of text (specifies text alignment, text size, text static or scrolling)
AastraIPPhoneInputScreen (for User Input screens)	Creates screens for which the user can input text where applicable.
AastraIPPhoneInputScreen Time and Date Attributes (for User Input screens)	Allows you to specify US (HH:MM:SS am/pm and MM/DD/YYYY) or International (HH:MM:SS and DD/MM/YYYY) time/date formats for an XML user input screen.

XML Object	Description
AastraIPPhoneDirectory (for Directory List screen)	Creates an online Directory List that a user can browse in real-time.
AastraIPPhoneStatus (for Idle screen)	Creates a screen that displays status messages when applicable.
AastraIPPhoneExecute (for executing XML commands)	Allows the phone to execute commands (i.e., “reset”, “NoOp”, etc.) using XML .
AastraIPPhoneConfiguration (for pushing a configuration to the phone)	Allows the server to push a configuration to the phone.(See page 201 for more information).
AastraIPPhoneImageScreen (Standard Bitmap Image)	Creates a display with a single bitmap image according to alignment, height, and width specifications.
AastraIPPhoneImageMenu (Menu Image)	Creates a display with a bitmap image as a menu. Menu selections are linked to keypad keys (0-9, *, #).
AastraIPPhoneTextMenu (Icon Menu) (Icon Menu Image)	Creates a display that has a small icon before each item in the menu.

For more information about creating customized XML applications, see Appendix G, “Creating an XML Application.”

You can also use the following attributes/options with the XML objects to further customize your XML applications:

Attribute/Option	Description/Usage	Valid Values
Beep	Enables or disables a BEEP option to indicate a status on the phone. **Use with:** XML object (See Appendix G) Configuration files (See page 200) Aastra Web UI (See page 200)	yes | no Default = no **Note**: This value is case sensitive.
xml status scroll delay (config files) **Status Scroll Delay (seconds)** (Web UI)	Allows you to set the time delay, in seconds, between the scrolling of each status message on the phone. **Use with:** Configuration files (See page 201) Aastra Web UI (See page 201)	1 to 25 Default = 5
Timeout	Specifies a timeout value for the LCD screen display. **Use with:** XML object (See Appendix G)	0, 30, 45, 60 Default =45
XML Get Timeout	Specifies a timeout value, in seconds, that the phone waits for the far side to return a response after accepting the HTTP GET connection. **Use with:** Configuration Files (See page 202)	0 to 214748364 seconds Default =0 (never timeout)

Attribute/Option	Description/Usage	Valid Values
LockIn	Specifies whether or not the information on the LCD screen stays displayed when other events occur (such as pressing buttons on the keypad). **Use with:** XML object (See Appendix G)	yes | no Default = no
CancelAction	Specifies a URI that the phone executes a GET on when the user presses the default CANCEL key. **Use with:** XML object (See Appendix G)	Fully qualified URI **For example**: `cancelAction= http:// 10.50.10.117/ ft.xml`

Enabling/Disabling a Beep for Status Message Displays

You can enable or disable a BEEP option using the Status Message object (AastraIPPhoneStatus), the configuration files, or the Aastra Web UI.

Note: For enabling/disabling a status message beep using the Status Message object, see Appendix G, "Creating an XML Application."

When the phone receives a status message, the BEEP notifies the user that the message is displaying.

You can use the following to enable/disable a status message beep:

- **AastraIPPhoneStatus** object (via XML object; see Appendix G)
- **xml beep notification** (via configuration files)
- **XML Beep Support** (via the Aastra Web UI)

Enabling the beep is an indication to the phone to sound a beep when it receives an AastraIPPhoneStatus object. If you disable the beep, or no AastraIPPhoneStatus object appears in the status message, then the default behavior is no beep is heard when the object arrives to the phone.

The value set in the configuration files and Aastra Web UI override the attribute you specify for the AastraIPPhoneStatus object.

For example, if the AastraIPPhoneStatus object has the attribute of **Beep="yes"**, and you uncheck (disable) the "**XML Beep Support**" in the Aastra Web UI, the phone does not beep when it receives an AastraIPPhoneStatus object.

Setting the BEEP option in the configuration files and the Aastra Web UI applies to the phone immediately.

Reference

For information about enabling/disabling the XML beep in the Aastra Web UI, see "XML Beep Support" on page 5-48.

Scroll Delay Option for Status Messages

The IP phones support a scroll delay option that allows you to set the time delay, in seconds, between the scrolling of each status message on the phone. The default time is 5 seconds for each message to display before scrolling to the next message. You can configure this option via the configuration files or the Aastra Web UI.

You can use the following to set the scroll delay for status messages:

- **xml status scroll delay** (via the configuration files)
- **Status Scroll Delay (seconds)** (via the Aastra Web UI)

Changes apply to the phone immediately.

Reference

For more information about configuring status scroll delay, see "Status Scroll Delay" on page 5-50.

XML Configuration Push from the Server

The IP phones provide an XML feature that allows you to make configuration changes to the phone that take affect immediately, without having to reboot the phone. This feature involves creating XML scripts that push the changed configuration parameter(s) from the server to the IP phones.

You can use the **AastraIPPhoneConfiguration** object in the XML scripts to change configuration parameters or configure new parameters. However, since the IP phone does not save **new** parameters created in XML scripts to the *local.cfg* file, when the phone reboots, it does not save the new parameters on the phone. In order for the phone to apply **new** configuration parameters, you have to enter the parameters via the user interfaces (Telephone User Interface, Web User Interface, or configuration files), or reapply the new parameters using the XML scripts after every boot.

Specific configuration parameters are dynamic on the phone when pushed from XML scripts on the server. See Appendix G, "Creating an XML Application" for more information about XML configuration scripts and dynamic configuration parameters.

For more information about creating XML configuration scripts and for XML script examples, see Appendix G, "Creating an XML Application".

Configuring the Phone to use XML

You can configure the phone to request the XML objects you create by configuring specific parameters via the configuration files or the Aastra Web UI.

Users can access XML applications via softkeys configured on the IP phones. The phone performs an HTTP GET on the URI configured in the Aastra Web UI or configuration 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 Get Timeout

The IP phone has a parameter called, "**xml get timeout**" that allows you to specify a timeout value, in seconds, that the phone waits for the far side to return a response after accepting the HTTP GET connection. If the far side accepts the GET connection but never returns a response, it blocks the phone until it is rebooted. If you enter a value greater than 0 for this parameter, the phone times out and will not be blocked.

For more information about configuring this parameter, see Appendix A, the section, "XML Settings" on page A-89.

XML Push Requests

In addition to initiating a request to an XML application from a softkey, 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 about adding "xml" lines in HTTP packets, see Appendix G, "Creating an XML Application.".

Example Configuration of XML application

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
```

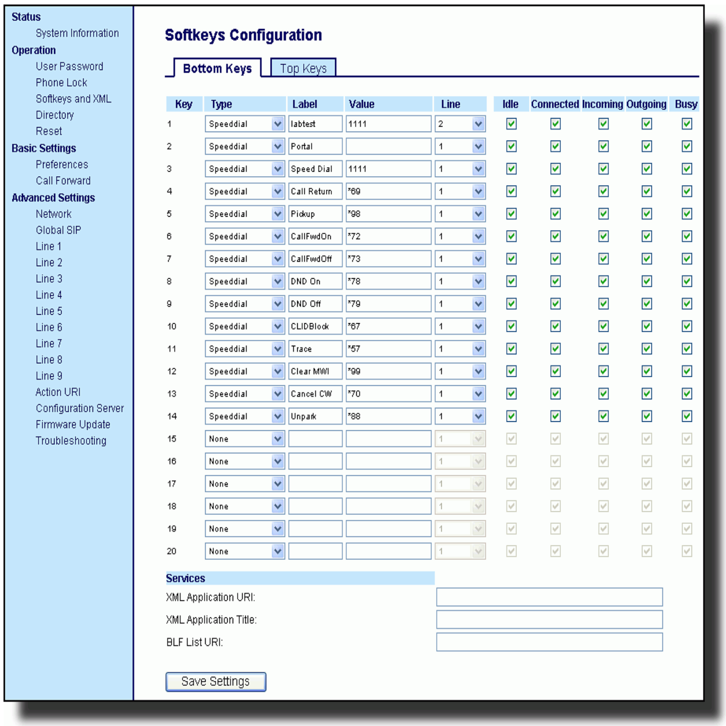
Configuring for XML on the IP Phone

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 Appendix A, the section, "XML Settings" on page A-89.

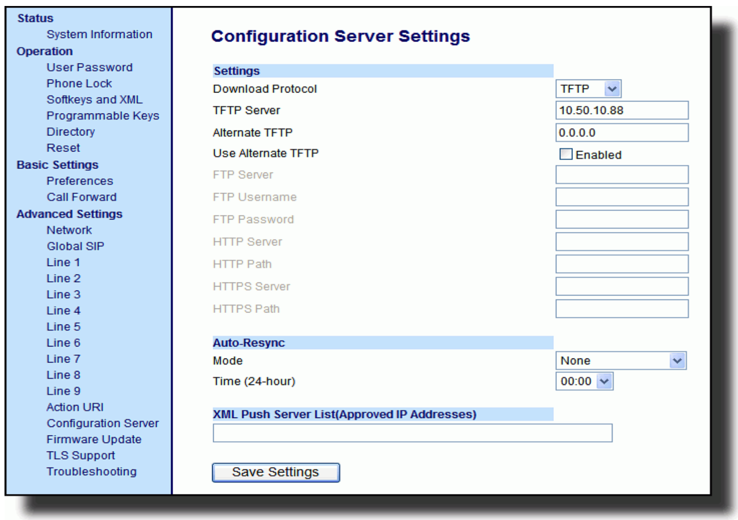
Aastra Web UI

For the 57i/57i 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 "**Value**" field, enter the IP address or qualified domain name of the XML application.
6	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, you could enter an XML application called "`http://172.16.96.63/aastra/internet.php`" in the applicable field.
7	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**".

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>8</td><td>Click Save Settings to save your changes.

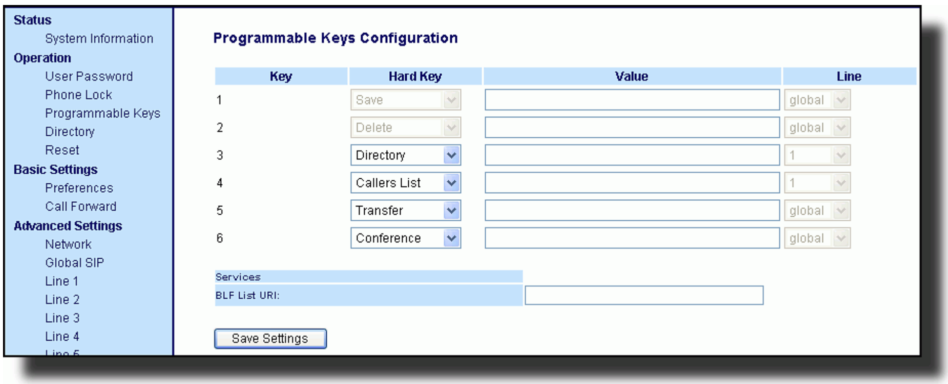
The XML application is applied to the IP phone immediately.

When the XML application is pushed to the phone via an HTTP POST, a host IP address or domain name server is required.</td></tr>
<tr><td>9</td><td>Click on Advanced Settings->Configuration Server.

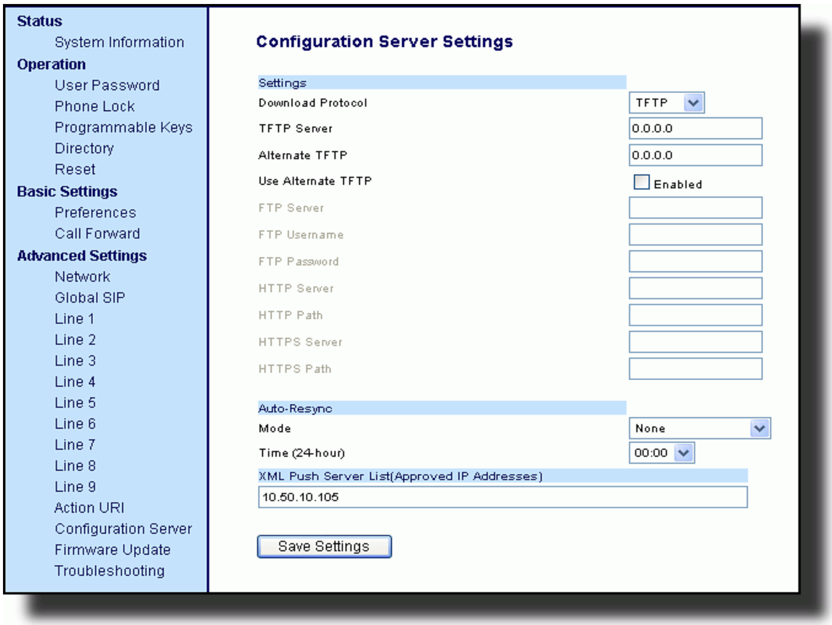


</td></tr>
<tr><td>10</td><td>In the "XML Push Server List (Approved IP Addresses)" field, enter the host IP address and/or domain name server. You can enter multiple IP address and/or domain name servers (separated by commas). In the example in Step 8, the illustration shows a host IP address of "10.50.10.53, 10.50.10.54" in the applicable field.</td></tr>
<tr><td>11</td><td>Click Save Settings to save your changes.

Note: No posting is performed if a session times out.</td></tr>
</table>

Aastra Web UI	
For the 53i	
1	Click on **Operation->Programmable Keys.** **53i Screen** 
2	For the 53i, select a key from keys 3 through 6.
3	In the "**Hard Key**" field, select **XML** from the list box.
4	In the "**Value**" field, enter the IP address or qualified domain name of the XML application.
5	Click Save Settings to save your changes. The XML application is applied to the IP phone immediately. When the XML application is pushed to the phone via an HTTP POST, a host IP address or domain name server is required.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>6</td><td>Click on Advanced Settings->Configuration Server.



</td></tr>
<tr><td>7</td><td>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). In the example in Step 6, the illustration shows a host IP address of "10.50.10.53, 10.50.10.54" in the applicable field.</td></tr>
<tr><td>8</td><td>Click 

to save your changes.

Note: No posting is performed if a session times out.</td></tr>
</table>

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

Step	Action
For the 55i/57i/57i CT:	
1	Press the **Services** key 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 "**Custom Features**" screen, press **Exit**.

IP Phone UI	
Step	**Action**
For the 53i:	
1	Press the programmable key configured on the phone for XML services. A "**Custom Features**" screen displays.
2	Use the and to scroll through the customized features.
3	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 "**Custom Features**" screen, press the XML programmable key again.

XML Action URIs

The IP phones have a feature that allows an administrator to specify a uniform resource identifier (URI) that triggers a GET when certain events occur. The IP phone events that support this feature are:

- Startup
- Successful registration
- Incoming call
- Outgoing call
- Offhook
- Onhook

The following table identifies the configurable action URI parameters in the configuration files and the Aastra Web UI. This table also identifies the variables that apply to specific parameters.

Configuration File Parameters	Aastra Web UI Parameters at Advanced Settings->Action URI	Applicable Variables
action uri startup	Startup	-
action uri registered	Successful Registration	$$SIPUSERNAME$$ $$SIPAUTHNAME$$ $$PROXYURL$$
action uri incoming	Incoming Call	$$REMOTENUMBER$$ $$DISPLAYNAME$$ $$SIPUSERNAME$$ $$INCOMINGNAME$$
action uri outgoing	Outgoing Call	$$REMOTENUMBER$$ $$SIPUSERNAME$$
action uri offhook	Offhook	-
action uri onhook	Onhook	-

How it works

When a startup, successful registration, incoming call, outgoing call, offhook, or onhook call event occurs on the phone, the phone checks to see if the event has an action URI configured. If the phones finds a URI configured, any variables configured (in the form $$VARIABLENAME$$) are replaced with the value of the appropriate variable. After all of the variables are bound, the phone executes a GET on the URI. The Action URI binds all variables and is not dependant on the state of the phone.

For example, if you enter the following string for the **action uri outgoing** parameter:

```
action uri outgoing: http://10.50.10.140/
outgoing.pl?number=$$REMOTENUMBER$$
```

and you dial out the number 5551212, the phone executes a GET on:

```
http://10.50.10.140/outgoing.pl?number=5551212
```

Note: If the phone can't find the Action URI you specify, it returns a "NULL" response. For example,
`http://10.50.10.140/outgoing.pl?number=`

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

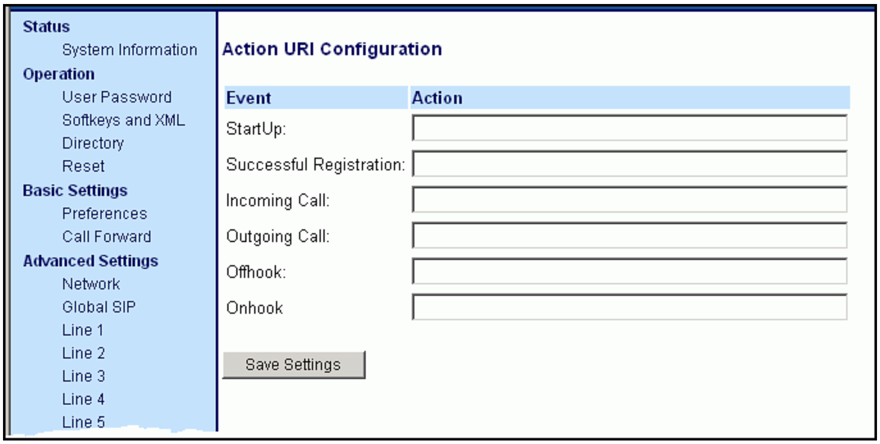
Configuring XML Action URIs

Use the following procedures to configure XML Action URIs using the configuration files or the Aastra Web UI.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Action URI Settings" on page A-92.

	Aastra Web UI
1	Click on **Advanced Settings->Action URI.**



2	Enter an XML URI for a startup event in the “**Startup**” field. For example: `http://10.50.10.140/startup` This parameter specifies the URI for which the phone executes a GET on when a startup event occurs.
3	Enter an XML URI for a successful registration in the “**Successful Registration**” field. For example: `http://10.50.10.14/registered.php?auth name=$$SIPAUTHNAME$$` This parameter specifies the URI for which the phone executes a GET on when a successful registration event occurs. **Note:** For a successful registration event, you can use the following variables in the URI: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ The “**Successful Registration**” parameter executes on the first successful registration of each unique line configured on the phone.

	Aastra Web UI
4	Enter an XML URI for an incoming call event in the "**Incoming Call**" field. For example: `http://10.50.10.140/incoming.php?number=$$REMOTENUMBER$$` This parameter specifies the URI for which the phone executes a GET on when an incoming call event occurs. **Note:** For an incoming call event, you can use the following variables in the URI: • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$SIPUSERNAME$$ • $$INCOMINGNAME$$
5	Enter an XML URI for an outgoing call event in the "**Outgoing Call**" field. For example: `http://10.50.10.140/outgoing.php?number=$$REMOTENUMBER$$` This parameter specifies the URI for which the phone executes a GET on when an outgoing call event occurs. **Note:** For an outgoing call event, you can use the following variables in the URI: • $$REMOTENUMBER$$ • $$SIPUSERNAME$$
6	Enter an XML URI for an offhook event in the "**Offhook**" field. For example: `http://10.50.10.140/offhook` This parameter specifies the URI for which the phone executes a GET on when an offhook event occurs.
7	Enter an XML URI for an onhook event in the "**Onhook**" field. For example: `http://10.50.10.140/onhook` This parameter specifies the URI for which the phone executes a GET on when an onhook event occurs.
8	Click **Save Settings** to save your changes.

XML Softkey URI

In addition to specifying variables for the Action URIs, you can also specify variables in the XML softkey URIs that are bound when the key is pressed. These variables are the same as those used in the Action URIs.

When an administrator enters an XML softkey URI either via the Aastra Web UI or the configuration files, they can specify the following variables:

- $$SIPUSERNAME$$
- $$SIPAUTHNAME$$
- $$PROXYURL$$
- $$REMOTENUMBER$$
- $$DISPLAYNAME$$
- $$INCOMINGNAME$$

When the softkey is pressed, if the phone finds a URI configured with variables (in the form $$VARIABLENAME$$), they are replaced with the value of the appropriate variable. After all of the variables are bound, the softkey executes a GET on the URI.

Example

For example, if the administrator specifies an XML softkey with the value:

```
http://10.50.10.140/script.pl?name=$$SIPUSERNAME$$
```

This softkey executes a GET on:

```
http://10.50.10.140/script.pl?name=42512
```

assuming that the sip username of the specific line is 42512.

You can configure the XML softkey URI variables via the configuration files or the Aastra Web UI.

Configuring XML Softkey URIs

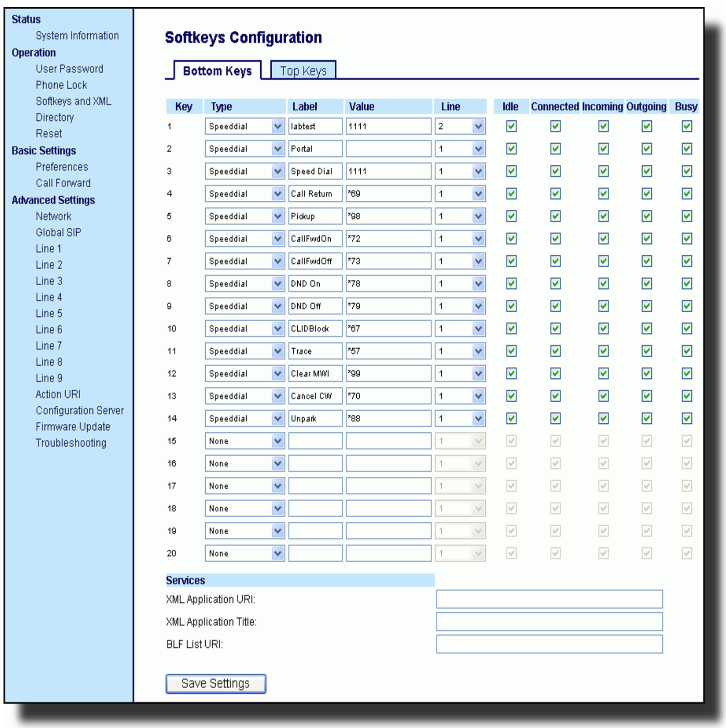
Use the following procedures to configure XML Softkey URIs using the configuration files or the Aastra Web UI.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-136.

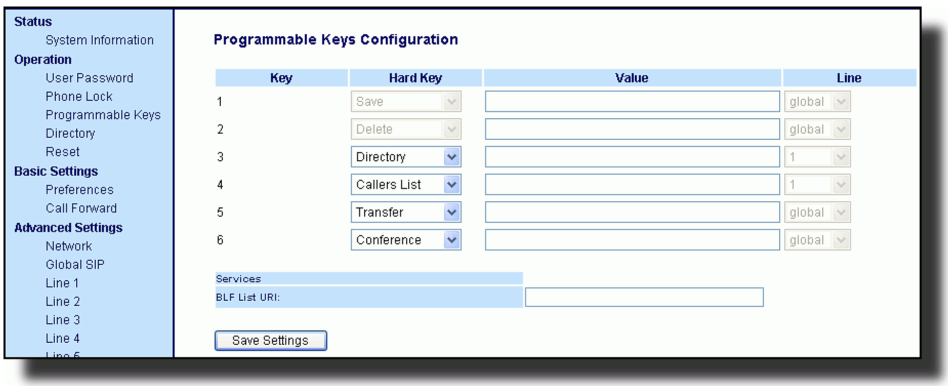
Aastra Web UI

For the 57i/57i CT:

1	Click on **Operation->Softkeys and XML**. 
2	Select a key from keys 1 through 20.

	Aastra Web UI
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 XML softkey. For example, "**aastra**".
5	In the "**Value**" field, enter the URI that the phone performs a GET on when the key is pressed. For example: `http://10.50.10.140/script.pl?name=$$SIPUSERNAME$$` **Note:** You can use the following variables in the URI: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$INCOMINGNAME$$
6	Click **Save Settings** to save your changes.

Aastra Web UI

For the 53i:	
1	Click on **Operation->Programmable Keys.** **53i Screen** 
2	On the 53i, select a key from keys 1 through 6.
3	In the "**Type**" field, select **XML** from the list box.
4	On the 53i, in the "**Label**" field, enter a label that displays on the IP phone for the XML softkey. For example, "**aastra**".
5	In the "**Value**" field, enter the URI that the phone performs a GET on when the key is pressed. For example: `http://10.50.10.140/script.pl?name=$$SIPUSERNAME$$` **Note:** You can use the following variables in the URI: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$INCOMINGNAME$$
6	Click **Save Settings** to save your changes.

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.

The following table describes each parameter.

Parameter	Description
Headset tx gain	The increased (+db) or decreased (-db) amount of signal transmitted from the headset microphone to t he far-end party.
Headset sidetone gain	The increased (+db) or decreased (-db) amount of sidetone signal from the headset microphone to the headset speaker.
Handset tx gain	The increased (+db) or decreased (-db) amount of signal transmitted from the handset microphone to the far-end party.
Handset sidetone gain	The increased (+db) or decreased (-db) amount of sidetone signal from the handset microphone to the handset speaker.

(continued)

Parameter	Description
Handsfree tx gain	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 **d/f** key (handsfree key) works. Audio mode has 4 options: **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 d /fkey. 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 d /fkey. **2 (Speaker/Headset)** - Incoming calls are sent to the speakerphone. By pressing the d /fkey, 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 d /fkey, 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 Appendix A, the section, "Audio Transmit and Receive Gain Adjustment Settings" on page A-125.

Centralized Conferencing (for Sylantro and Broadsoft Servers)

The IP phones include support for centralized conferencing (Ad-Hoc conferencing) for Sylantro and Broadsoft servers. This feature provides centralized conferencing on the SIP server (versus localized, on the phone) and allows IP phone users to do these tasks:

- Conference two active calls together into a conference call.
- When on an active conference call, invite another party into the call.
- Create simultaneous conference calls on the same IP phone (Sylantro servers only). For example, the IP phone user at extension 2005 could create these two conferences, and put one conference on hold while conversing with the other party:
  - Line 1: conference together extensions 2005, 2010, and 2020.
  - Line 2: conference together extensions 2005, 2011 and 2021.

  When an IP phone user is connected to multiple conference calls, some outbound proxies have maximum call "hold" time set from 30-90 seconds. After this time, the call that is on hold is disconnected.
- Disconnect from an active conference call while allowing the other callers to remain connected.
- Ability to create N-way conference.
- Join two active calls together into a conference call.
- Incoming or outgoing active call can join any of the existing conferences.

If the administrator does not configure centralized conferencing, then the IP phone uses localized conferencing by default.

➡ **Note:** When you configure centralized conferencing globally for an IP Phone, the global settings apply to all lines. Although, for the global setting to work on soft lines, the user must configure the lines with the applicable phone number.

An Administrator can configure centralized conferencing on a global or per-line basis using the configuration files or the Aastra Web UI.

To use the centralized conferencing after it is enabled, see your Model-specific IP Phone *User Guide*.

Configuring Centralized Conferencing Using the Configuration Files

You use the following parameters to configure centralized conferencing in the configuration files:

Global Parameter

- `sip centralized conf`

Per-Line Parameter

`sip lineN centralized conf`

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Centralized Conferencing Settings" on page A-60.

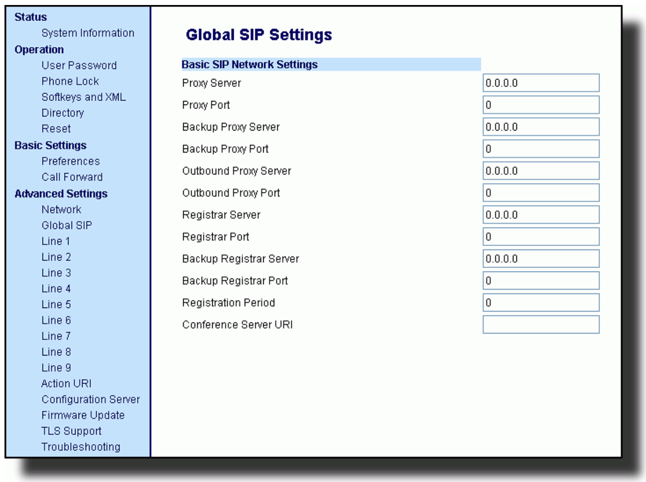
Configuring Centralized Conferencing Using the Aastra Web UI

Use the following procedure to configure centralized conferencing using the Aastra Web UI.

Aastra Web UI

Global Configuration

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



2 In the "**Conference Server URI**" field, do one of the following actions:

- To disable centralized conferencing on the IP phone, leave this field empty (blank).
- To enable SIP centralized conferencing on the IP phone, do one of the following actions:
  - — If you have specified a proxy server/registrar server, then to reach the media server via the proxy server, set this field to one of the following:

    conf (Sylantro server), or

    Conference (Broadsoft server)

    By setting this field to **conf** or **Conference**, you specify conf@<proxy_server _address>: <proxy_port>. For example, if the proxy server address is 206.229.26.60 and the port used is 10060, then by setting this parameter to **conf**, you are specifying the following: conf@206.229.26.60:10060.
  - — To reach the media server using a different address/port than that specified by the proxy, set this field to the following:

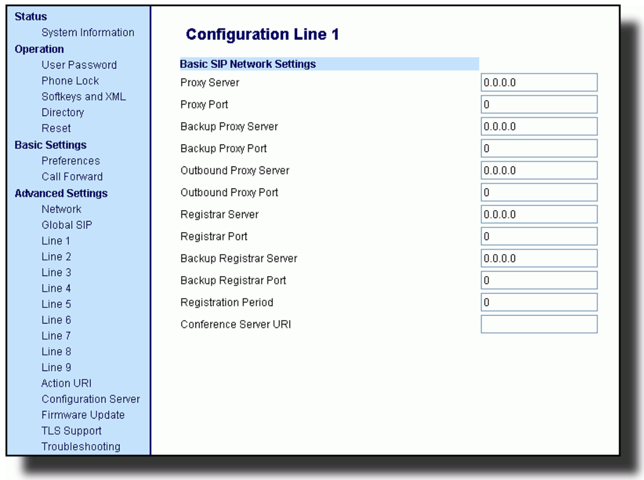
    conf@<media_server _address>: <media_port>

Aastra Web UI

3 Click **Save Settings** to save your changes.

Per-Line Configuration

1 Click on **Advanced Settings->Line <#>->Basic SIP Network Settings**



2 In the "**Conference Server URI**" field, do one of the following actions:

- To disable centralized conferencing on this line, leave this field empty (blank).
- To enable SIP centralized conferencing on this line, do one of the following actions:
  - — If you have specified a proxy server/registrar server, then to reach the media server via the proxy server, set this field to one of the following:

    conf (Sylantro server), or

    Conference (Broadsoft server)

    By setting this field to **conf** or **Conference**, you specify conf@<proxy_server _address>:<proxy_port>.
  - — To reach the media server using a different address/port than that specified by the proxy, set this field to the following:

    conf@<media_server _address>: <media_port>

3 Click **Save Settings** to save your changes.

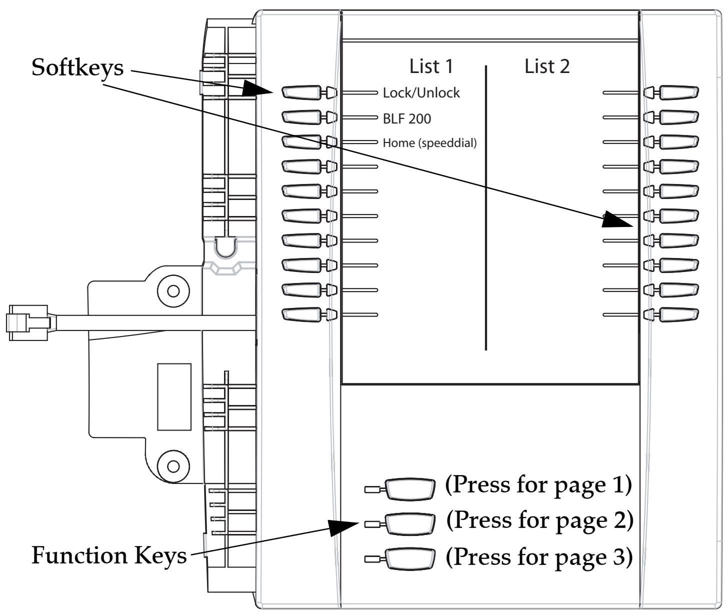
Customizing the Display Columns on the 560M Expansion Module

The 560M Expansion Module screen displays softkeys in column format. The function keys on the bottom left of the Module allow you to display 3 full screens of softkeys. Each screen consists of 2 columns with the following default headings on each page:

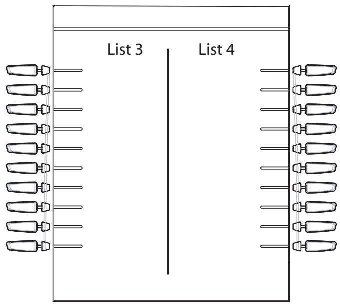
Page 1 "List 1" and "List 2"

Page 2 "List 3" and "List 4"

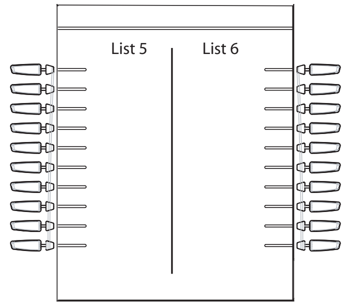
Page 3 "List 5" and "List 6"



Page 1 Screen



Page 2 Screen



Page 3 Screen

To use the 560M, press the function key for the page you want to display to the LCD (page 1, page2, or page 3), and press the applicable softkey.

You can customize the headings on each 560M Expansion Module screen using the configuration files. You use the following parameters to customize the column headings:

Expansion Module 1 (3 pages)

- expmod1page1left
- expmod1page1right
- expmod1page2left
- expmod1page2right
- expmod1page3left
- expmod1page3right

Expansion Module 2 (3 pages)

- expmod2page1left
- expmod2page1right
- expmod2page2left
- expmod2page2right
- expmod2page3left
- expmod2page3right

Expansion Module 3 (3 pages)

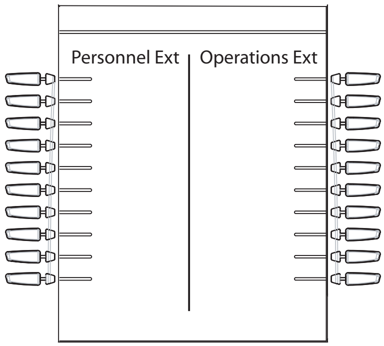
- expmod3page1left
- expmod3page1right
- expmod3page2left
- expmod3page2right
- expmod3page3left
- expmod3page3right

Example

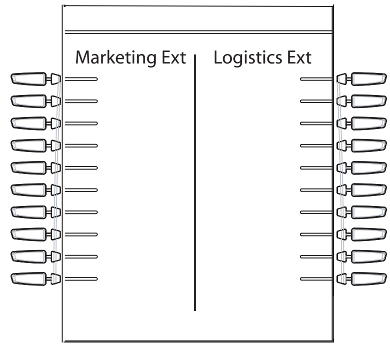
The following is an example of configuring Expansion Module 1 column headings.

```
expmod1page1left: Personnel Ext
expmod1page1right: Operations Ext
expmod1page2left: Marketing Ext
expmod1page2right: Logistics Ext
expmod1page3left: Engineering Ext
expmod1page3right: Shipping Ext
```

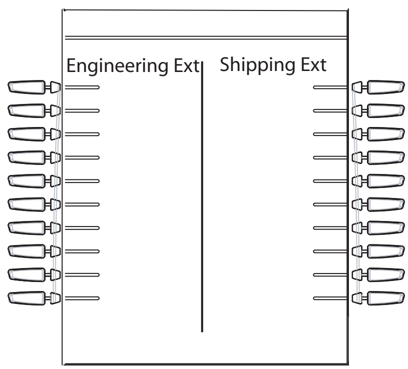
These settings display to the Expansion Module as shown in the following illustrations.



Page 1 Screen



Page 2 Screen



Page 3 Screen

Cuztomizing the 560M Expansion Module Column Display.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Customizing 560M Expansion Module Column Display" on page A-166.

Chapter 6
Configuring Advanced Operational Features

About this chapter

Introduction

The IP phones have advanced operational features you can configure using the configuration files and/or the Aastra Web UI.

This chapter describes each of these features and provides procedures for configuring each feature.

Topics

This chapter covers the following topics:

Advanced Operational Features

Description

This section provides the following information about advanced features of the IP phones:

Feature	Description
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.
Auto-discovery Using mDNS	The IP phones automatically perform an auto-discovery of all servers on a network using mDNS. When the IP phone discovers a TFTP server, it is automatically configured by that TFTP server.
Single Call Restriction (57i CT only)	Allows you to enable or disable a single call restriction between the 57i CT base unit and a call server.
Missed Call Summary Subscription	Allows missed calls that have been redirected by the server, to be incremented in the missed calls indicator on the phone it was initially directed to.
Blacklist Duration (Broadsoft Servers)	Allows you to specifies the length of time, in seconds, that a failed server remains on the server blacklist. The IP phone avoids sending a SIP message to a failed server (if another server is available) for this amount of time.
Whitelist Proxy	Allows you to configure the phone to either accept or reject call requests from a trusted proxy server.

Feature	Description
Transport Layer Security (TLS)	Allows you to enable or disable the use of Persistent Transport Layer Security (TLS). Persistent TLS sets up the connection to the server once and re-uses that connection for all calls from the phone. The setup connection for Persistent TLS is established during the registration of the phone. If the phones are set to use Persistent TLS, and a call is made from the phone, this call and all subsequent calls use the same authenticated connection. This significantly reduces the delay time when placing a call.
Symmetric UDP Signaling	Allows you to enable or disable the phone to use port 5060 to send SIP UDP messages.
Removing UserAgent and Server SIP Headers	Allows you to enable or disable the addition of the User-Agent and Server SIP headers in the SIP stack.
Multi-Stage Digit Collection (Billing Codes) Support (for Sylantro Servers)	IP Phones support Sylantro Server features, like mandatory and optional billing codes that require the application server to notify the phone to collect more digits before completing the call. The IP phone is able to collect digits in two stages to support the billing code feature.

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.

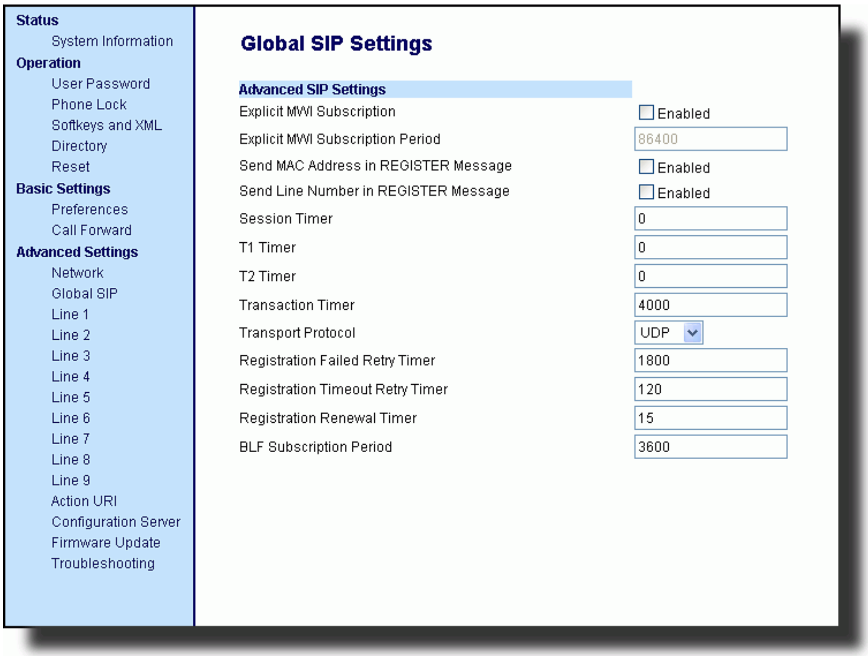
Configuring the MAC address/Line Number in REGISTER Message

Configuration Files

For specific parameters you can set in the configuration files for enabling/disabling MAC address and line number, see Appendix A, the section, "Advanced Operational Parameters" on page A-172.

.

Aastra Web UI

Step	Action
1	Click on **Advanced Settings->Global SIP->Advanced SIP Setting.** 
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. **Note:** The session prompts you to restart the IP phone to apply the configuration settings.
5	Select **Operation->Reset** and click Restart .

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.

Configuring SIP Message Sequence for Blind Transfer

Configuration Files

For the specific parameter you can set in the configuration files for enabling/disabling the blind transfer method, see Appendix A, the section, "Blind Transfer Setting" on page A-172.

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.

Configuring Update Caller ID During a Call

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 Appendix A, the section, "Update Caller ID Setting." on page A-173.

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

Configuration Files

For the specific parameters you can set in the configuration files for boot sequence recovery mode, see Appendix A, the section, "Boot Sequence Recovery Mode." on page A-173.

Auto-discovery Using mDNS

The IP phones can perform an auto-discovery of all servers on a network using mDNS. When the IP phone discovers a TFTP server, it is automatically configured by that TFTP server.

An unconfigured phone (phone right out of the box) added to a network, attempts to auto-discover a configuration server on the network without any end-user intervention. When it receives DHCP option 66 (TFTP server), it automatically gets configured by the TFTP server.

An already configured phone (either previously configured by auto-discovery or manually configured) added to a network, uses its predefined configuration to boot up.

Notes:

1. Configuration parameters received via DHCP do not constitute configuration information, with the exception of a TFTP server. Therefore, you can plug a phone into a DHCP environment, still use the auto-discovery process, and still allow the use of the TFTP server parameter to set the configuration server.

2. DHCP option 66 (TFTP server details) overrides the mDNS phase of the auto-discovery. Therefore, the DHCP option takes priority and the remaining process of auto-discovery continues.

3. As the phone performs auto-discovery, all servers in the network (including the TFTP server), display in the phone window. However, only the server configured for TFTP automatically configures the phone.

Single Call Restriction (57i CT only)

On the 57i CT, an administrator can enable or disable a single call restriction between the 57i CT base unit and a call server.

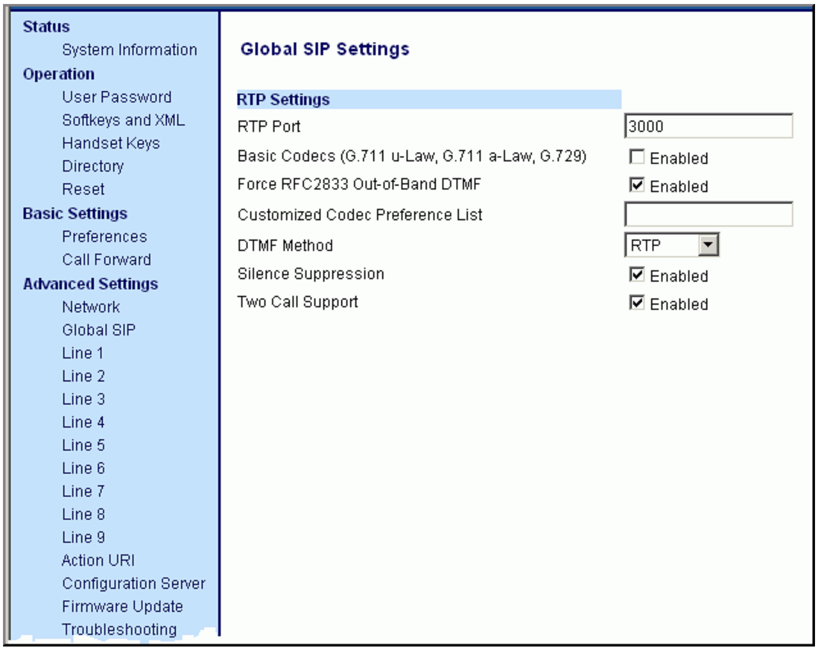
When this feature is enabled (set to 1), you can make separate active calls from the 57i CT base unit and from the cordless handset. If this feature is disabled (set to 0), only one call can be active at a time either from the base unit or from the handset. When this feature is disabled, and you make an active call on either the base unit or the handset, any other attempt to make an active call is put on hold. Also, when this feature is disabled, more than one call can negotiate complex audio codecs since only a single call is decoding audio at a time.

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

Configuring Single Call Restriction.

Configuration Files
For the specific parameters you can set in the configuration files for single call restriction on the 57i CT, see Appendix A, the section, "Single Call Restriction" on page A-174.

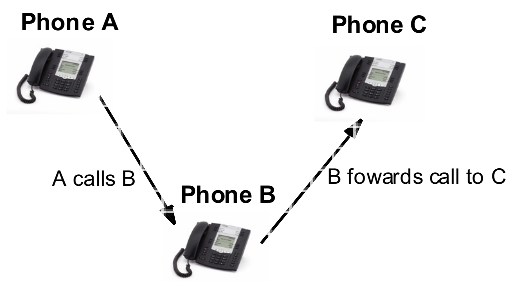
Aastra Web UI

Step	Action
1	Click on **Advanced Settings->Global SIP->RTP Settings.** 
2	Enable the "**Two Call Support**" field by checking the check box. (Disable this field by unchecking the box).
3	Click Save Settings to save your settings. **Note:** The session prompts you to restart the IP phone to apply the configuration settings.
4	Select **Operation->Reset** and click Restart.

Missed Call Summary Subscription

The "**Missed Call Summary Subscription**" feature allows missed calls that have been redirected by the server, to be incremented in the missed calls indicator on the phone it was initially directed to. This feature is called the Missed Call Summary Subscription and can be set with a timer that allows the phone to use the feature for a period of time before the timer expires. For this feature to work, you must configure voicemail on the phone that the call was initially directed to.

For example, phones A, B, and C are connected to the server. You configure the server to direct calls coming into phone B (which has voicemail configured) to be forwarded to phone C. When phone A calls phone B, the server forwards the call to phone C. With the new feature in 2.1, phone B receives notification from the server that the call was forwarded and the missed calls indicator is incremented on phone B.



Missed calls indicator increments on phone B.
Note:Voicemail must be configured on phone B.

An Administrator can configure this feature on a global or per-line basis, using the configuration files or the Aastra Web UI

Configuring Missed Call Summary Subscription using the Configuration Files

In addition to enabling/disabling the Misses Call Summary Subscription, You can also configure the amount of time, in seconds, that the phone uses this feature. The timer is configurable on a global basis only.

You use the following parameters to configure Missed Call Summary Subscription feature on a global basis:

Global Parameters

- **`sip missed call summary subscription`**
- **`sip missed call summary subscription period`**

Use the following parameters to configure Missed Call Summary Subscription feature on a per-line basis:

Per-Line Parameter

- **`sip lineN missed call summary subscription`**

Configuration Files

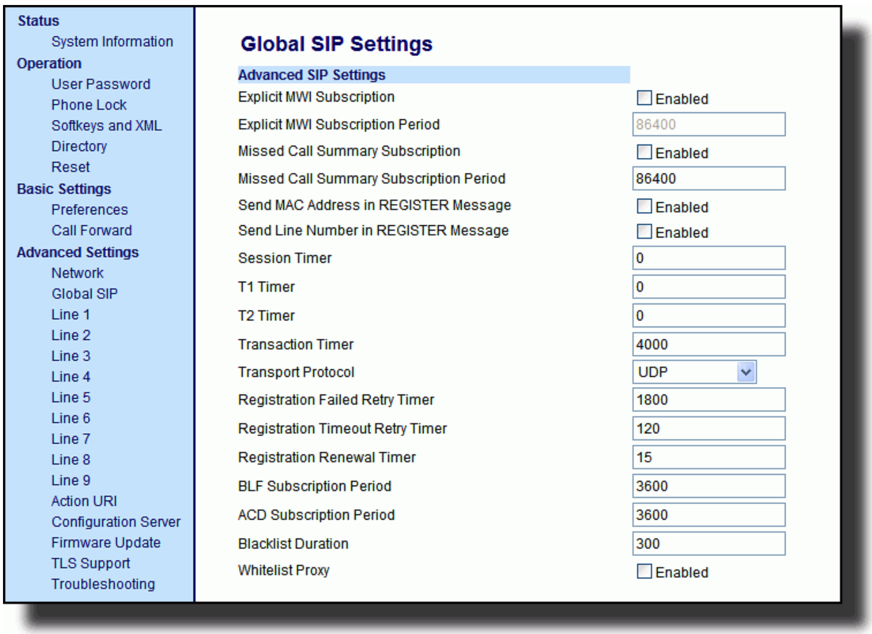
For the specific parameters you can set in the configuration files for Missed Call Summary Subscription, see Appendix A, the section, "Missed Call Summary Subscription Settings" on page A-70.

Configuring Missed Call Summary Subscription using the Aastra Web UI

Use the following procedure to configure the Missed Call Summary Subscription feature using the Aastra Web UI.

Aastra Web UI

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

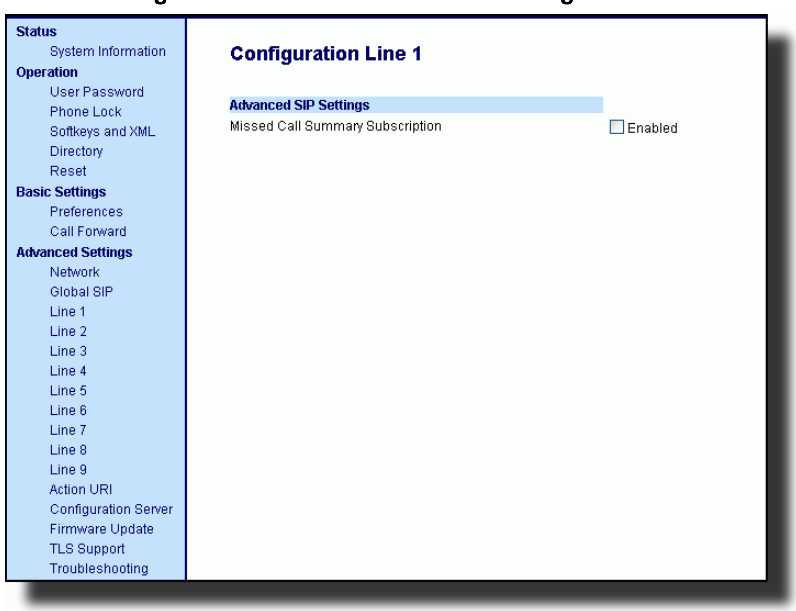


Aastra Web UI

2	The "**Missed Call Summary Subscription**" field is disabled by default. To enable this field, check the box. This feature allows missed calls that have been redirected by the server, to be incremented in the missed calls indicator on the phone it was initially directed to. For example, phones A, B, and C are connected to the server. You configure the server to direct calls coming into phone B (which has voicemail configured) to be forwarded to phone C. When phone A calls **sip missed call summary subscription** parameter, phone B receives notification from the server that the call was forwarded and the missed calls indicator is incremented on phone B. **Note:** You must configure voicemail on the phone that the call was initially directed to (phone B in the above example).
3	The "**Missed Call Summary Subscription Period**" field is enabled with a default value of 86400. To disable this field, enter zero (0), or leave the field blank.
4	Click Save Settings to save your changes.

Per-Line Configuration

1	Click on **Advanced Settings->Line <N>->Advanced SIP Settings.**



2	Click Save Settings to save your changes.

Blacklist Duration (Broadsoft Servers)

The Blacklist Duration feature on Broadsoft Servers helps to reduce unnecessary delays during proxy/registrar server failures, caused by the IP phone repeatedly sending SIP messages to a failed server. If you enable this feature, then whenever the IP phone sends a SIP message to a server, and does not get a response, the phone automatically adds the server to the blacklist. The IP phone avoids sending messages to any servers on the blacklist. If all servers are on the blacklist, then the IP phone attempts to send the message to the first server on the list.

You can specify how long failed servers remain on the blacklist in the IP phone's configuration file or in the Aastra Web UI. The default setting is 300 seconds. If you set the duration to 0 seconds, then you disable the blacklist feature.

Configuring Blacklist Duration Using the Configuration Files

Use the following parameter to configure the Blacklist Duration in the configuration files:

- **`sip blacklist duration`**

Configuration Files

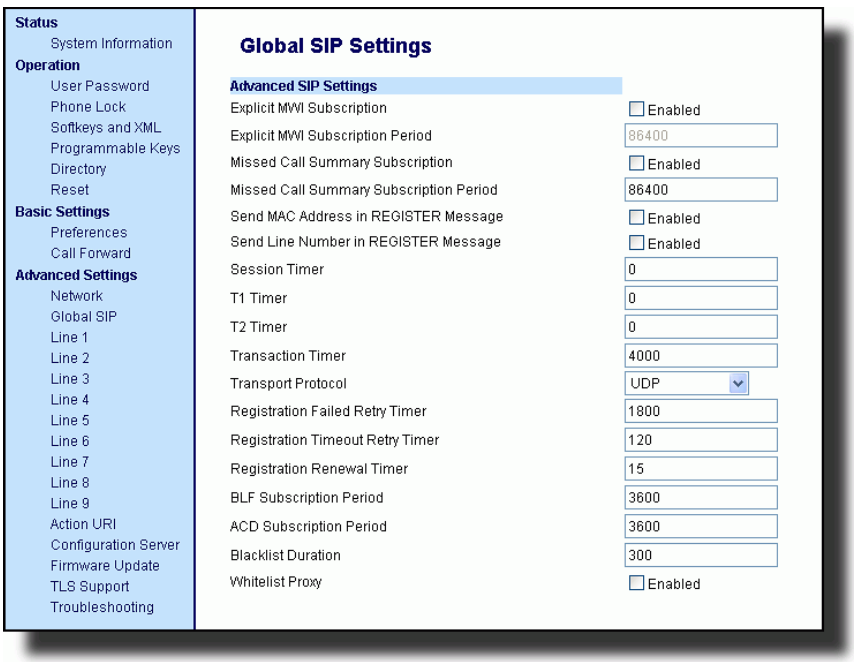
For the specific parameters you can set in the configuration files for setting Blacklist Duration, see Appendix A, the section, "Transport Layer Security (TLS) Settings" on page A-72.

Configuring a Server Blacklist Using the Aastra Web UI

You use the following procedure to configure Blacklist Duration using the Aastra Web UI.

Aastra Web UI

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



2 In the "**Blacklist Duration**" field, specify the length of time, in seconds, that a failed server remains on the server blacklist. The IP phone avoids sending a SIP message to a failed server (if another server is available) for this amount of time. Valid values are 0 to 9999999. Default is 300 seconds (5 minutes).

For example: **600**

Note: The value of "**0**" disables the blacklist feature.

3 Click **Save Settings** to save your changes.

Whitelist Proxy

To protect your IP phone network, you can configure a "**Whitelist Proxy**" feature that screens incoming call requests received by the IP phones. When this feature is enabled, an IP phone accepts call requests from a trusted proxy server *only*. The IP phone rejects any call requests from an untrusted proxy server

Configuring Whitelist Proxy Using the Configuration Files

You use the following parameter to configure the whitelist proxy feature using the configuration files:

- **`sip whitelist`**

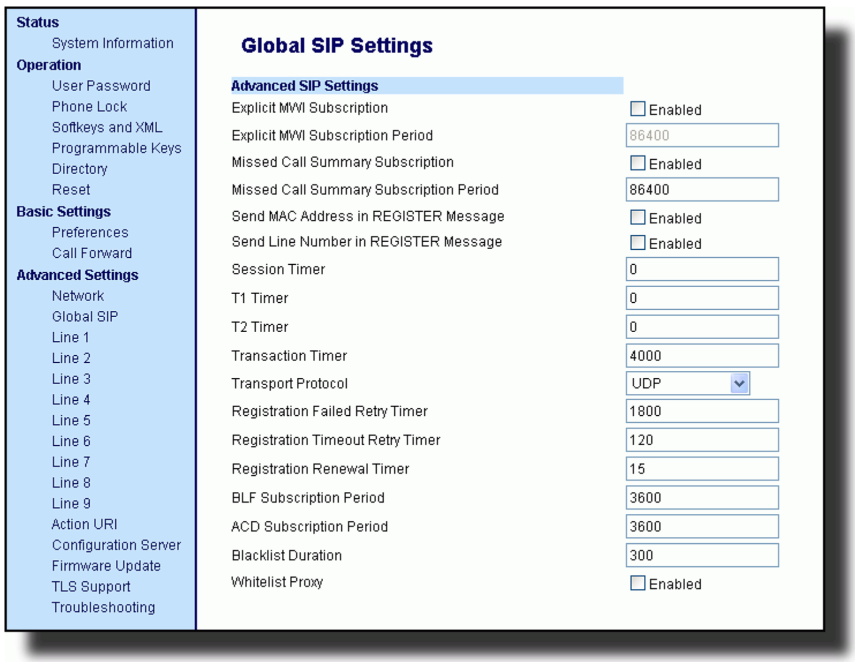
Configuration Files

For the specific parameters you can set in the configuration files for setting Whitelist Proxy, see Appendix A, the section, "Whitelist Proxy" on page A-175.

Configuring Whitelist Proxy Using the Aastra Web UI

Use the following procedure to configure the Whitelist Proxy feature using the Aastra Web UI.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Global SIP->Advanced SIP Settings



</td></tr>
<tr><td>2</td><td>The "Whitelist Proxy" field is disabled by default. To enable this field, check the box.

When this feature is enabled, an IP phone accepts call requests from a trusted proxy server only. The IP phone rejects any call requests from an untrusted proxy server.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your changes.</td></tr>
</table>

Transport Layer Security (TLS)

The IP Phones support a transport protocol called **Transport Layer Security (TLS)** and **Persistent TLS**. TLS is a protocol that ensures communication privacy between the SIP phones and the Internet. TLS ensures that no third party may eavesdrop or tamper with any message.

TLS is composed of two layers: the TLS Record Protocol and the TLS handshake protocol. The TLS Record Protocol provides connection security with some encryption method such as the Data Encryption Standard (DES). The TLS Handshake Protocol allows the server and client to authenticate each other and to negotiate an encryption algorithm and cryptographic keys before data is exchanged. TLS requires the use of specific security certificate files to perform TLS handshake:

- Root and Intermediate Certificates
- Local Certificate
- Private Key
- Trusted Certificate

When the phones use **TLS** to authenticate with the server, each individual call must setup a new TLS connection. This can take more time when placing each call. Thus, the IP phones also have a feature that allows you to setup the connection to the server once and re-use that one connection for all calls from the phone. It is called **Persistent TLS**. The setup connection for Persistent TLS is established during the registration of the phone. If the phones are set to use Persistent TLS, and a call is made from the phone, this call and all subsequent calls use the same authenticated connection. This significantly reduces the delay time when placing a call.

Notes:
1. Persistent TLS requires the **outbound proxy server** and **outbound proxy port** parameters be configured in either the configuration files or the Aastra Web UI (*Advanced Settings->Global SIP->Basic SIP Network Settings*). There can be only one persistent TLS connection created per phone. The phone establishes the TLS connection to the configured outbound proxy.
2. If you configure the phone to use Persistent TLS, you must also specify the Trusted Certificate file to use. The Root and Intermediate Certificates, Local Certificate, and Private Key files are optional.

On the IP phones, an Administrator can configure TLS and Persistent TLS on a global-basis only, using the configuration files or the Aastra Web UI.

SIP Asserted Identity (for Sylantro Servers)

The IP Phones support a private extension to the SIP, Asserted Identity (SAI) within Trusted Networks (as defined in RFC 3325), inside the User Agent Server (UA) in the Aastra IP phones.

This feature allows a network of trusted SIP servers to assert the identity of authenticated users, and verify that phone messages originate from a Trusted Identity. Upon receiving a message from a caller in the Trust Network, the IP phone reads the contents of the P-Asserted-Identity (PAI) header field and displays it on the phone UI. This field contains a more accurate description of the caller identity (extension/phone number) than is contained in the SIP message.

When an IP phone receives an incoming call, the IP phone does the following actions:

- Checks to see if the incoming call is from a registered proxy server.
- If the call is forwarded via a registered proxy server, then the message has already been verified and authenticated by the server. The caller is part of the Trust Network. The IP phone UI displays the caller information contained in the PAI header.
- If the call is not forwarded via a registered proxy server - and therefore is not a “Trusted Entity” - the IP phone UI does not display any trust information contained in the PAI header.

Configuring TLS Using the Configuration Files

You use the following parameters to configure TLS in the configuration files:

- **`sip transport protocol`**
- **`sips persistent tls`**
- **`sips root and intermediate certificates`**
- **`sips local certificate`**
- **`sips private key`**
- **`sips trusted certificates`**

Configuration Files

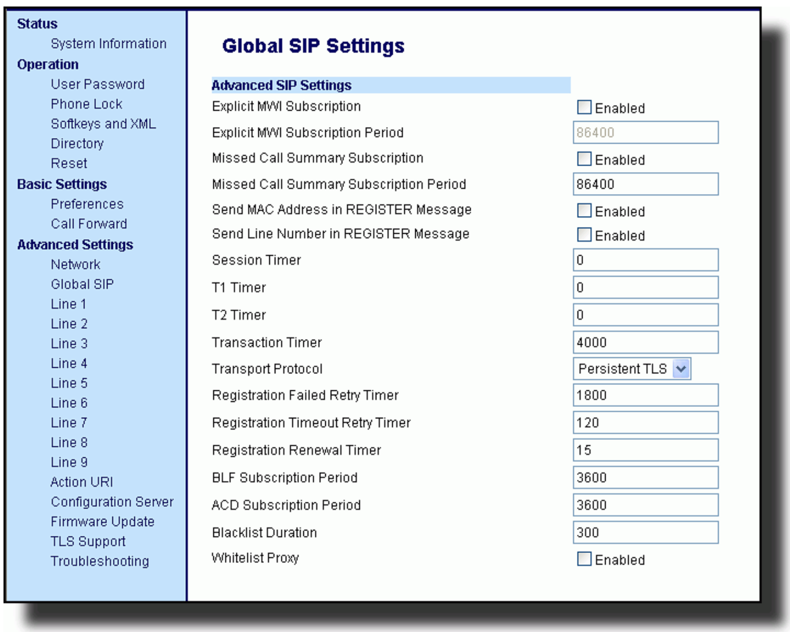
For the specific parameters you can set in the configuration files for setting TLS, see Appendix A, the section, “Transport Layer Security (TLS) Settings” on page A-72.

Configuring TLS Using the Aastra Web UI

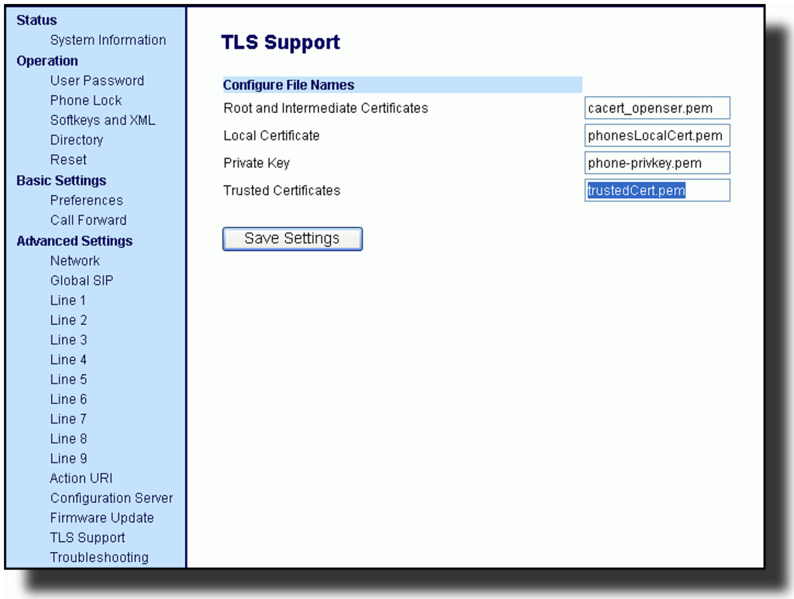
To configure TLS using the Aastra Web UI, you must enable TLS or Persistent TLS first. Then you must define the TLS certificate file names that you want the phone to use. Use the following procedure to configure TLS using the Aastra Web UI.

Aastra Web UI

1	Click on **Advanced Settings->Global SIP->Advanced SIP Settings**.
2	In the "**Transport Protocol**" field, select **TLS** or **Persistent TLS**. **Note:** If configuring **Persistent TLS**, you must go to *Advanced Settings->Global SIP->Basic Network Settings* and configure the "**Outbound Proxy Server**" and "**Outbound Proxy Port**" parameters.
3	Click  Save Settings to save your changes.



Aastra Web UI

4	Click on **Advanced Settings->TLS Support.** 

Advanced Operational Features

	Aastra Web UI
5	Enter the certificate file names and the private key file name in the appropriate fields. The Root and Intermediate Certificate files contain one root certificate and zero or more intermediate certificates which must be placed in order of certificate signing with root certificate being the first in the file. If the local certificate is signed by some well known certificate authority, then that authority provides the user with the Root and Intermediate Certificate files (most likely just CA root certificate). The Trusted Certificate files define a list of trusted certificates. The phone's trusted list must contain the CA root certificates for all the servers it is connecting to. For example, if the phone is connecting to server A which has a certificate signed by CA1, and server B which has a certificate signed by CA2, the phone must have CA1 root certificate and CS2 root certificate in its Trusted Certificate file. **Notes**: **1.** If configuring TLS, you must specify the files for Root and Intermediate Certificates, the Local Certificate, the Private Key, and the Trusted Certificates in order for the phone to receive calls. **2.** If configuring Persistent TLS, you must specify the Trusted Certificates (which contains the trusted certificate list). All other certificates and the Private Key are optional. **3.** The certificate files and Private Key file names must use the format ".pem". **4.** To create custom certificate files and private key files to use on your IP phone, contact Aastra Technical Support.
6	Click **Save Settings** to save your changes.

Symmetric UDP Signaling

By default, the IP phones use symmetric UDP signaling for outgoing UDP SIP messages. When symmetric UDP is enabled, the IP phone generates and listens for UDP messages using port 5060.

You can manually disable symmetric UDP signaling using the IP phone's configuration file. When you disable symmetric UDP signaling, then the IP phone chooses a random source port for UDP messages.

The IP phone also chooses a random source port for UDP messages if you configure a backup proxy server, registrar server, or outbound proxy server.

An Administrator can configure symmetric UDP signaling using the configuration files only.

Configuring Symmetric UDP Signaling Using the Configuration Files

You use the following parameter to enable or disable Symmetric UDP Signaling in the configuration files:

- **`sip symmetric udp signaling`**

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "RTP, Codec, DTMF Global Settings" on page A-77.

Removing UserAgent and Server SIP Headers

Currently, the phone always configures the SIP UserAgent/Server headers to contain:

Aastra <PhoneModel>/<FirmwareVersion>

You can suppress the addition of these headers by using the following parameter in the configuration files:

- **sip user-agent**

Setting this parameter allows you to enable or disable the addition of the User-Agent and Server SIP headers from the SIP stack.

You can configure this feature using the configuration files only.

Configuring UserAgent/Server SIP Headers

You use the following parameter to specify whether the UserAgent and Server SIP header is added to the SIP stack.

- `sip user-agent`

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "User-Agent Setting" on page A-176.

Multi-Stage Digit Collection (Billing Codes) Support (for Sylantro Servers)

The IP Phones support Multi-Stage Digit Collection (billing codes) for Sylantro Servers. Sylantro Server features, like mandatory and optional billing codes, requires that the application server notify the phone to collect more digits before completing the call. The IP phone is able to collect digits in two stages to support the billing code feature.

Aastra IP Phone users are prompted to enter the correct billing code when they dial these numbers:

- External numbers.
- Eternal numbers dialed using a Speeddial key.

Billing Codes Implementation Notes

Note the following implementation information:

- IP phone users may enter a 2-9 digit billing code. Billing codes may not start with either 0 (Operator) or 9 (external calls).
- When using Sylantro Click-to-Call, IP phone users select a billing code from a pull-down menu.
- When placing a call, a secondary dial tone alerts IP phone users to enter the billing code. The IP phone UI also displays a "Enter Billing Code" message.
- If an IP phone user redials a number, they do not have to re-enter the billing code. The billing code information is maintained and processed accordingly.
- If an IP phone user enters an invalid billing code, the call fails.

Mandatory versus Optional Billing Codes

This release of the Aastra IP phones supports two types of billing codes: Mandatory and Optional. The Sylantro server configuration determines which type of billing code is used on the IP phones.

- **Mandatory billing codes**: Calls are not connected until the user enters a valid billing code. The user dials the phone number. When prompted for billing codes, user dials the billing code.

For example, suppose the IP phone user is using billing code 300, and dialing the external number 617-238-5500. The IP user then enters the number using the following format:

6172385000#300

Using mandatory billing codes, if the user is configuring a Speeddial number, then they enter the number using the following format:

<phonenumber>%23<billingcode>

To use this format with the default dial plan terminator (#), the # sign required by Sylantro as a delimiter should be represented as an escaped character by using the sequence %23. The speed dial format for an external number that includes a mandatory billing code becomes:

<phonenumber>%23<billing code>

- **Optional billing codes:** The user dials an optional billing code by dialing *50, followed by the billing code digits. When prompted for additional digits, user enters the phone number.

For example, suppose the IP phone user is using billing code 500, and dialing the external number 617-238-5000. The IP user then enters the number using the following format:

***50500#6172385000**

If the user is dialing configuring a Speeddial number, then they enter the number using the following format:

***50<billingcode>#<phonenumber>**

To use this format with the default dial plan terminator (#), the # sign required by Sylantro as a delimiter should be represented as an escaped character by using the sequence %23. The speed dial format for an external number that includes an optional billing code becomes:

***50<billing code>%23<phone number>**

Numbers Not Requiring Billing Codes

Billing codes are not required for the following two types of calls:

- Emergency calls (E911)
- Calls between extensions

Chapter 7
Encrypted Files on the IP Phone

About this chapter

Introduction

This chapter provides information about encryption on the IP phones and provides methods an administrator can use to store encrypted files to a server.

Topics

This chapter covers the following topics:

Encrypted Files on 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 encrypted 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.

Encrypted Files on the IP Phone

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).
2. 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.
```

Encryption and the IP Phone

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"**

Encrypted Files on the IP Phone

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

Chapter 8
Upgrading the Firmware

About this chapter

Introduction

This chapter provides information about upgrading the IP phone firmware.

Topics

This chapter covers the following topics:

Firmware Upgrade

Upgrading the Firmware

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 Chapter 1, the section, "Configuration Server Requirement" on page 1-20.

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)

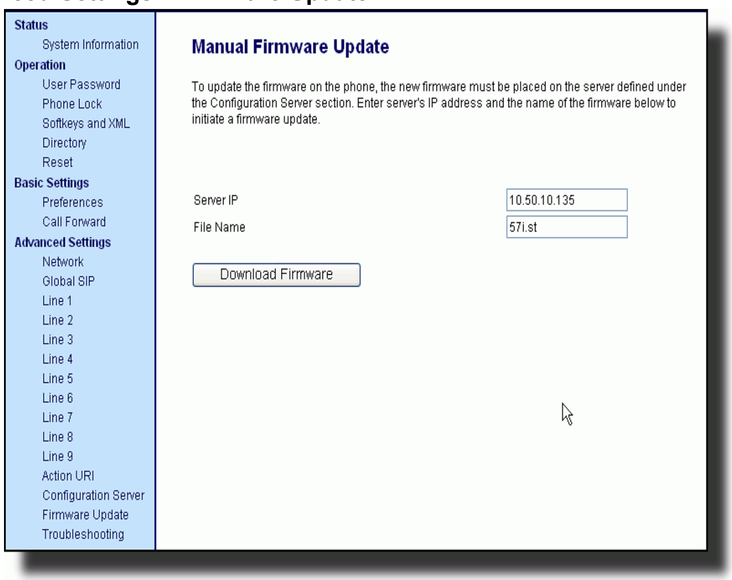
Use the following procedure to activate a firmware download using TFTP.

Warning: Do not reset or turn off the phone until the download is complete.

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.

Upgrading the Firmware

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Firmware Update.


</td></tr>
<tr><td>2</td><td>Enter the TFTP server IP address or qualified domain name in the "TFTP Server IP" field.</td></tr>
<tr><td>3</td><td>Enter the firmware file name (<phone model>.st) that you want to download to your IP phone in the "File Name" field. For example, 53i.st.

This file name must match the actual name of the firmware file residing on your configuration server.</td></tr>
<tr><td>4</td><td>Click 

.

This starts the upgrade process. If the upgrade is successful the following message displays on the screen: "Firmware Upgrade Successful".</td></tr>
</table>

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.

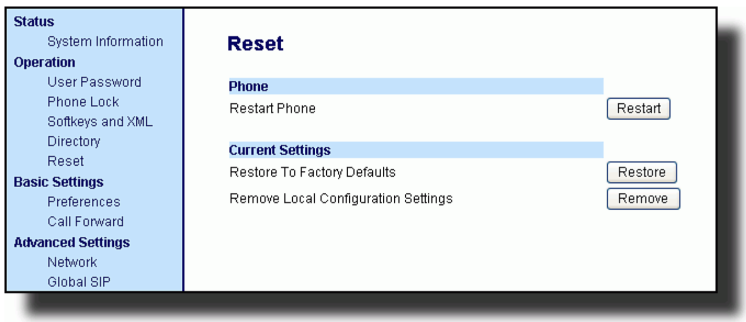
Restarting the Phone Using the IP Phone UI

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Restart Phone**.
3	***For 53i:*** Press **#** to confirm. **Note:** To cancel the Restart, press the ◀ key. ***For 55i, 57i, 57i CT:*** Press **Restart**. **Note:** To cancel the Restart, press **Cancel**.

Upgrading the Firmware

Restarting the Phone Using the Aastra Web UI

	Aastra Web UI
1	Click on **Operation->Reset->Phone.** 
2	Click  to restart the phone.

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**

Upgrading the Firmware

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.

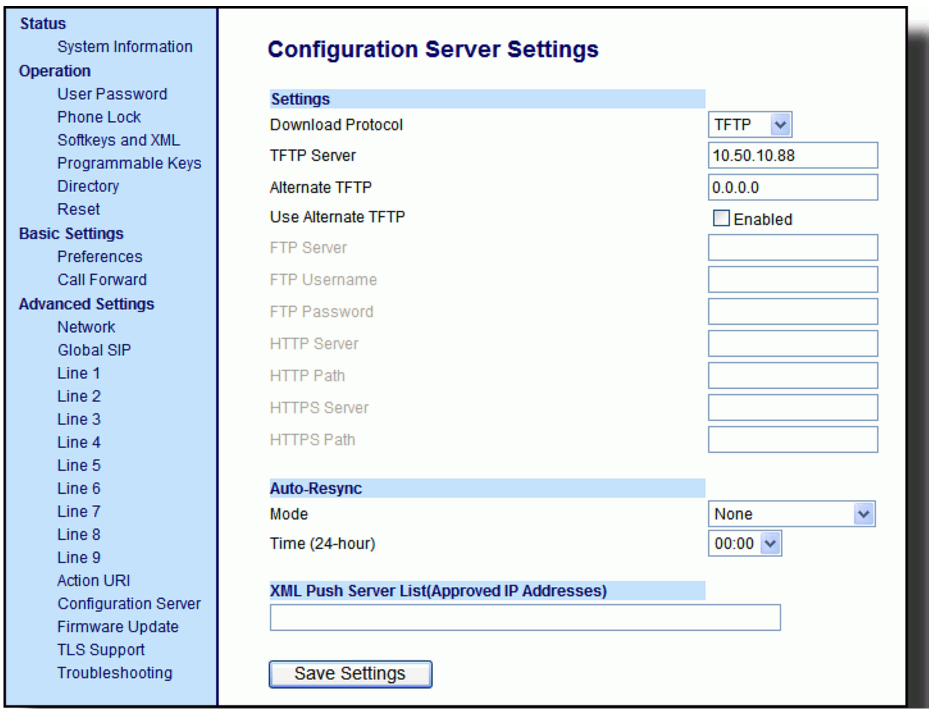
4. The resync time is based on the local time of the IP phone.

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 Appendix A, the section, "Configuration Server Settings" on page A-13.

Aastra Web UI

Step	Action
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.

Upgrading the Firmware

Reference

For more information about setting automatic update on the IP phone, see the "auto resync mode" and "auto resync time" parameters, see Appendix A, the section, "Configuration Server Settings" on page A-13.

Chapter 9
Troubleshooting

About this chapter

Introduction

This chapter describes tasks that a system administrator can perform on the IP phones for troubleshooting purposes. It also includes answers to questions you may have while using the IP phones.

Topics

This chapter covers the following topics:

Troubleshooting

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
- Filter the logs according to severity that get reported to 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 configuration files or the Aastra Web UI, you can specify the location for which to save files for troubleshooting purposes.

In the configuration files, you use the following parameters to configure log settings:

- **log server ip** - The IP address for which to save log files for troubleshooting purposes.
- **log server port** - The IP port to use to save log files for troubleshooting purposes.
- **log level** - The severity level of the logs to be reported to a log file. Log Level default is Error (3). (Changes to this parameter via the Aastra Web UI require a reboot).

Reference

For more information about the log setting configuration parameters, see Appendix A, the section, “Troubleshooting Parameters” on page A-177.

In the Aastra Web UI, you use the following parameters to configure log settings:

- **Log IP -** The IP address for which to save log files for troubleshooting purposes.
- **Log Port -** The IP port to use to save log files for troubleshooting purposes.

Module/Debug Level Settings

The Aastra IP phones provide blog module support that allows enhanced severity filtering of log calls sent as blog output.

The blog, as used on the IP phones, is a an online debugging tool that can be frequently updated and intended for technical support analyzation. Blogs are defined by their format: a series of entries posted to a single page in reverse-chronological order. The IP Phone blogs are separated into modules which allow you to log specific information for analyzing.

You can set the Module/Debug Levels using the configuration files or the Aastra Web UI. The following table identifies the blog modules you can set.

Aastra Web UI Parameters	Configuration File Parameters
LINMGR (Line Manager information)	log module linemgr
UI (User Interface (UI) related)	log module user interface
MISC (Miscellaneous)	log module misc
SIP (Call control SIP stack)	log module sip
DIS (Display drivers)	log module dis
DSTORE (Delayed storage)	log module dstore
EPT (Endpoint module)	log module ept
IND (Indicator module)	log module ind
KBD (Keyboard module	log module kbd
NET (Network module	log module net
PROVIS (Provisioning module	log module provis
RTPT (Realtime Transport module	log module rtpt
SND (Sound module	log module snd

Aastra Web UI Parameters	Configuration File Parameters
PROF (Profiler module)	log module prof
XML (Extension Markup Lanaguage)	log module xml

In the configuration files and the Aastra Web UI, you can enable or disable each module by setting the values of 1 (enable) or 0 (disable).

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.

In the "**Support Information**" section, you can:

- **Get local.cfg**
- **Get server.cfg**
- **Show task and Stack Status**

Aastra Technical Support uses this support information for troubleshooting the IP phone when required.

Performing Troubleshooting Tasks

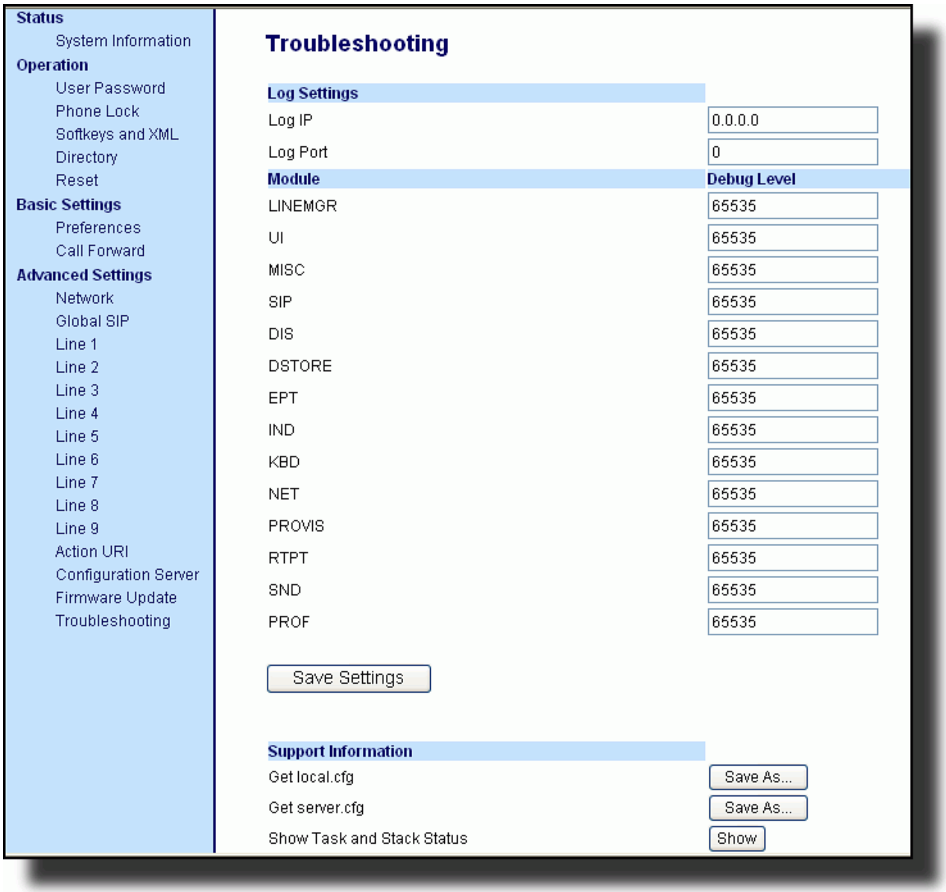
Use the following procedures to perform troubleshooting on the IP phone via the configuration files or the Aastra Web UI.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Troubleshooting Parameters" on page A-177.

Troubleshooting
Troubleshooting

Aastra Web UI

Step	Action
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 location.

Aastra Web UI

Step	Action
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 set blog modules:	
7	Select a module for which you want to have log files created and sent to a blog.
8	Enter a value (in decimal format) in the “**Debug Level**” field. The value of “**1**” enables the logging of the module. The value of “0” disables the logging of the module. The default value is “**1**”.
9	Click Save Settings to save your settings.
10	Click on **Operation->Reset**.
11	In the "**Restart Phone**" field click Restart to restart the IP phone.
To perform support tasks:	
12	To store the local configuration file to the specified location, click on Save As... in the "**Get local.cfg**" field.
13	To store the server configuration file to the specified location, click on Save As... in the "**Get server.cfg**" field.
14	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 the next section, “Troubleshooting Solutions” on page 9-7.

Troubleshooting Solutions

Description

This section describes solutions to some most common problems that can occur while using the IP phones.

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 auto-assigns 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 needs 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 launches the **Aastra IP Phone Firmware Recovery** page.
3. Call Customer Support and request a <*phone model*>*.st* file.
4. Copy the file to your TFTP server.
5. Enter the <*phone model*>*.st* 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:

Reason:

The site-specific password in *security.tuz* does not match the password used to encrypt the *<mac>.tuz* or *aastra.tuz* files.

Fix:

Encrypt the *.cfg* files to *.tuz* using the correct password, or replace the *security.tuz* with the correct encrypted file.

Reason:

Neither of the <*mac*>*.tuz* and *aastra.tuz* files are present on the configuration server (TFTP/FTP/HTTP).

Fix:

Create the encrypted files using *anacrypt.exe* and copy them to the configuration server.

Reason:

The encrypted <*mac*>*.tuz* or *aastra.tuz* file is encrypted using a different version of *anacrypt.exe* than the phone firmware.

Fix:

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 Chapter 4, the section, "Configuring the Configuration Server Protocol" on page 4-79.

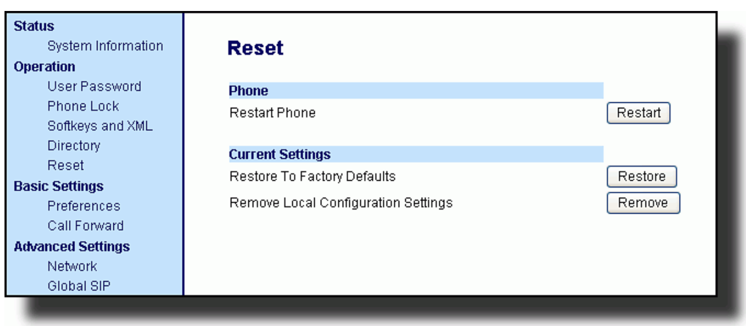
For specific protocol parameters you can set in the configuration files, see Appendix A, the section, "Configuration Server Settings" on page A-13.

How do I restart the IP phone?

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Restart Phone**.
3	***For 53i:*** Press **#** to confirm. **Note:** To cancel the Restart, press the ◀ key. ***For 55i, 57i, 57i CT:*** Press **Restart**. **Note:** To cancel the Restart, press **Cancel**.

Aastra Web UI

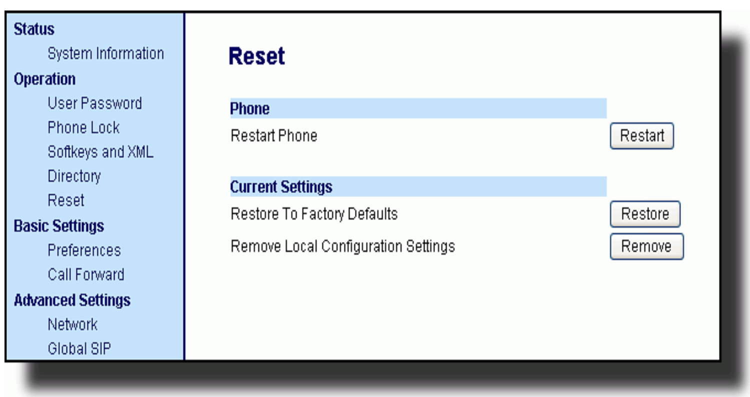
1	Click on **Operation->Reset->Phone.** 
2	Click Restart to restart the phone.

How do I set the IP phone to factory default?

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu** and enter your Administrator Password (default is **22222**).
3	Select **Factory Default.**
4	***For 53i:*** The "*Restore Defaults?*" prompt displays. Press **#** to confirm. ***For 55i/57i/57i CT:*** The "*Reset phone to factory defaults?*" prompt displays. Press **Default** to confirm.

Aastra Web UI

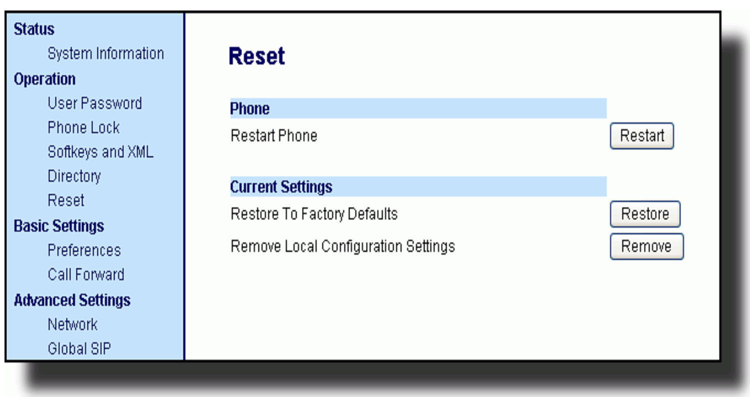
Step	Action
1	Click on **Operation->Reset->Current Settings.** 
2	In the "**Restore to Factory Defaults**" field, click Restore . This restores all factory defaults, and removes any saved configuration and directory list files.

How do I erase the phone's local configuration?

IP Phone UI

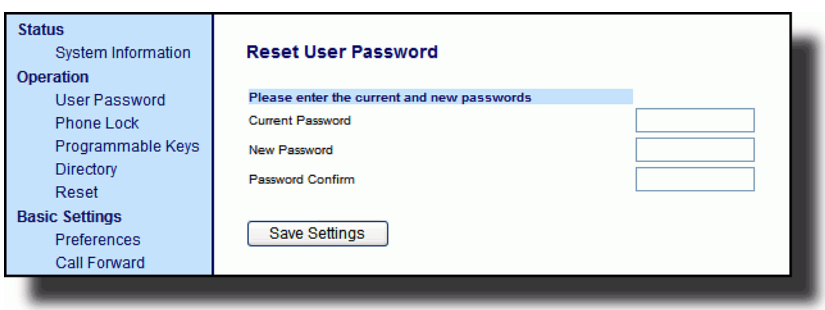
Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu** and enter your Administrator Password (default is **22222**).
3	Select **Erase Local Config.**
4	***For 53i:*** The "*Erase local config?*" prompt displays. Press **#** to confirm. ***For 55i/57i/57i CT:*** The "*Erase local config?*" prompt displays. Press **Erase** to confirm.

Aastra Web UI

Step	Action
1	Click on **Operation->Reset->Current Settings.** 
2	In the "**Remove Local Configuration Settings**" field, click Remove . This removes the last customized configuration settings made on the phone.

How to reset a user's password?

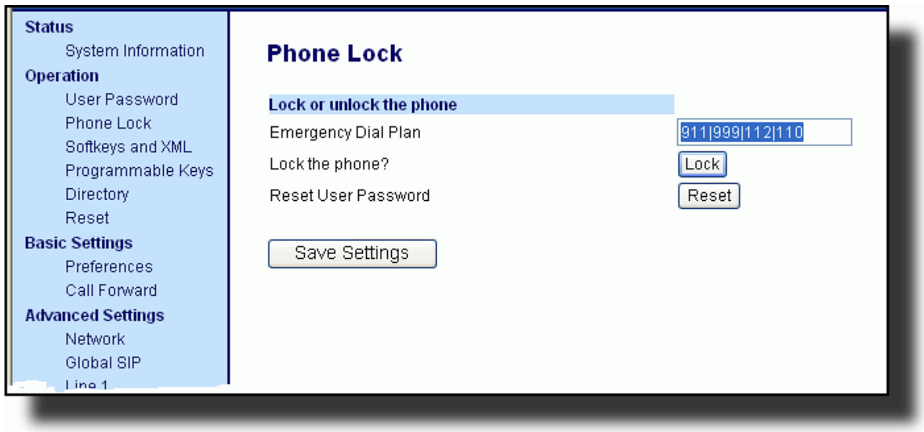
IP Phone UI	
1	Press on the phone to enter the Options List.
2	Select **User Password**.
3	Enter the current user password.
4	Press **Enter**.
5	Enter the new user password. **Note:** The IP phones support numeric characters only in passwords. If you enter a password with alpha characters, the phone uses the default password instead.
6	Press **Enter**.
7	Re-enter the new user password.
8	Press **Enter**. A message, "Password Changed" displays on the screen.

Aastra Web UI	
Step	**Action**
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.

How do I lock and unlock the phone?

IP Phone UI

Step	Action
Lock the phone:	
1	Press on the phone to enter the Options List.
2	Select **Phone Lock.** The prompt, "*Lock the phone*?" displays.
3	Press **Lock** to lock the phone.
Unlock the phone:	
1	Press on the phone to enter the Options List. The prompt, "*To unlock the phone...Password:*"
2	Enter the user or administrator password and press **Enter**. The phone unlocks.

	Aastra Web UI
1	Click on **Operation->Phone Lock**. 
Lock the phone:	
2	In the "**Lock the Phone?**" field, click Lock . The phone locks dyanamically and displays the following message: "*Phone is locked*".
Unlock the phone:	
3	Click on **Operation->Phone Lock**.
4	In the "**Unlock the Phone?**" field, click Unlock . The phone unlocks dynamically and displays the following message: "*Phone is unlocked*".

Appendix A
Configuration Parameters

About this appendix

Introduction

This appendix describes the parameters you can set in the configuration files for the IP phones. The configuration files include <mac.cfg> and config.cfg.

Topics

This appendix covers the following topics:

Appendix A

Setting Parameters in Configuration Files

Appendix A

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.

Note: Configuration parameters that you enter in the configuration files are NOT case sensitive.

Reference

For information about configuration file precedence, see Chapter 1, the section, "Configuration File Precedence" on page 1-22.

This section includes the following types of configurable parameters:

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).

Simplified IP Phone UI Options Menu

Parameter – *options simple menu*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to enable a simplified options menu or enable the full menu on the IP Phone UI. **Full Options Menu** / **Simplified Options Menu**: Call Forward / **Call Forward** Preferences / **Preferences** Phone Status / **Phone Status** User Password / Removed Administrator Menu / Removed Restart Phone / Removed Phone Lock / **Phone Lock** Handset Pairing (CT models only) / **Handset Pairing** (CT models only) **Warning**: When using the simplified menu, you cannot change the Network settings from the IP Phone UI. If the network settings become misconfigured, you must “factory default” the phone and use the full menu to recover the network settings from the Phone UI **OR** use the Aastra Web UI to configure the network settings.
Format	Boolean
Default Value	0 (full options menu)
Range	0 (full options menu) 1 (simplified options menu)
Example	options simple menu: 1

Nested table within Description:

Full Options Menu	Simplified Options Menu
Call Forward	**Call Forward**
Preferences	**Preferences**
Phone Status	**Phone Status**
User Password	Removed
Administrator Menu	Removed
Restart Phone	Removed
Phone Lock	**Phone Lock**
Handset Pairing (CT models only)	**Handset Pairing** (CT models only)

Network Settings

Parameter – *dhcp* *DHCP* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings Advanced Settings->Network-> Basic Network Settings aastra.cfg, <mac>.cfg
Description	Enables or disables DHCP. Enabling DHCP populates the required network information. The DHCP server serves the network information that the IP phone requires. If the IP phone is unable to get any required information, then you must enter it manually. DHCP populates the following network information: IP Address, Subnet Mask, Gateway, Broadcast Address, Domain Name Servers (DNS), TFTP, HTTP HTTPS, and FTP servers, and Timer Servers. For DHCP to automatically populate the IP address or qualified domain name for the TFTP server, your DHCP server must support Option 66.	
Format	Integer	
Default Value	1 (enabled)	
Range	0 (disabled) 1 (enabled)	
Example	dhcp: 1	

Parameter – *ip* *Ip Address* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings Advanced Settings->Network-> Basic Network Settings 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** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings Advanced Settings->Network-> Basic Network Settings 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->Administrator Menu-> Network Settings Advanced Settings->Network-> Basic Network Settings 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->Administrator Menu-> Network Settings Advanced Settings->Network-> Basic Network Settings 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->Administrator Menu-> Network Settings Advanced Settings->Network-> Basic Network Settings aastra.cfg, <mac>.cfg
Description	A service that translates domain names into IP addresses. To assign static DNS addresses, disable DHCP.	
Format	IP address	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	dns2: 192.168.0.6	

Parameter – *ethernet port 0* *LAN Port* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings Advanced Settings->Network-> Basic Network Settings aastra.cfg, <mac>.cfg
Description	The send (TX) and receive (RX) method to use on Ethernet port 0 to transmit and receive data over the LAN.	
Format	Integer	
Default Value	0	
Range	0 - auto-negotiate 1 - full-duplex, 10Mbps 2 - full-duplex, 100Mbps 3 - half-duplex, 10Mbps 4 - half-duplex, 100Mbps	
Example	lan port: 1	

Parameter – *ethernet port 1* *PC Port* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings Advanced Settings->Network-> Basic Network Settings aastra.cfg, <mac>.cfg
Description	The send (TX) and receive (RX) method to use on Ethernet port 1 to transmit and receive data over the LAN.	
Format	Integer	
Default Value	0	
Range	0 - auto-negotiate 1 - full-duplex, 10Mbps 2 - full-duplex, 100Mbps 3 - half-duplex, 10Mbps 4 - half-duplex, 100Mbps	
Example	ethernet port 1: 2	

Password Settings

Parameter – *admin password*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	Allows you to set a new administrator password for the IP phone. **Note:** The IP phones support numeric characters only in passwords. If you enter a password with alpha characters, the phone uses the default password instead.	
Format	Integer	
Default Value	22222	
Range	0 to 4294967295	
Example	admin password: 1234567890	

Parameter – *user password* *Current Password* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->User Password Operation->User Password aastra.cfg, <mac>.cfg
Description	Allows you to set a new user password for the IP phone. **Note:** The IP phones support numeric characters only in passwords. If you enter a password with alpha characters, the phone uses the default password instead.	
Format	Integer	
Default Value	Default value is an empty string "" (left blank)	
Range	0 to 4294967295	
Example	user password: 123	

Emergency Dial Plan Settings

Parameter – *emergency dial plan*	**Configuration Files** aastra.cfg, <mac>.cfg **Aastra Web UI** Operation->Phone Lock
Description	Allows you to specify an emergency number to use on your IP phone so a caller can contact emergency services in the local area when required. The default emergency numbers on the IP phones is 911, 999, 112, and 110. 911 - A United States emergency number. 999 - A United Kingdom emergency number. 112 - An international emergency telephone number for **GSM mobile phone** networks. In all European Union countries it is also the emergency telephone number for both mobile and fixed-line telephones. 110 - A police and/or fire emergency number in Asia, Europe, Middle East, and South America. **Note:** Contact your local phone service provider for available emergency numbers in your area.
Format	Integer
Default Value	911|999|112|110
Range	Up to 128 characters
Example	emergency dial plan: 911|999

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** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings Advanced Settings->Configuration Server 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 HTTPS	
Example	download protocol: HTTPS	

Parameter – *tftp server* *TFTP Server* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->TFTP Server->Primary TFTP Advanced Settings->Configuration Server 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	

Appendix A

Parameter – *alternate tftp server* *Alternate TFTP* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->TFTP Server-> Alternate 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	alternate tftp server: 192.168.0.132	

Parameter – *use alternate tftp* *Use Alternate TFTP* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->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 alternate tftp: 1	

Parameter – *ftp server* *FTP Server* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->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** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->FTP Server Advanced Settings->Configuration Server 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. **Note:** The IP Phones support usernames containing dots (".").	
Format	Text	
Default Value	Not Applicable	
Range	Up to 63 alphanumeric characters	
Example	ftp username: 57iaastra	

Parameter – *ftp password* *FTP Password* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->FTP Server Advanced Settings->Configuration Server 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: 1234	

Parameter – *http server* *HTTP Server* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->HTTP Server Advanced Settings->Configuration Server 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 next parameter (**http path**).	
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->Administrator Menu-> Network Settings->HTTP Server Advanced Settings->Configuration Server aastra.cfg, <mac>.cfg
Description	The HTTP path name to enter. If the IP phone's configuration and firmware 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/57i	

Parameter – *https server* *HTTPS Server* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->HTTPS-> HTTPS Client->Download Server Advanced Settings->Configuration Server aastra.cfg, <mac>.cfg
Description	The HTTPS server's IP address. This will become effective after this configuration file has been downloaded into the phone. **Optional**: You can also assign an HTTPS relative path to the HTTPS server. See the next parameter (**https path**)	
Format	IP address or fully qualified Domain Name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	https server: 192.168.0.143	

Parameter – *https path* *HTTPS Path* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->HTTPS-> HTTPS Client->Download Path Advanced Settings->Configuration Server aastra.cfg, <mac>.cfg
Description	The HTTPS path name to enter. If the IP phone's configuration and firmware files are located in a sub-directory beneath the server's HTTPS 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	https path: ipphones/55i	

Parameter – *auto resync mode* *Mode* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Configuration Server-> Auto-Resync 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 the IP 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	

Network Address Translation (NAT) Settings

Parameter – *sip nat ip* *NAT IP* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu->SIP Settings Advanced Settings->Network 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 SIP Port* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu->SIP Settings Advanced Settings->Network 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: 51620	

Parameter – *sip nat rtp port* *NAT RTP Port* (in Phone UI and Web UI)	**IP Phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> SIP Settings->RTP Port Base Advanced Settings->Network-> Advanced Network Settings aastra.cfg, <mac>.cfg
Description	Indicates the port through which the RTP packets are sent. This value must specify the beginning of the RTP port range on the gateway or router. 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	51720	
Range	Not Applicable	
Example	sip nat rtp port: 51730	

Parameter – *sip nortel nat support* *Nortel NAT Traversal Enabled* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu->SIP Settings Advanced Settings->Network 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** **Configuration Files**	Advanced Settings->Network 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	

HTTPS Client and Server Settings

Parameter – *https client method* *HTTPS Client Method* (in Web UI)	**IP Phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->HTTPS->HTTPS Client Advanced Settings->Network-> Advanced Network Settings aastra.cfg, <mac>.cfg
Description	Defines the security method that the client advertises to the server during the Secure Socket Layer (SSL) handshake. Available options are: **TLS 1.0** - Transport Layer Security version 1 (TLS 1.0) is a protocol that ensures privacy between communicating applications and their users on the Internet. TLS is the successor to SSL. **SSL 3.0** - Secure Socket Layer version 3 (SSL 3.0) is a commonly-used protocol for managing the security of a message transmission on the Internet.	
Format	Alphanumeric characters	
Default Value	SSL 3.0	
Range	TLS 1.0 SSL 3.0 (default)	
Example	https client method: TLS 1.0	

Parameter – *https redirect http get* *HTTPS Server - Redirect HTTP to HTTPS* (in Web UI)	**IP Phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->HTTPS-> HTTPS Server->Redirect Advanced Settings->Network-> Advanced Network Settings aastra.cfg, <mac>.cfg
Description	Allows or disallows redirection from the HTTP server to the HTTPS server.	
Format	Boolean	
Default Value	1 (enables redirection)	
Range	0 (disables redirection) 1 (enables redirection)	
Example	https redirect http get: 0	

Parameter – *https block http post xml* *HTTPS Server - Block XML HTTP POSTs* (in Web UI)	**IP Phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->HTTPS-> HTTPS Server->XML Advanced Settings->Network-> Advanced Network Settings aastra.cfg, <mac>.cfg
Description	Enables or disables the blocking of XML scripts from HTTP POSTs. Some client applications use HTTP POSTs to transfer XML scripts. The phones's HTTP server accepts these POSTs even if server redirection is enabled, effectively bypassing the secure connection. When this parameter is enabled (blocking is enabled), receipt of an HTTP POST containing an XML parameter header results in the following response: "403 Forbidden". This forces the client to direct the POSTs to the HTTPS server through use of the "https://" URL.	
Format	Boolean	
Default Value	0 (disables blocking of XML HTTP POSTs)	
Range	0 (disables blocking of XML HTTP POSTs) 1 (enables blocking of XML HTTP POSTs)	
Example	https block http post xml: 1	

UPnP Settings

Parameter – *upnp manager* *UPnP* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->UPnP Advanced Settings->Network-> Advanced Network Settings aastra.cfg, <mac>.cfg
Description	Enables or disables Universal Plug and Play (UpnP) on the IP phone. If you set this parameter to “0”, you can manually configure NAT on the IP phone and the UPnP manager will not start.	
Format	Boolean	
Default Value	0 (false)	
Range	0 (false) 1 (true)	
Example	upnp manager: 1	

Parameter – *upnp gateway*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	IP address or fully qualified Domain Name of the Internet gateway or router. This parameter stores the IP address of the gateway or router in the event that only non-default UPnP gateways get discovered on the network. The UPnP port mappings are saved to this IP address so even if the phone reboots, it will still have the correct port mappings.	
Format	IP address or fully qualified Domain Name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	upnp gateway: 120.400.003.2	

Parameter – *upnp mapping lines* *UPnP Mapping Lines* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences->General aastra.cfg, <mac>.cfg
Description	Enables or disables the use of Universal Plug and Play (UpnP) on a specific line on the IP phone. **Note:** For this feature to work, UPnP must be enabled on the phone (**upnp manager** and **upnp gateway** must be set).	
Format	Integer	
Default Value	0 (UPnP is not mapped to a specific line)	
Range	0 to 10	
Example	upnp mapping lines: 5 This example indicates that line 5 allows UPnP mapping.	

Virtual Local Area Network (VLAN) Settings

Global Parameters

Parameter – *tagging enabled* *VLAN Enable* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->VLAN->VLAN Enable Advanced Settings->Network->VLAN->Global aastra.cfg, <mac>.cfg
Description	Enables or disables VLAN on the IP phones. This is a global setting.	
Format	Boolean	
Default Value	0 (false)	
Range	0 (false) 1 (true)	
Example	tagging enabled: 1	

Parameter – *priority non-ip* *Priority, Non-IP Packet* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->VLAN->Phone->Priority ->Other Advanced Settings->Network->VLAN->Global aastra.cfg, <mac>.cfg
Description	Specifies the priority value for non-IP packets. This is a global setting.	
Format	Integer	
Default Value	5	
Range	0 to 7	
Example	priority non-ip: 7	

LAN Port (Ethernet Port 0) Parameters

Parameter – *vlan id* *VLAN ID* (for LAN Port in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->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 LAN Port in Web UI)	**IP phone UI**	Options->Administrator Menu-> Network Settings->VLAN->Phone-> Priority->SIP Options->Administrator Menu-> Network Settings->VLAN->Phone-> Priority->RTP Options->Administrator Menu-> Network Settings->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 26) 5 (based on the default ToS DSCP RTP setting of 46) 5 (based on the default ToS DSCP RTCP setting of 46)	
Range	0 to 63 (for DSCP) 0 to 7 (for SIP, RTP, and RTCP priorities)	
Example	tos priority map: (26,7)	

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

PC Port (Ethernet Port 1) Parameters

Parameter – *vlan id port 1* *VLAN ID* (for PC Port in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->VLAN-> Passthrough->VLAN ID Advanced Settings->Network->VLAN->Port 1 aastra.cfg, <mac>.cfg

Description	Specifies the VLAN ID used to pass packets through to a PC via Port 1.
	
	Note: If you set the VLAN id port 1 (passthrough port) to **4095**, all untagged packets are sent to this port. The following is an example of configuring the phone on a VLAN where all untagged packets are sent to the passthrough port.
	
	<u>Example</u>
	You enable tagging on the phone port as normal but set the passthrough port (VLAN id port 1) to 4095. The following example sets the phone to be on VLAN 3 but the passthrough port is configured as untagged.
	
	`tagging enabled: 1`
	`VLAN id: 3`
	`VLAN id port 1: 4095`	

Format	Integer	
Default Value	1	
Range	1 to 4095	
Example	VLAN id port 1: 3	

Parameter – *QoS eth port 1 priority* *Priority* (for PC Port in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->VLAN->Passthrough ->Priority Advanced Settings->Network->VLAN->Port 1 aastra.cfg, <mac>.cfg
Description	Specifies the priority value used for passing VLAN packets through to a PC via Port 1.	
Format	Integer	
Default Value	0	
Range	0 to 7	
Example	QoS eth port 1 priority: 3	

Type of Service (ToS)/DSCP Settings

Parameter – *tos sip* *SIP* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->Type of Service->SIP Advanced Settings->Network-> Type of Service,DSCP aastra.cfg, <mac>.cfg
Description	The Differentiated Services Code Point (DSCP) for SIP packets.	
Format	Integer	
Default Value	26	
Range	0 to 63	
Example	tos sip: 3	

Parameter – *tos rtp* *RTP* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->Type of Service->RTP Advanced Settings->Network-> Type of Service,DSCP aastra.cfg, <mac>.cfg
Description	The Differentiated Services Code Point (DSCP) for RTP packets.	
Format	Integer	
Default Value	46	
Range	0 to 63	
Example	tos rtp: 2	

Parameter – *tos rtcp* *RTCP* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->Type of Service->RTCP Advanced Settings->Network-> Type of Service,DSCP aastra.cfg, <mac>.cfg
Description	The Differentiated Services Code Point (DSCP) for RTCP packets.	
Format	Integer	
Default Value	46	
Range	0 to 63	
Example	tos rtcp: 3	

Time Server Settings

Parameter – *time server disabled* *NTP Time Servers* (in Web UI)	**Aastra Web UI** Advanced Settings->Network->Advanced Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables the time server. This parameter affects the **time server1**, **time server2**, and **time server3** parameters. Setting this parameter to **0** allows the use of the configured Time Server(s). Setting this parameter to **1** prevents the use of the configured Time Server(s).
Format	Integer
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	time server disabled: 0

Parameter – *time server1* *Time Server 1* (in Web UI)	**Aastra Web UI** Advanced Settings->Network->Advanced Network Settings **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-> Advanced Network Settings 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-> Advanced Network Settings 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	

Time and Date Settings

Parameter – *time format* *Time Format* (in Phone UI)	**IP phone UI** **Configuration Files**	Options->Time and Date->Time Format 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 (12 hr format)) 1 (24 hr format)	
Example	time format: 0	

Parameter – *date format* *Date Format* (in Phone UI)	**IP phone UI** **Configuration Files**	Options->Time and Date->Date Format aastra.cfg, <mac>.cfg
Description	This parameter allows the user to change the date to various formats.	
Format	Integer	
Default Value	0	
Range	0 (WWW MMM DD) (default) 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 – *dst config* *Daylight Savings* (in Phone UI)	**IP phone UI** **Configuration Files**	Options->Time and Date->Daylight Savings aastra.cfg, <mac>.cfg
Description	Enables/disables the use of daylight savings time.	
Format	Integer	
Default Value	3	
Range	0 - OFF 1 - 30 min summertime 2 - 1 hr summertime 3 - automatic	
Example	dst config: 0	

Parameter – *time zone name* *Time Zone* (in Phone UI)	**IP phone UI** **Configuration Files**	Options->Time and Date->Time Zone aastra.cfg, <mac>.cfg
Description	Assigns a time zone name to the time server.	
Format	Text	
Default Value	US-Eastern	
Range	See "Time Zone Name/Time Zone Code Table" below.	
Example	time zone name: US-Central	

Time Zone Name/Time Zone Code Table

Time Zone Name	Time Zone Code
AD-Andorra	CET
AG-Antigua	AST
AI-Anguilla	AST
AL-Tirane	CET
AN-Curacao	AST
AR-Buenos Aires	ART
AS-Pago Pago	BST
AT-Vienna	CET
AU-Lord Howe	LHS
AU-Tasmania	EST
AU-Melbourne	EST
AU-Sydney	EST
AU-Broken Hill	CST
AU-Brisbane	EST
AU-Lindeman	EST
AU-Adelaide	CST
AU-Darwin	CST
AU-Perth	WST
AW-Aruba	AST
BA-Sarajevo	EET
BB-Barbados	AST
BE-Brussels	CET
BG-Sofia	EET
BM-Bermuda	AST
BO-La Paz	BOT
BR-Noronha	FNT
BR-Belem	BRT
BR-Fortaleza	BRT
BR-Recife	BRT
BR-Araguaina	BRS
BR-Maceio	BRT
BR-Sao Paulo	BRS
BR-Cuiaba	AMS
BR-Porto Velho	AMT
BR-Boa Vista	AMT
BR-Manaus	AMT
BR-Eirunepe	ACT
BR-Rio Branco	ACT
BS-Nassau	EST
BY-Minsk	EET
BZ-Belize	CST

Time Zone Name	Time Zone Code
CA-Newfoundland	NST
CA-Atlantic	AST
CA-Eastern	EST
CA-Saskatchewan	EST
CA-Central	CST
CA-Mountain	MST
CA-Pacific	PST
CA-Yukon	PST
CH-Zurich	CET
CK-Rarotonga	CKS
CL-Santiago	CLS
CL-Easter	EAS
CN-China	CST
CO-Bogota	COS
CR-Costa Rica	CST
CU-Havana	CST
CY-Nicosia	EES
CZ-Prague	CET
DE-Berlin	CET
DK-Copenhagen	CET
DM-Dominica	AST
DO-Santo Domingo	AST
EE-Tallinn	EET
ES-Madrid	CET
ES-Canary	WET
FI-Helsinki	EET
FJ-Fiji	NZT
FK-Stanley	FKS
FO-Faeroe	WET
FR-Paris	CET
GB-London	GMT
GB-Belfast	GMT
GD-Grenada	AST
GF-Cayenne	GFT
GI-Gibraltar	CET
GP-Guadeloupe	AST
GR-Athens	EET
GS-South Georgia	GST
GT-Guatemala	CST
GU-Guam	CST
GY-Guyana	GYT

Time Zone Name	Time Zone Code
HK-Hong Kong HN-Tegucigalpa HR-Zagreb HT-Port-au-Prince HU-Budapest	HKS CST CET EST CET
IE-Dublin IS-Reykjavik IT-Rome	GMT GMT CET
JM-Jamaica JP-Tokyo	EST JST
KY-Cayman	EST
LC-St Lucia LI-Vaduz LT-Vilnius LU-Luxembourg LV-Riga	AST CET EET CET EET
MC-Monaco MD-Chisinau MK-Skopje MQ-Martinique MS-Montserrat MT-Malta MX-Mexico City MX-Cancun MX-Merida MX-Monterrey MX-Mazatlan MX-Chihuahua MX-Hermosillo MX-Tijuana	CET EET CET AST AST CET CST CST CST CST MST MST MST PST
NI-Managua NL-Amsterdam NO-Oslo NR-Nauru NU-Niue NZ-Auckland NZ-Chatham	CST CET CET NRT NUT NZS CHA

Time Zone Name	Time Zone Code
PA-Panama	EST
PE-Lima	PES
PL-Warsaw	CET
PR-Puerto Rico	AST
PT-Lisbon	WET
PT-Madeira	WET
PT-Azores	AZO
PY-Asuncion	PYS
RO-Bucharest	EET
RU-Kaliningrad	EET
RU-Moscow	MSK
RU-Samara	SAM
RU-Yekaterinburg	YEK
RU-Omsk	OMS
RU-Novosibirsk	NOV
RU-Krasnoyarsk	KRA
RU-Irkutsk	IRK
RU-Yakutsk	YAK
RU-Vladivostok	VLA
RU-Sakhalin	SAK
RU-Magadan	MAG
RU-Kamchatka	PET
RU-Anadyr	ANA
SE-Stockholm	CET
SG-Singapore	SGT
SI-Ljubljana	CET
SK-Bratislava	CET
SM-San Marino	CET
SR-Paramaribo	SRT
SV-El Salvador	CST
TR-Istanbul	EET
TT-Port of Spain	AST
TW-Taipei	CST
UA-Kiev	EET
US-Eastern	EST
US-Central	CST
US-Mountain	MST
US-Pacific	PST
US-Alaska	AKS
US-Aleutian	HAS
US-Hawaii	HST
UY-Montevideo	UYS

Time Zone Name	Time Zone Code
VA-Vatican	CET
YU-Belgrade	CET

SIP Local Dial Plan Settings

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Parameter – *sip dial plan* *Local Dial Plan* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences **Configuration Files** 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; # can terminate a dial string | — 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] “,” (open/close quotes) — In the configuration files, enter the sip dial plan value using quotes. **Note:** You can configure prefix dialing by adding a prepend digit to the dial string. For example, if you add a prepend map of “[2-9]XXXXXXXXX,91”, the IP phone adds the digits “91” to any 10-digit number beginning with any digit from 2 to 9 that is dialed out. Other examples of prepend mappings are: • **1X+#,9** (Prepends 9 to any digit string beginning with “1” and terminating with “#”.) • **6XXX,579** (Prepends “579” to any 4-digit string starting with “6”.) • **[4-6]XXXXXX,78** (Prepends “78” to any 7-digit string starting with “4”, “5”, or “6”.
Format	Alphanumeric characters
Default Value	X+#|XX+*
Range	Up to 127 alphanumeric characters
Example	sip dial plan: "X+#|XXX+*"

Appendix A

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	Allows you to enable or disable a dial plan terminator. If you enable the use of this parameter, when you configure the IP phone's dial plan with 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 after to finish dialing the number on the keypad before making the call.	
Format	Boolean	
Default Value	0	
Range	0 (Disable) 1 (Enable)	
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, between consecutive key presses on the IP phone. The default for this parameter is 4 seconds. If you press a key on the phone and wait 4 seconds before pressing the next key, the key times out and cancels the digit selection. You must press consecutive keys before the timeout occurs.	
Format	Integer	
Default Value	4	
Range	Not Applicable	
Example	sip digit timeout: 6	

SIP Basic, Global Settings

SIP Global Authentication Settings

Parameter – *sip screen name* *Screen Name* (in Web UI)	**IP Phone UI** Options->Administrator Menu->SIP Settings **Aastra Web UI** Advanced Settings->Global SIP->Basic SIP Settings **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 user's name of the phone.
Format	Text
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip screen name: Joe Smith

Parameter – *sip screen name 2* *Screen Name 2* (in Web UI)	**IP Phone UI** Options->Administrator Menu->SIP Settings **Aastra Web UI** Advanced Settings->Global SIP->Basic SIP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Used to display text on a second line on the screen of the phone. **Notes**: **1.** If other status messages display on the phone, such as "Network Disconnected", the Screen Name 2 value does not display. **2.** Symbol characters are allowed (such as "#"). **3.** If the text is longer than the display width, than the display truncates the text to fit the display.
Format	Alphanumeric characters.
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters.
Example	sip screen name 2: Lab Phone

Parameter – *sip user name* *Phone Number* (in Web UI)	**IP Phone UI** Options->Administrator Menu->SIP Settings **Aastra Web UI** Advanced Settings->Global SIP-> Basic SIP Settings **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:** The IP Phones support Usernames containing dots (“.”).
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)	**IP Phone UI** Options->Administrator Menu->SIP Settings **Aastra Web UI** Advanced Settings->Global SIP-> Basic SIP Settings **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 display name: Joe Smith

Parameter – *sip auth name* *Authentication Name* (in Web UI)	**IP Phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu->>SIP Settings Advanced Settings->Global SIP-> Basic SIP Settings 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** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu->SIP Settings Advanced Settings->Global SIP-> Basic SIP Settings aastra.cfg, <mac>.cfg
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 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

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

SIP Global Network Settings.

Parameter – *sip proxy ip* *Proxy Server* (in Web UI)	**IP Phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu->SIP Settings Advanced Settings->Global SIP-> Basic SIP Settings 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** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu->SIP Settings Advanced Settings->Global SIP-> Basic SIP Settings 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 backup proxy ip* *Backup Proxy Server* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Basic SIP Network Settings aastra.cfg, <mac>.cfg
Description	The IP address of the backup SIP proxy server for which the IP phone uses when the primary SIP proxy is unavailable.	
Format	IP address or fully qualified Domain Name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	sip backup proxy ip: 192.168.0.102	

Parameter – *sip backup proxy port* *Backup Proxy Port* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Basic SIP Network Settings aastra.cfg, <mac>.cfg
Description	The backup proxy's port number.	
Format	Integer	
Default Value	0	
Range	Not Applicable	
Example	sip backup proxy port: 5060	

Parameter – *sip outbound proxy* *outbound proxy server* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings 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

Parameter – *sip registrar ip* *Registrar Server* (in Web UI)	**IP Phone UI** Options->Administrator Menu->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->Administrator Menu->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 backup registrar ip* *Backup Registrar Server* (in Web UI)	**Aastra Web UI** Advanced Settings->Global SIP->Basic SIP Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	The address of the backup registrar (typically, the backup SIP proxy) for which the IP phone uses to send *REGISTER* requests if the primary registrar is unavailable. A global value of 0.0.0.0 disables backup registration. However, the phone is still active and you can dial using username@ip address of the phone. If the backup registrar IP address is set to 0.0.0.0 for a per-line basis (i.e, line 1, line 2, etc.), then the backup 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 backup registrar ip: 192.168.0.102

Parameter – *sip backup registrar port* *Backup Registrar Port* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Basic SIP Network Settings aastra.cfg, <mac>.cfg
Description	The backup registrar's (typically the backup SIP proxy) port number.	
Format	Integer	
Default Value	0	
Range	Not Applicable	
Example	sip backup registrar port: 5060	

Parameter – *sip registration period* *Registration Period* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Basic SIP Network Settings 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	

Appendix A

SIP Basic, Per-Line Settings

The following parameters are SIP per-line settings. The value of "*N*" is 1 - 9 for 53i, 55i, 57i, and 57i CT.

SIP Per-Line Authentication Settings

Parameter – *sip lineN screen name* *Screen Name* (in Web UI)	**Aastra Web UI** Advanced Settings->Line 1 thru 9 **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 screen name 2* *Screen Name 2* (in Web UI)	**Aastra Web UI** Advanced Settings->Line 1 thru 9 **Configuration Files** aastra.cfg, <mac>.cfg
Description	Used to display text on a second line on the screen of the phone. **Notes**: **1.** If other status messages display on the phone, such as “Network Disconnected”, the Screen Name 2 value does not display. **2.** Characters are allowed (such as “#”). **3.** If the text is longer than the display width, than the display truncates the text to fit the display.
Format	Alphanumeric characters
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip line1 screen name 2: Lab Phone

Parameter – *sip lineN user name* *Phone Number* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 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. 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 A-53. **Note:** The IP Phones support Usernames containing dots (“.”).	
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** **Configuration Files**	Advanced Settings->Line 1 thru 9 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 auth name* *Authentication Name* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 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)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 aastra.cfg, <mac>.cfg
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* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 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: 1010appearance of phone 3) sip line1 user name: 10102(# for phone 2 with) sip line1 bla number: 1010appearance 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	

Appendix A

Parameter – *sip lineN mode* *Line Mode* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 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. • If the softkeys on the 57i/57i CT or the programmable keys on the 53i 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	

SIP Per-Line Network Settings.

Parameter – *sip lineN proxy ip* *Proxy Server* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 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)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 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 linex backup proxy ip* *Backup Proxy Server* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 aastra.cfg, <mac>.cfg
Description	The IP address of the backup SIP proxy server for which the IP phone uses when the primary SIP proxy is unavailable.	
Format	IP address or fully qualified Domain Name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	sip line1 backup proxy ip: 192.168.0.102	

Parameter – *sip linex backup proxy port* *Backup Proxy Port* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 aastra.cfg, <mac>.cfg
Description	The backup proxy's port number.	
Format	Integer	
Default Value	0	
Range	Not Applicable	
Example	sip line1 backup proxy port: 5060	

Parameter – *sip lineN outbound proxy* *Outbound Proxy Server* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 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** **Configuration Files**	Advanced Settings->Line 1 thru 9 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	

Parameter – *sip lineN registrar ip* *Registrar Server* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 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)	**Aastra Web UI** Advanced Settings->Line 1 thru 9 **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 linex backup registrar ip* *Backup Registrar Server* (in Web UI)	**Aastra Web UI** Advanced Settings->Line*N*-> Basic SIP Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	The address of the backup registrar (typically, the backup SIP proxy) for which the IP phone uses to send *REGISTER* requests if the primary registrar is unavailable. A global value of 0.0.0.0 disables backup registration. However, the phone is still active and you can dial using username@ip address of the phone. If the backup registrar IP address is set to 0.0.0.0 for a per-line basis (i.e, line 1, line 2, etc.), then the backup 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 backup registrar ip: 192.168.0.102

Parameter – *sip linex backup registrar port* *Backup Registrar Port* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line*N*-> Basic SIP Network Settings aastra.cfg, <mac>.cfg
Description	The backup registrar's (typically the backup SIP proxy) port number.	
Format	Integer	
Default Value	0	
Range	Not Applicable	
Example	sip line1 backup registrar port: 5060	

Parameter – *sip lineN registration period* *Registration Period* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line 1 thru 9 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	

Centralized Conferencing Settings

Global Settings

Parameter – *sip centralized conf* *Conference Server URI* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced->Global SIP Settings-> Basic SIP Network Settings aastra.cfg, <mac>.cfg
Description	Globally enables or disables SIP centralized conferencing for an IP phone as follows: • To disable centralized conferencing, leave this field empty (blank). • To enable SIP centralized conferencing, then do one of the following actions: — If you have specified a proxy server/registrar server, then to reach the media server via the proxy server, set this field to one of the following: **conf** (Sylantro server), or **Conference** (Broadsoft server) By setting this field to **conf**, you specify conf@<proxy_server _address>: <proxy_port>. For example, if the proxy server address is 206.229.26.60 and the proxy port used is 10060, then by setting this parameter to **conf**, you are specifying the following: conf@206.229.26.60:10060 — To reach the media server using a different address/port than that specified by the proxy, set this field to the following: **conf@<media_server _address>: <media_port>**	
Format	String	
Default Value	Blank	
Example	sip centralized conf: conf	

Per-Line Settings

Parameter – *sip lineN centralized conf* *Conference Server URI* (in Web UI)	**Aastra Web UI** Advanced->Line <1 thru 9>-> Basic SIP Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enable or disable per-line SIP centralized conferencing for an IP phone as follows: • To disable centralized conferencing, leave this field empty (blank). • To enable SIP centralized conferencing on a specific line, do one of the following actions: — If you have specified a proxy server/registrar server, then to reach the media server via the proxy server, set this field to one of the following: **conf** (Sylantro server), or **Conference** (Broadsoft server) By setting this field to **conf**, you specify conf@<proxy_server _address>: <proxy_port>. For example, if the proxy server address is 206.229.26.60 and the proxy port used is 10060, then by setting this parameter to **conf**, you are specifying the following: conf@206.229.26.60:10060. — To reach the media server using a different address/port than that specified by the proxy, set this field to the following: **conf@<media_server _address>: <media_port>**
Format	String
Default Value	Blank
Examples	sip line3 centralized conf: conf

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 explicit mwi subscription period* *Explicit MWI Timeout* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	The requested duration, in seconds, before the MWI subscription times out. The phone re-subscribes to MWI before the subscription period ends.	
Format	Integer	
Default Value	86400	
Range	30 - 214748364	
Example	sip explicit mwi timeout: 30	

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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	

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** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	This timer is a SIP transaction layer timer defined in RFC 3261. Timer 1 is an estimate, in milliseconds, of the round-trip time (RTT).	
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** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	This timer is a SIP transaction layer timer defined in RFC 3261. Timer 2 represents the amount of time, in milliseconds, a non-INVITE server transaction takes to respond to a request.	
Format	Integer	
Default Value	0	
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 designated 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 Real-Time Transport Protocol (RTP) port on the IP phone uses to send out SIP signaling packets. **Notes**: **1.** If you set the value of this parameter to 4 (TLS), the phone checks to see if the "**sips persistent tls**" is enabled. If it is enabled, the phone uses Persistent TLS on the connection. If "**sips persistent tls**" is disabled, then the phone uses TLS on the connection. If TLS is used, you must specify the Root and Intermediate Certificates, the Local Certificate, the Private Key, and the Trusted Certificates. **2.** If the phone uses Persistent TLS, you MUST specify the Trusted Certificates; the Root and Intermediate Certificates, the Local Certificate, and the Private Key are optional. For more information about Persistent TLS, see "Transport Layer Security (TLS) Settings" on page A-72.	
Format	Integer	
Default Value	1 - UDP	
Range	Valid values are: 0 - User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) 1 - UDP 2 - TCP 4 - Transport Layer Security (TLS)	
Example	sip transport protocol: 4	

Parameter – *sip registration retry timer* *Registration Failed Retry Timer* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> 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	

Parameter – *sip registration timeout retry timer* *Registration Timeout Retry Timer* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	Specifies the length of time, in seconds, that the phone waits until it re-attempts to register after a REGISTER message times out. If this parameter is set lower than 30 seconds, the phone uses a minimum timer of 30 seconds.	
Format	Integer	
Default Value	120	
Range	30 to 214748364	
Example	sip registration timeout retry timer: 150	

Parameter – *sip registration renewal timer* Registration Renewal Timer (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	The length of time, in seconds, prior to expiration, that the phone renews registrations. For example, if the value is set to 20, then 20 seconds before the registration is due to expire, a new REGISTER message is sent to the registrar to renew the registration.	
Format	Integer	
Default Value	15	
Range	0 to 214748364 The value set for this parameter should be between 0 and the value set for the registration period.	
Example	sip registration renewal timer: 10	

Parameter – *sip blf subscription period* *BLF Subscription Period* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP ->Advanced SIP Settings 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 is the minimum value)	
Example	sip blf subscription period: 2000	

Parameter – *sip acd subscription period* *ACD Subscription Period* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP ->Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	Specifies the time period, in seconds, that the IP phone resubscribes the Automatic Call Distribution (ACD) subscription service after a software/ firmware upgrade or after a reboot of the IP phone.	
Format	Integer	
Default Value	3600	
Range	120 (2 minutes is the minimum value)	
Example	sip acd subscription period: 2000	

Missed Call Summary Subscription Settings

Global Parameters

Parameter – *sip missed call summary subscription* *Missed Call Summary Subscription* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	Enables or disables the Missed Call Summary Subscription feature. This feature allows missed calls that have been redirected by the server, to be incremented in the missed calls indicator on the phone it was initially directed to. For example, phones A, B, and C are connected to the server. You configure the server to direct calls coming into phone B (which has voicemail configured) to be forwarded to phone C. When phone A calls **sip missed call summary subscription** parameter, phone B receives notification from the server that the call was forwarded and the missed calls indicator is incremented on phone B. **Note:** You must configure voicemail on the phone that the call was initially directed to (phone B in the above example).	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	sip missed call summary subscription: 1	

Parameter – *sip missed call summary subscription period* *Missed Call Summary Subscription Period* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	Specifies the amount of time, in seconds, that the phone uses the Missed Calls Summary Subscription feature. This parameter is always enabled with a default value of 86400 seconds. When the phone reaches the limit set for this parameter, it sends the subscription again. To disable this parameter, leave the field blank or set the field to zero (0).	
Format	Integer	
Default Value	86400	
Range	0 to 99999999	
Example	sip missed call summary subscription period: 70000	

Per-Line Parameter

Parameter – *sip lineN missed call summary subscription* *Missed Call Summary Subscription* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	Enables or disables the Missed Call Summary Subscription feature. This feature allows missed calls that have been redirected by the server, to be incremented in the missed calls indicator on the phone it was initially directed to. For example, phones A, B, and C are connected to the server. You configure the server to direct calls coming into phone B (which has voicemail configured) to be forwarded to phone C. When phone A calls phone B, the server forwards the call to phone C. If you enable the **sip missed call summary subscription** parameter, phone B receives notification from the server that the call was forwarded and the missed calls indicator is incremented on phone B. **Note:** You must configure voicemail on the phone that the call was initially directed to (phone B in the above example).	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	sip line1 missed call summary subscription: 1	

Transport Layer Security (TLS) Settings

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	Specifies the protocol that the RTP port on the IP phone uses to send out SIP signaling packets. **Notes**: **1.** If you set the value of this parameter to 4 (TLS), the phone checks to see if the "**sips persistent tls**" is enabled. If it is enabled, the phone uses Persistent TLS on the connection. If "**sips persistent tls**" is disabled, then the phone uses TLS on the connection. If TLS is used, you must specify the Root and Intermediate Certificates, the Local Certificate, the Private Key, and the Trusted Certificates. **2.** If the phone uses Persistent TLS, you MUST specify the Trusted Certificates; the Root and Intermediate Certificates, the Local Certificate, and the Private Key are optional.	
Format	Integer	
Default Value	1 - UDP	
Range	Valid values are: 0 - User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) 1 - UDP 2 - TCP 4 - Transport Layer Security (TLS)	
Example	sip transport protocol: 4	

Parameter – *sips persistent tls*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables the use of Persistent Transport Layer Security (TLS). Persistent TLS sets up the connection to the server once and re-uses that connection for all calls from the phone. The setup connection for Persistent TLS is established during the registration of the phone. If the phones are set to use Persistent TLS, and a call is made from the phone, this call and all subsequent calls use the same authenticated connection. This significantly reduces the delay time when placing a call. **Notes:** **1.** Persistent TLS requires the **outbound proxy server** and **outbound proxy port** parameters be configured in either the configuration files or the Aastra Web UI (*Advanced Settings->Global SIP->Basic SIP Network Settings*). There can be only one persistent TLS connection created per phone. The phone establishes the TLS connection to the configured outbound proxy. **2.** If you configure the phone to use Persistent TLS, you must also specify the **Trusted Certificate** file to use. The Root and Intermediate Certificates, Local Certificate, and Private Key files are optional.
Format	Boolean
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	sips persistent tls: 1

Parameter – *sips root and intermediate certificates* *Root and Intermediate Certificates* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->TLS Support aastra.cfg, <mac>.cfg
Description	Allows you to specify the SIP Root and Intermediate Certificate files to use when the phone uses the TLS transport protocol to setup a call. The Root and Intermediate Certificate files contain one root certificate and zero or more intermediate certificates which must be placed in order of certificate signing with root certificate being the first in the file. If the local certificate is signed by some well known certificate authority, then that authority provides the user with the Root and Intermediate Certificate files (most likely just CA root certificate). This parameter is required when configuring TLS (optional for Persistent TLS.) **Note:** The certificate files must use the format ".pem". To create custom certificate files to use on your IP phone, contact Aastra Technical Support.	
Format	<file name>.pem	
Default Value	N/A	
Range	N/A	
Example	sips root and intermediate certificates: cacert_openser.pem	

Parameter – *sips local certificate* *Local Certificate* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->TLS Support aastra.cfg, <mac>.cfg
Description	Allows you to specify the Local Certificate file to use when the phone uses the TLS transport protocol to setup a call. This parameter is required when configuring TLS (optional for Persistent TLS.) **Note:** The certificate file must use the format ".pem". To create specific certificate files to use on your IP phone, contact Aastra Technical Support.	
Format	<file name>.pem	
Default Value	N/A	
Range	N/A	
Example	sips local certificate: phonesLocalCert.pem	

Parameter – *sips private key* *Private Key* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->TLS Support aastra.cfg, <mac>.cfg
Description	Allows you to specify a Private Key file to use when the phone uses the TLS transport protocol to setup a call. This parameter is required when configuring TLS (optional for Persistent TLS.) **Note:** The key file must use the format ".pem". To create specific private key files to use on your IP phone, contact Aastra Technical Support.	
Format	<file name>.pem	
Default Value	N/A	
Range	N/A	
Example	sips private key: phone-privkey.pem	

Parameter – *sips trusted certificates* *Trusted Certificates* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->TLS Support aastra.cfg, <mac>.cfg
Description	Allows you to specify the Trusted Certificate files to use when the phone uses the TLS transport protocol to setup a call. The Trusted Certificate files define a list of trusted certificates. The phone's trusted list must contain the CA root certificates for all the servers it is connecting to. For example, if the phone is connecting to server A which has a certificate signed by CA1, and server B which has a certificate signed by CA2, the phone must have CA1 root certificate and CA2 root certificate in its Trusted Certificate file. This parameter is required when configuring TLS or Persistent TLS. **Note:** The certificate files must use the format ".pem". To create custom certificate files to use on your IP phone, contact Aastra Technical Support.	
Format	<file name>.pem	
Default Value	N/A	
Range	N/A	
Example	sips trusted certificates: trustedCert.pem	

RTP, Codec, DTMF Global Settings

Global Settings

Parameter – *sip rtp port* *RTP Port Base* (in IP Phone UI) *RTP Port* (in Web UI)	**IP Phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> SIP Settings->RTP Port Base Advanced Settings->Global SIP->RTP Settings aastra.cfg, <mac>.cfg
Description	Indicates the port through which the RTP packets are sent. This value must specify the beginning of the RTP port range on the gateway or router. 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: 3000	

Parameter – *sip use basic codecs* *Basic Codecs* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP->RTP Settings 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 m-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** **Configuration Files**	Advanced Settings->Global SIP->RTP Settings 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	

Parameter – *sip srtp mode* *RTP Encryption* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> RTP Settings aastra.cfg, <mac>.cfg
Description	This parameter determines if SRTP is enabled on this IP phone, as follows: • If set to 0, then disable SRTP. • If set to 1 then SRTP calls are preferred. • If set to 2, then SRTP calls only are generated/accepted.	
Format	Integer	
Default Value	0 (SRTP Disabled)	
Range	0 (SRTP Disabled) 1 (SRTP Preferred) 2 (SRTP Only)	
Example	sip srtp mode: 1	

Parameter – *sip silence suppression* *Silence Suppression* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP->RTP Settings 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	

Per-Line Settings

Parameter – *sip lineN dtmf method* *DTMF Method* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP->RTP Settings 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	

Parameter – *sip lineN srtp mode*	**Aastra Web UI** Advanced Settings->Line <1-9>->RTP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	This parameter determines if SRTP is enabled on this line, as follows: • If set to -1, then use the global setting for this line. (This is the default setting.) • If set to 0, then disable SRTP. • If set to 1 then SRTP calls are preferred. • If set to 2, then SRTP calls only are generated/accepted.
Format	Integer
Default Value	0 (disabled)
Range	-1 0 1 2
Example	sip line1 mode: 1

Autodial Settings

Global Settings

Parameter – *sip autodial number* *Autodial Number* (in Web UI)	**Aastra Web UI** Advanced Settings>Global SIP> Autodial Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Globally specifies the SIP phone number that the IP phone autodials when the handset is lifted from the phone cradle. An empty (blank) value disables autodial on the phone.
Format	Integer
Default Value	Blank
Range	Any valid SIP number
Examples	sip autodial number: 8500

Appendix A

Parameter – *sip autodial timeout* *Autodial Timeout* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings>Global SIP> Autodial Settings aastra.cfg, <mac>.cfg
Description	Globally specifies the time, in seconds, that the phone waits to dial a preconfigured number after the handset is lifted from the IP phone cradle. If this parameter is set to 0 (hotline), the phone immediately dials a preconfigured number when you lift the handset. If this parameter is set to a value greater than 0, the phone waits the specified number of seconds before dialing the preconfigured number (warmline) when you lift the handset. Default is 0 (hotline).	
Format	Integer	
Default Value	0	
Range	0 to 120	
Examples	sip autodial timeout: 30	

Per-Line Settings

Parameter – *sip lineN autodial number* *Autodial Number* (in Web UI)	**Aastra Web UI** Advanced Settings>LineN>Autodial Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	On a per-line basis, this parameter specifies the SIP phone number that the IP phone autodials when the handset is lifted from the phone cradle. Valid values can be: -1 (Default) The phone uses the global autodial setting for this line. Blank (Empty field) Disables autodial on this line. Valid SIP Number Dials the SIP number specified for this line.
Format	Integer
Default Value	-1
Range	Any valid SIP number.
Examples	sip line1 autodial number: 8500

Parameter – *sip lineN autodial timeout* *AutoDial Timeout* (in Web UI)	**Aastra Web UI** Advanced Settings>LineN>Autodial Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	On a per-line basis, this parameter specifies the time, in seconds, that the phone waits to dial a preconfigured number after the handset is lifted from the IP phone cradle. If this parameter is set to 0 (hotline), the phone immediately dials a preconfigured number when you lift the handset. If this parameter is set to a value greater than 0, the phone waits the specified number of seconds before dialing the preconfigured number (warmline) when you lift the handset. Default is 0 (hotline).
Format	Integer
Default Value	0
Range	0 to 120
Examples	sip line1 autodial timeout: 30

Appendix A

Voicemail Settings

Parameter – *sip lineN vmail* **Note:** The value of "*N*" is 1 - 9 for 53i, 55i, 57i, 57i CT.	**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** **Configuration Files**	Operation->Directory 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** **Configuration Files**	Operation->Directory 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 57i and 57i 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 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 57i and 57i 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

Customize Callers List and Services Key

Parameter – *services script*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a specific URI for accessing services after pressing the Services key. When this parameter is set, it overrides the standard function of the Services key.
Format	Alphanumeric characters
Default Value	N/A
Range	N/A
Example	services script: http://10.50.100.234/test.xml

Parameter – *callers list script*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a specific URI for accessing the Callers List after pressing the Callers List key. When this parameter is set, it overrides the standard function of the Callers List key.
Format	Alphanumeric characters
Default Value	N/A
Range	N/A
Example	callers list script: http://10.50.100.234/test.xml

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 Forwarding via the Aastra Web UI and the IP Phone UI using the "Call Forward" 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 Forwarding.
Format	Boolean
Default Value	0 (false)
Range	0 (false), 1 (true)
Example	call forward 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 (false)
Range	0 (false), 1 (true)
Example	missed calls indicator disabled: 1

Appendix A

XML Settings

Parameter – *xml get timeout*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a timeout value, in seconds, that the phone waits for the far side to return a response after accepting the HTTP GET connection. If the far side accepts the GET connection but never returns a response, it blocks the phone until it is rebooted. If you enter a value greater than 0 for this parameter, the phone times out and will not be blocked.
Format	Integer
Default Value	0 (never timeout)
Range	0 to 214748364 seconds
Example	xml get timeout: 20

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** **Configuration Files**	Operation->Softkeys and XML->Services 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 (Approved IP Addresses)* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Configuration Server 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	

Parameter – *xml beep notification* *XML Beep Support* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables a BEEP notification on the phone when a status message object (AastraIPPhoneStatus) containing a “beep” attribute arrives to the phone. Changes to this parameter are applied immediately.
Format	Boolean
Default Value	1 (ON)
Range	0 (OFF)No beep is audible even if the beep attribute is present in the XML object. 1 (ON)The phone beeps when an XML object with the “beep” attribute arrives to the phone.
Example	xml beep notification: 0

Parameter – *xml status scroll delay* *Status Scroll Delay (seconds)* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the length of time, in seconds, that each XML status message displays on the phone. **Note:** Changes to this parameter are applied immediately.
Format	Integer
Default Value	5
Range	1 to 25
Example	xml status scroll delay: 3

Action URI Settings

Parameter – *action uri startup* *Startup* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Action URI aastra.cfg, <mac>.cfg
Description	Specifies the URI for which the phone executes a GET on when a startup event occurs.	
Format	Fully qualified URI	
Default Value	Not Applicable	
Range	Up to 128 ASCII characters	
Example	action uri startup: http://10.50.10.140/startup	

Parameter – *action uri registered* *Successful Registration* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Action URI aastra.cfg, <mac>.cfg
Description	Specifies the URI for which the phone executes a GET on when a successful registration event occurs. This parameter can use the following variables: $$SIPUSERNAME$$ $$SIPAUTHNAME$$ $$PROXYURL$$ **Note:** The "**action uri registered**" parameter executes on the first successful registration of each unique line configured on the phone.	
Format	Fully qualified URI	
Default Value	Not Applicable	
Range	Up to 128 ASCII characters	
Example	action uri registered: http://10.50.10.14/registered.php?auth name=$$SIPAUTHNAME$$	

Parameter – *action uri incoming* *Incoming Call* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Action URI aastra.cfg, <mac>.cfg
Description	Specifies the URI for which the phone executes a GET on when an incoming call event occurs. This parameter can use the following variables: $$REMOTENUMBER$$ $$DISPLAYNAME$$ $$SIPUSERNAME$$ $$INCOMINGNAME$$	
Format	Fully qualified URI	
Default Value	Not Applicable	
Range	Up to 128 ASCII characters	
Example	action uri incoming: http://10.50.10.140/ incoming.php?number=$$REMOTENUMBER$$	

Parameter – *action uri outgoing* *Outgoing Call* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Action URI aastra.cfg, <mac>.cfg
Description	Specifies the URI for which the phone executes a GET on when an outgoing call event occurs. This parameter can use the following variables: $$REMOTENUMBER$$ $$SIPUSERNAME$$	
Format	Fully qualified URI	
Default Value	Not Applicable	
Range	Up to 128 ASCII characters	
Example	action uri outgoing: http://10.50.10.140/ outgoing.php?number=$$REMOTENUMBER$$	

Parameter – *action uri offhook* *Offhook* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Action URI aastra.cfg, <mac>.cfg
Description	Specifies the URI for which the phone executes a GET on when an offhook event occurs.	
Format	Fully qualified URI	
Default Value	Not Applicable	
Range	Up to 128 ASCII characters	
Example	action uri offhook: http://10.50.10.140/offhook	

Parameter – *action uri onhook* *Onhook* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Action URI aastra.cfg, <mac>.cfg
Description	Specifies the URI for which the phone executes a GET on when an onhook event occurs.	
Format	Fully qualified URI	
Default Value	Not Applicable	
Range	Up to 128 ASCII characters	
Example	action uri onhook: http://10.50.10.140/onhook	

Ring Tone and Tone Set Global Settings

Parameter – *ring tone* *Global Ring Tone* (in Web UI)	**IP Phone UI** **Aastra Web UI:** **Configuration Files**	Options->Tones->Set Ring Tone Basic Settings->Preferences->Ring Tones 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)	**IP Phone UI** Options->Tones->Tone Set **Aastra Web UI:** Basic Settings->Preferences->Ring Tones **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 Mexico United Kingdom (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:** **Configuration Files**	Basic Settings->Preferences->Ring Tones 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: Configuration Files:	Global -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	

Incoming Call Interrupts Dialing Setting

Parameter – *incoming call interrupts dialing* *Incoming Call Interrupts Dialing* (in Web UI)	**Aastra Web UI:** Basic Settings->Preferences->General **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enable or disables how the phone handles incoming calls while the phone is dialing out. When you enable this parameter (1 = true), an incoming call interrupts the outgoing call during dialing and allows the phone to ring for the user to answer the incoming call. When you disable this parameter (0 = false), which is the default, the phone does not interrupt the outgoing call during dialing and instead rings the incoming call on another free line (or sends busy signal if all remaining lines are busy). You have a choice to ignore the incoming call, or answer the incoming call on another line, via the **Ignore** and **Answer** softkeys that display. If you choose to answer the incoming call, you can answer the call, finish the call, and then hang up. You can still go back to the original outgoing call and finish dialing out. **Notes**: **1.** On a 53i, you must use the up and down arrow keys to ignore or answer the call. **2.** On all phone models, if you disable this parameter (0=disable), and the phone receives an incoming call while you are dialing an outgoing call, you can pick up the call and perform transfer or conference as required. After enabling or disabling this feature, it takes affect on the phone immediately.
Format	Boolean
Default Value	0 (false)
Range	0 (false) 1 (true)
Example	incoming call interrupts dialing: 1

Goodbye Key Cancels Incoming Call

Parameter – *goodbye cancels incoming call* *Goodbye Key Cancels Incoming Call* (in Web UI)	**Aastra Web UI:** Basic Settings->Preferences->General **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enable or disables the behavior of the Goodbye Key on the IP phone. When you enable this parameter (1 = enable), the Goodbye key rejects the incoming call. When you disable this parameter (0 = disable), the Goodbye key hangs up the active call. **For the 55i, 57i, and 57i CT:** If you enable this parameter, and the phone receives another call when an active call is already present, the phone displays softkey 1 as "**answer**" and softkey 2 as "**ignore**". You can press the required softkey as applicable. **For the 53i:** If you enable this parameter, and the phone receives another call when an active call is already present, the "**ignore**" option *only* displays in the LCD window. The phone ignores the incoming call. If you press the DOWN arrow key, the phone answers the incoming call. **Note:** After enabling or disabling this feature, it takes affect on the phone immediately.
Format	Boolean
Default Value	1 (true)
Range	0 (false) 1 (true)
Example	goodbye cancels incoming call: 0

Stuttered Dial Tone Setting

Parameter – *stutter disabled* *Stuttered Dial Tone* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences->General 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 Settings

Parameter – *call waiting* *Call Waiting* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences->General **Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to enable or disable Call Waiting on the IP phone. If you enable call waiting (default), the user has the option of accepting a second call while currently on the first call. If you disable call waiting, and a user is currently on a call, a second incoming call is automatically rejected by the phone with a busy message. If you disable call waiting on the phone, and the user is on a call, any further incoming calls will receive busy unless "Call Forward Busy" or "Call Forward No Answer and Busy" is configured on the phone. It will then forward the call according to the rule configured. The phone can only: • -transfer the currently active call or • accept transferred calls if there is no active calls. If call waiting is disabled: • on the 57i CT base, and the handset is currently on a call, all additional incoming calls are rejected on the handset. • intercom calls are treated as regular incoming calls and are rejected. • pre-dialing with live dial pad disabled still accepts incoming calls. • the "Incoming Call Cancels Dialing" parameter is ignored because the incoming call is automatically rejected. • the Missed Calls List does not get updated with details of calls. • the Blind Transfer feature on the phone may not work if two calls are made to the phone at one time.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	call waiting: 0

Appendix A

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. **Note:** The Call Waiting Tone feature works only if the Call Waiting parameter is enabled.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disable) 1 (enabled)
Example	call waiting tone: 0

Message Waiting Indicator Settings

Parameter – *mwi led line* *Message Waiting Indicator Line* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences->General **Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to enable the Message Waiting Indicator (MWI) on a single line or on all lines on the phone. For example, if you set this parameter to 3, the LED illuminates if a voice mail is pending on line 3. If you set this parameter to 0, the LED illuminates if a voice mail is pending on any line on the phone (lines 1 through 9). **Note:** To enable MWI for all lines in the configuration files, set this parameter to zero (0). The enable MWI for all lines in the Aastra Web UI, select “All” in the “Message Waiting Indicator Line” field.
Format	Integer
Default Value	0 (all lines)
Range	0 to 9
Example	mwi led line: 3

Priority Alert Settings

Parameter – *priority alerting enabled* *Enable Priority Alerting* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings 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 auto call distribution* *auto call distribution* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings aastra.cfg, <mac>.cfg
Description	When an "alert-acd" 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 auto call distribution: 2	

Appendix A

Parameter – *alert community 1* *community-1* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings aastra.cfg, <mac>.cfg
Description	When an "alert community-1" 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 community 1: 3	

Parameter – *alert community 2* *community-2* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings aastra.cfg, <mac>.cfg
Description	When an "alert community-2" 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 community 2: 4	

Parameter – *alert community 3* *community-3* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings aastra.cfg, <mac>.cfg
Description	When an "alert community-3" 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 community 3: 1	

Parameter – *alert community 4* *community-4* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings aastra.cfg, <mac>.cfg
Description	When an "alert community-4" 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 community 4: 2	

Parameter – *alert external* *alert external* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings 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 emergency* *alert emergency* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings 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 group* *Group* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings 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 internal* *alert internal* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings 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 priority* *alert priority* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Priority Alerting Settings 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* *Webpage Language* (in Web UI)	**IP Phone UI** Options->Language **Aastra Web UI** Basic Settings->Preferences-> Language Settings->Webpage Language **Configuration File** aastra.cfg, <mac>.cfg
Description	The language you want to display in the IP Phone UI and the Aastra Web UI. Valid values for 53i, 55i, 57i are: 0 (English) 1 (French) 2 (Spanish) 3 (German) 4 (Italian) Valid values for 57i CT are: 0 (English) 1 (French) 2 (Spanish) **Note:** All languages may not be available for selection. The available languages are dependant on the language packs currently loaded to the IP phone. For more information about loading language packs, see "Loading Language Packs" on page 5-21.
Format	Integer
Default Value	0
Range	0 to 4 (for 53i, 55i, 57i) 0 to 2 (for 57i CT)
Example	language: 2 language: 3 language: 4

Language Pack Settings

Parameter – *language N* *Language N* (in Web UI) Where "N" can be 1, 2, 3, or 4	**Aastra Web UI** Basic Settings->Preferences-> Language Settings **Configuration File** <mac>.cfg
Description	The language pack you want to load to the IP phone. Valid values are: **lang_fr_ca.txt** **lang_es_mx.txt** **lang_de.txt** **lang_it.txt** **Notes**: **1.** The languages packs you load are dependant on available language packs from the configuration server. For more information about loading language packs, see "Loading Language Packs" on page 5-21. **2.** You must reboot the phone to load a language pack.
Format	lang_<ISO 639>-<ISO 3166>.txt or lang_<ISO 639>.txt **Note:** For valid values for <ISO 639> and <ISO 3166>, see "Language Codes (from Standard ISO 639)" on page A-111 and "Country Codes (from Standard ISO 3166)" on page A-111.
Default Value	N/A
Range	N/A
Example	language 1: lang_fr_ca.txt language 2: lang_es_mx.txt language 3: lang_de.txt language 4: lang_it.txt

The following table identifies the language code to use for the IP phone language packs.

Language Codes (from Standard ISO 639)

Language	Language Code
English	en
French	fr-ca
Spanish	es
German	de
Italian	it

The following table identifies the country codes to use for the IP phone language packs.

Country Codes (from Standard ISO 3166)

Country	Country Code
AFGHANISTAN	AF
ÅLAND ISLANDS	AX
ALBANIA	AL
ALGERIA	DZ
AMERICAN SAMOA	AS
ANDORRA	AD
ANGOLA	AO
ANGUILLA	AI
ANTARCTICA	AQ
ANTIGUA AND BARBUDA	AG
ARGENTINA	AR
ARMENIA	AM
ARUBA	AW
AUSTRALIA	AU
AUSTRIA	AT
AZERBAIJAN	AZ

Country	Country Code
BAHAMAS	BS
BAHRAIN	BH
BANGLADESH	BD
BARBADOS	BB
BELARUS	BY
BELGIUM	BE
BELIZE	BZ
BENIN	BJ
BERMUDA	BM
BHUTAN	BT
BOLIVIA	BO
BOSNIA AND HERZEGOVINA	BA
BOTSWANA	BW
BOUVET ISLAND	BV
BRAZIL	BR
BRITISH INDIAN OCEAN TERRITORY	IO
BRUNEI DARUSSALAM	BN
BULGARIA	BG
BURKINA FASO	BF
BURUNDI	BI
CAMBODIA	KH
CAMEROON	CM
CANADA	CA
CAPE VERDE	CV
CAYMAN ISLANDS	KY
CENTRAL AFRICAN REPUBLIC	CF
CHAD	TD
CHILE	CL
CHINA	CN
CHRISTMAS ISLAND	CX
COCOS (KEELING) ISLANDS	CC
COLOMBIA	CO
COMOROS	KM
CONGO	CG
CONGO, THE DEMOCRATIC REPUBLIC OF THE	CD
COOK ISLANDS	CK
COSTA RICA	CR
CÔTE D'IVOIRE	CI
CROATIA	HR
CUBA	CU
CYPRUS	CY
CZECH REPUBLIC	CZ

Country	Country Code
DENMARK DJIBOUTI DOMINICA DOMINICAN REPUBLIC	DK DJ DM DO
ECUADOR EGYPT EL SALVADOR EQUATORIAL GUINEA ERITREA ESTONIA ETHIOPIA	EC EG SV GQ ER EE ET
FALKLAND ISLANDS (MALVINAS) FAROE ISLANDS FIJI FINLAND FRANCE FRENCH GUIANA FRENCH POLYNESIA FRENCH SOUTHERN TERRITORIES	FK FO FJ FI FR GF PF TF
GABON GAMBIA GEORGIA GERMANY GHANA GIBRALTAR GREECE GREENLAND GRENADA GUADELOUPE GUAM GUATEMALA GUERNSEY GUINEA GUINEA-BISSAU GUYANA	GA GM GE DE GH GI GR GL GD GP GU GT GG GN GW GY
HAITI HEARD ISLAND AND MCDONALD ISLANDS HOLY SEE (VATICAN CITY STATE) HONDURAS HONG KONG HUNGARY	HT HM VA HN HK HU

Country	Country Code
ICELAND INDIA INDONESIA IRAN, ISLAMIC REPUBLIC OF IRAQ IRELAND ISLE OF MAN ISRAEL ITALY	IS IN ID IR IQ IE IM IL IT
JAMAICA JAPAN JERSEY JORDAN	JM JP JE JO
KAZAKHSTAN KENYA KIRIBATI KOREA, DEMOCRATIC PEOPLE'S REPUBLIC OF KOREA, REPUBLIC OF KUWAIT KYRGYZSTAN	KZ KE KI KP KR KW KG
LAO PEOPLE'S DEMOCRATIC REPUBLIC LATVIA LEBANON LESOTHO LIBERIA LIBYAN ARAB JAMAHIRIYA LIECHTENSTEIN LITHUANIA LUXEMBOURG	LA LV LB LS LR LY LI LT LU

Country	Country Code
MACAO	MO
MACEDONIA, THE FORMER YUGOSLAV REPUBLIC OF	MK
MADAGASCAR	MG
MALAWI	MW
MALAYSIA	MY
MALDIVES	MV
MALI	ML
MALTA	MT
MARSHALL ISLANDS	MH
MARTINIQUE	MQ
MAURITANIA	MR
MAURITIUS	MU
MAYOTTE	YT
MEXICO	MX
MICRONESIA, FEDERATED STATES OF	FM
MOLDOVA, REPUBLIC OF	MD
MONACO	MC
MONGOLIA	MN
MONTENEGRO	ME
MONTSERRAT	MS
MOROCCO	MA
MOZAMBIQUE	MZ
MYANMAR	MM
NAMIBIA	NA
NAURU	NR
NEPAL	NP
NETHERLANDS	NL
NETHERLANDS ANTILLES	AN
NEW CALEDONIA	NC
NEW ZEALAND	NZ
NICARAGUA	NI
NIGER	NE
NIGERIA	NG
NIUE	NU
NORFOLK ISLAND	NF
NORTHERN MARIANA ISLANDS	MP
NORWAY	NO
OMAN	OM

Country	Country Code
PAKISTAN PALAU PALESTINIAN TERRITORY, OCCUPIED PANAMA PAPUA NEW GUINEA PARAGUAY PERU PHILIPPINES PITCAIRN POLAND PORTUGAL PUERTO RICO	PK PW PS PA PG PY PE PH PN PL PT PR
QATAR	QA
RÉUNION ROMANIA RUSSIAN FEDERATION RWANDA	RE RO RU RW

Country	Country Code
SAINT HELENA	SH
SAINT KITTS AND NEVIS	KN
SAINT LUCIA	LC
SAINT PIERRE AND MIQUELON	PM
SAINT VINCENT AND THE GRENADINES	VC
SAMOA	WS
SAN MARINO	SM
SAO TOME AND PRINCIPE	ST
SAUDI ARABIA	SA
SENEGAL	SN
SERBIA	RS
SEYCHELLES	SC
SIERRA LEONE	SL
SINGAPORE	SG
SLOVAKIA	SK
SLOVENIA	SI
SOLOMON ISLANDS	SB
SOMALIA	SO
SOUTH AFRICA	ZA
SOUTH GEORGIA AND THE SOUTH SANDWICH ISLANDS	GS
SPAIN	ES
SRI LANKA	LK
SUDAN	SD
SURINAME	SR
SVALBARD AND JAN MAYEN	SJ
SWAZILAND	SZ
SWEDEN	SE
SWITZERLAND	CH
SYRIAN ARAB REPUBLIC	SY
TAIWAN, PROVINCE OF CHINA	TW
TAJIKISTAN	TJ
TANZANIA, UNITED REPUBLIC OF	TZ
THAILAND	TH
TIMOR-LESTE	TL
TOGO	TG
TOKELAU	TK
TONGA	TO
TRINIDAD AND TOBAGO	TT
TUNISIA	TN
TURKEY	TR
TURKMENISTAN	TM
TURKS AND CAICOS ISLANDS	TC
TUVALU	TV

Country	Country Code
UGANDA UKRAINE UNITED ARAB EMIRATES UNITED KINGDOM UNITED STATES UNITED STATES MINOR OUTLYING ISLANDS URUGUAY UZBEKISTAN	UG TA AE GB US TM UY UZ
VANUATU Vatican City State VENEZUELA VIET NAM VIRGIN ISLANDS, BRITISH VIRGIN ISLANDS, U.S.	VU see HOLY SEE VE VN VG VI
WALLIS AND FUTUNA WESTERN SAHARA	WF EH
YEMEN	YE
Zaire ZAMBIA ZIMBABWE	see CONGO, THE DEMOCRATIC REPUBLIC OF THE ZM ZW

Suppress DTMF Playback Setting

Parameter – *suppress dtmf playback* *Suppress DTMF Playback* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences **Configuration Files** 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

Display DTMF Digits Setting

Parameter – *display dtmf digits* *Display DTMF Digits* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences->General aastra.cfg, <mac>.cfg
Description	Enables and disables the display of DTMF digits when dialing on the IP phone. DTMF is the signal sent from the phone to the network that you generate when you press the phone's touch keys. This is also known as "touchtone" dialing. Each key you press on your phone generates two tones of specific frequencies. One tone is generated from a high-frequency group of tones and the other from a low frequency group. If enabled, this parameter displays the digits on the IP phone display if you are dialing from the keypad, or from a softkey or programmable key. This parameter is disabled by default (no digits display when dialing).	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	display dtmf digits: 1	

Intercom, Auto-Answer, and Barge In Settings

Outgoing Intercom Settings

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 **For Configuration Files:** 3 - Off	
Range	**For Aastra Web UI:** Phone-Side Server-Side Off **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** **Configuration Files**	Basic Settings->Preferences-> Outgoing Intercom Settings 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 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	

Parameter – *sip intercom mute mic* *Microphone Mute* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences-> Incoming Intercom Settings 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 (true)	
Range	0 (false - microphone is not muted) 1 (true - microphone is muted)	
Example	sip intercom mute mic: 1	

Parameter – *sip play warning tone* *Play Warning Tone* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences-> Incoming Intercom Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables a warning tone to play when the phone receives an incoming intercom call on an active line.
Format	Integer
Default Value	1 (true)
Range	0 (false - warning tone will not play) 1 (true - warning tone will play)
Example	sip play warning tone: 0

Parameter – *sip intercom allow barge in* *Allow Barge In* (in Web UI)	**Aastra Web UI:** Basic Settings->Preferences-> Incoming Intercom Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enable or disables how the phone handles incoming intercom calls while the phone is on an active call. When you enable this parameter (1 = enable), which is the default value, an incoming intercom call takes precedence over any active call, by placing the active call on hold and automatically answering the intercom call. When you disable this parameter (0 = disable), and there is an active call, the phone treats an incoming intercom call like a normal call and plays the call warning tone. **Note:** After enabling or disabling this feature, it takes affect on the phone immediately.
Format	Boolean
Default Value	1 (true)
Range	0 (false) 1 (true)
Example	sip intercom allow barge in: 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 d /fkey. 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 d /fkey. **2** **Speaker/headset** - Incoming calls are sent to the speakerphone . By pressing the d /f 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 d /fkey, you can switch between the headset, the handsfree speakerphone, and the handset.
Example	audio mode: 2

Directed Call Pickup (BLF or XML 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 – *directed call pickup prefix* *Directed Call Pickup by Prefix* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences-> Directed Call Pickup Settings aastra.cfg, <mac>.cfg
Description	Allows you to enter a specific prefix string (depending on what is available on your server), that the phone automatically dials when dialing the Directed Call Pickup number. For example, for Broadsoft servers, you can enter a value of ***98** for the "**directed call pickup prefix**". When the phone performs the Directed Call Pickup after pressing a BLF or BLF/List softkey, the phone prepends the *98 value to the designated extension of the BLF or BLF/List softkey when dialing out. **Notes:** **1.** The default method for the phone to use is Directed Call Pickup over BLF if the server provides applicable information. If the Directed Call Pickup over BLF information is missing in the messages to the server, the Directed Call Pickup by Prefix method is used if a value for the prefix code exists in the configuration. **2.** You can define only one prefix at a time for the entire BLF/List. **3.** The phone that picks up displays the prefix code + the extension number (for example, *981234 where prefix key = *98, extension = 1234). **4.** Symbol characters are allowed (for example "*").	
Format	Integer	
Default Value	Not Applicable	
Range	Not Applicable	
Example	directed call pickup prefix: *98	

Parameter – *play a ring splash* *Play a Ring Splash* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences ->Directed Call Pickup Settings 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	

ACD Auto-Available Timer Settings

Parameter – *acd auto available* *Auto Available* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences ->Auto Call Distribution Settings aastra.cfg, <mac>.cfg
Description	Enables or disables the use of the ACD Auto-Available Timer.	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	acd auto available: 1	

Parameter – *acd auto available timer* *Auto Available Timer* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences ->Auto Call Distribution Settings aastra.cfg, <mac>.cfg
Description	Specifies the length of time, in seconds, before the IP phone status switches back to “available.”	
Format	Integer	
Default Value	60 (seconds)	
Range	0 to 120 (seconds)	
Example	acd auto available timer: 60	

Park and Pickup Global Settings (57i/57i CT only)

Parameter – *sprecode* *Park Call* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the code to enter before entering the extension for where you want to park an incoming call. The applicable value is dependant on the type of server in the network: **Server** **Park Values*** **Asterisk** 70 **Sylantro** *98 **BroadWorks** *68 **ININ PBX** callpark ***Leave "value" fields blank to disable the park and pickup feature.
Format	Alphanumeric characters
Default Value	<Blank>
Range	See applicable values in the table above.
Example	sprecode: *68

Parameter – *pickupsprecode* *Pick Up Parked Call* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the code to enter before entering the extension for where you want to pickup a parked call. The applicable value is dependant on the type of server in the network: **Server** **Pickup Values*** **Asterisk** 70 **Sylantro** *99 **BroadWorks** *88 **ININ PBX** pickup ***Leave "value" fields blank to disable the park and pickup feature.
Format	Alphanumeric characters
Default Value	<Blank>
Range	See applicable values in the table above.
Example	pickupsprecode: *88

Mapping 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 speeddial 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 57i CT Base Station, the **Redial** key on the 57i 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** **Configuration Files**	Basic Settings->Preferences->Key Mapping aastra.cfg, <mac>.cfg
Description	Sets the **Conf** key as a speeddial 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 57i CT Base Station, the **Conf** key on the 57i 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	

Appendix A

Softkey/Programmable Key/Feature Key/ Expansion Module Key Parameters

This section provides the softkey, programmable key, feature key, and expansion module key parameters you can configure on the IP phones. The following table provides the number of keys you can configure for each model phone and expansion module, and the number of lines available for each type of phone.

IP Phone Model	Softkeys	Expansion Module Keys	Programmable Keys	Lines Available	Handset Keys Available
53i	-	36 to 108* (Model 536M)	6 (Up to 4 functions)	9	-
55i	6 Bottom Keys (Up to 20 functions)	36 to 108* (Model 536M) 60 to 180** (Model 560M)	6 Top Keys (Up to 6 functions)	9	-
57i	12 Top and Bottom Softkeys (Up to 10 functions on top keys; Up to 20 functions on bottom keys)	36 to 108* (Model 536M) 60 to 180** (Model 560M)	-	9	-
57i CT	12 Top and Bottom Softkeys (Up to 10 functions on top keys; Up to 20 functions on bottom keys)	36 to 108* on Base Station 60 to 180** on Base Station (Model 560M)	-	9	15

*The 536M expansion module consists of 36 softkeys. You can have up to 3 expansion modules on an IP phone totaling 108 softkeys. Valid for 53i, 55i, 57i, and 57i CT phones.

**The 560M expansion module consists of 60 softkeys. You can have up to 3 expansion modules on an IP phone totaling 180 softkeys. Valid for 55i, 57i, and 57i CT phones.

Note: When entering definitions for softkeys, the "#" sign must be enclosed in quotes.

Softkey Settings for 55i, 57i, 57i CT

The value of "*N*" for the following parameters is dependent on the number of softkeys available on the55i, 57i, and 57i 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: • **none** - Indicates softkey and/or programmable key is disabled. • **line** - Indicates softkey and/or programmable key is configured for line use. • **speeddial** - Indicates softkey and/or programmable key is configured for speeddial use. Speeddial is applicable to the 536M and 560M also. You can configure a softkey to speeddial a specific number by pressing that softkey. Optionally, you can also configure a speeddial key to dial prefix numbers. With this option, the prefix numbers automatically dial when you press the softkey, and the phone waits for you to enter the remaining numbers to dial out. **Note**: When there is an active call, the speeddial keys send DTMF digits through the active voice path. To dial out, you have to first put the active call on hold and then press the speeddial key. • **dnd** - Indicates softkey and/or programmable key is configured for do not disturb on the phone. This option is "**Do Not Disturb**" in the Aastra Web UI). • **blf** - Indicates softkey and/or programmable key is configured for Busy Lamp Field (BLF) use. User can dial out on a BLF configured key. Maximum of 50 BLFs are applicable to the 536M and 560M also. • **list** - Indicates softkey and/or programmable key 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. • **acd** - (for Sylantro Servers only) Indicates the programmable key is configured for auto call distribution (called "**Auto call distribution**" in the Aastra Web UI). The ACD feature allows the Sylantro server to distribute calls from a queue to registered IP phone users (agents). • **dcp** - (for Sylantro Servers only) Indicates the programmable key is configured for either directed call pickup or group call pickup (called "**Directed Call Pickup**" in the Aastra Web UI). The Directed Call Pickup/Group Call Pickup feature allows you to intercept or pickup a call on a monitored extension or a group of monitored extensions. • **xml** - Indicates the softkey and/or programmable key is configured to accept an XML application for accessing customized XML services. You can also specify an XML softkey URL for this option. •

	• **flash** - Indicates the softkey and/or programmable key is set to generate a flash event when it is pressed on the 57i and 57i CT, or a feature key is pressed on the 57i CT handset. 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). • **sprecode** - Indicates the softkey and/or programmable key is configured to automatically activate specific services offered by the server. For example, if the sprecode value of *82 is configured, then by pressing the softkey, *82 automatically activates a service provided by the server. • **park** - Indicates the softkey and/or programmable key is configured to park incoming calls when pressed. • **pickup** - Indicates the softkey and/or programmable key is configured to pick up parked calls when pressed. • **lcr** - Indicates the softkey and/or programmable key is configured for “last call return” when pressed. • **directory** - Indicates the softkey and/or programmable key is set for accessing the Directory List. • **callers list** - Indicates the softkey and/or programmable key is set for accessing the Callers List. • **icom** - Indicates the softkey and/or programmable key is set to be used as the Intercom key. • **services** - Indicates the softkey and/or programmable key is set to be used as the Services key. • **phonelock** - Indicates the softkey and/or programmable key is set to be used to lock/unlock the phone. • **empty** - Indicates the softkey and/or programmable key is configured to force a blank entry on the IP phone display for a specific softkey. The softkeys 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	none line speeddial dnd ("do not disturb" in the Aastra Web UI) blf list ("BLF\List" in the Aastra Web UI) acd (“Auto call distribution” in the Aastra Web UI) dcp (“Directed Call Pickup” in the Aastra Web UI) xml flash sprecode park pickup lcr callers list directory icom services phonelock empty
Example	softkey1 type: line softkey2 type: speeddial softkey3 type: lcr softkey4 type: xml **Directed Call Pickup on Extension 2200** softkey2 type: dcp softkey2 label: dcp2200 softkey2 value: 2200 softkey2 states: incoming outgoing idle connected **Group Call Pickup on group_A** softkey3 type: dcp softkey3 label: gcp_A softkey3 value: groupcallpickup softkey3 states: incoming outgoing idle connected

Parameter – *softkeyN label* *Label* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Softkeys and XML aastra.cfg, <mac>.cfg
Description	The text label that displays on the IP phone for the softkey. The "**softkeyN label**" parameter can be set for the following softkey types only: • speeddial • BLF • acd • dcp • XML • Flash • sprecode • Park • Pickup • Directory • Callers List • Icom • Services **Notes**: **1.** For the 57i and 57i 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" softkey3 label: flash softkey4 label: "johnsmith"	

Parameter – *softkeyN value* *Value* (In WEb UI)	**Aastra Web UI** Operation->Softkeys and XML **Configuration Files** aastra.cfg, <mac>.cfg
Description	This is the value you assign to the softkey. The "**softkeyN value**" parameter can be set for the following softkey types only: • speeddial (you can enter a speeddial number for this field; optionally, you can also enter a prefix for the speeddial value to allow the phone to dial the prefix after you press the speeddial softkey; you then enter the rest of the number from the keypad on the phone.) • BLF • sprecode • Park • Pickup • dcp • XML **Notes**: **1.** For speeddial - Value is the phone number, extension, or prefix number to enter for the softkey. **2.** For blf - Value is the extension you want to monitor. **3.** For sprecode - Value is dependent on services offered by server. **4.** For Park, Pickup - For valid values, see Chapter 5, the section, "Park/Pickup Call Server Configuration Values" on page 5-157. For Park/Pickup examples, see Chapter 5, the section, "Model 57i/57i CT Examples" on page 5-155. **5.** For xml - You can specify a URI to use for this XML softkey. The variables you can use with the XML softkey URI are: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$INCOMINGNAME$$
Format	Integer
Default Value	Not Applicable
Range	N/A
Example	softkey1 value: 9 softkey2 value: 411 softkey4 value: http://10.50.10.140 script.pl?name=$$SIPUSERNAME$$ softkey5 value: 123456+ (example of a speeddial prefix)

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. The "**softkeyN line**" parameter can be set for the following softkey types only: • speeddial • BLF • BLF/List • acd • dcp • Park • Pickup • lcr	
Format	Integer	
Default Value	1	
Range	1 through 9	
Example	softkey1 line: 1 softkey2 line: 5	

Parameter – *softkeyN states* *Idle, Connected, Incoming, Outgoing, Busy* (in Web UI)	**Aastra Web UI** Operation->Softkeys and XML **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, busy**) 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 type - None:** All states disabled **For softkey types - Line, DND, speeddial, BLF, BLF List, dcp, XML, lcr, Directory, Callers List, Icom, Services, empty**: idle, connected, incoming, outgoing **For softkey type - Flash**: All states disabled **For softkey type - Sprecode, Park**: connected **For softkey type - Pickup:** idle, outgoing **For softkey type - acd:** idle

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. **busy** The current line is busy because the line is in use or the line is set as “Do Not Disturb” **Note:** For softkey type, Pickup, values can be: just idle, just outgoing, or idle outgoing.
Example	softkey1 states: idle incoming outgoing softkey2 states: connected

Programmable Key Settings for 53i and 55i

- The value of "*N*" for the following parameters is dependent on the number of programmable keys available on the 53i models. See the table on page 136 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: • **none -** Indicates no setting for programmable key. • **line** - Indicates programmable key is configured for line use. • **speeddial** - Indicates programmable key is configured for speeddial use. You can configure a programmable key to speeddial a specific number by pressing that key. Optionally, you can also configure a speeddial key to dial prefix numbers. With this option, the prefix numbers automatically dial when you press the programmable key, and the phone waits for you to enter the remaining numbers to dial. **Note**: When there is an active call, the speeddial keys send DTMF digits through the active voice path. To dial out, you have to first put the active call on hold and then press the speeddial key. • **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** - Indicates programmable key is configured for Busy Lamp Field (BLF) use. User can dial out on a BLF configured key. • **list** - Indicates programmable key is configured for BLF list use. User can dial out on a BLF List configured key. • **acd** - (for Sylantro Servers only) Indicates the programmable key is configured for auto call distribution (called "**Auto Call Distribution**" in the Aastra Web UI). The ACD feature allows the Sylantro server to distribute calls from a queue to registered IP phone users (agents). • **dcp** - (for Sylantro Servers only) Indicates the programmable key is configured for either directed call pickup or group call pickup (called "**Directed Call Pickup**" in the Aastra Web UI). The Directed Call Pickup/Group Call Pickup feature allows you to intercept or pickup a call on a monitored extension or a group of monitored extensions. • **xml** - Indicates programmable key is configured to accept an XML application for accessing customized XML services. You can also specify an XML softkey URL for this option. • **flash** - Indicates programmable key is set to generate a flash event when it is pressed on the 53i. 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).

	• **sprecode** - Indicates programmable key is configured to automatically activate specific services offered by the server. For example, if the sprecode value of *82 is configured, then by pressing the softkey, *82 automatically activates a service provided by the server. • **park** - Indicates programmable key is configured to park incoming calls when pressed. • **pickup** - Indicates programmable key is configured to pick up parked calls when pressed. • **lcr** - Indicates programmable key is configured for “last call return” when pressed. • **directory** - Indicates programmable key is configured to access the Directory List. • **callers list** - Indicates programmable key is configured to access the Callers List. • **conference** - Indicates programmable key is configured as a conference key. Enter as “**conf**” in configuration files. • **transfer** - Indicates programmable key is configured as a Transfer key for transferring calls. Enter as “**xfer**” in configuration files. • **phonelock** - Indicates the programmable key is set to be used to lock/unlock the phone. • **empty** - Indicates programmable key is configured to force a blank entry on the IP phone display for a specific programmable key. **Note:** Keys 1 and 2 on the 53i are hardcoded as the **Save** and **Delete** keys, respectively, and are not configurable.
Format	Text
Default Value	Not Applicable

Range	none line speeddial dnd ("do not disturb" in the Aastra Web UI) blf list (“BLF/List” in the Aastra Web UI) acd (“Auto call distribution” in the Aastra Web UI) dcp (“Directed Call Pickup” in the Aastra Web UI) xml flash sprecode park pickup lcr directory callers list conf xfer phonelock empty
Example	prgkey3 type: speeddial

Appendix A

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. The "**prgkeyN value**" parameter can be set for the following softkey types only: • speeddial (you can enter a speeddial number for this field; optionally, you can also enter a prefix for the speeddial value to allow the phone to dial the prefix after you press the speeddial programmable key; you then enter the rest of the number from the keypad on the phone.) • line • BLF • sprecode • dcp • XML • Park • Pickup **Notes**: **1.** For speeddial - Value is the phone number, extension, or prefix number to enter for the programmable key. **2.** For line - Value is optional; for example L4. **3.** For blf - Value is the extension you want to monitor. **4.** For sprecode - Value is dependent on services offered by server. **5.** For xml - Value is IP address of the XML application. **6.** For Park, Pickup - For valid values, see Chapter 5, the section, "Park/Pickup Call Server Configuration Values" on page 5-157. For Park/Pickup examples, see Chapter 5, the section "Model 53i Examples" on page 5-156.
Format	Integer
Default Value	N/A
Range	N/A
Example	prgkey3 value: 411 prgkey4 value: 123456+ (example of a speeddial prefix)

Parameter– *prgkeyN line* *Line* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Programmable Keys aastra.cfg, <mac>.cfg
Description	This is the line associated with the programmable key you are configuring. The "**prgkeyN line**" parameter can be set for the following softkey types only: • speeddial • BLF • BLF/List • acd • dcp • Park • Pickup • lcr	
Format	Integer	
Default Value	1	
Range	1 through 9	
Example	prgkey3 line: 1 prgkey4 line: 5	

Top Softkey Settings for 57i and 57i CT

Parameter – *topsoftkeyN type* *Top Softkeys->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: • **none** - Indicates softkey is disabled. • **line** - Indicates softkey is configured for line use. • **speeddial** - Indicates softkey is configured for speeddial use. You can configure a softkey to speeddial a specific number by pressing that softkey. Optionally, you can also configure a speeddial key to dial prefix numbers. With this option, the prefix numbers automatically dial when you press the softkey, and the phone waits for you to enter the remaining numbers to dial out. **Note**: When there is an active call, the speeddial keys send DTMF digits through the active voice path. To dial out, you have to first put the active call on hold and then press the speeddial key. • **dnd** - Indicates softkey is configured for do not disturb on the phone. This option is "**Do Not Disturb**" in the Aastra Web UI). • **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. • **acd** - (for Sylantro Servers only) Indicates the programmable key is configured for auto call distribution (called "**Auto call distribution**" in the Aastra Web UI). The ACD feature allows the Sylantro server to distribute calls from a queue to registered IP phone users (agents). • **dcp** - (for Sylantro Servers only) Indicates the programmable key is configured for either directed call pickup or group call pickup (called "**Directed Call Pickup**" in the Aastra Web UI). The Directed Call Pickup/Group Call Pickup feature allows you to intercept or pickup a call on a monitored extension or a group of monitored extensions. • **xml** - Indicates the softkey is configured to accept an XML application for accessing customized XML services. You can also specify an XML softkey URL for this option. • **flash** - Indicates the softkey is set to generate a flash event when it is pressed on the 57i and 57i CT, or a feature key is pressed on the 57i CT handset. 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).

	• **sprecode** - Indicates the softkey is configured to automatically activate specific services offered by the server. For example, if the sprecode value of *82 is configured, then by pressing the softkey, *82 automatically activates a service provided by the server. • **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. • **lcr** - Indicates the softkey is configured for “last call return” when pressed. • **callers list** - Indicates the softkey is set for accessing the Callers List. • **directory** - Indicates the softkey is set for accessing the Directory List. • **icom** - Indicates the softkey is set to be used as the Intercom key. • **services** - Indicates the softkey is set to be used as the Services key. • **phonelock** - Indicates the softkey is set to be used to lock/unlock the phone. • **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	none line speeddial dnd blf list ("BLF\List" in the Aastra Web UI) acd (“Auto call distribution” in the Aastra Web UI) dcp (“Directed Call Pickup” in the Aastra Web UI) xml flash sprecode park pickup lcr callers list directory icom services phonelock empty

Example	topsoftkey1 type: line topsoftkey2 type: speeddial topsoftkey3 type: lcr topsoftkey4 type: xml

Parameter – *topsoftkeyN label* *Top Softkeys->Label* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Softkeys and XML aastra.cfg, <mac>.cfg
Description	The text label that displays on the IP phone for the softkey. The "**topsoftkeyN label**" parameter can be set for the following softkey types only: • speeddial • BLF • acd • dcp • XML • Flash • sprecode • Park • Pickup • Directory • Callers List • Icom • Services **Notes**: **1.** For the 57i and 57i CT phones, an icon appears beside the soft key label that indicates the status of the line. **2.** If the *topsoftkeyN type* parameter is set to "**flash**", and no label value is entered for the top*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	topsoftkey1 label: "Line 9" topsoftkey2 label: "info" topsoftkey4 label: "johnsmith"	

Parameter – *topsoftkeyN value* *Top Softkeys->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. The "**topsoftkeyN value**" parameter can be set for the following softkey types only: • speeddial (you can enter a speeddial number for this field; optionally, you can also enter a prefix for the speeddial value to allow the phone to dial the prefix after you press the speeddial programmable key; you then enter the rest of the number from the keypad on the phone.) • BLF • sprecode • Park • Pickup • dcp • XML **Notes**: **1.** For speeddial - Value is the phone number, extension, or prefix number to enter for the softkey. **2.** For blf - Value is the extension you want to monitor. **3.** For sprecode - Value is dependent on services offered by server. **4.** For Park, Pickup - For valid values, see Chapter 5, the section, "Park/Pickup Call Server Configuration Values" on page 5-157. For Park/Pickup examples, see Chapter 5, the section, "Model 57i/57i CT Examples" on page 5-155. **5.** For xml - You can specify a URI to use for this XML softkey. The variables you can use with the XML softkey URI are: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$INCOMINGNAME$$	
Format	Integer	
Default Value	Not Applicable	
Range	N/A	
Example	topsoftkey1 value: 9 topsoftkey2 value: 411 topsoftkey4 value: http://10.50.10.140 script.pl?name=$$SIPUSERNAME$$ topsoftkey5 value: 12345+ (example of a speeddial prefix)	

Parameter – *topsoftkeyN line* *Top Softkeys->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. The "**topsoftkeyN line**" parameter can be set for the following softkey types only: • speeddial • BLF • BLF/List • acd • dcp • Park • Pickup • lcr	
Format	Integer	
Default Value	1	
Range	1 through 9	
Example	topsoftkey1 line: 1 topsoftkey2 line: 5	

Handset Feature Key Settings for the 57i CT

Parameter – *featurekeyN type* *Type* (in Web UI)	**Aastra Web UI** Operation->Handset Keys **Configuration Files** aastra.cfg, <mac>.cfg
Description	The type of feature key to configure. Valid types are: • **none** - Indicates feature key is disabled. • **line** - Indicates the feature key is configured for line use. • **transfer**- Indicates feature key is configured for transferring a call. • **conference** - Indicates feature key is configured for conference calling. • **public -** Indicates feature key is configured to toggle from public to private mode. A public and private softkey can be used when at a line item in the Directory List. 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 57i CT accepts a maximum of 50 entries with the public attribute. • **icom** - Indicates the feature key is set to be used to make an intercom call. • **directory** - Indicates the feature key is set for accessing the Directory List. • **callers list** - Indicates the feature key is set for accessing the Callers List. • **park**- Indicates the feature key is configured to park incoming calls when pressed. • **pickup**- Indicates the feature key is configured to pick up parked calls when pressed. • **flash** - Indicates the feature key is set to generate a flash event when it is pressed on the 57i CT handset. 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).
Format	Text
Default Value	None

Range	none line transfer conference public icom directory callers list park pickup flash
Example	featurekey1 type: line featurekey2 type: public featurekey3 type: park featurekey4 type: pickup

Parameter – *featurekeyN label* *Label* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Handset Keys aastra.cfg, <mac>.cfg
Description	The text label that displays on the IP phone for the feature key. **Notes**: **1.** For the 57i CT phones, an icon appears beside the feature key label that indicates the status of the line. **2.** If a feature key is configured but no label is set, the IP phone sets the label to the English, French, or Spanish translation of the chosen action. The language used is based on the current language of the cordless handset.	
Format	Text	
Default Value	Not Applicable	
Range	Not Applicable	
Example	featurekey1 label: Line 9 featurekey2 label: Public featurekey4 label: John Smith	

Appendix A

Expansion Module Key Settings for 536M (for all model phones) and 560M (for 55i, 57i, 57i CT phones only)

<table>
<tr><td>Parameter –
expmodX keyN type

Type
(in Web UI)</td><td>Aastra Web UI
Configuration Files</td><td>Operation->Expansion Module N
aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td colspan="2">The type of softkey to configure. Valid types are:
• none - Indicates softkey is disabled.
• line - Indicates softkey is configured for line use.
• speeddial - Indicates softkey is configured for speeddial use. You can configure a softkey to speeddial a specific number by pressing that softkey. Optionally, you can also configure a speeddial key to dial prefix numbers. With this option, the prefix numbers automatically dial when you press the softkey, and the phone waits for you to enter the remaining numbers to dial out.
Note: When there is an active call, the speeddial keys send DTMF digits through the active voice path. To dial out, you have to first put the active call on hold and then press the speeddial key.
• dnd - Indicates softkey is configured for do not disturb on the phone. This option is "Do Not Disturb" in the Aastra Web UI).
• 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.
• acd - (for Sylantro Servers only) Indicates the programmable key is configured for auto call distribution (called “Auto call distribution” in the Aastra Web UI). The ACD feature allows the Sylantro server to distribute calls from a queue to registered IP phone users (agents).
• dcp - (for Sylantro Servers only) Indicates the programmable key is configured for either directed call pickup or group call pickup (called “Directed Call Pickup” in the Aastra Web UI). The Directed Call Pickup/Group Call Pickup feature allows you to intercept or pickup a call on a monitored extension or a group of monitored extensions.
• xml - Indicates the softkey is configured to accept an XML application for accessing customized XML services. You can also specify an XML softkey URL for this option.
• lcr - Indicates the softkey is configured for “last call return” when pressed.</td></tr>
</table>

Format	• **callers list** - Indicates the softkey is set for accessing the Callers List. • **directory** - Indicates the softkey is set for accessing the Directory List. • **icom** - Indicates the softkey is set to be used as the Intercom key. • **services** - (not available on the 53i) Indicates the softkey is set to be used as the Services key. • **phonelock** - Indicates the softkey is set to be used to lock/unlock the phone. • **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.
Default Value	none
Range	none line speeddial dnd blf list ("BLF\List" in the Aastra Web UI) acd (“Auto call distribution” in the Aastra Web UI) dcp (“Directed Call Pickup” in the Aastra Web UI) xml lcr callers list directory icom services (not available on the 53i) phonelock empty
Example	expmod1 key1 type: line expmod1 key2 type: speeddial expmod1 key3 type: blf expmod1 key4 type: list

Parameter – *expmodX keyN label* *Label* (in Web UI)	**Aastra Web UI** Operation->Expansion Module N **Configuration Files** aastra.cfg, <mac>.cfg
Description	The text label that displays on the softkey for the Expansion Module. The "**expmodX keyN label**" parameter can be set for the following softkey types only: • speeddial • BLF • acd • dcp • XML • Directory • Callers List • Icom • Services **Note:** For the 57i and 57i CT phones, an icon appears beside the soft key label that indicates the status of the line.
Format	Text
Default Value	Not Applicable
Range	For line, blf types - Up to 9 characters. For speeddial type - Up to 11 characters.
Example	expmod1 key 1 label: "Line 9" expmod2 key 1 label: "info" expmod3 key 1 label: "johnsmith"

Parameter – *expmodX keyN value* *Value* (In WEb UI)	**Aastra Web UI** **Configuration Files**	Operation->Expansion Module N aastra.cfg, <mac>.cfg
Description	The text label that displays on the IP phone for the softkey on the Expansion Module. The "**expmodX keyN value**" parameter can be set for the following softkey types only: • speeddial • blf • dcp • XML • directory • callers list • Icom • Services (not available on 53i) **Notes**: **1.** For the 57i and 57i CT phones, an icon appears beside the soft key label that indicates the status of the line. **2.** For blf - Value is the extension you want to monitor.	
Format	Integer	
Default Value	Not Applicable	
Range	N/A	
Example	expmod1 key1 value: 9 expmod1 key2 value: 411 expmod1 key3 value: 123456+ (example of a speeddial prefix)	

Parameter – *expmodX keyN line* *Line* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Expansion Module N aastra.cfg, <mac>.cfg
Description	This is the line associated with the softkey you are configuring on the Expansion Module. The number of applicable lines available is dependent on the specific IP phone model. The "**expmodX keyN line**" parameter can be set for the following softkey types only: • speeddial • BLF • BLF/List • acd • dcp • lcr	
Format	Integer	
Default Value	1	
Range	1 through 9	
Example	expmod1 key1 line: 1 expmod1 key2 line: 5	

Locking Softkeys and Programmable Keys

Parameter– *softkeyN locked*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Locks the specified softkey on the 55i, 57i, or 57i CT IP phone. Locking the key prevents a user from changing or configuring the softkey. When a key is locked, the phone uses the server settings and ignores any previous local configuration. Affects the following parameters: • softkeyN type • softkeyN label • softkeyN value • softkeyN line • softkeyN states
Format	Boolean
Default Value	0 (disable)
Range	0 (disable) 1 (enable)
Example	softkey1 locked: 1

Parameter– *topsoftkeyN locked*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Locks the specified top softkey on the 57i or 57i CT IP phone. Locking the key prevents a user from changing or configuring the softkey. When a key is locked, the phone uses the server settings and ignores any previous local configuration. Affects the following parameters: • topsoftkeyN type • topsoftkeyN label • topsoftkeyN value • topsoftkeyN line
Format	Boolean
Default Value	0 (disable)
Range	0 (disable) 1 (enable)
Example	topsoftkey1 locked: 1

Parameter– *prgkeyN locked*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Locks the specified top softkey on the 53i or 55i IP phone. Locking the key prevents a user from changing or configuring the programmable key. When a key is locked, the phone uses the server settings and ignores any previous local configuration. Affects the following parameters: • prgkeyN type • prgkeyN value • prgkeyN line
Format	Boolean
Default Value	0 (disable)
Range	0 (disable) 1 (enable)
Example	prgkey1 locked: 1

Parameter– *featurekeyN locked*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Locks the specified top softkey on the 57i CT IP phone. Locking the key prevents a user from changing or configuring the feature key. When a key is locked, the phone uses the server settings and ignores any previous local configuration. Affects the following parameters: • featurekeyN type • featurekeyN label
Format	Boolean
Default Value	0 (disable)
Range	0 (disable) 1 (enable)
Example	featurekey1 locked: 1

Parameter– *expmodX key N locked*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Locks the specified top softkey on the 5-Series Expansion Module attached to the IP phone. Locking the key prevents a user from changing or configuring the softkey on the expansion module. When a key is locked, the phone uses the server settings and ignores any previous local configuration. Affects the following parameters: • expmodX keyN type • expmodX keyN value • expmodX keyN line
Format	Boolean
Default Value	0 (disable)
Range	0 (disable) 1 (enable)
Example	expmod1 key4: 1

Appendix A

Customizing 560M Expansion Module Column Display

Expansion Module 1

Parameter– *expanmod1page1left*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the first 560M expansion module, first page, left column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod1page1left: Personnel Ext

Parameter– *expanmod1page1right*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the first 560M expansion module, first page, right column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod1page1right: Operations Ext

Parameter– *expanmod1page2left*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the first 560M expansion module, second page, left column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod1page2left: Marketing Ext

Parameter– *expanmod1page2right*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the first 560M expansion module, second page, right column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod1page2right: Logistics Ext

Parameter– *expanmod1page3left*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the first 560M expansion module, third page, left column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod1page3left: Engineering Ext

Parameter– *expanmod1page3right*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the first 560M expansion module, third page, right column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod1page3right: Shipping Ext

Appendix A

Expansion Module 2

Parameter– *expanmod2page1left*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the second 560M expansion module, first page, left column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod2page1left: Personnel Ext

Parameter– *expanmod2page1right*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the second 560M expansion module, first page, right column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod2page1right: Operations Ext

Parameter– *expanmod2page2left*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the second 560M expansion module, second page, left column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod2page2left: Marketing Ext

Parameter– *expanmod2page2right*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the second 560M expansion module, second page, right column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod2page2right: Logistics Ext

Parameter– *expanmod2page3left*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the second 560M expansion module, third page, left column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod2page3left: Engineering Ext

Parameter– *expanmod2page3right*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the second 560M expansion module, third page, right column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod2page3right: Shipping Ext

Expansion Module 3

Parameter– *expanmod3page1left*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the third 560M expansion module, first page, left column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod3page1left: Personnel Ext

Parameter– *expanmod3page1right*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the third 560M expansion module, first page, right column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod3page1right: Operations Ext

Parameter– *expanmod3page2left*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the third 560M expansion module, second page, left column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod3page2left: Marketing Ext

Parameter– *expanmod3page2right*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the third 560M expansion module, second page, right column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod3page2right: Logistics Ext

Parameter– *expanmod3page3left*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the third 560M expansion module, third page, left column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod3page3left: Engineering Ext

Parameter– *expanmod3page3right*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the third 560M expansion module, third page, right column.
Format	Text String
Default Value	N/A
Range	N/A
Example	expanmod3page3right: Shipping Ext

Appendix A

Advanced Operational Parameters

The following parameters in this section allow the system administrator to set advanced operational features on the IP phones.

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 Zero (0) disables the max boot count feature.
Example	max boot count: 0

Single Call Restriction

Parameter – *two call support* *Two Call Support* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP->RTP Settings aastra.cfg, <mac>.cfg
Description	Enables or disables the single media path restriction between the 57i CT base unit and the handset. When this feature is enabled (set to 1), you can make separate active calls from the 57i CT base unit and from the cordless handset. If this feature is disabled (set to 0), only one call can be active at a time either from the base unit or from the handset. When this feature is disabled, and you make an active call on either the base unit or the handset, any other attempt to make an active call is put on hold. Also, when this feature is disabled, more than one call can negotiate complex audio codecs since only a single call is decoding audio at a time.	
Format	Boolean	
Default Value	1	
Range	0 - Disable 1 - Enable	
Example	two call support: 0	

Blacklist Duration

Parameter – *sip blacklist duration* *Blacklist Duration (Aastra Web UI)*	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP Settings-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	Specifies the length of time, in seconds, that a failed server remains on the server blacklist. The IP phone avoids sending a SIP message to a failed server (if another server is available) for this amount of time. **Note:** The value of "**0**" disables the blacklist feature.	
Format	Integer	
Default Value	300 (5 minutes)	
Range	0 to 9999999	
Example	sip blacklist duration: 600	

Whitelist Proxy

Parameter – *sip whitelist* *Whitelist Proxy* (Aastra Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	This parameter enables/disables the whitelist proxy feature, as follows: • Set to 0 to disable the feature. • Set to 1 to enable the feature. When this feature is enabled, an IP phone accepts call requests from a trusted proxy server *only*. The IP phone rejects any call requests from an untrusted proxy server.	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	sip whitelist: 1	

Symmetric UDP Signaling Setting

Parameter – *sip symmetric udp signaling*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to enable or disable the phone to use port 5060 to send SIP UDP messages. The value "**1**" (which is the default) enables the phone to use port 5060. The value "**0**" (zero) disables the phone from using port 5060 and allows the phone to choose a random port to send SIP UDP messages.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	sip symmetric udp signaling: 0

User-Agent Setting

Parameter – *sip user-agent*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to enable or disable the addition of the User-Agent and Server SIP headers in the SIP stack. The value of "0" prevents the UserAgent and Server SIP header from being added to the SIP stack. The value of "1" allows these headers to be added.
Format	Boolean
Default Value	1 (true)
Range	0 (false) 1 (true)
Example	sip user-agent: 0

Troubleshooting Parameters

The following parameters in this section allow the system administrator to set logging and support settings for troubleshooting purposes.

Log Settings

Parameter – *log server ip* *Log IP* (in Web UI)	**Aastra Web UI** Advanced Settings->Troubleshooting->Log Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the IP address for which to save log files for troubleshooting purposes.
Format	IP address
Default Value	0.0.0.0
Range	Not Applicable
Example	log server ip: 192.168.3.2

Parameter – *log server port* *Log Port* (in Web UI)	**Aastra Web UI** Advanced Settings->Troubleshooting->Log Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the IP port to use to save log files for troubleshooting purposes. This is the IP port that transmits information from the IP phone to the IP address location.
Format	Integer
Default Value	0
Range	Any valid IP port
Example	log server port: 2

Parameters – *log module <module name>* *Module/Debug Level* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Troubleshooting-> Module/Debug Level aastra.cfg, <mac>.cfg
Description	Allows enhanced severity filtering of log calls sent as blog output. The blog, as used on the IP phones, is a an online debugging tool that can be frequently updated and intended for technical support analyzation. Blogs are defined by their format: a series of entries posted to a single page in reverse-chronological order. The IP Phone blogs are separated into modules which allow you to log specific information for analyzing: **Module Name (configuration files)** • linemgr • user interface • misc • sip • dis (display driver) • ept (endpoint) • ind (indicator) • kbd (keyboard) • net (network) • provis (provisioning) • rtpt • snd (sound)	
Format	Boolean	
Default Value	1 (enable)	
Range	0 - Disable 1 - Enable	
Examples	log module sip: 0 log module kbd: 1	

Appendix B
Configuration Server Setup

About this appendix

Introduction

This appendix describes how to setup the TFTP protocol configuration server in your network.

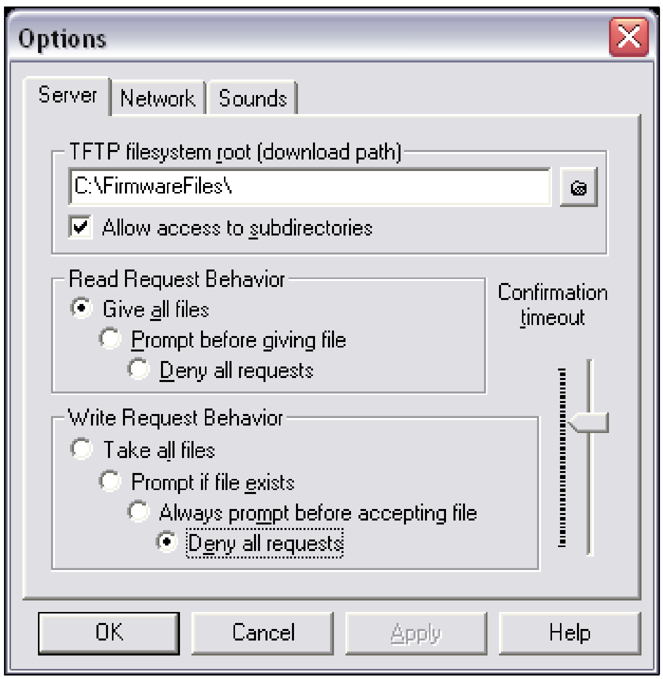
Topics

This appendix covers the following topics:

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 C Configuring the IP Phone at the Asterisk IP PBX

About this appendix

Introduction

This appendix describes how to setup a user's phone with an extension to make and receive calls using the Asterisk as the PBX.

Topics

This appendix covers the following topics:

Appendix C

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 57i 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 57i phone.

```
exten -> 1234,1,Dial(SIP/phone1,20)
```

This definition completes configuring the 57i 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/phone1192.168.1.1(D) 255.255.255.255 5060
Unmonitored
```

This completes the basic set-up for the 57i 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 D
Sample Configuration Files

Appendix D

About this appendix

Introduction

This appendix provides sample configuration files for the 57i, 57i CT, and 53i.

Topics

This appendix covers the following topics:

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.

57i Sample Configuration File

```
# Sample Configuration File
# ===========================


# Date: October 20th, 2005

# Phone Model: 57i

# 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 57i, 57iCT, and 53i 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
```

Sample Configuration Files


```
#  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
#alternate tftp server:
#use alternate tftp server: 1         # If your DHCP server assigns
                                        # a TFTP server address which
                                     # you do not use, you can use
                                        # the alternate tftp server.

## FTP server settings
#ftp server: 192.168.0.131   # can be IP or FQDN
#ftp username: aastra
#ftp password: 57iaastra


## 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 57i and 57iCT
#     L1 to L3 on the 53i
#
#   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
```

Sample Configuration Files


```
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
#  57i or 57iCT 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
```

57i CT Sample Configuration File

```
# Sample Configuration File
# =========================


# Date: October 26th, 2005

# Phone Model: 57iCT

# 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 57i, 57iCT, and 53i phones will download 2
# configuration files from the TFTP server while restarting, the
```

Sample Configuration Files


```
# "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.
```

Sample Configuration Files


```
#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
#alternate tftp server:
#use alternate tftp server: 1         # If your DHCP server assigns
                                        # a TFTP server address which
                                        # you do not use, you can use
                                        # the alternate tftp server.

## FTP server settings
#ftp server: 192.168.0.131   # can be IP or FQDN
#ftp username: aastra
#ftp password: 57iaastra


## 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 57i and 57iCT
#     L1 to L3 on the 53i
#
#   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
#  57i or 57iCT 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 57i
#  phone, following are the parameters specific to the 57i Cordless
#  phones' handset. These parameters can be defined either int the
```

Appendix D

```
#  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
```

Sample Configuration Files


```
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
```

Sample Configuration Files

53i Sample Configuration File

```
# Sample Configuration File
#= = = = = = = = = = = = = = = = = = = = = =

# Date: October 26th, 2005

# Phone Model: 53i

# 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 57i, 57iCT, and 53i 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
#alternate tftp server:
#use alternate tftp server: 1    # If your DHCP server assigns
                                 # a TFTP server address which
                                 # you do not use, you can use
                                 # the alternate tftp server.

## FTP server settings
#ftp server: 192.168.0.131   # can be IP or FQDN
#ftp username: aastra
#ftp password: 57iaastra


## 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 57i and 57iCT
#     L1 to L3 on the 53i
#
#   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 53i phone, and only 2 on the 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, 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).

# 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 E
Sample BLF Softkey Settings

About this appendix

Introduction

This appendix provides sample BLF softkey settings for both the Asterisk server and the BroadSoft BroadWorks server.

Topics

This appendix covers the following topics:

Appendix E

Sample BLF Softkey Settings

Asterisk BLF

The following are sample softkey and programmable key configurations to enable Asterisk BLF support on Aastra IP phones.

57i and 57i CT Configuration Parameters for Asterisk BLF

```
softkey1 type: blf
softkey1 value: 9995551212
softkey1 label: John
softkey1 line: 1
```

53i 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
```

Sample BLF Softkey Settings

BroadSoft BroadWorks BLF

The following are sample softkey and programmable key configurations to enable Broadsoft BroadWorks Busy Lamp Field support on Aastra IP phones.

57i and 57i 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:my57i-blf-list@as.broadsoft.com
```

53i 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:my53i-blf-list@as.broadsoft.com
```

Appendix F
Sample Multiple Proxy Server Configuration

Appendix F

About this appendix

Introduction

This appendix provides a sample multiple proxy server configuration.

Topics

This appendix covers the following topics:

Appendix F

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 G
Creating an XML Application

About this appendix

Introduction

This appendix provides information required to create an XML application for use on the IP phones.

Topics

This appendix covers the following topics:

How to Create an XML Application

Appendix G

Overview

This Appendix describes how to create an XML application for your IP phones. Sections in this appendix include:

- XML format
- Creating XML Objects
- Creating Custom Softkeys
- HTTP Post
- 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	>

Support of Virtual Web Servers

The IP phones support the configuration of a web sever to host multiple websites with different content. With this feature, the webserver looks at the Host header in the incoming HTTP request sent by the browser to determine which website the phone is trying to access.

Creating XML Objects

This section describes how to create XML objects.

The IP phone XML API supports the following proprietary objects that allow for the customization of the IP phone's display.

You can also use the following attributes/options with the XML objects:

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 57i/57i 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	
Done		X	X
Cancel		X	X
Exit	X	X	X
Dial		X	X
Submit			X
Backspace			X
Nextspace			X
Dot			X
ChangeCase			X
Numeric/Alpha			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. You can also turn off automatic numbering of menu items using the **style=”none**" attribute or use bullets for lists instead of numbers using the **style=”bullet”**.

TheText Menu object allows users to navigate the application, by linking HTTP requests to menu items.

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".

Softkeys:

- 1=Select
- 6=Done

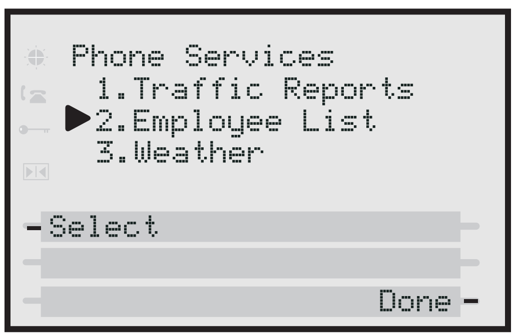
XML Description:

```
<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.

Turning Off Automatic Numbering of Menu Items

When implementing the AastraIPPhoneTextMenu object, you can turn off automatic numbering of the items in a list using the **style=”none**" attribute.

XML Description:

You use this attribute as follows.

```
<AastraIPPhoneTextMenu style=”none”>
   <Title>Example</Title>
   <MenuItem>
      <Prompt>Item 1</Prompt>
      <URI>http://somepage.xml</URI>
   </MenuItem>
</AastraIPPhoneTextMenu>
```

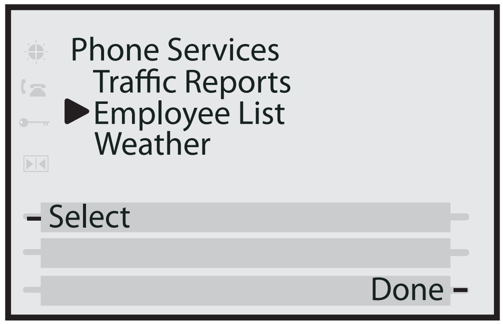
XML example:

The following is an example of using the "**style=none**" attribute with the AastraIPPhoneText Menu object.

```
<AastraIPPhoneTextMenu style=”none”>
   <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:

The following example shows the items in the Text Menu without the item numbering.



Appendix G

Using Bullets in Menu Items

When implementing the AastraIPPhoneTextMenu object, you can use bullets to list items using the **style=”bullet**" attribute.

XML Description:

You use this attribute as follows.

```
<AastraIPPhoneTextMenu style=”bullet”>
   <Title>Example</Title>
   <MenuItem>
      <Prompt>Item 1</Prompt>
      <URI>http://somepage.xml</URI>
   </MenuItem>
</AastraIPPhoneTextMenu>
```

XML example:

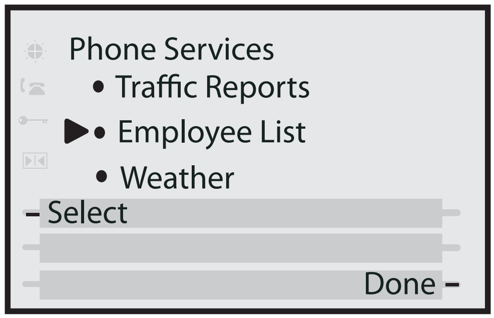
The following is an example of using the "**style=bullet**" attribute with the AastraIPPhoneText Menu object.

```
<AastraIPPhoneTextMenu style=”bullet”>
   <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>
```

Appendix G

XML Screen Example:

The following example shows the items in the Text Menu without the item numbering.



Text Screen Object (Text Screens)

You use the **AastraIPPhoneTextScreen** object to display text to the LCD screen on the IP Phone. The screen text wraps appropriately and can scroll to display a message longer then four lines.

After implementing this object, text displays to the LCD on the IP phone. A user can scroll through the screens as required. If you use the "**destroyOnExit**" attribute in the XML script, when the user exits the XML screeens, the screens are destroyed. You can also allow specific text screens to redisplay for redirection to a new page by using the "**doneAction**" attribute and specifying the new page to go to in the XML script.

Notes:
1. You can use the "**destroyOnExit**" attribute with any XML object as required.
2. You can use the "**doneAction**" attribute with the **AastraIPPhoneTextScreen** and **AastraIPPhoneFormattedTextScreen** objects only.
3. 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*".

Implementation

The following is how you would implement the Text Screen object.

Softkey:

- 6=Done

XML Description:

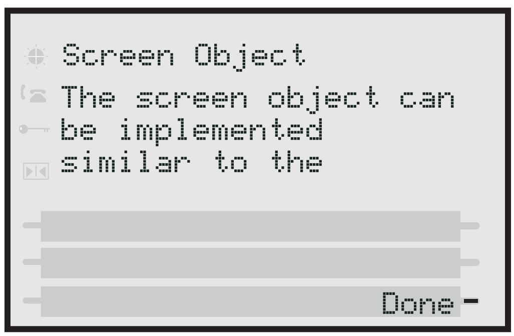
```
<AastraIPPhoneTextScreen destroyOnExit = "yes/no">
   <Title>Screen Title</Title>
   <Text>The screen text goes here</Text>
</AastraIPPhoneTextScreen>

or

<AastraIPPhoneTextScreen doneAction="HTTP address">
   <Title>Screen Title</Title>
   <Text> The screen text goes here</Text>
</AastraIPPhoneTextScreen>
```

***XML Example 1*:**

```
<AastraIPPhoneTextScreen destroyOnExit = "yes">
   <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>
```

Note: This example displays text that you can scroll through on the LCD screen. As you scroll the screen, the previous text is destroyed.

***XML Screen Example 1*:**

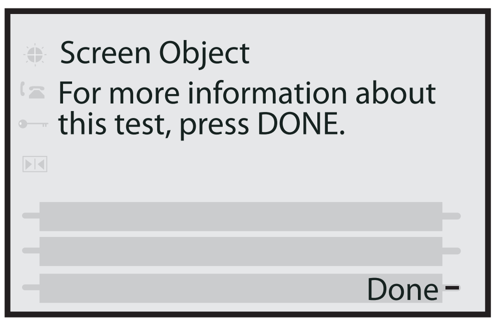


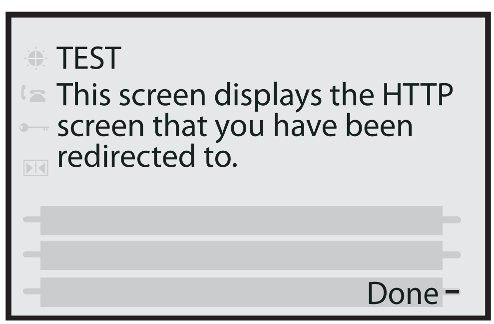
***XML Example 2*:**

```
<AastraIPPhoneTextScreen doneAction="http://10.50.10.117/
test.xml">
   <Title>Screen Object</Title>
   <Text>For more information about this test, press Done.</Text>
</AastraIPPhoneTextScreen>
```

Note: This example displays text that you can scroll through on the LCD screen. As you scroll the screen, and then press **DONE** (55i, 57i, 57i CT) or the **RIGHT ARROW** key (53i), the screen redirects you to the location specified in the script. After pressing **DONE** or the **RIGHT ARROW** key, the phone checks if a "**doneAction**" exists in the XML script. If it does, the screen gets redirected to the location specified. If it does not exist, then the scrolled screens use the "**destroyOnExit**" attribute and destroy the screens.

***XML Screen Example 2*:**





XML Support for Answer and Ignore Softkeys

When the IP phone receives an XML application (either via a post or an incoming action URI) while a call is coming into the phone, the user can either answer or ignore the call with new softkeys that display (55i, 57i, and 57i CT), or press the left and right arrow keys (53i), without canceling the XML application.

For a 55i, 57i, and 57i CT, an Administrator can use the "**Answer**" and "**Ignore**" attributes in an XML script to implement this feature. For a 53i, an Administrator can use the "**allowAnswer**" attribute with the **AastraIPPhoneTextScreen** XML object. Valid values for the "allowAnswer" attribute are "yes" or "no" (default).

For 55i, 57i, and 57i CT:

- The **Answer** and **Ignore** softkeys display on the LCD when the phone has an incoming call at the same time it receives an XML application.
- XML applications are destroyed if the phone receives a call after the XML has been rendered.

When the **Answer** softkey displays, you can press it to answer the incoming call without disturbing the current XML application. When you answer the call, the softkey disappears from the LCD. Pressing the **Ignore** softkey ignores the incoming call without disturbing the current XML application.

Implementation (55i, 57i, 57i CT)

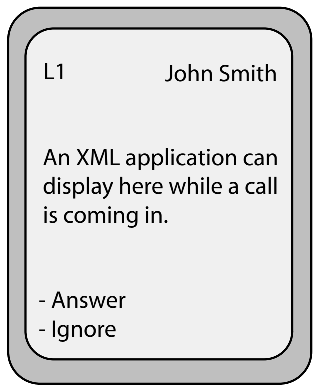
***Softkeys*:**

- 1=Answer
- 2=Ignore

***XML Example*:**

```
<SoftKey index="1">
   <Label>Answer</Label>
   <URI>SoftKey:Answer</URI>
</SoftKey>

<SoftKey index="2">
   <Label>Answer</Label>
   <URI>SoftKey:Ignore</URI>
</SoftKey>
```

***XML Screen Example*:**



For 53i:

- An **<Ignore Answer>** line displays on the LCD when the phone receives an incoming call at the same time it receives an XML application.
- XML applications are destroyed if the phone receives a call after the XML has been rendered.

When the **<Ignore Answer>** line displays, you can press the **Right Arrow** key (Answer) to answer the incoming call without disturbing the current XML application. When you answer the call, the <Ignore Answer> line disappears from the LCD. Pressing the **Left Arrow** key ignores the incoming call without disturbing the current XML application.

Implementation (53i)

***XML Example*:**

```
<AastraIPPhoneTextScreen allowAnswer = "yes">
   <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*:**

Text Screen Format Object (for Text Screens)

The **AastraIPPhoneFormattedTextScreen** object allows you to specify a format for the text that displays in the LCD window on the phone. Using this object, you can specify the following for the text that displays:

- alignment (using the "**Align**" attribute and specifying right, left, or center)*
- text size (using the "**Line Size**" attribute and specifying normal or double height)*
- display type (using the "**Line**" or "**Scroll**" attributes to specify static or scrolling)*

*See the AastraIPPhoneFormattedTextScreen structure example on page G-18.

The phone's LCD screen allows up to a total of 5 lines for displaying text. The formatted text displays in three distinct blocks in the order it is written in the XML object:

- The first block displays text at the top of the LCD window. By default, this text block is static. This block can contain as many lines as the XML object specifies and can range from zero (0) up to the LCD screen size.

- The second block displays below the first block. By default, the second text block displays as scrolling text. This block takes up as many lines as the XML developer specifies, up to the LCD screen size.
- The third block displays below the second block. By default, the third block displays static text, and takes up whatever lines remain blank on the LCD screen.

Note: Any of the three blocks can be set as static text or scrolling text but the text displays on a total of 5 lines only. Any lines that display after line 5 are lost.

Using the AastraIPPhoneFormattedTextScreen object, the display truncates a line after the last word in that line and continues to wrap the text to the next line.The phone ignores any lines that display after the 5th line on the LCD screen.

AastraIPPhoneFormattedTextScreen Structure

The **AastraIPPhoneFormattedTextScreen** object describes the structure of the XML document that you can use to format the text that displays on the phone's LCD screen. The default structure of the AastraIPPhoneFormattedTextScreen object is:

```
<AastraIPPhoneFormattedTextScreen>
    <Line Size="<normal/double>" Align="<right/left/center>">A line
        of static text</Line>
    <Scroll Height=n>
        <Line Size="<normal/double>" Align="<right/left/center>">
            Scrolling text</Line>
        <Line>Another line of Scrolling Text</Line>
        <Line>Yet another scrolling line</Line>
    </Scroll>
    <Line>Some static footer text</Line>
</AastraIPPhoneFormattedTextScreen>
```

Note: The text in bold in the above structure indicate the options you can use for text size (normal, double), alignment (right, left, center), and scrolling height (from 1 to 5 lines). Setting a scrolling height less than "1" automatically sets the scroller height to "1".

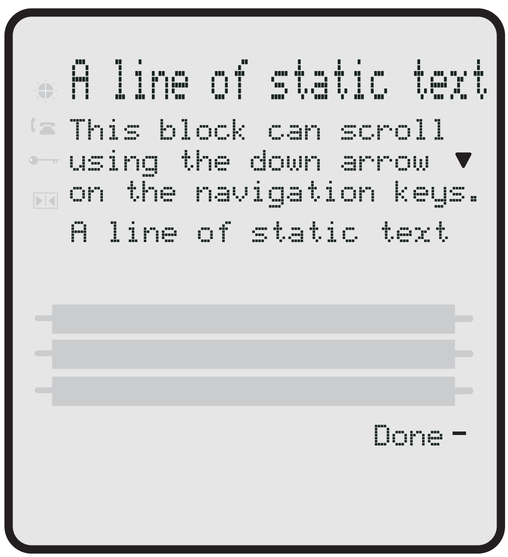
***XML Example*:**

The following is an example of using the AastraIPPhoneFormattedTextScreen object with a static line at the top and bottom blocks and 2 scrolling lines in the middle block.

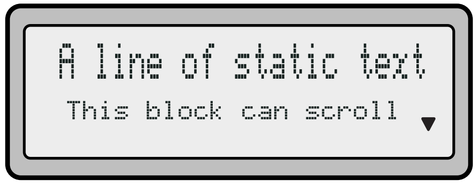
```
<AastraIPPhoneFormattedTextScreen>
    <Line Size="double" Align="left">A line of static text</Line>
    <Scroll Height=2>
        <Line Size="normal" Align="right"> Scrolling text</Line>
        <Line>Another line of Scrolling Text</Line>
        <Line>Yet another scrolling line</Line>
     </Scroll>
     <Line>Some static footer text</Line>
</AastraIPPhoneFormattedTextScreen>
```

***XML Screen Examples*:**

IP Phones 55i, 57i, and 57i CT



*IP Phone 53*i



Note: You can view all 5 lines at once on the 55i, 57i and 57i CT IP phones. You can view only 2 lines at a time (up to 5 lines) on the 53i IP phone.

UserInput Object (User Input Screens)

The **AastraIPPhoneInputScreen** object allows application developers to create screens for which the user can input text 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.

For User input screens, you can also create the following:

- Time and Date Format User input screens. For more information, see "Time and Date Formats (User Input Screens)" on page G-28.
- Multiple input fields per AastraIPPhoneInputScreen object. For more information, see "Multiple Input Fields (User Input Screens)" on page G-31.

Implementation (IP Addresss)

The following is how you would implement the **AastraIPPhoneInputScreen** object using an IP Address.

Note: For all available parameters you can use for the **AastraIPPhoneInputScreen** 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

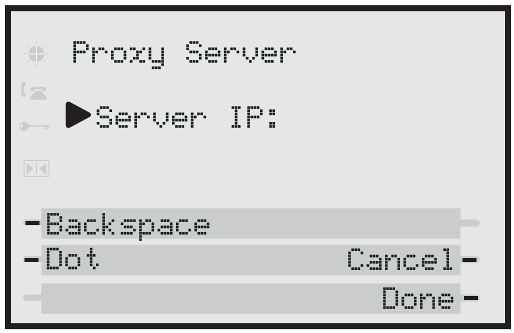
***XML Description*:**

```
<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*:**



Implementation (Number)

The following is how you would implement the UserInput object using Numbers.

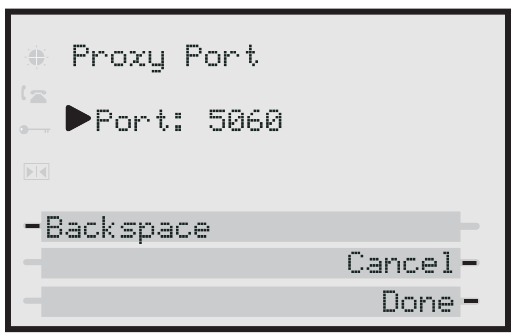
***Softkeys*:**

- 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:



Implementation (String)

The following is how you would implement the UserInput object using Strings in XML.

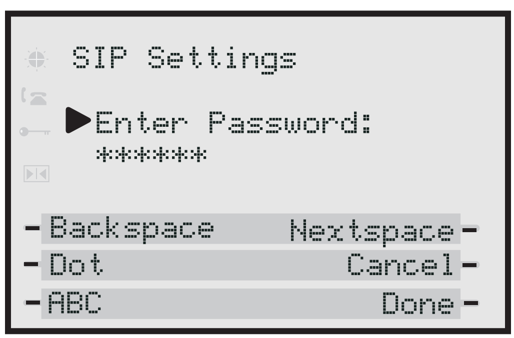
Softkeys:

- 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 the user entered 12345, then the URL sent back to the server is http://10.50.10.53/script.pl?passwd=12345.

XML Softkey for Special Characters (User Input Screens for 55i, 57i, and 57i CT only)

In Release 2.1, the IP Phone can dynamically receive a Symbol List when it receives the **AastraIPPhoneInputScreen** XML object. You can have a single symbol specified for the softkey, or you can have a list of symbols. When there is only one symbol in the list, the symbol displays with no delay. When there is a list of symbols, you can keep pressing the symbol softkey to cycle through the list of symbols to select the one you want to use.

To display a list of customized symbols to the phone's softkey, the server must include the list of characters in the URI field of the XML softkey script. The URI must be in the format:

```
SymbolList=”<Symbol List content>”
```

The content of the Symbol List must be encapsulated by quotes. You can specify multiple symbols in one URI. For example, the **SymbolList="@#”** specifies the @ and # symbols.

Note: You can have multiple Symbol List softkeys with different lists of symbols. The maximum length of the data in a Symbol List is 230 characters.

There are some special characters that needed to be encoded due to XML limitations. The following table specifies these characters.

Symbol	XML Encoding
single quote (‘)	'
double quote (“)	"
greater-than sign (>)	>
less-than sign (<)	<
ampersand (&)	&

The following is an example XML URI using the characters in the table above:

```
SymbolList="@#&amp;&gt;&lt"
```

The Symbol List content for this URI is @, #, &, >, <.

Implementation

Softkeys:

- 1 = <Single Symbol or Symbol List>

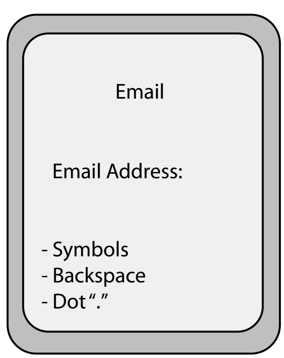
XML Softkey Example:

```
<SoftKey index="1">
   <Label>Symbols</Label>
   <URI>SoftKey:SymbolList="@#=&amp;"</URI>
</SoftKey>
```

XML Object and Softkey Example:

```
<AastraIPPhoneInputScreen type = "IP">
   <Title>Email</Title>
   <SoftKey index="1">
   <Label>Symbols</Label>
   <URI>SoftKey:SymbolList="@"</URI>
   </SoftKey>
   <SoftKey index = "2">
      <Label> Backspace </Label>
      <URI>SoftKey:Exit</URI>
   </Softkey>
   <SoftKey index = "3">
      <Label> Dot </Label>
      <URI>SoftKey:Exit</URI>
   </Softkey>
<Prompt>Email Address:</Prompt>
   <URL>http://myserver.com/myscript.com</URL>
   <Parameter>email</Parameter>
   <Default></Default>
<AastraIPPhoneInputScreen>
```

XML Screen Example:



Time and Date Formats (User Input Screens)

The **AastraIPPhoneInputScreen** object allows you to specify US ((HH:MM:SS am/pm and MM/DD/YYYY) or International (HH:MM:SS and DD/MM/YYYY) time/date formats for an XML user input screen on the IP phone.

Note: You use this time/date format in an XML script that displays in an XML window. This time/date format does NOT affect the time/date format configured under the Options Menu on the phone.

Using this AastraIPPhoneInputScreen object, you can specify the following attributes for date/time format to create the user input screen:

- “**timeUS**”
- “**timeInt**”
- **“US Date”**
- **“Int Date”**

You can use these XML attributes in an application that requires the user to specify a time or a date, for example, when scheduling a meeting. The user could press a softkey that gets the XML script asking you to enter a time/date. This time/date is then forwarded to a script that schedules the meeting or reports a conflict.

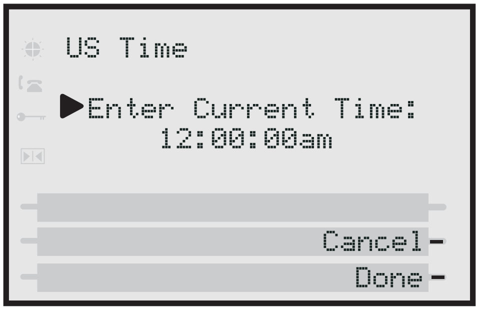
<u>XML Examples:</u>

The following example illustrates the AastraIPPhoneInputScreen object using the time/date attributes.

<u>United StatesTime</u>

```
<AastraIPPhoneInputScreen type="timeUS">
    <Title>US Time</Title>
    <Prompt>Enter Current Time:</Prompt>
    <Parameter>Time</Parameter>
 </AastraIPPhoneInputScreen>
```

IP Phones 55i, 57i/57i CT



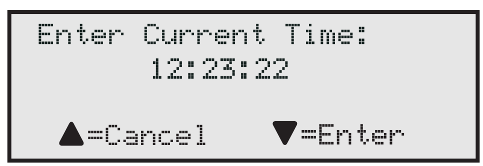
IP Phone 53i

<u>International Time</u>

```
<AastraIPPhoneInputScreen type="timeInt">
    <Title>Int Time</Title>
    <Prompt>Enter Current Time:</Prompt>
    <Parameter>Time</Parameter>
    <Default>14:23:22</Default>
 </AastraIPPhoneInputScreen>
```

IP Phones 55i, 57i/57i CT

IP Phone 53i

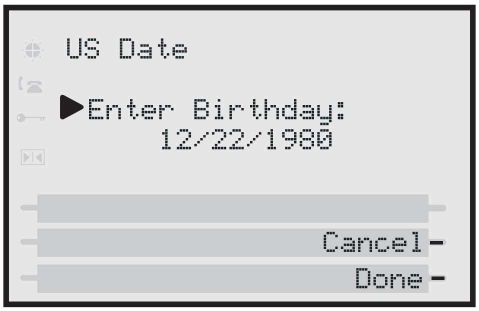


Note: If you don't specify a time, the XML script uses the following default values:
- 12:00:00am (US)
- 00:00:00 (International)

United States Date

```
<AastraIPPhoneInputScreen type="dateUS">
    <Title>US Date</Title>
    <Prompt>Enter Birthday:</Prompt>
    <Parameter>Date</Parameter>
    <Default>12/22/1980</Default>
 </AastraIPPhoneInputScreen>
```

IP Phones 55i, 57i/57i CT



IP Phone 53i

International Date

```
<AastraIPPhoneInputScreen type="dateInt">
    <Title>Int Date</Title>
    <Prompt>Enter Birthday:</Prompt>
    <Parameter>Date</Parameter>
 </AastraIPPhoneInputScreen>
```

IP Phones 55i, 57i/57i CT

IP Phone 53i

Note: If you don't specify a date, the XML script uses the current date in the specified format.

Multiple Input Fields (User Input Screens)

You can have multiple input fields per AastraIPPhoneInputScreen object.The following specifications apply to this feature:

- Multiple input fields are applicable to 55i, 57i, and 57i CT IP phones only.
- Each AastraIPPhoneInputScreen object can support up to 6 input fields.
- Each input field supports its own set of custom softkeys (these take precedance over any softkeys configured outside of the <InputField> tags.
- The phone treats softkey information specified outside of the <InputField> tags as a default value.
- Each AastraIPPhoneInputScreen object supports only 1 URL.

The following tags support this feature.

Tags for Multiple Input Fields	
<InputField>.......</InputField> **Note:** This tag supports the type, password, and editable attributes. If specified, they override the default values provided in the root tag.	**New tags within the <InputField> tags**: <Prompt> <Parameter> <Default> <Softkey> **Note:** These tags override the values specified outside of the <InputField> tags.
<Selection>.......</Selection>	Any data specified between the <Selection> tags is appended to the URI in the form Selection = <DATA>.

You use the following attributes to create multiple input fields using the AastraIPPhoneInputScreen object.

New AastraIPPhoneInputScreen Attributes	
defaultIndex	Allows you to set an integer value from 1 - 6 that specifies which input field the edit cursor starts in. Default is 1.
displayMode	Allows you to set a specific display mode for the screen: "**normal**" or "**condensed**". Default is "**normal**" mode. **Normal Mode** - Prompts and inputs each appear on separate lines on the display. The user can scroll through the lines. **Notes**: **1.** An <InputField> tag with a type set to "**empty**" creates two blank lines in "normal" mode. **2.** All inputs fields are left justfied in "normal" mode. **Condensed Mode** - Prompts and inputs appear on the same line (up to 5 inputs on a single screen). **Notes**: **1.** Title tag wrapping limits the number of fields per screen to 1 or 3 in condensed mode. **2.** An <InputField> tag with a type set to "**empty**" creates a blank line in "condensed" mode.

Example

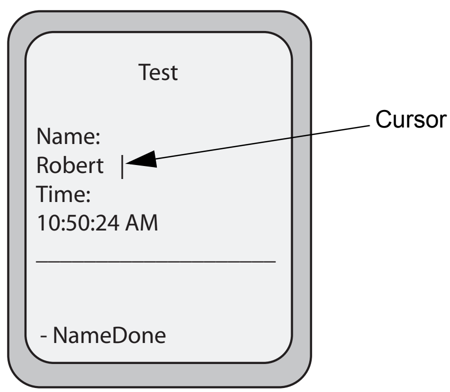
The following is an example AastraIPPhoneInputScreen script using the multiple input field feature:

```
<AastraIPPhoneInputScreen type="string">
   <Title>Test</Title>
   <URL>http://10.50.10.117/test.pl</URL>
   <Default/>
   <InputField>
      <Prompt>Name:</Prompt>
      <Parameter>name</Paramter>
      <Default>Robert</Default>
      <SoftKey index="3">
         <Label>NameDone</Label>
         <URI>Softkey:Submit</URI>
      </SoftKey>
```

```
    </InputField>
    <InputField type="timeUS">
        <Prompt>Time:</Prompt>
        <Parameter>time</Paramter>
        <Default>10:50:24AM</Default>
    </InputField>
    <InputField type="dateUS>
        <Prompt>Date:</Prompt>
        <Parameter>date</Paramter>
        <Default>01/24/07</Default>
        <SoftKey index="5">
            <Label>DateDone</Label>
            <URI>Softkey:Submit</URI>
        </SoftKey>
    </InputField>
</AastraIPPhoneInputScreen>
```

Posting the above AastraIPPhoneInputScreen XML script displays the following screens on the IP phone.

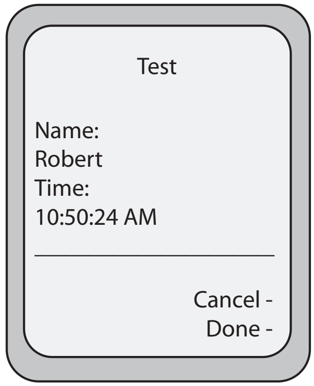
Screen 1



Screen 1 Note: Notice that the cursor is active within the first element field and that the phone is displaying the InputField #1 softkey.

Scolling down changes to the next input field.

Screen 2



Screen 2 Note: Notice that the softkeys change to reflect the new input type, timeUS.

Scrolling down again goes to the next page of inputs.

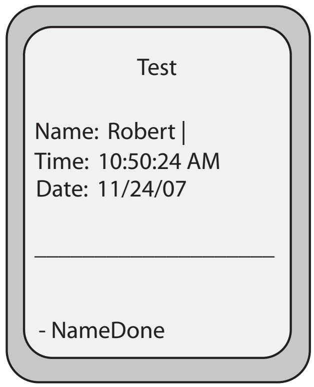
Screen 3

Screen 3 Note: Notice that the softkeys change again to reflect the current input item on the screen (Date).

Condensed Mode Screen

If you use the attribute "**displayMode = "condensed**" in the root tag (**Screen 4**), the display changes to have prompt and inputs on the same line. The behavior of the screens using "condensed" mode is the same as Screens 1 through 3 but without the scrolling.

Screen 4



Directory Object (Directory List Screen) *(57i 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.

Implementation

The following is how you would implement the Directory object in XML.

Note:

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>57i - John Doe 1</Prompt>
      <URI> 10.50.10.49</URI>
   </MenuItem>
   <MenuItem>
      <Prompt>53i - John Doe 2</Prompt>
      <URI>4326</URI>
   </MenuItem>
<MenuItem>
      <Prompt>57i 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.

Status Message Object (Idle Screen)

The IP phones support an XML **AastraIPPhoneStatus** object for displaying status messages on a single designated line on the phone's idle screen. The messages display when the server pushes XML information to the phone.

The 57i/57i CT phones display messages on the second line in the phone window. (where "No Service" would display if there was no service. If there is no service on the phone, the "No Service" message overrides the XML object message). The 53i phone displays messages on the first line (overriding the DisplayName). The phone truncates long messages that are wider then the phone screen.

If the phone receives multiple messages, the first message received displays first and the remaining messages scroll consecutively one at a time. Messages remain displayed until they are removed (by the server) or the phone reboots. The AastraIPPhoneStatus object feature is always enabled.

Note: You can set the amount of time, in seconds, that a message displays to the phone before scrolling to the next message. For more information about this feature, see "Scroll Delay Option (configurable via config files and Aastra Web UI only)" on page G-58.

AastraIPPhoneStatus Structure

The **AastraIPPhoneStatus** object describes the structure of the XML document that you can use to send status messages to the phone. The basic structure of the AastraIPPhoneStatus object is:

```
<AastraIPPhoneStatus>
   <Session>My session ID</Session>
   <Message index="Msg index">Message</Message>
<! -- Additional status messages may be added under new Message tags-->
<AastraIPPhoneStatus/>
```

The "**My Session ID**" attribute must be unique to the application sending the XML object to the phone. The application generates the session ID, which could be a combination of letters and numbers. There is a maximum of one **<Session>** tag per PhoneStatus object, so the **<Session>** tag is optional.

XML Examples

Example 1: The following is an example of using the AastraIPPhoneStatus object:

```
<AastraIPPhoneStatus>
   <Session>abc12345</Session>
<Message index="3">Server side call forwarding disabled</Message>
<AastraIPPhoneStatus/>
```

In this example, the AastraIPPhoneStatus object sends the default behavior with the status message (i.e., the status message is added to the scroll list).

Example 2: You can also use the AastraIPPhoneStatus object to remove status messages from the display, by setting an empty tag for the <Message index> tag.

The following example removes the status message that was posted to the phone in Example 1.

```
<AastraIPPhoneStatus>
   <Session>abc12345</Session>
<Message index="3"/>
<AastraIPPhoneStatus/>
```

Appendix G

Execute Commands Object (for executing XML commands)

An **AastraIPPhoneExecute** object on the IP phones allows the phone to execute commands using XML. The phones support the following execute object commands:

- **Reset -** This command waits until the phone is idle and then executes a reset. You can use this command in an XML script to enable the server to force phone firmware changes and/or for troubleshooting purposes.
- **NoOp** - This command has no affect on the IP phone. It is made up of a blank URI. You can use this feature when you need to press a key on the phone to access a feature, and it is not necessary to display anything.
- **FastReboot** - This command allows an administrator to perform a fast reboot of the IP phone when required.

Note: For FastReboot commands, you must set the XML push server list in the Aastra Web UI from where the XML HTTP post is being sent.

- **XML Softkey/Programmable Key LED Behavior -** The **URI=”Led: <softkey or programmable key type>=<execute command>”/** controls the behavior of XML softkey or programmable key LEDs. The softkey or programmable key must be configured for XML before you enable this feature.

The following paragraphs describe the AastraIPPhoneExecute object and the commands you can use with this object.

AastraIPPhoneExecute Object Structure

The **AastraIPPhoneExecute** object describes the structure of the XML document that you can use to send a command to the phone. It delivers multiple execution requests to the phone. The basic structure of the AastraIPPhoneExecute object is:

```
<AastraIPPhoneExecute>
   <ExecuteItem URI ="the URL or URI to be executed"/>
<! -- Additional execution items may be added under new ExecuteItem tag-->
</AastraIPPhoneExecute>
```

Using the Reset Command

The <ExecuteItem URI =""/> tag can be entered with the command the phone should execute. Upon receiving an AastraIPPhoneExecute object, the phone begins executing the URL or URI specified.

The following example shows an AastraIPPhoneExecute object using the **Reset** command:

```
<AastraIPPhoneExecute>
       <ExecuteItem URI="Command: Reset"/>
</AastraIPPhoneExecute>
```

➡ **Note:** If you specify a command as a URI attribute (instead of a URL), the keyword "**Command**" must be prepended in the value of the URI attribute so that the phone recognizes it as a URI attribute value. If you enter a URI and leave out the “**Command**” keyword, the phone interprets the value in the URI attribute as a URL containing network resources.

The following example shows the AastraIPPhoneExecute object using a URL:

```
<AastraIPPhoneExecute>
       <ExecuteItem URI="http://aastraserver/message.xml"/>
</AastraIPPhoneExecute>
```

When the phone receives this object, it displays the specified XML URI page.

Using the NoOp Command

You can use the AastraIPPhoneExecute object as an object to create a blank display (it has no affect on the IP phone). It is made up of a blank URI. You can use this feature when you need to press a key on the phone to access a feature, and it is not necessary to display anything. You can also use the AastraIPPhoneExecute object and this command with other objects in an XML script.

The following example shows an AastraIPPhoneExecute object using a blank URI:

```
<AastraIPPhoneExecute>
       <ExecuteItem URI=””/>
<AastraIPPhoneExecute>
```

Using the FastReboot Command

Using an XML post, the FastReboot feature forces a phone to reboot in the idle state. It does not check for new firmware and only downloads language packs if there is a change in the supported filenames of the language packs. The phone DOES do a DHCP lookup on every “fast reboot”. It also only downloads the Directory files if the names have changed.

Note: For FastReboot commands, you must set the XML push server list in the Aastra Web UI from where the XML HTTP post is being sent.

The following example shows an AastraIPPhoneExecute object using the **FastReboot** command.

```
<AastraIPPhoneExecute>
       <ExecuteItem URI="Command:FastReboot"/>
</AastraIPPhoneExecute>
```

XML Softkey or Programmable Key LED Behavior Command

You can control the behavior of XML softkey or programmable key LEDs using the command **URI=”Led: <softkey or programmable key type>=<execute command>”/** with the **AastraIPPhoneExecute** object.

This LED feature is applicable to XML softkeys (top or bottom), programmable keys, and expansion module softkeys. The softkey or programmable key must be configured for XML before you enable this feature.

The following table describes the behaviors you can assign to the XML key LEDs, and the attribute you enter in the AastraIPPhoneExecute script.

LED Behavior	XML Attribute
on	URI = “Led: <softkey type>= on”/
off	URI = “Led: <softkey type>=off”/
fastFlash	URI = “Led: <softkey type>=fastflash”/
slowFlash	URI = “Led: <softkey type>=slowflash”/

The command for LED control is sent using the XML HTTP post from one of the XML push server lists. The following example shows an XML LED command in an AastraIPPhoneExecute script.

```
<AastraIPPhoneExecute>
   <ExecuteItem URI=”Led: topsoftkey20=on”/>
   <ExecuteItem URI=”Led: softkey1=off”/>
   <ExecuteItem URI=”Led: prgkey2=fastflash”/>
   <ExecuteItem URI=”Led: expmod2 key20=slowflash”/>
</AastraIPPhoneExecute>
```

Notes:

1. Using the Aastra Web UI, you must set the XML push server list from where the XML HTTP post is being sent.

2. The LED states specified in the AastraIPPhoneExecute script are not saved after a reboot.

Dynamic Configuration Object (to push a configuration to the phone)

The IP phones provide an XML feature that allows you to make configuration changes to the phone that take affect immediately, without having to reboot the phone. This feature involves creating XML scripts that push the changed configuration parameter(s) from the server to the IP phones.

You can use the **AastraIPPhoneConfiguration** object in the XML scripts to change configuration parameters or configure new parameters. However, since the IP phone does not save **new** parameters created in XML scripts to the *local.cfg* file, when the phone reboots, it does not save the new parameters on the phone. In order for the phone to apply **new** configuration parameters, you have to enter the parameters via the user interfaces (Telephone User Interface, Web User Interface, or configuration files), or reapply the new parameters using the XML scripts after every boot.

Specific configuration parameters are dynamic on the phone when pushed from XML scripts on the server. See "Dynamic Configuration Parameters" on page G-46 for more information about dynamic configuration parameters.

AastraIPPhoneConfiguration Object Structure

The **AastraIPPhoneConfiguration** object describes the structure of the XML document that you can use to push the configuration to the IP phone. The basic structure of the AastraIPPhoneConfiguration object is:

```
<AastraIPPhoneConfiguration>
   <ConfigureItem>
      <Parameter>Parameter name</Parameter>
      <Value>Parameter value</Value>
   </ConfigureItem>

   <ConfigureItem>
      <Parameter>Parameter name</Parameter>
      <Value>Parameter value</Value>
   </ConfigureItem>
   .
   .
   .
<AastraIPPhoneConfiguration>
```

Note: You can add more configuration items as applicable. There is a limit of 5000 bytes (less than 5k) on the size of XML configuration objects.

XML Example:

The following is an example of an XML configuration script that creates a softkey you can press on the IP phone to push the configuration from the server to the phone:

```
<AastraIPPhoneConfiguration>
   <ConfigureItem>
      <Parameter>softkey1 label</Parameter>
      <Value>John Doe</Value>
   </ConfigureItem>
   <ConfigureItem>
      <Parameter>softkey1 type</Parameter>
      <Value>speeddial</Value>
   </ConfigureItem>
   <ConfigureItem>
      <Parameter>softkey1 value</Parameter>
      <Value>5551234</Value>
   </ConfigureItem>
</AastraIPPhoneConfiguration>
```

Dynamic Configuration Parameters

The following table identifies which configuration parameters change dynamically when using XML configuration scripts to push the configuration to the phone.

Dynamic Configuration Parameters	
softkeyN type*	line7 ring tone
softkeyN label*	line8 ring tone
softkeyN value*	line9 ring tone
softkeyN line*	suppress dtmf playback
softkeyN states*	redial disabled**
prgkeyN type*	call transfer disabled
prgkeyN name*	conference disabled
prgkeyN value*	directory disabled**
prgkeyN line*	callers list disabled**
live dialpad	options password enabled
tone setv	time server disabled
ring tone	time reserved
audio mode	dst config
language	time server1
ringback timeout	time server2
headset tx gain	time server3
headset sidetone gain	time format
handset tx gain	date format
handset sidetone gain	time zone name
handsfree tx gain	time zone code
line1 ring tone	time zone minutes
line2 ring tone	dst minutes
line3 ring tone	dst start relative date
line4 ring tone	dst start month
line5 ring tone	dst start day
line6 ring tone	dst start week
dst start hour	sip intercom line
dst end relative date	sip allow auto answer

Dynamic Configuration Parameters	
dst end month	sip silence suppression
dst end day	sip send mac
dst end week	sip send line
dst end hour	xml application URI
tftp server	xml application title
alternate tftp server	xml beep notification
use alternate tftp server	action uri registered
admin password	action uri incoming
user password	action uri outgoing
sip nat ip	action uri onhook
sip nat port	action uri offhook
sip dial plan	action uri startup
sip dial plan terminator	play a ring splash
sip digit timeout	map redial key to
sip blf subscription period	map conf key to
sip registration retry timer	download protocol
sip forward number	ftp server
sip forward mode	ftp username
sip ring number	ftp password
sip vmail	http server
sip dtmf method	http path
sip lineN forward number	directory 1***
sip lineN forward mode	directory 2***
sip lineN ring number	directory script
sip lineN vmail	auto resync time
sip lineN dtmf method	auto resync mode
sip intercom type	
sip intercom prefix code	
sip intercom mute mic	

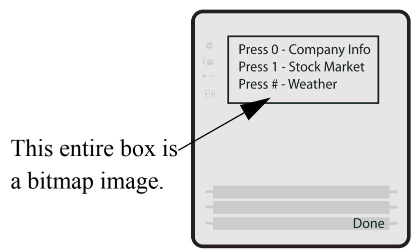
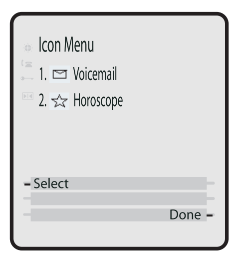
*Changes to subscriptions (BLF or BLA) require a reboot.

**This parameter is dynamic so a user can't access it or add to it. However, you need to reboot the phone to clear the list.

*** You need to reboot the phone to download new directories.

XML Image Objects (55i, 57i/57i CT only)

The 55i and 57i/57i CT IP phones provide an XML feature that allows you to load images in XML applications that display to the LCD screen. The following table describes the three types of image objects that you can use in an XML script:

XML Image Object	Image Type	Description
AastraIPPhoneImageScreen	Standalone Bitmap Image	Displays a single bitmap image according to alignment, height, and width specifications. 
AastraIPPhoneImageMenu	Menu Image	Displays a bitmap image as a menu. Menu selections are linked to keypad keys (0-9, *, #). 
AastraIPPhoneTextMenu (Icon Menu)	Icon Menu Image	Displays a small icon before each item in the menu. 

You can include images with an XML object in two ways:

- Using pixel data stored with specified tags
- Using an internal bitmap loaded via a specified URI

Note: The actual resolution of the image on the LCD screen is dependent on the phone model. The 55i screen has a resolution of 144 x 75 pixels. The 57i/57i CT screens have a resolution of 144 x 128 pixels. However, the display of the image is limited to 40 x 144 pixels high.

The following paragraphs describe each of the image objects and how you can use them in the XML scripts.

AastraIPPhoneImageScreen Object

The **AastraIPPhoneImageScreen** object displays an image as a single bitmap image on the LCD screen. You can specify the placement of the image on the screen by setting the following attributes:

- **verticalAlign**
- **horizontalAlign**
- **height**
- **width**

The image is specified as a series of hexadecimal characters. Two hex characters map to one byte of pixel data, where each bit represents a pixel. The image data describes the bitmap from left to right and top to bottom. The data is padded on an 8-bit boundary, so if the height and width do not match the pixel information, then the image will not display correctly. The character strings map to the middle and four corners of the screen. If desired, you can specify an integer as an absolute pixel for customized displays.

Note: A special URI of "Image:Logo" is used to load the current logo bitmap for the phone (Aastra or branded). In this case, the height and width can be ignored.

XML Example:

The XML script for the AastraIPPhoneImageScreen object is as follows:

```
<AastraIPPhoneImageScreen>
 <Image verticalAlign ="top,middle,bottom,'pixel'",
         horizontalAlign = "left,middle,right,'pixel'"
         height="'integer'", width="'integer'">
    FF00F800FF00FFFFFFFFFFFF00FF00FF00FFF8F200
  </Image>
  <!-vertical and horizontal default to "middle" ->
  <!-height and width must match the image data ->
</AastraIPPhoneImageScreen>
```

Note: This object supports the destroyOnExit attribute.

XML Screen Example:

Appendix G

AastraIPPhoneImageMenu

The **AastraIPPhoneImageMenu** object allows you to use a bitmap image to display as a menu. Each menu item is linked to a keypad key (0-9, *, #). You can use this type of image object when you want to display menu choices as a non-ASCII character set or with pictures only. You can specify the placement of the image on the screen by setting the following attributes:

- **verticalAlign**
- **horizontalAlign**
- **height**
- **width**

The image is specified as a series of hexadecimal characters. Two hex characters map to one byte of pixel data, where each bit represents a pixel. The image data describes the bitmap from left to right and top to bottom. The data is padded on an 8-bit boundary, so if the height and width do not match the pixel information, then the image will not display correctly. The character strings map to the middle and four corners of the screen. If desired, you can specify an integer as an absolute pixel for customized displays.

XML Example:

The XML script for the AastraIPPhoneImageMenu object is as follows:

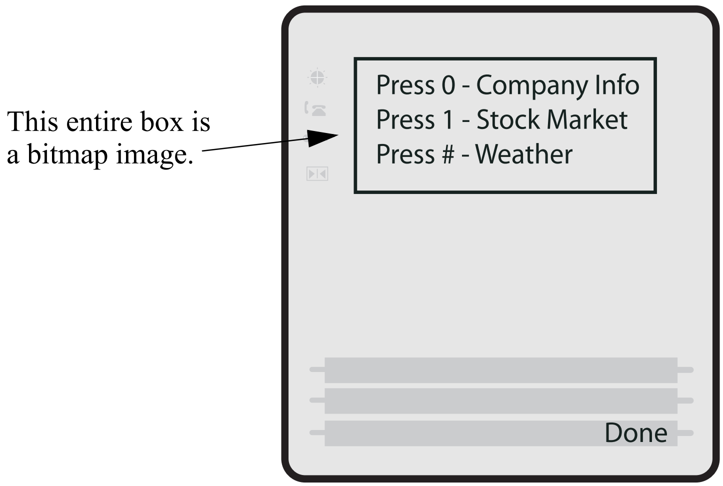
```
<AastraIPPhoneImageMenu>
 <Image verticalAlign ="top,middle,bottom,'pixel'",
         horizontalAlign = "left,middle,right,'pixel'"
         height="'integer'", width="'integer'">
    FF00F800FF00FFFFFFFFFFFF00FF00FF00FFF8F200
  </Image>
  <!-Base attribute is optional-->
  <URIList base = "http://someserver/">
    <URI key = "0">link1.php</URI>
    <URI key = "1">link2.php</URI>
    <URI key = "#">link3.php</URI>
</URIList>
</AastraIPPhoneImageMenu>
```

Appendix G

Each URI is appended to the "base" URI if one exists. The default softkey is a "Done" key at the bottom right position. It is a parse error if two URIs specify the same index.

Note: This object supports the destroyOnExit attribute.

XML Screen Example:



Appendix G

AastraIPPhoneTextMenu (Icon Menu)

The **AastraIPPhoneTextMenu** (Icon Menu) object is identical to the "Text Menu Object (Menu Screens)" described on page G-6, except a small icon image appears after the menu number and before the text item. The Text Menu object allows application developers to create a numerical list of menu items on the IP phones. The AastraIPPhoneTextMenu (Icon Menu) object allows users to navigate the application, by linking HTTP requests and icons to menu items.

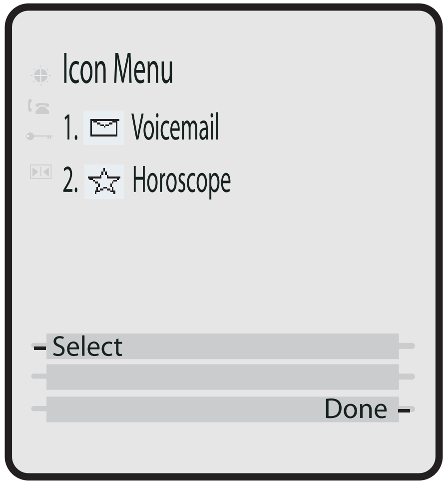
XML Example:

The XML script for the AastraIPPhoneTextMenu (Icon Menu) object is as follows:

```
<AastraIPPhoneTextMenu>
  <Title> Icon Menu </Title>
  <MenuItem base = http://someserver/,
            icon="1">
    <Prompt> Voicemail </Prompt>
    <URI>scripts/voicemail.php</URI>
  </MenuItem>
  <MenuItem icon="2">
    <Prompt> Horoscope</Prompt>
    <URI> scripts/horoscope.php </URI>
  </MenuItem>
  <IconList>
    <Icon index = "1"> Icon:Envelope </Icon>
    <Icon index = "2"> FFFF0000FFFF0000 </Icon>
  </IconList>
</AastraIPPhoneTextScreen>
```

The "**icon=1**" attribute in the above example tells the phone to display the icon specified in the IconList called "**<Icon index = "1"> Icon:Envelope </Icon>**". For more information about the "IconList", see "Using the <IconList>" on page G-54.

XML Screen Example:



Using the <IconList>

You can incorporate the use of an **<IconList>** attribute in the XML script to define the images you will use with the AastraIPPhoneTextMenu object. After you define the icons in the <IconList>, you can then call those icons as you need them throughout the script. This also allows you to assign the icons to softkeys on the phone.

You can load images in the <IconList> in two ways:

- By specifying the image after a special prefix called “Icon:”
  For example:

```
<IconList>
   <Icon index = “1”> Icon:Envelope </Icon>)
<IconList>
```

- By specifying the image in hexidecimal format
  For example:

```
<IconList>
   <Icon index = “2”> FF00F3 </Icon>)
<IconList>
```

Note: For a list of icons that are internal to the IP phone when specifying the "Icon:<uri name>", see the section "IP Phone Internal Icon Images" on page G-56.

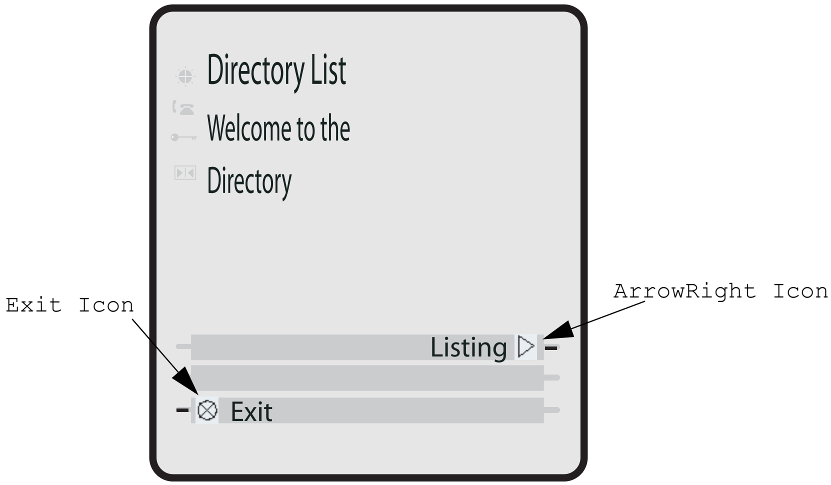
You can use one or both ways to load images within the same XML script. The softkey icons display at the edge of the screen. Any XML object that supports softkeys can support the optional <IconList> tag.

XML Example:

The following example illustrates the use of the <IconList> attribute to create images for softkey prompts.

```
<AastraIPPhoneTextScreen>
  <Title> Directory List </Title>
  <Text> Welcome to the Directory. </Text>
  <SoftKey index = "4" icon="2">
    <Label> Listing </Label>
    <URI> http://server/page.php </URI>
  </SoftKey>
  <SoftKey index = "3" icon="1">
    <Label> Exit </Label>
    <URI> http://server/otherpage.php </URI>
  </SoftKey>
  <IconList>
    <Icon index = "1"> FF00F3 </Icon>
    <Icon index = "2"> Icon:ArrowRight </Icon>
  </IconList>
</AastraIPPhoneTextScreen>
```

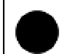
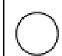
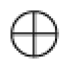
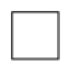
XML Screen Example:



IP Phone Internal Icon Images

The IP phones have internal icon images you can use when specifying the “Icon:<uri name>” attribute in the <IconList>. The following table lists the internal icons you can specify.

URI Name	Displays this:
PhoneOnHook	
PhoneOffHook	
PhoneRinging	
DND	None
ArrowRight	
ArrowLeft	
Speaker	

URI Name	Displays this:
ArrowUp	
ArrowDown	
ArrowsUpandDown	
FilledCircle	
EmptyCircle	
Envelope	
Prohibit	
Square	
Ellipse	
TailArrowUp	
TailArrowDown	

Attributes/Options to Use with XML Objects

Beep Attribute (configurable via XML objects, config files, or Aastra Web UI)

You can enable or disable a **"Beep"** option to indicate a status on the phone using the Status Message object (AastraIPPhoneStatus), the configuration files, or the Aastra Web UI.

Note: For enabling/disabling a status message beep using the configuration files and the Aastra Web UI, see Chapter 5, the section, "Enabling/Disabling a Beep for Status Message Displays" on page 5-200.

When the phone receives a status message, the BEEP notifies the user that the message is displaying. The following attribute in the AastraIPPhoneStatus object enables/disables the BEEP from being heard:

`< AastraIPPhoneStatus Beep="yes|no">` (case sensitive)

This attribute is optional. If notification is required, the attribute must be in the ROOT. If the BEEP attribute is set to "**yes**" (i.e. Beep="yes") then it is an indication to the phone to sound a beep when it receives the object. If the Beep attribute is set to "**no**" (i.e. Beep="no") or not present, then the default behavior is no beep is heard when the object arrives to the phone.

Note: The value set in the configuration files and Aastra Web UI override the attribute you specify for the AastraIPPhoneStatus object.

Scroll Delay Option (configurable via config files and Aastra Web UI only)

The IP phones support a scroll delay option that allows you to set the time delay, in seconds, between the scrolling of each status message on the phone. The default time is 5 seconds for each message to display before scrolling to the next message. You can configure this option via the configuration files or the Aastra Web UI. Changes are dynamic and apply to the phone immediately.

Note: For more information about setting the scroll delay option, see Chapter 5, the section, "Scroll Delay Option for Status Messages" on page 5-201.

Timeout Attribute (configurable via XML objects only)

The XML "**Timeout"** attribute allows you to specify a timeout value for the LCD screen display. You must use the "**Timeout**" aattribute in the ROOT. When the phone receives an XML object with this attribute, it overrides the default 45 second timeout specified for custom applications. Setting the "Timeout" attribute to zero (0) disables this feature.

***<u>XML Example</u>*:**

The following example illustrates the use of the "Timeout" attribute with the AastraIPPhoneFormattedTextScreen object. The result would cause the LCD screen to timeout in 30 seconds.

```
<xs:element name="AastraIPPhoneFormattedTextScreen">
       <xs:attribute name="Beep" 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:AastraIPPhoneFormattedTextScreen Timeout= "30" default = "45">
           <xs:simpleType>
             <xs:restriction base="xs:number/>
           </xs:simpleType>
        </xs:attribute>
```

Appendix G

LockIn Attribute (configurable via XML objects only)

The XML "**LockIn"** attribute allows you to specify whether or not the information on the LCD screen stays displayed when other events occur (such as pressing buttons on the keypad).

Note: This attribute is ignored during incoming calls. If this attribute is set, and the phone receives an incoming call, the LCD screen exits the XML information and displays the information about the incoming call.

You must use the "LockIn" aattribute in the ROOT. The settings for the LockIn attribute are "Yes" for enabled, and "No" for disabled.

***XML Example*:**

The following example illustrates the use of the "LockIn" attribute with the AastraIPPhoneFormattedTextScreen object. The result would cause the LCD screen to lock in the display of the XML information even if other events occur (except for an incoming call).

```
<xs:element name="AastraIPPhoneFormattedTextScreen">
       <xs:attribute name="Beep" 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:AastraIPPhoneFormattedTextScreen LockIn= "yes" default="no">
           <xs:simpleType>
             <xs:restriction base="xs:string>
                 <xs:pattern value="yes|no"/>
             </xs:restriction>
           </xs:simpleType>
        </xs:attribute>
```

Appendix G

CancelAction Attribute (configurable via XML objects only)

The XML "**cancelAction"** attribute allows you to specify a URI that a GET is executed on when the user presses the default CANCEL key.

Note: The URI must be fully qualified.

You can add this optional feature at the end of any of the XML objects. The format is:

```
<Hardkey action="Cancel">
   <Label>Label to Display</Label>
   <URI>URI to Get</Label>
</Hardkey>
```

XML Example:

The following illustrates the AastraIPPhoneTextScreen object using the "Cancel" attribute:

```
<AastraIPPhoneTextScreen cancelAction=”http://10.50.10.117/
ft.xml”>
<Title>Test</Title>
<Text>This is a test of the cancel action</Text>
</AastraIPPhoneTextScreen>
```

When this XML script is sent to the phone, and the user presses the default CANCEL key, the script executes a GET on http://10.50.10.117/ft.xml.

Note: If the "cancelAction" attribute was not used in the above script, then pressing the CANCEL key would simply cancel the current screen.

Appendix G

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%3E57i+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. The phone denies any posts larger than this limit.

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);
?>
```

HTTP Refresh Header

You can use an HTTP refresh header with the XML screen objects on the IP phones. This feature provides the following:

- All current XML screen objects have the ability to be refreshed by adding a **Refresh** and **URL** setting to the HTTP headers. (see Refresh setting format below)
- The Refresh setting is set by the XML application and it is up to the application to decide which objects it wants to refresh.

Note: This HTTP refresh header feature only applies to objects that display to the screen.

The Refresh setting must be included in the HTTP header. The XML application decides which objects it wants to use with this setting. The phone recognizes this setting when parsing the HTTP header. If the setting is present, then it passes along the refresh timeout and the URL to the ParserData object, which all XML screen objects inherit from. The ParserData class also has a timer, which must be set to expire at the next refresh time. When the timer expires (time to refresh the screen), the phone requests the URL again and displays the refreshed screen.

Refresh Setting Format

The following is the Refresh setting format for the HTTP header:

```
Refresh: <timeout>; URL=<page to load>
```

The following example is a Refresh setting for use in an HTTP header:

```
Refresh: 3; URL=http://10.50.10.140/cgi-bin/update.xml
```

Note: You must use the **Refresh** and **URL** parameters in order for this feature to work in the HTTP header.

HTTP Language Header

You can use an HTTP language header with the XML screen objects on the IP phones. By including the language information in the HTTP header, the HTTP GET includes the phone model, the firmware and the MAC address of the phone. When the user selects a language on the phone, the selected language is sent to the XML application and the application displays to the screen in that language.

Note: This HTTP language header feature only applies to objects that display to the screen.

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/xsdvalidator/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:AastraIPPhoneTextScreen LockIn="yes" default="no">
     <xs:simpleType>
       <xs:restriction base="xs:string>
          <xs:pattern value="yes|no"/>
       </xs:restriction>
     </xs:simpleType>
</xs:attribute>
    <xs:AastraIPPhoneTextScreen Timeout="30" default="45">
       <xs:simpleType>
    </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.

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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SIP IP Phone Administrator Guide

Models 53i, 55i, 57i, and 57i CT

If you've read this owner's manual and consulted the Troubleshooting section and still have problems, please visit our website at www.aastratelecom.com or call 1-800-574-1611 for technical assistance.

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